



EZLoop® IP-PBX Enterprise SIP Server



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CH1. Overview

The EZLoop® IP-PBX Enterprise SIP Server is the next generation all-in-one IP PBX system for small to medium enterprise. It is also designed to operate on a variety of VoIP applications, such as auto-attendant, voice conference, call transfer, call pick up and IP-based communications. With the tiny box, small to medium enterprise or homes can use it to access the Internet and to make VoIP phone calls.

Customers can select different suite and optional products to meet their request. It can be integrated with EZLoop® PSTN Gateway FX04/06 and can provide PSTN access function; it can also provide extensions. With flexible and full functionality, EZLoop® IP-PBX Enterprise SIP Server can give a complete transition from traditional PABX to the new generation IP-PBX.



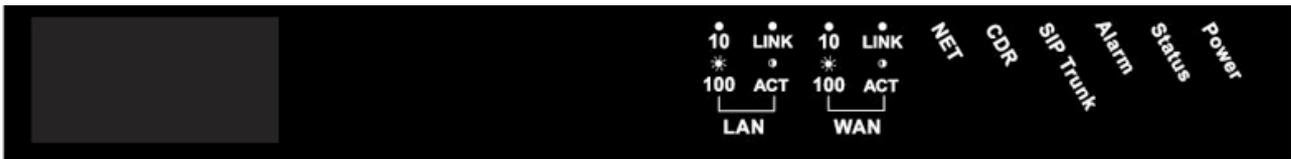
1.1 Specifications

- **Protocol**
 - SIP (Session Initiation Protocol)
- **Call Features**
 - Authentication
 - Automated Attendant
 - Call Transfer (CPE based)
 - Blind Transfer (CPE based)
 - Call Forward on Busy (CPE based)
 - Call Forward on No Answer (CPE based)
 - Call Forward Unconditional (CPE based)
 - Call Hold/Retrieval (CPE based)
 - Call Routing
 - Call Waiting (CPE based)
 - Caller ID
 - Do Not Disturb (CPE based)
 - Flexible Extension Logic
 - Music On Hold
 - Music On Transfer
 - Call Pickup
 - Three-way Conference ()
 - Time and Date
 - Trunking (EZLoop® PSTN Gateway FX04/06)
 - VoIP Gateways (EZLoop® PSTN Gateway FX04/06)
 - Voice Mail to e-mail
 - Call Detail Records

- **Codecs**
 - G.711 (A-Law & _-Law)
 - G.729
- **Technical Features**
 - Subscriber NAT transversal
 - Phone set record Greeting
 - Management: Web Browser Management
 - HTTP upgrade firmware and ring back tone file
 - Export/Import configuration
 - Network Interface: 1WAN 1LAN
 - DTMF: in-band, RFC2833, SIP-Info
 - Network: Support Fixed IP, DHCP, and PPPoE mode
- **Capacity**
 - 100 register users
 - 20 concurrent calls
- **Dimension**
 - 17.5 x 12.5 x 3.2 cm

1.2 Hardware Overview

1.2.1 Front Panel and LED Indication



- **Power:** Light on when EZLoop® IP-PBX Enterprise SIP Server is powered on.
- **Status:** Light on when system is ready.
- **Alarm:** Light Flash when system is upgrading software, please do not unplug power when Alarm is flashing.
- **SIP Trunk:** Light on when EZLoop® IP-PBX Enterprise SIP Server successfully registered to all of the enabled SIP Trunks; Light flash when EZLoop® IP-PBX Enterprise SIP Server failed to register to one of the enabled SIP Trunks; light off when there is no SIP Trunk has enabled.
- **CDR:** EZLoop® IP-PBX Enterprise SIP Server can output Call Detail Records to external computer. User has to execute CDR program on computer, when EZLoop® IP-PBX Enterprise SIP Server is ready to connect with CDR server and output data, this indication will light on.

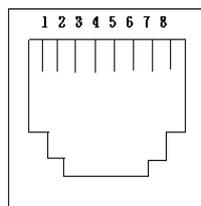
Note:

- CDR Function can only work in local area network. Please prepare the CDR server under LAN.
 - The CDR server is proprietary, for more information about CDR, please contact with the sales person of .
- **NET:** Display Network status. If WAN port of EZLoop® IP-PBX Enterprise SIP Server is under Fixed IP mode, LCD will light on. If WAN port is under DHCP or PPPoE mode, and EZLoop® IP-PBX Enterprise SIP Server succeeds in getting IP, LED will be flashing. If WAN port is under DHCP or PPPoE mode, and EZLoop® IP-PBX Enterprise SIP Server fails to get IP, LED will light off.
 - **WAN**
 - **LINK/ACT:** Light on when WAN port is connected to network. Flash when data is transmitting or receiving.
 - **10/100:** Light on when network rate is 100 Mb/s, and light off when network rate is 10 Mb/s.
 - **LAN**
 - **LINK/ACT:** Light on when LAN port is connected to network. Flash when data is transmitting or receiving.
 - **10/100:** Light on when network rate is 100 Mb/s, and light off when network rate is 10 Mb/s.

1.2.2 Back Panel



- **Reset:** Network and Login information will return to default values.
- **LAN/WAN:** RJ-45 socket, complied with Ethernet 10/100base-T.
The pin-out is as following:



PIN 1, 2: Transmit
PIN 3, 6: Receive

- **12V DC:** Input AC 100V~240V;output DC12V

CH2. Start to configure EZLoop® IP-PBX Enterprise SIP Server

2.1 Step 1

Connect **LAN** port of EZLoop® IP-PBX Enterprise SIP Server with PC via crossover cable or connect with Switch/ Hub via straight through cable.

2.2 Step 2

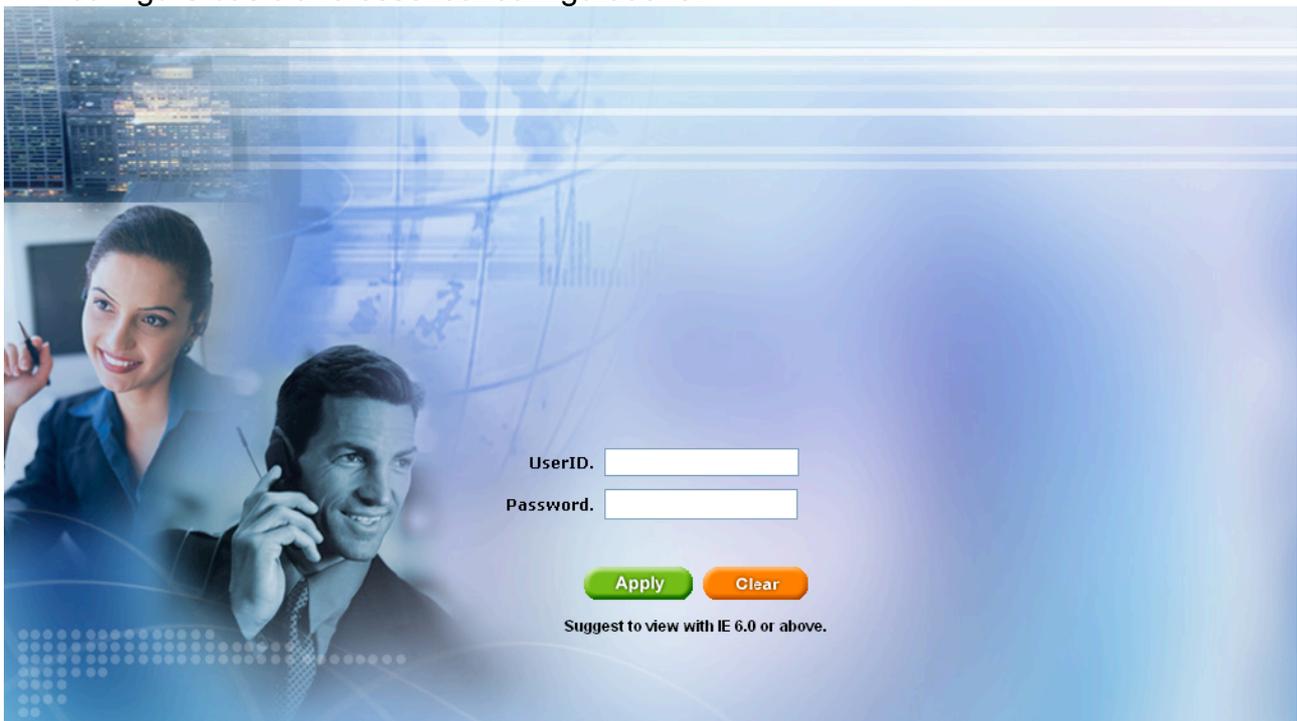
Prepare one computer, and change the IP address to be **192.168.123.12x** with **subnet mask 255.255.255.0**.

2.3 Step 3

Open browser and link to default LAN IP address of EZLoop® IP-PBX Enterprise SIP Server "**192.168.123.123**" with default port number **10087**, i.e. **http://192.168.123.123:10087**

2.4 Step 4

Login EZLoop® IP-PBX Enterprise SIP Server with default userID: "**root**", and **no password**. After login EZLoop® IP-PBX Enterprise SIP Server, user can start to configure basic and essential configurations.



2.5 Step 5: To configure basic and essential configurations

To make EZLoop® IP-PBX Enterprise SIP Server work have to set some basic and essential configurations, those include Network, Extension (FXS and IP Phone devices), and Trunk (FXO devices).

2.5.1 Network Configuration

Enter **Management** → **Network** to configure WAN and LAN IP.

The screenshot shows the 'NetWork Setting' page in the IP-PBX management interface. The page has a blue header with 'IP-PBX' and navigation tabs for 'Configuration', 'Information', 'Management', and 'Reboot System'. A 'Language' dropdown menu is also present. The main content area is titled 'NetWork Setting' and contains two sections: 'WAN' and 'LAN'. Each section has a table of configuration fields. The WAN section includes fields for Mode (Fixed, DHCP, PPPoE), IP Address, Netmask, Gateway, DNS, PPPoE ID, PPPoE PWD, and Mac. The LAN section includes fields for IP Address, Netmask, and Mac. At the bottom of the page, there are 'Apply' and 'Cancel' buttons.

WAN	
Mode:	<input checked="" type="radio"/> Fixed <input type="radio"/> DHCP <input type="radio"/> PPPoE
IP ADDRESS:	192.168.13.96
NETMASK:	255.255.248.0
GATEWAY:	192.168.8.254
DNS:	61.220.126.2
PPPoE ID:	none
PPPoE PWD:	****
Mac:	0001a8131233

LAN	
IP ADDRESS:	192.168.123.123
NETMASK:	255.255.255.0
Mac:	0001a8131316

■ WAN

- **Mode:** Select EZLoop® IP-PBX Enterprise SIP Server WAN port network mode to be Fixed IP, DHCP or PPPoE.
- **IP Address/NETMASK/GATEWAY:** If user has set EZLoop® IP-PBX Enterprise SIP Server to be fixed IP mode. User need to input IP address/Subnet Mask/Default Gateway.
- **DNS:** Input DNS address.
- **PPPoE ID:** If user select PPPoE mode, here can input PPPoE account ID.
- **PPPoE PWD:** If user select PPPoE mode, here can input PPPoE account password.
- **Mac:** Mac address of EZLoop® IP-PBX Enterprise SIP Server WAN port. The Mac address cannot be modified.

■ LAN

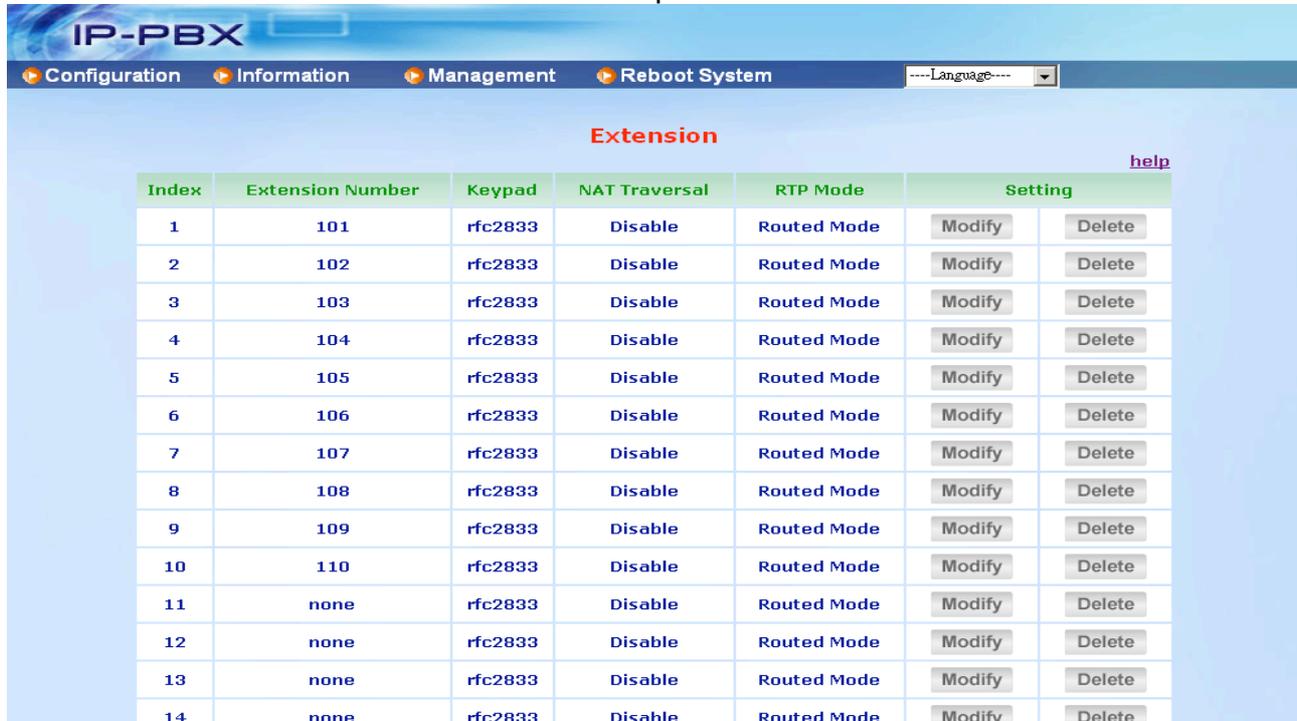
- **IP Address:** Input IP address for LAN port of EZLoop® IP-PBX Enterprise SIP Server.
- **NETMASK:** Input Subnet Mask for LAN port of EZLoop® IP-PBX Enterprise SIP Server.
- **Mac:** Mac address of EZLoop® IP-PBX Enterprise SIP Server LAN port. The Mac address cannot be modified.

Press Apply to save configuration, or press Cancel to quit configuration.

2.5.2 Extension Configuration

User has to set Extension account for other device to register on EZLoop® IP-PBX Enterprise SIP Server.

Enter **Configuration** → **Extension** to configure Extension data. On screen will show 100 sets Extension. User can press **Modify** to add new Extension or modify configured Extension data. Press **Delete** will delete the specified Extension.



Index	Extension Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	101	rfc2833	Disable	Routed Mode	Modify	Delete
2	102	rfc2833	Disable	Routed Mode	Modify	Delete
3	103	rfc2833	Disable	Routed Mode	Modify	Delete
4	104	rfc2833	Disable	Routed Mode	Modify	Delete
5	105	rfc2833	Disable	Routed Mode	Modify	Delete
6	106	rfc2833	Disable	Routed Mode	Modify	Delete
7	107	rfc2833	Disable	Routed Mode	Modify	Delete
8	108	rfc2833	Disable	Routed Mode	Modify	Delete
9	109	rfc2833	Disable	Routed Mode	Modify	Delete
10	110	rfc2833	Disable	Routed Mode	Modify	Delete
11	none	rfc2833	Disable	Routed Mode	Modify	Delete
12	none	rfc2833	Disable	Routed Mode	Modify	Delete
13	none	rfc2833	Disable	Routed Mode	Modify	Delete
14	none	rfc2833	Disable	Routed Mode	Modifv	Delete

After press Modify can input detail setting for Extension.

The screenshot shows the 'IP-PBX' web interface with a navigation bar containing 'Configuration', 'Information', 'Management', and 'Reboot System'. A 'Language' dropdown menu is visible. The main content area displays the 'Extension Setting' form with the following fields and values:

Extension Setting	
Extension Number:	101
Password:	***
Call Group:	1
Pickup Group:	1
DialPlan:	ext+allroutes
Keypad:	RFC2833
NAT Traversal:	Disable
RTP Mode:	Routed Mode
MailBox:	Disable

At the bottom of the form are two buttons: 'Apply' (green) and 'Cancel' (blue).

- **Extension Number:** Assign the number of Extension. This number is also the register name for device.
- **Password:** Assign the register password for device to register on EZLoop® IP-PBX Enterprise SIP Server.
- **Call Group:** You can use the Call Group parameter to assign an Extension to one or more groups.
- **Pickup Group:** You can use the Pickup Group option in conjunction with this parameter to allow a ringing phone to be answered from another extension.

Note:

- The Pickup Group option is used to control which Call Groups a channel may pick up—a channel is given authority to answer another ringing channel if it is assigned to the same Pickup Group as the ringing channel's Call Group. By default, remote ringing extensions can be answered with *8.
- You can define multiple Call Groups and Pickup Groups for one Extension by a "comma". For example, you can input "1,3,5" into Call Group or Pickup Group.

- **DialPlan:** Define the dialing plan for Extension. It specifies the location of the instruction used to control what the phone is allowed to do, and what to do with incoming calls for this extension. In this field, you can Choose 5 dial level for Extension, including [ext-only], [ext+R1], [ext+R12], [ext+R123], [ext+allroutes]. You can define an "Outgoing call" record, to a certain Route Level, as R1, R2..., etc. [ext-only] means this subscriber can only call to Extension. [ext+R1] means the subscriber with such DialPlan can call to Extension and Route Level with R1. [ext+R12] means the subscriber with such DialPlan can call to Extension and Route Level with R1 and R2. [ext+R123] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2 and R3. [ext+allroutes] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2, R3 and R4.

Note:

- For more information about Route Level, please refer to the user manual: [3.1.5.1 Outgoing Call Rule](#).

- **Keypad:** User can select Keypad type to be RFC2833, In-band, or SIP-Info. The

setting should be also match the Keypad setting of Extension device.

- **NAT Traversal:** If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to EZLoop® IP-PBX Enterprise SIP Server on public network, then user has to enable this function. Enable NAT Traversal to force EZLoop® IP-PBX Enterprise SIP Server to ignore the contact information for the Extension and use the address from which the packets are being received.
- **RTP Mode:** User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode. The voice media will be routed “Peer-to-Peer” if two clients are both setting to Direct Mode. This way will improve the voice quality and reduce the performance wastage of the EZLoop® IP-PBX Enterprise SIP Server.

Note:

- If one peer set to Direct Mode but another peer set to Routed Mode, the result will become to Routed Mode.
- Voice media will still go through the EZLoop® IP-PBX Enterprise SIP Server if the EZLoop® IP-PBX Enterprise SIP Server needs to detect DTMF.
- If you enable the NAT Traversal function for Extension, the RTP mode will change to Routed Mode directly; this way will avoid the “one-way voice” or “no voice issue” of VoIP.
- If the both peers are under different subnet, or one peer is under Public IP but another one is under Private IP, **we strongly suggest you to set the RTP mode to Routed Mode to avoid some unexpected voice problems.**

- **Mail Box:** User can select to disable or enable mail box function. If this function is enabled, user has to input e-mail address for the Extension. When having voice mail of incoming call, system will send this voice mail to the specified e-mail address.

Note:

- **Please remember set the SMTP in the page of Management → SMTP Setting to activate the Voice Mail to E-mail.**
- If the EZLoop® IP-PBX Enterprise SIP Server got a new message, it will send the message to the user by email immediately.

Press Apply to save configuration, or press Cancel to quit configuration.

Note:

- For more information about Extension setting, please refer to user’s manual **CH3. Full Web Configurations**

2.5.3 Trunk Configuration

User has to set Trunk account for Trunk (FXO device, e.g.) to register to EZLoop® IP-PBX Enterprise SIP Server or set some necessary configuration for SIP trunk (For more application, please go to.....). Enter **Configuration** → **Trunk** to configure Trunk data. On screen will show **20** sets Trunks. User can press **Modify** to add new Trunk or modify configured Trunk data. Press **Delete** will delete the specified Trunk.

Trunk

[help](#)

Index	Trunk Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	888	rfc2833	Disable	Routed Mode	Modify	Delete
2	889	rfc2833	Disable	Routed Mode	Modify	Delete
3	070070	rfc2833	Disable	Routed Mode	Modify	Delete
4	none	none	Disable	Routed Mode	Modify	Delete
5	none	none	Disable	Routed Mode	Modify	Delete
6	none	none	Disable	Routed Mode	Modify	Delete
7	none	none	Disable	Routed Mode	Modify	Delete
8	none	none	Disable	Routed Mode	Modify	Delete
9	none	none	Disable	Routed Mode	Modify	Delete
10	none	none	Disable	Routed Mode	Modify	Delete
11	none	none	Disable	Routed Mode	Modify	Delete
12	none	none	Disable	Routed Mode	Modify	Delete
13	none	none	Disable	Routed Mode	Modify	Delete
14	none	none	Disable	Routed Mode	Modify	Delete

After press Modify can input detail setting for Trunk.

The screenshot shows the 'Trunk Setting' configuration page in an IP-PBX web interface. The page has a blue header with 'IP-PBX' and navigation tabs for Configuration, Information, Management, and Reboot System. A language dropdown is in the top right. The main content area is light blue and contains a 'Trunk Setting' form with the following fields and values:

Field	Value
Trunk Number	888
Password	***
Host	Dynamic
DialPlan	from-pstn
Keypad	RFC2833
NAT Traversal	Disable
RTP Mode	Routed Mode
Port	
External Server Address	
Maximum Channels	
Outbound Caller ID	
Hot-Key Tran	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
Music RBT	<input checked="" type="radio"/> Disable <input type="radio"/> Enable

At the bottom of the form are 'Apply' and 'Cancel' buttons.

- **Trunk Number:** Assign the number of Trunk. This number is also the register name for Trunk device.

Note:

- The Trunk Number can also be a “Trunk ID”. In the Routing Table page, you should define the destination of prefix route. When you define the prefix route, you should set the Trunk ID (Trunk Number) in the Trunk page first; then you could input the correct Trunk ID in the Destination field.

- **Password:** Assign the register password for device to register on EZLoop® IP-PBX Enterprise SIP Server.
- **Host:** Setting the Host to Dynamic will require the trunk to register the EZLoop® IP-PBX Enterprise SIP Server so that the EZLoop® IP-PBX Enterprise SIP Server knows how to reach the trunk. You can also set the Host to an IP address or FQDN if you set the Host to [Pre-define]. There will be a field called [Address] appear when you choose Host to [Pre-define]. This limits only where you place calls to, as the user is allowed to place calls from anywhere.
- **DialPlan:** Define the dialing plan for Trunk. It specifies the location of the instruction used to control what the phone is allowed to do, and what to do with incoming calls for this extension. In this field, you can Choose 6 dial level for Extension, including [from-pstn], [ext-only], [ext+R1], [ext+R12], [ext+R123], [ext+allroutes]. You can define an “Outgoing call” record, to a certain route level, as R1, R2..., etc. [from-pstn] is used for Trunk only. [ext-only] means this subscriber can only call to Extension. [ext+R1] means the subscriber with such DialPlan can call to Extension and Route Level with R1. [ext+R12] means the subscriber with such DialPlan can call to Extension and Route Level with R1 and R2. [ext+R123] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2 and R3. [ext+allroutes] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2, R3 and R4.

Note:

- For more information about Route Level, please refer to the user manual: [3.1.5.1 Outgoing Call Rule](#).

- **DTMF:** User can select DTMF type to be RFC2833, In-band, or SIP-Info.
- **NAT Traversal:** If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to EZLoop® IP-PBX Enterprise SIP Server on public network, then user has to enable this function. Enable NAT Traversal to force EZLoop® IP-PBX Enterprise SIP Server to ignore the contact information for the Trunk and use the address from which the packets are being received.
- **RTP Mode:** User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode. The voice media will be routed “Peer-to-Peer” if two clients are both setting to Direct Mode. This way will improve the voice quality and reduce the performance wastage of the EZLoop® IP-PBX Enterprise SIP Server.

Note:

- If one peer set to Direct Mode but another peer set to Routed Mode, the result will become to Routed Mode.
 - Voice media will still go through the EZLoop® IP-PBX Enterprise SIP Server if the EZLoop® IP-PBX Enterprise SIP Server needs to detect DTMF.
 - If you enable the NAT Traversal function for Extension, the RTP mode will change to Routed Mode directly; this way will avoid the “one-way voice” or “no voice issue” of VoIP.
 - If the both peers are under different subnet, or one peer is under Public IP but another one is under Private IP, **we strongly suggest you to set the RTP mode to Routed Mode to avoid some unexpected voice problems.**
- **Port:** You can use this to define the SIP signal port if you want to listen on a nonstandard SIP signal port.
 - **External Server Address:** This field will allow you to set the domain in the SIP From URI. Setting this will avoid some unexpected issue if the service provider needs this for authentication.
 - **Maximum Channels:** This will limit the maximum channels for this Trunk. For example, you set 2 into this field; only 2 outgoing calls could go via this Trunk. Default is no limit.
 - **Outbound Caller ID:** Some service provider will require the correct registered caller ID if it got an incoming call. Default the EZLoop® IP-PBX Enterprise SIP Server will send the Extension’s caller ID to this Trunk, if you set empty here.

Note:

- Normally, SIP From URI will contain the Extension’s calling ID and EZLoop® IP-PBX Enterprise SIP Server’s IP address, but some ITSP may reject this call due to some security issue. You can modify the Calling ID and IP/ Domain in the fields of [External Server Address] and [Outbound Caller ID] when the call is going via the EZLoop® IP-PBX Enterprise SIP Server to the Destination (Trunk) to avoid such security issue.
 - If you set a FXO gateway as the Trunk, you can just use the default Trunk 888 and 889 as the FXO’s register number.
 - For the FXO gateway, you may just only configure Trunk Number, Password, Host, DialPlan, Keypad, NAT Traversal and RTP Mode.
 - If you set the ITSP as the Trunk, you may need to set the following configure: Port, External Server Address and Outbound Caller ID.
 - **For more information, please refer to the CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.**
- **Hot-Key Tran:** Enable this feature will permits the calling party or called party to transfer a call by pressing the ***0 (For Blind Transfer)** or ***9 (For consultant**

Transfer) key if the call is Between Extension and Trunk. Default is disabled.

Note:

- If you enable this feature in Trunk page, we suggest you also enable Hot-Key Tran of IP PBX page.
- Please note that if this option is used, the RTP Mode will always be Routed Mode, as EZLoop® IP-PBX Enterprise SIP Server needs to monitor the call to detect when the caller presses the *0 or *9 key.

- **Music RBT:** Provides music to the calling party until the call is answered

Press Apply to save configuration, or press Cancel to quit configuration.

Note:

- For more information about Trunk setting, please refer to **[CH3. Full WEB Configurations.](#)**

CH3. Full Web Configurations

After Login EZLoop® IP-PBX Enterprise SIP Server will see screen as below, and there are four main categories, user can click on each category to extend detail items.



- Configuration: Include all telephony configuration of EZLoop® IP-PBX Enterprise SIP Server.
 - IP PBX
 - Extension
 - Trunk
 - SIP Trunk Reg.
 - Routing Table
 - Dial Group
 - Speed Dial
 - Others
- Information: To show related information.
 - Subscriber
- Management: Include all system management of EZLoop® IP-PBX Enterprise SIP Server.
 - Network
 - TimeZone
 - SMTP Setting
 - User Account
 - Firmware Upload
 - Music Upload
 - Import Setting
 - Export Setting
 - Flash Clean
- Reboot System: To reboot system of EZLoop® IP-PBX Enterprise SIP Server.

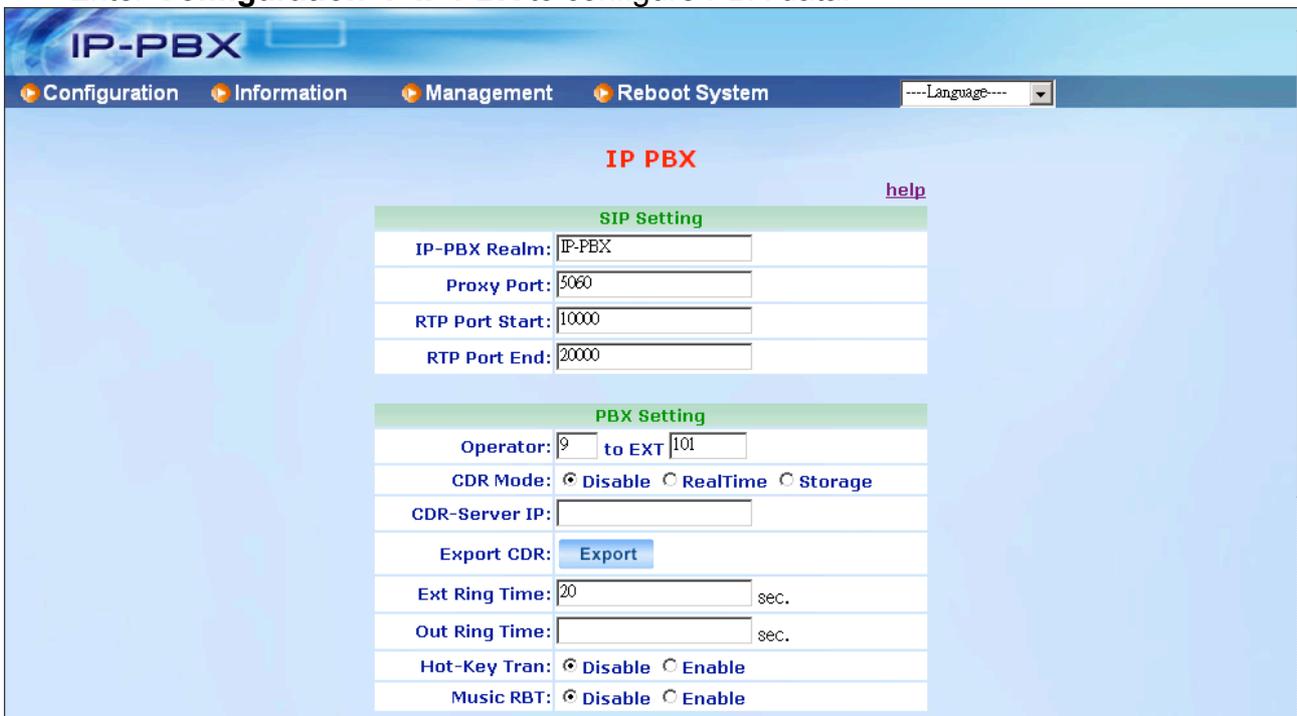
3.1 Configuration

User can set EZLoop® IP-PBX Enterprise SIP Server telephony configuration under Configuration category.



3.1.1 IP PBX

Enter **Configuration** → **IP PBX** to configure PBX data.



System Announcement	
Scheduled Greeting	
Office Hour AM:	09 : 00 - 11 : 59
Lunch Break:	12 : 00 - 12 : 59
Office Hour PM:	13 : 00 - 17 : 59
Day:	Monday - Friday
Fixed Greeting	
<input checked="" type="radio"/> Disable <input type="radio"/> Working Time <input type="radio"/> Non-working Time <input type="radio"/> Special Greeting	
Behind NAT	
Behind NAT:	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
External IP:	<input type="text"/>
External Host:	<input type="text"/>
Local Net:	<input type="text"/> / <input type="text"/>
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

SIP Setting

- **IP-PBX Realm:** Configure Realm of EZLoop® IP-PBX Enterprise SIP Server. This parameter is essential when there is more than one EZLoop® IP-PBX Enterprise SIP Server, and user wants to have inter-calls between EZLoop® IP-PBX Enterprise SIP Servers. Please refer to SIP Trunk configuration.
- **Proxy Port:** These optional parameters allow you to control the port on which you wish the EZLoop® IP-PBX Enterprise SIP Server to accept SIP connections. Default is 5060.
- **RTP Port Start:** The voice media will use RTP as the transport protocol. You can define the RTP port range that EZLoop® IP-PBX Enterprise SIP Server opened. Default start port is 10000.
- **RTP Port End:** The voice media will use RTP as the transport protocol. You can define the RTP port range that EZLoop® IP-PBX Enterprise SIP Server opened. Default end port is 20000.

Note:

- Default RTP port range is 10000 to 20000 and default proxy port is 5060. If your EZLoop® IP-PBX Enterprise SIP Server is behind a firewall, please make sure you have already open the RTP port (10000-20000) and proxy port (5060). And you should also make sure the proxy port (5060) has already mapped to EZLoop® IP-PBX Enterprise SIP Server.

PBX Setting

- **Operator:** Assign operator access code. When caller dial in EZLoop® IP-PBX Enterprise SIP Server, press this assigned code can reach operator.
- **to EXT:** Assign operator 's extension number. When caller press operator's access code, EZLoop® IP-PBX Enterprise SIP Server will transfer this call to the assigned Extension.
- **CDR Mode:** Chose the mode for CDR. You can disable the CDR or send the CDR record to a certain CDR server. You can also store the CDR records within EZLoop® IP-PBX Enterprise SIP Server.
 - **Disable:** Chose this one to Disable CDR function.
 - **RealTime:** You can install a CDR program to collect and store CDR records. The

CDR program is proprietary. For more information about such CDR program, please contact with your contact window of .

Note:

- If you already setup the CDR program in a PC. Please also input the PC's IP address to [CDR-Server IP] field. Every 5 seconds, EZLoop® IP-PBX Enterprise SIP Server will send a CDR record to CDR-Server. And CDR-Server will collect such records as a CSV file.

- **Storage:** If you do not prepare a PC as a CDR server. You can also define the CDR Mode to Storage. EZLoop® IP-PBX Enterprise SIP Server will store the C D R records within itself.

Note:

- If you chose the CDR Mode to Store, you can download the CDR file by pressing Export button of Export CDR field. When you export CDR files, EZLoop® IP-PBX Enterprise SIP Server will clean the CDR record from it. EZLoop® IP-PBX Enterprise SIP Server can only store 500 CDR records within itself. If you do not export the CDR file but the records is over than 500, the oldest one will be instead by newest CDR record.

- **CDR-Server IP:** If you choose the CDR Mode to RealTime, here you can input the IP address of CDR server which you installed the CDR program.
- **Export CDR:** If you chose the CDR Mode to Storage, you can press Export button to download the CDR file. The CDR file is within a CSV format.
- **Ext Ring Time:** This field defines the timeout value if the call is between Extension and Extension. Default is 20 seconds.
- **Out Ring Time:** This field defines the timeout value if the call is from Extension to outside (define by routing table). Default is no limitation.
- **Hot-Key Tran:** User can enable or disable Hot-key transfer function. If the call is establish between Extensions. Enable this feature will permits the calling party or called party to transfer a call by pressing the ***0 (For Blind Transfer) or *9 (For consultant Transfer)** key. Default is disabled.

Note:

- Please note that if this option is used, the RTP Mode will always be Routed Mode, as EZLoop® IP-PBX Enterprise SIP Server needs to monitor the call to detect when the caller presses the *0 or *9 key.

- **Music RBT:** Enabling this option will provide music to the calling party until the call is answered.

System Announcement

- **Scheduled Greeting:** You can define a business time for the company announcement.
 - **Office Hour AM:** Define the work time for AM. Default is 09:00-11:59. The sound file for greeting of work time is: greeting-day.gsm. You can change it by dialing to **111.
 - **Lunch Break:** Define the time for lunch break. Default is 12:00-12:59. The sound file for greeting of lunch break is: greeting-noon.gsm. You can change it by dialing to **112.
 - **Office Hour PM:** Define the work time for PM. Default is 13:00-17:59. The sound file for greeting of work time is: greeting-day.gsm. You can change it by dialing to **111.
 - **Day:** Define the work days. Default is Monday to Friday.

Note:

- If the time is a non-working time, the greeting files will be greeting-night.gsm. You can change it by dialing to **113.
- For more information about the greeting record, please refer to user manual: [4.1.3 How to record the other announcements](#)

- **Fixed Greeting:** You can enforce the company announcement as a fixed greeting.
 - **Disable:** If you disable the Fixed Greeting, the announcement will base on the above work time setting.
 - **Working Time:** Chose this one will enforce the company announcement as a greeting for work time: greeting-day.gsm.
 - **Non-working Time:** Chose this one will enforce the company announcement as a greeting for non-work time: greeting-night.gsm.
 - **Special Greeting:** Chose this one will enforce the company announcement as a special greeting: greeting-temporary.gsm. You can record it by dialing **115.

Behind NAT

- **Behind NAT:** If your EZLoop® IP-PBX Enterprise SIP Server is behind NAT, we strongly suggest u to enable Behind NAT to avoid some unexpected issue, such as “one way voice”.
- **External IP:** If you input External IP, EZLoop® IP-PBX Enterprise SIP Server will take that IP address as its argument. If EZLoop® IP-PBX Enterprise SIP Server is behind NAT, the SIP header will normally use the private IP address assigned to the server. The remote device will not know how to route back to this address; thus, it must be replaced with a valid, routable address.
- **External Host:** External Host takes a fully qualified domain name as its argument. If EZLoop® IP-PBX Enterprise SIP Server is behind NAT, the SIP header will normally use the private IP address assigned to the server. If you set this option, EZLoop® IP-PBX Enterprise SIP Server will perform periodic DNS lookups on the hostname and replace the private IP address with the IP address returned from the DNS lookup.

Note:

- You should not set both of External IP and External Host together; otherwise there will be some unexpected problems appeared. That means you can only chose one for External IP or External Host for “Behind NAT”
- **Local Net:** Local Net is used to tell EZLoop® IP-PBX Enterprise SIP Server which IP addresses are considered local. If one of caller or callee is not under Local Net, EZLoop® IP-PBX Enterprise SIP Server will set the address in the SIP header that can be translated to that specified by External IP or the IP address can be looked up with External Host. The format will be **IP/ Subnet Mask**. Example: **192.168.1.0/255.255.255.0**

3.1.2 Extension

User has to set Extension account for other device to register on EZLoop® IP-PBX Enterprise SIP Server.

Enter **Configuration** → **Extension** to configure Extension data. On screen will show 100 sets Extension. User can press **Modify** to add new Extension or modify configured Extension data. Press **Delete** will delete the specified Extension.

Extension

[help](#)

Index	Extension Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	101	rfc2833	Disable	Routed Mode	Modify	Delete
2	102	rfc2833	Disable	Routed Mode	Modify	Delete
3	103	rfc2833	Disable	Routed Mode	Modify	Delete
4	104	rfc2833	Disable	Routed Mode	Modify	Delete
5	105	rfc2833	Disable	Routed Mode	Modify	Delete
6	106	rfc2833	Disable	Routed Mode	Modify	Delete
7	107	rfc2833	Disable	Routed Mode	Modify	Delete
8	108	rfc2833	Disable	Routed Mode	Modify	Delete
9	109	rfc2833	Disable	Routed Mode	Modify	Delete
10	110	rfc2833	Disable	Routed Mode	Modify	Delete
11	none	rfc2833	Disable	Routed Mode	Modify	Delete
12	none	rfc2833	Disable	Routed Mode	Modify	Delete
13	none	rfc2833	Disable	Routed Mode	Modify	Delete
14	none	rfc2833	Disable	Routed Mode	Modifv	Delete

After press Modify can input detail setting for Extension.

The screenshot shows the 'IP-PBX' web interface with a navigation bar containing 'Configuration', 'Information', 'Management', and 'Reboot System'. A 'Language' dropdown menu is visible on the right. The main content area displays the 'Extension Setting' form with the following fields and values:

Extension Setting	
Extension Number:	101
Password:	***
Call Group:	1
Pickup Group:	1
DialPlan:	ext+allroutes
Keypad:	RFC2833
NAT Traversal:	Disable
RTP Mode:	Routed Mode
MailBox:	Disable

At the bottom of the form are two buttons: 'Apply' (green) and 'Cancel' (blue).

- **Extension Number:** Assign the number of Extension. This number is also the register name for device.
- **Password:** Assign the register password for device to register on EZLoop® IP-PBX Enterprise SIP Server.
- **Call Group:** You can use the Call Group parameter to assign an Extension to one or more groups.
- **Pickup Group:** You can use the Pickup Group option in conjunction with this parameter to allow a ringing phone to be answered from another extension.

Note:

- The Pickup Group option is used to control which Call Groups a channel may pick up—a channel is given authority to answer another ringing channel if it is assigned to the same Pickup Group as the ringing channel's Call Group. By default, remote ringing extensions can be answered with *8.
- You can define multiple Call Groups and Pickup Groups for one Extension by a "comma". For example, you can input "1,3,5" into Call Group or Pickup Group.

- **DialPlan:** Define the dialing plan for Extension. It specifies the location of the instruction used to control what the phone is allowed to do, and what to do with incoming calls for this extension. In this field, you can Choose 5 dial level for Extension, including [ext-only], [ext+R1], [ext+R12], [ext+R123], [ext+allroutes]. You can define an "Outgoing call" record, to a certain Route Level, as R1, R2..., etc. [ext-only] means this subscriber can only call to Extension. [ext+R1] means the subscriber with such DialPlan can call to Extension and Route Level with R1. [ext+R12] means the subscriber with such DialPlan can call to Extension and Route Level with R1 and R2. [ext+R123] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2 and R3. [ext+allroutes] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2, R3 and R4.

Note:

- For more information about Route Level, please refer to the user manual: [3.1.5.1 Outgoing Call Rule](#).

- **Keypad:** User can select Keypad type to be RFC2833, In-band, or SIP-Info. The setting should be also match the Keypad setting of Extension device.
- **NAT Traversal:** If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to EZLoop® IP-PBX Enterprise SIP Server on public network, then user has to enable this function. Enable NAT Traversal to force EZLoop® IP-PBX Enterprise SIP Server to ignore the contact information for the Extension and use the address from which the packets are being received.
- **RTP Mode:** User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode. The voice media will be routed “Peer-to-Peer” if two clients are both setting to Direct Mode. This way will improve the voice quality and reduce the performance wastage of the EZLoop® IP-PBX Enterprise SIP Server.

Note:

- If one peer set to Direct Mode but another peer set to Routed Mode, the result will become to Routed Mode.
- Voice media will still go through the EZLoop® IP-PBX Enterprise SIP Server if the EZLoop® IP-PBX Enterprise SIP Server needs to detect DTMF.
- If you enable the NAT Traversal function for Extension, the RTP mode will change to Routed Mode directly; this way will avoid the “one-way voice” or “no voice issue” of VoIP.
- If the both peers are under different subnet, or one peer is under Public IP but another one is under Private IP, **we strongly suggest you to set the RTP mode to Routed Mode to avoid some unexpected voice problems.**

- **Mail Box:** User can select to disable or enable mail box function. If this function is enabled, user has to input e-mail address for the Extension. When having voice mail of incoming call, system will send this voice mail to the specified e-mail address.

Note:

- Please remember set the SMTP in the page of Management → SMTP Setting to activate the Voice Mail to E-mail.
- If the EZLoop® IP-PBX Enterprise SIP Server got a new message, it will send the message to the user by email immediately.

Press Apply to save configuration, or press Cancel to quit configuration.

3.1.3 Trunk

User has to set Trunk account for Trunk (FXO device) to register to EZLoop® IP-PBX Enterprise SIP Server or set some necessary configuration for SIP trunk (For more application, please go to.....). Enter **Configuration** → **Trunk** to configure Trunk data.

On screen will show **20** sets Trunks. User can press **Modify** to add new Trunk or modify configured Trunk data. Press **Delete** will delete the specified Trunk.

IP-PBX

Configuration Information Management Reboot System Language

Trunk

[help](#)

Index	Trunk Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	888	rfc2833	Disable	Routed Mode	Modify	Delete
2	889	rfc2833	Disable	Routed Mode	Modify	Delete
3	070070	rfc2833	Disable	Routed Mode	Modify	Delete
4	none	none	Disable	Routed Mode	Modify	Delete
5	none	none	Disable	Routed Mode	Modify	Delete
6	none	none	Disable	Routed Mode	Modify	Delete
7	none	none	Disable	Routed Mode	Modify	Delete
8	none	none	Disable	Routed Mode	Modify	Delete
9	none	none	Disable	Routed Mode	Modify	Delete
10	none	none	Disable	Routed Mode	Modify	Delete
11	none	none	Disable	Routed Mode	Modify	Delete
12	none	none	Disable	Routed Mode	Modify	Delete
13	none	none	Disable	Routed Mode	Modify	Delete
14	none	none	Disable	Routed Mode	Modify	Delete

After press Modify can input detail setting for Trunk.

Example 1: Set Trunk for FXO gateway

IP-PBX

Configuration Information Management Reboot System Language

Trunk Setting

Trunk Number:

Password:

Host:

DialPlan:

Keypad:

NAT Traversal:

RTP Mode:

Port:

External Server Address:

Maximum Channels:

Outbound Caller ID:

Hot-Key Tran: Disable Enable

Music RBT: Disable Enable

Example 2: Set Trunk ID for SIP Trunk

IP-PBX

Configuration Information Management Reboot System Language

Trunk Setting

Trunk Number: 070070

Password: ***

Host: Pre-define

Address: 218.32.223.140

DialPlan: from-pstn

Keypad: RFC2833

NAT Traversal: Disable

RTP Mode: Routed Mode

Port: 5060

External Server Address: 218.32.223.140

Maximum Channels: 5

Outbound Caller ID: 070070

Hot-Key Tran: Disable Enable

Music RBT: Disable Enable

Apply Cancel

- **Trunk Number:** Assign the number of Trunk. This number is also the register name for Trunk device.

Note:

- The Trunk Number can also be a “Trunk ID”. In the Routing Table page, you should define the destination of prefix route. When you define the prefix route, you should set the Trunk ID (Trunk Number) in the Trunk page first; then you could input the correct Trunk ID in the Destination field.

- **Password:** Assign the register password for device to register on EZLoop® IP-PBX Enterprise SIP Server.
- **Host:** Setting the Host to Dynamic will require the trunk to register the EZLoop® IP-PBX Enterprise SIP Server so that the EZLoop® IP-PBX Enterprise SIP Server knows how to reach the trunk. You can also set the Host to an IP address or FQDN if you set the Host to [Pre-define]. There will be a field called [Address] appear when you choose Host to [Pre-define]. This limits only where you place calls to, as the user is allowed to place calls from anywhere.
- **Address:** If user set Host type as Pre-define, here need to input the Address of server where Trunk register to.
- **DialPlan:** Define the dialing plan for Trunk. It specifies the location of the instruction used to control what the phone is allowed to do, and what to do with incoming calls for this extension. In this field, you can Choose 6 dial level for Extension, including [from-pstn], [ext-only], [ext+R1], [ext+R12], [ext+R123], [ext+allroutes]. You can define an “Outgoing call” record, to a certain route level, as R1, R2..., etc. [from-pstn] is used for Trunk only. [ext-only] means this subscriber can only call to Extension. [ext+R1] means the subscriber with such DialPlan can call to Extension and Route Level with R1. [ext+R12] means the subscriber with such DialPlan can call to Extension and Route Level with R1 and R2. [ext+R123] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2 and R3. [ext+allroutes] means the subscriber with such DialPlan can call to Extension and Route Level with R1, R2, R3

and R4.

Note:

- For more information about Route Level, please refer to the user manual: [3.1.5.1 Outgoing Call Rule](#).

- **Keypad:** User can select DTMF type to be RFC2833, In-band, or SIP-Info.
- **NAT Traversal:** If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to EZLoop® IP-PBX Enterprise SIP Server on public network, then user has to enable this function. Enable NAT Traversal to force EZLoop® IP-PBX Enterprise SIP Server to ignore the contact information for the Trunk and use the address from which the packets are being received.
- **RTP Mode:** User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode. The voice media will be routed “Peer-to-Peer” if two clients are both setting to Direct Mode. This way will improve the voice quality and reduce the performance wastage of the EZLoop® IP-PBX Enterprise SIP Server.

Note:

- If one peer set to Direct Mode but another peer set to Routed Mode, the result will become to Routed Mode.
 - Voice media will still go through the EZLoop® IP-PBX Enterprise SIP Server if the EZLoop® IP-PBX Enterprise SIP Server needs to detect DTMF.
 - If you enable the NAT Traversal function for Extension, the RTP mode will change to Routed Mode directly; this way will avoid the “one-way voice” or “no voice issue” of VoIP.
 - If the both peers are under different subnet, or one peer is under Public IP but another one is under Private IP, **we strongly suggest you to set the RTP mode to Routed Mode to avoid some unexpected voice problems.**
- **Port:** You can use this to define the SIP signal port if you want to listen on a nonstandard SIP signal port.
 - **External Server Address:** This field will allow you to set the domain in the SIP From URI. Setting this will avoid some unexpected issue if the service provider needs this for authentication.
 - **Maximum Channels:** This will limit the maximum channels for this Trunk. For example, you set 2 into this field; only 2 outgoing calls could go via this Trunk. Default is no limit.
 - **Outbound Caller ID:** Some service provider will require the correct registered caller ID if it got an incoming call. Default the EZLoop® IP-PBX Enterprise SIP Server will send the Extension’s caller ID to this Trunk, if you set empty here.

Note:

- Normally, SIP From URI will contain the Extension’s calling ID and EZLoop® IP-PBX Enterprise SIP Server’s IP address, but some ITSP may reject this call due to some security issue. You can modify the Calling ID and IP/ Domain in the fields of [External Server Address] and [Outbound Caller ID] when the call is going via the EZLoop® IP-PBX Enterprise SIP Server to the Destination (Trunk) to avoid such security issue.
- If you set a FXO gateway as the Trunk, you can just use the default Trunk 888 and 889 as the FXO’s register number.
- For the FXO gateway, you may just only configure Trunk Number, Password, Host, DialPlan, Keypad, NAT Traversal and RTP Mode.
- If you set the ITSP as the Trunk, you may need to set the following configure: Port, External Server Address and Outbound Caller ID.
- **For more information, please refer to the CH5. Appendix-Application**

between CPE device and EZLoop® IP-PBX Enterprise SIP Server.

- **Hot-Key Tran:** Enable this feature will permits the calling party or called party to transfer a call by pressing the ***0 (For Blind Transfer)** or ***9 (For consultant Transfer)** key if the call is Between Extension and Trunk. Default is disabled.

Note:

- If you enable this feature in Trunk page, we suggest you also enable Hot-Key Tran of IP PBX page.
- Please note that if this option is used, the RTP Mode will always be Routed Mode, as EZLoop® IP-PBX Enterprise SIP Server needs to monitor the call to detect when the caller presses the *0 or *9 key.

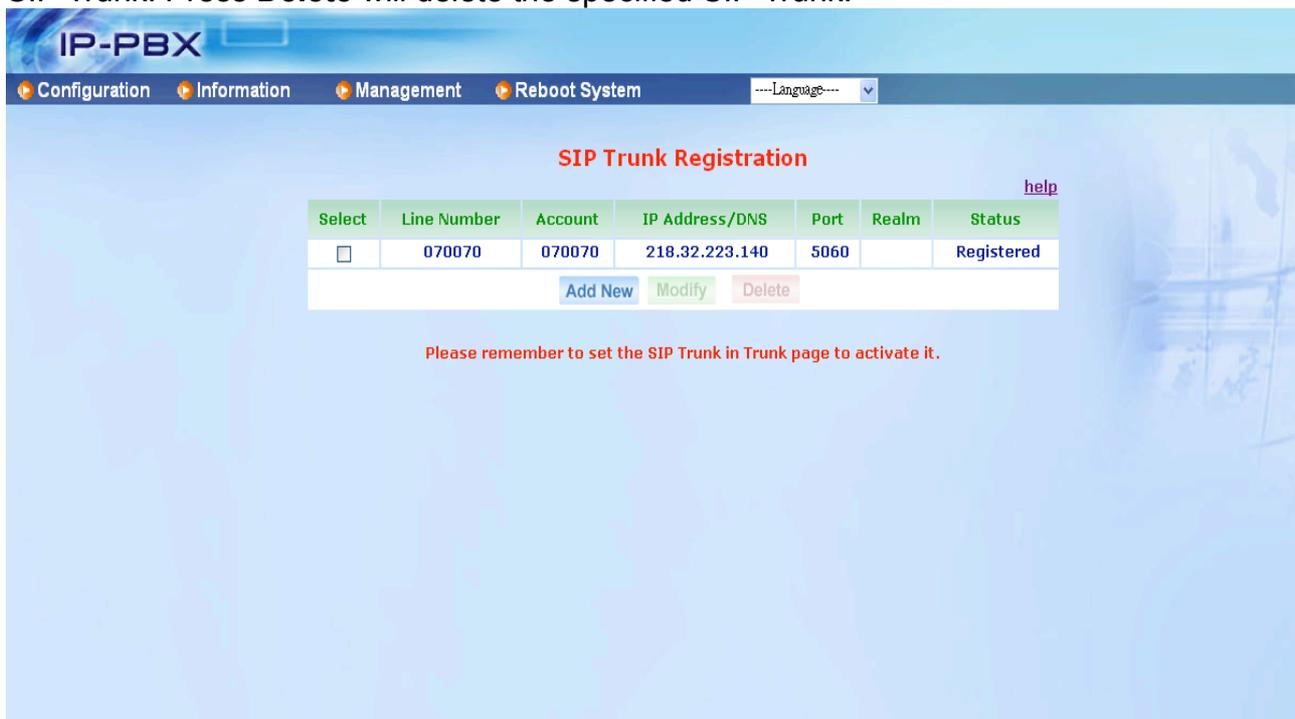
- **Music RBT:** Provides music to the calling party until the call is answered

Press Apply to save configuration, or press Cancel to quit configuration.

3.1.4 SIP Trunk Reg.

SIP Trunk is for EZLoop® IP-PBX Enterprise SIP Server **to register to other systems only**, such as ITSP or another EZLoop® IP-PBX Enterprise SIP Server.

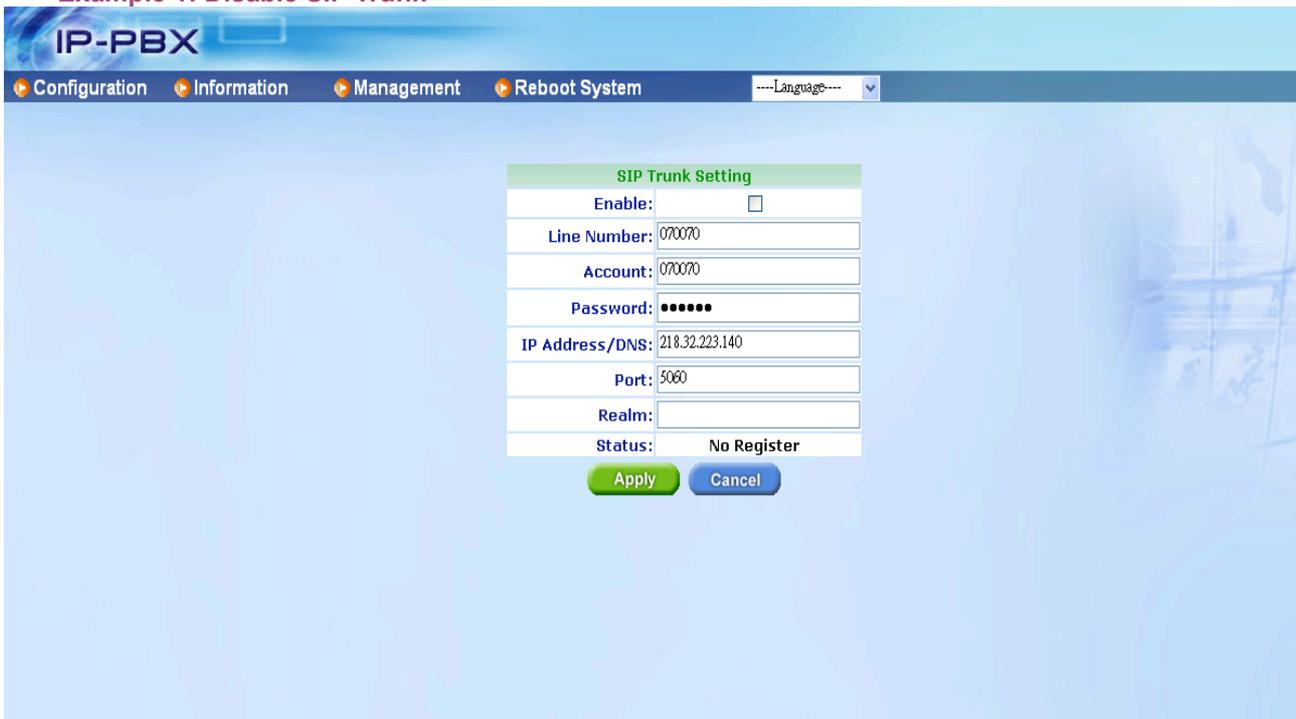
On screen of SIP Trunk will show all of the sets of SIP Trunks. You will find out the registered Account and registered server IP address, port number, Realm and the Register Status. User can press **Add New** to add new Trunk or **Modify** to configure the specified SIP Trunk. Press **Delete** will delete the specified SIP Trunk.



Select	Line Number	Account	IP Address/DNS	Port	Realm	Status
<input type="checkbox"/>	070070	070070	218.32.223.140	5060		Registered

Enter **Configuration**→**SIP Trunk-Add New** to configure EZLoop® IP-PBX Enterprise SIP Server register to ITSP. ITSP will provide related account information for EZLoop® IP-PBX Enterprise SIP Server to register. Please input the data here.

Example 1: Disable SIP Trunk

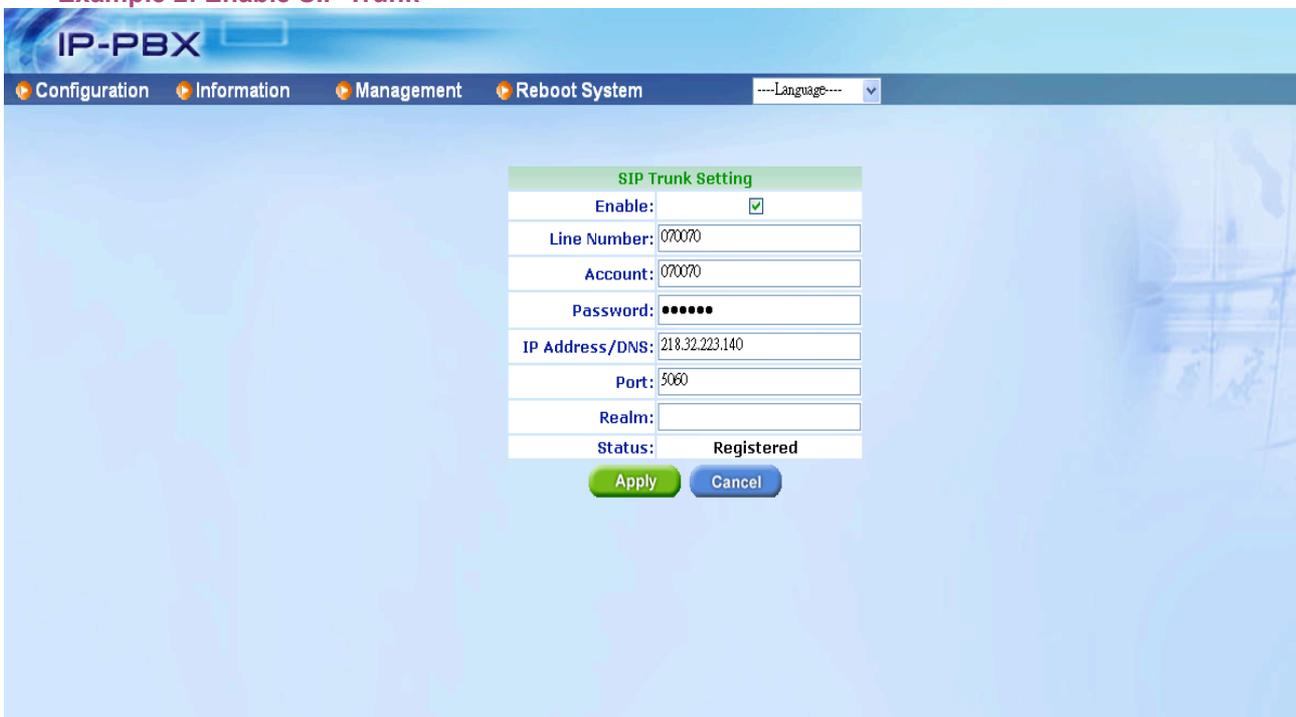


The screenshot shows the IP-PBX web interface. At the top, there is a navigation bar with 'Configuration', 'Information', 'Management', and 'Reboot System' buttons, and a language dropdown menu. The main content area displays the 'SIP Trunk Setting' form. The 'Enable' checkbox is unchecked. The other fields are: Line Number: 070070, Account: 070070, Password: masked with dots, IP Address/DNS: 218.32.223.140, Port: 5060, Realm: empty, and Status: No Register. There are 'Apply' and 'Cancel' buttons at the bottom of the form.

SIP Trunk Setting	
Enable:	<input type="checkbox"/>
Line Number:	070070
Account:	070070
Password:	••••••
IP Address/DNS:	218.32.223.140
Port:	5060
Realm:	
Status:	No Register

Apply Cancel

Example 2: Enable SIP Trunk



The screenshot shows the IP-PBX web interface. At the top, there is a navigation bar with 'Configuration', 'Information', 'Management', and 'Reboot System' buttons, and a language dropdown menu. The main content area displays the 'SIP Trunk Setting' form. The 'Enable' checkbox is checked. The other fields are: Line Number: 070070, Account: 070070, Password: masked with dots, IP Address/DNS: 218.32.223.140, Port: 5060, Realm: empty, and Status: Registered. There are 'Apply' and 'Cancel' buttons at the bottom of the form.

SIP Trunk Setting	
Enable:	<input checked="" type="checkbox"/>
Line Number:	070070
Account:	070070
Password:	••••••
IP Address/DNS:	218.32.223.140
Port:	5060
Realm:	
Status:	Registered

Apply Cancel

- **Enable:** Check to enable this SIP Trunk.
- **Line Number:** Line Number for registering to ITSP.
- **Account:** Account Name/ ID for registering to ITSP.
- **Password:** Account Password for registering to ITSP.
- **IP Address/DNS:** Enter IP or domain name of ITSP server.
- **Port:** Port number of ITSP server for registering.

- **Realm:** Realm of ITSP or another EZLoop® IP-PBX Enterprise SIP Server.

Note:

- When a call was sent form EZLoop® IP-PBX Enterprise SIP Server to a remote SIP-Trunk, the SIP-Trunk may attempt to authenticate the “call”. So EZLoop® IP-PBX Enterprise SIP Server should reply the correct Account ID and Password. How does the EZLoop® IP-PBX Enterprise SIP Server know which ID and Password it should send? When the call is going to SIP-Trunk via EZLoop® IP-PBX Enterprise SIP Server, the SIP-Trunk may reply a 407 code, which will contain a parameter called “Realm” for authentication, EZLoop® IP-PBX Enterprise SIP Server will re-send the call again and contains correct ID and Password based on the Realm. So the Realm should be unique. For more information about Realm, please contact with your ITSP.
- If you have multiple EZLoop® IP-PBX Enterprise SIP Server, you may hope those EZLoop® IP-PBX Enterprise SIP Server could call to each other. You should set the Extension to let those EZLoop® IP-PBX Enterprise SIP Server can register to each other, and you should also confirm the [Realm] in the page of **Configuration → IP PBX**. For more information, please refer to the user manual: **CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.**

- **Status:** Once SIP Trunk is configured and enabled, here will show the registration status.

Press Apply to save configuration, or press Cancel to quit configuration.

3.1.5 Routing Table

Routing Table is to set routing rule of EZLoop® IP-PBX Enterprise SIP Server. There are two directions to set rules: Outgoing Call Rule means from subscriber (Extension or Trunk registered on EZLoop® IP-PBX Enterprise SIP Server) to call out. Incoming Call Rule means call from other non-subscriber device to EZLoop® IP-PBX Enterprise SIP Server. Enter **Configuration → Routing Table-select direction and press Add New** to set routing table.

Outgoing Call Rule

Select	Prefix	Digits Length	Primary Dest.	Secondary Dest.	Add	Drop	Guest Allow
<input type="checkbox"/>	03	0	070070	889		2	Disable
			888				
<input type="checkbox"/>	2	8	888	070070	02		Enable

Incoming Call Rule

Select	Prefix	Digits Length	Add	Drop
<input type="checkbox"/>	070070	6	**999	6

3.1.5.1 Outgoing Call Rule

Outgoing Call Rule means from subscriber (Extension or Trunk registered on EZLoop® IP-PBX Enterprise SIP Server) to call out.

Example 1: Routing record with prefix 03 and no limit for Digits Length. Enable Route Password and Drop function.

IP-PBX

Configuration Information Management Reboot System ---Language---

Prefix: 03
 Digits Length: 0 Max Length:20

	Primary	Secondary	Third
Destination:	070070	889	888
Add:			
Drop:		2	
Route Password:	****	****	****
Guest Allow:	<input type="checkbox"/>		
Route Level:	<input type="checkbox"/> R1 <input type="checkbox"/> R2 <input type="checkbox"/> R3 <input checked="" type="checkbox"/> R4		

Apply Cancel

Example 2: Routing record with prefix 2 and Digits Length is 8. Enable Route Guest Allow and Add function.

IP-PBX

Configuration Information Management Reboot System ---Language---

Prefix: 2
 Digits Length: 8 Max Length:20

	Primary	Secondary	Third
Destination:	888	070070	none
Add:		02	
Drop:			
Route Password:			
Guest Allow:	<input checked="" type="checkbox"/>		
Route Level:	<input type="checkbox"/> R1 <input type="checkbox"/> R2 <input type="checkbox"/> R3 <input checked="" type="checkbox"/> R4		

Apply Cancel

- **Prefix:** Set prefix number for routing rule.
- **Digits Length:** Set the digit length of dialed number, if user doesn't want to limit the length, please set this parameter as 0. The maximum length is 20.

Note:

- If you set the Digits Length as a specific value, such as 10, the dialed number digits length should full match to 10, or you can set the Digits Length to 0 to ignore the digits length.

■ **Primary/Secondary/Third:** User can set three priorities for each routing rule, if EZLoop® IP-PBX Enterprise SIP Server fails to route to primary destination three times, it will try to route to secondary or third destination.

■ **Destination:** Here you can find some destination (Trunk) for choosing. You can define the destination for the prefix route.

Note:

- Before setting the Routing Table, you should set the Trunk info in the Trunk page first. So that this field will contain the Trunk ID for choosing.
- If the Trunk was setting to Dynamic in the Host field, but it doesn't register on EZLoop® IP-PBX Enterprise SIP Server, EZLoop® IP-PBX Enterprise SIP Server will skip this "priority" and route call to next priority immediately without trying. If the Trunk was setting to Address in the Host field, but the Address is not reachable, EZLoop® IP-PBX Enterprise SIP Server will try three times then route call to next priority.

■ **Add:** To add assigned number. For example, you set 02 here and the called number is 03123, the EZLoop® IP-PBX Enterprise SIP Server will add 02 then send 0203123 as the called number.

■ **Drop:** To drop **specified length of number**. For example, you set 2 here and the called number is 03123, the EZLoop® IP-PBX Enterprise SIP Server will drop 03 then send 123 as outgoing number.

Note:

- If you set both of the Drop and Add, EZLoop® IP-PBX Enterprise SIP Server will Drop first then Add.

Example:

If user set prefix as 002, digits length as 12, Primary destination as 888, Drop as 3, and Add as 0.

When caller called 002912345678, the prefix is 002; length is 12, so this call matches the routing rule.

002912345678_912345678(Drop 3 digits)_0912345678(Add 0)

Finally, EZLoop® IP-PBX Enterprise SIP Server will send 0912345678 to Trunk ID 888.

■ **Route Password:** Set password here so the EZLoop® IP-PBX Enterprise SIP Server will request password before sending the call to Trunk.

■ **Guest Allow:** Enable Guest Allow will allow user who is not your subscriber (Extension) to use such routing record. User can reach the Auto attendant (The default Auto Attendant of EZLoop® IP-PBX Enterprise SIP Server is **999) first then send call to Destination (Trunk). If you enable Guest Allow. If you disable Guest Allow, only the Extension can use this Routing record.

Note:

- For more information, please refer to the CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.

■ **Route Level:** You can define the Route Level for such route record. For example, you define a route record with prefix 0 and the Route Level to R3. That means only the subscriber with DialPlan [ext+R123] and [ext+allroutes] can use such record due to only these two DialPlan contain the Route Level of R3.

Press Apply to save configuration, or press Cancel to quit configuration.

3.1.5.2 Incoming Call Rule

Incoming Call Rule means call from other non-subscriber device to EZLoop® IP-PBX Enterprise SIP Server.

For example, you set the EZLoop® IP-PBX Enterprise SIP Server to register an ISTP as a SIP Trunk, so your EZLoop® IP-PBX Enterprise SIP Server could be as an “Extension” of ITSP. The other subscriber of ITSP could call to EZLoop® IP-PBX Enterprise SIP Server by the registered line number, when the EZLoop® IP-PBX Enterprise SIP Server got an incoming call, which is not its own subscriber, what will the EZLoop® IP-PBX Enterprise SIP Server do? The EZLoop® IP-PBX Enterprise SIP Server will perform the following example based on the “Incoming Call Rule”.

The following example means the EZLoop® IP-PBX Enterprise SIP Server got a called number 070070, which was sending from a non-subscriber of EZLoop® IP-PBX Enterprise SIP Server, EZLoop® IP-PBX Enterprise SIP Server will drop 6 digits then add **999 as the destination number. **999 is the default number of auto attendant. So the caller will hear greeting because the called number will be routed to auto attendant.

Incoming Call Rule Setting	
Prefix:	070070
Digits Length:	6
Add:	**999
Drop:	6

Apply Cancel

- **Prefix:** Set prefix number for routing rule.
- **Digits Length:** Set the length of dialed number, if user doesn't want to limit the length, please set this parameter as 0. The maximum length is 20.

Note:

- **If you set the Digits Length as a specific value, such as 10, the dialed number should full match to 10**, or you can set the Digits Length to 0 to ignore the digits length.
 - If the called number from another non-subscriber is equal to Prefix, you should set the Digits Length as a specific value.
 - If the called number is not equal to Prefix, you can set the Digits Length as a specific value or 0 to ignore Digits Length.
- **Add:** To add assigned number. For example, you set **999 here and you do not set

Drop. If the called number is 070070101, the EZLoop® IP-PBX Enterprise SIP Server will add **999 then send **999070070101 as the called number.

- **Drop:** To drop **specified length of number**. For example, you set 6 here and you do not set Add. If the called number is 070070101, the EZLoop® IP-PBX Enterprise SIP Server will drop 070070 then send 101 as called number.

Note:

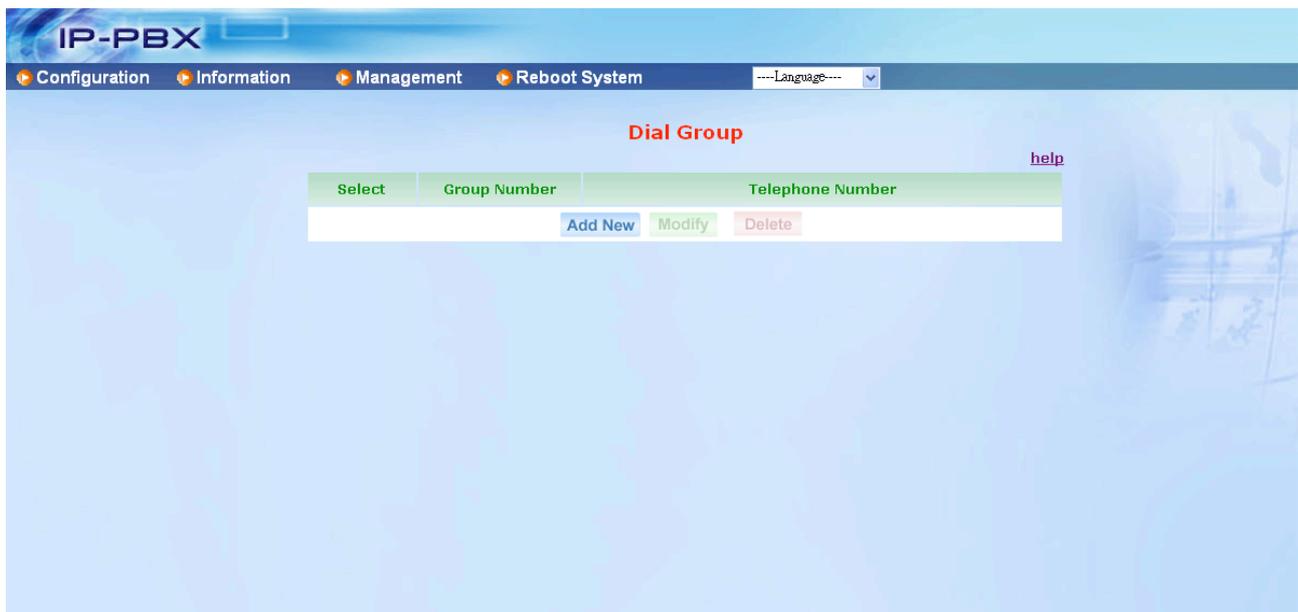
- If you set both of the Drop and Add, EZLoop® IP-PBX Enterprise SIP Server will Drop first then Add. For example, the EZLoop® IP-PBX Enterprise SIP Server received a number 070070, which was sending from a non-subscriber of EZLoop® IP-PBX Enterprise SIP Server, EZLoop® IP-PBX Enterprise SIP Server will drop 6 digits then add **999 as the destination number. **999 is the default number of auto attendant. So the caller will hear greeting because the called number will be routed to auto attendant.

Press Apply to save configuration, or press Cancel to quit configuration.

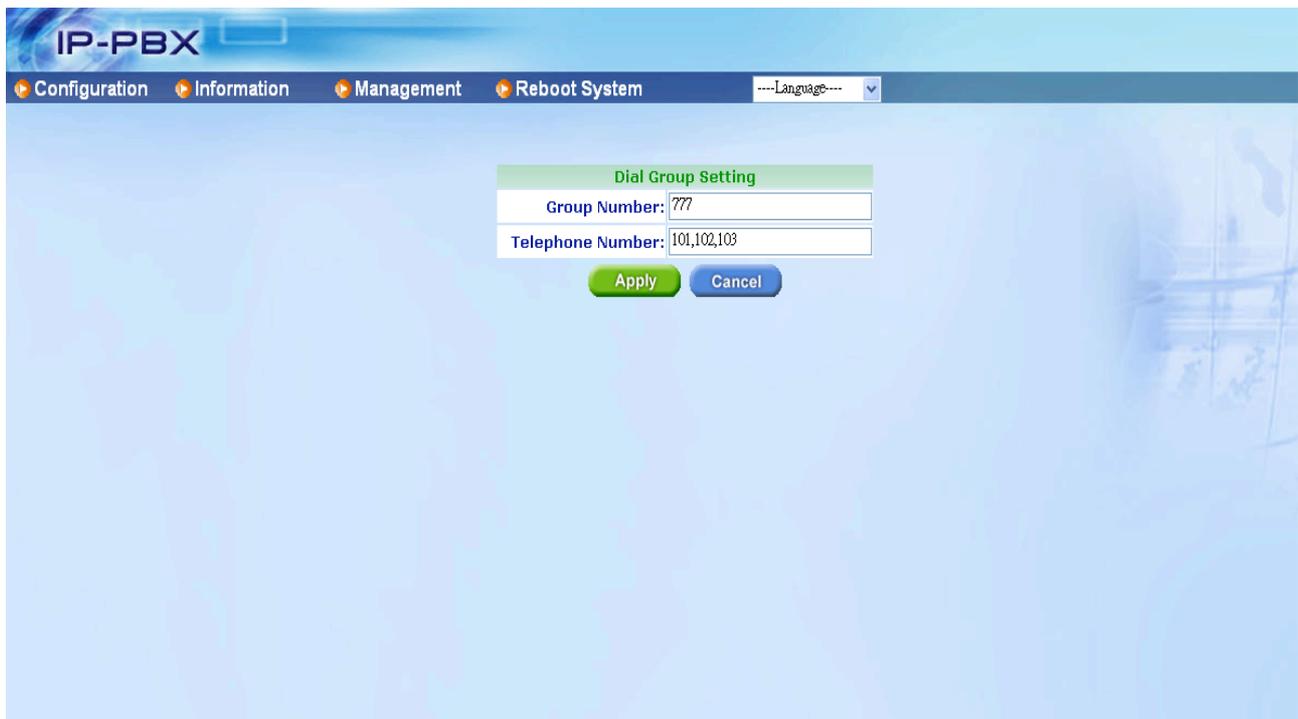
3.1.6 Dial Group

Dial Group is used to set the [group dialing], you can just input a group number and set specific members to join the group. Caller can dial the group number and **all of the members will ring**.

Enter **Configuration**→ **Dial Group** and press **Add New** to set Dial Group. Or Modify to configure the specified Dial Group. Press Delete will delete the specified Dial Group.



Enter **Configuration**→ **Dial Group** and press **Add New**→ **Press Apply** to configure Dial Group.



- **Group Number:** Input a specific Group Number here.
Note:
 - The Group number should NOT be the same with Extension, Trunk Number, and Speed Dial Number. All of the Numbers should be unique for EZLoop® IP-PBX Enterprise SIP Server system.
- **Telephone Number:** Set which members to join this Dial Group. You can input multiple members by a “comma”, such as [101,102,103]

3.1.7 Speed Dial

SpeedDial Table is used to set the Speed Dial function; you can just input a SpeedDial Number and set the destination number to Telephone Number field. Subscriber can just dial to the SpeedDial number and EZLoop® IP-PBX Enterprise SIP Server will switch the call to Telephone Number then call out.

Enter **Configuration** → **Speed Dial** to add Speed Dial record then press **Apply**.

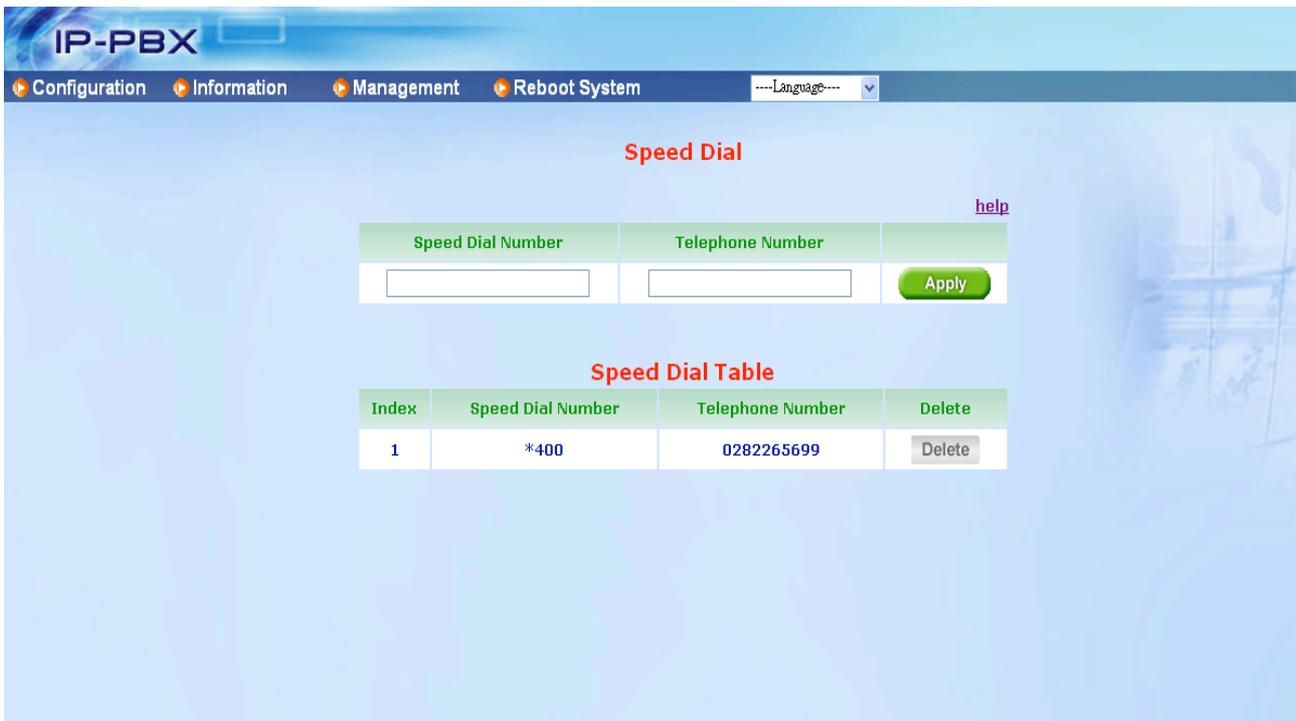
Speed Dial

[help](#)

Speed Dial Number	Telephone Number	
<input type="text" value="*400"/>	<input type="text" value="0082265699"/>	<input type="button" value="Apply"/>

Speed Dial Table

Index	Speed Dial Number	Telephone Number	Delete
-------	-------------------	------------------	--------



■ **SpeedDial Number:** You can input a specified SpeedDial Number here.

Note:

- The SpeedDial Number should not be the same with Extension, Trunk Number, and Group Number. All of the Numbers should be unique for EZLoop® IP-PBX Enterprise SIP Server system.
- If you want to modify a current SpeedDial record, you should delete the current record first and add another new one.

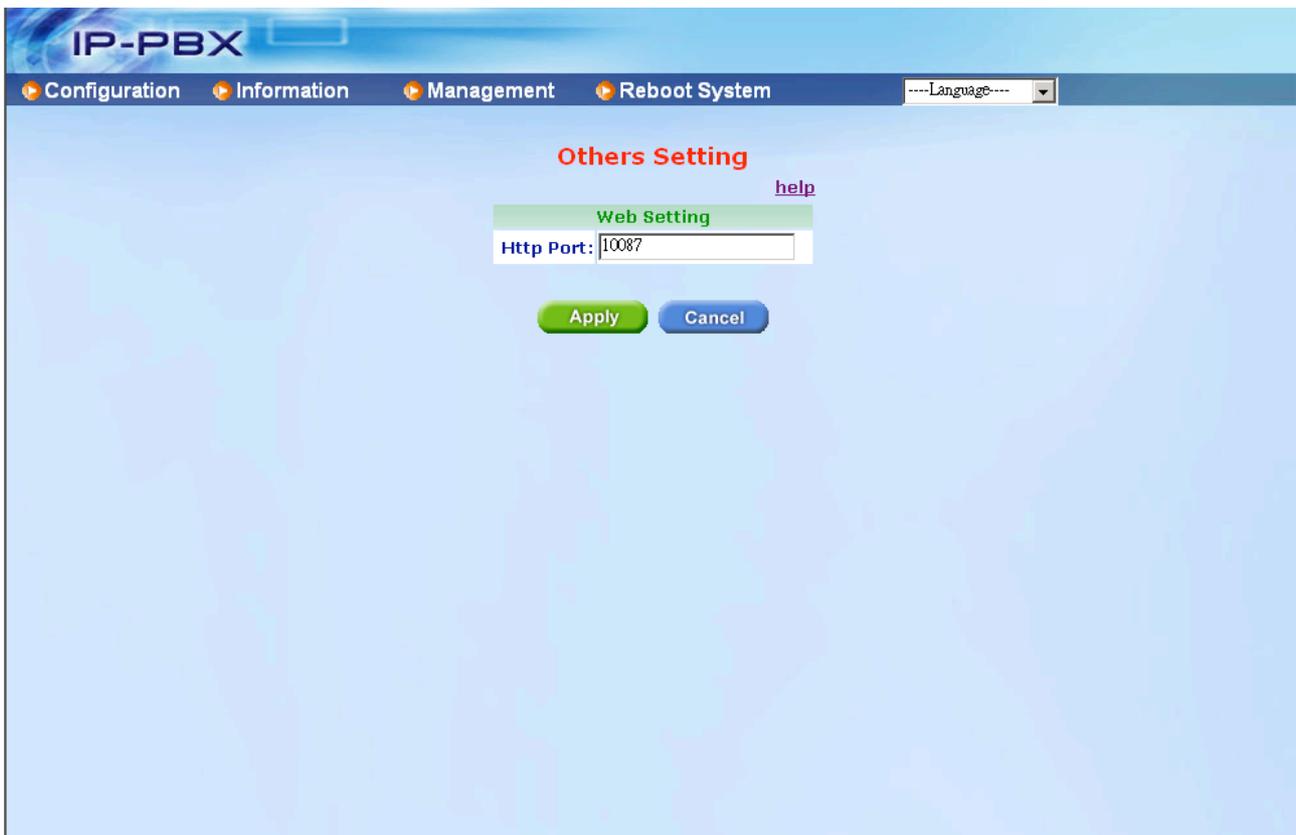
Telephone Number: Set the destination number for the Speed Dial.

Example:

User set SpeedDial number as *400 and Telephone Number as 0282265699. When caller called *400, EZLoop® IP-PBX Enterprise SIP Server will call to 0282265699 as destination number.

3.1.8 Other Setting

Enter **Configuration** → **Other Setting** to configure the other setting.



3.1.8.1 Web Setting

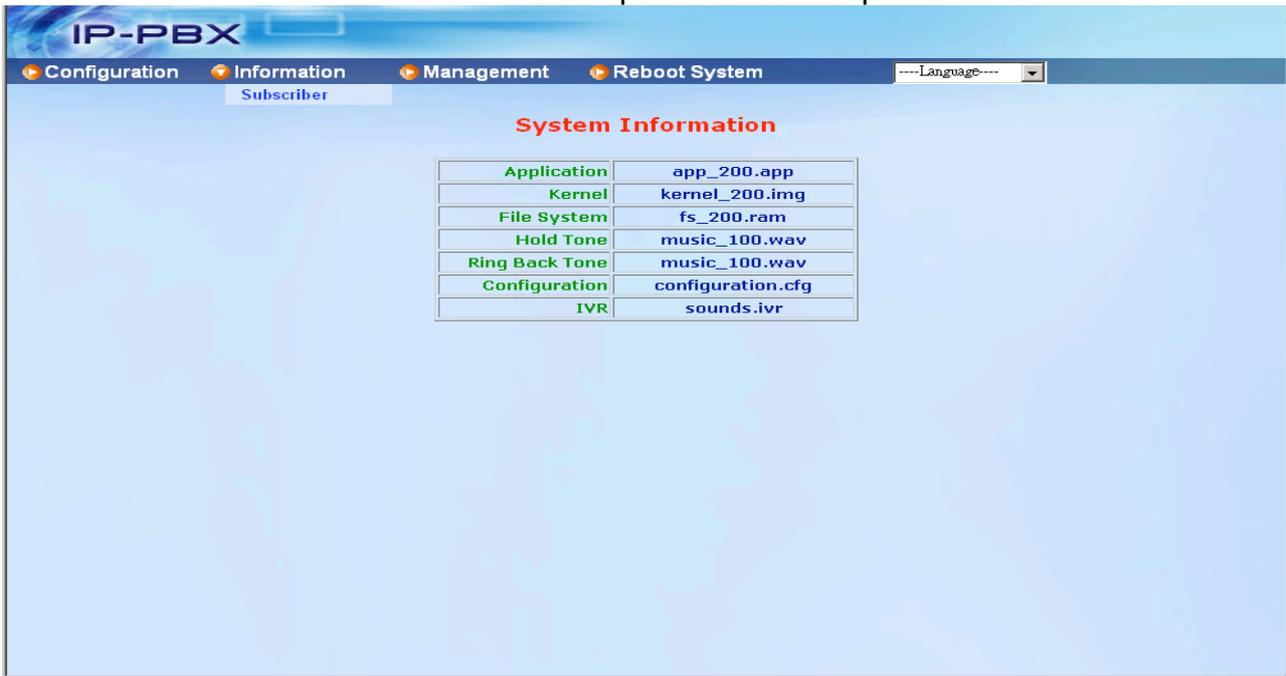
- **Http Port:** You can change the Http port for EZLoop® IP-PBX Enterprise SIP Server. Default is 10087.

Note:

- After Changing the Http port, you should reboot your EZLoop® IP-PBX Enterprise SIP Server manually to activate Http Port setting.

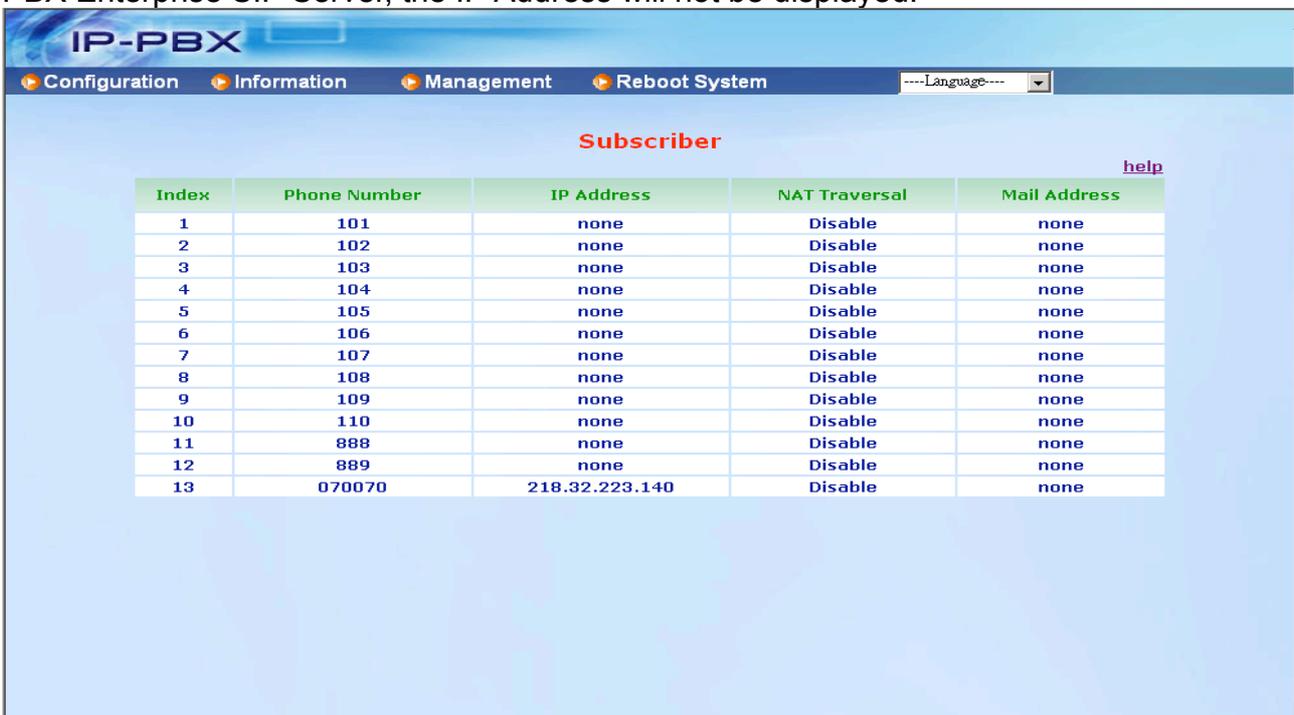
3.2 Information

User can check some information of EZLoop® IP-PBX Enterprise SIP Server here.



3.2.1 Subscriber

Enter **Information** → **Subscriber** to check information of Subscribers. User checks Phone Number, IP Address, Transversal, and Mail Address for Extension and Trunk here. If subscriber registered on EZLoop® IP-PBX Enterprise SIP Server, the IP Address will show up, on the other hand, if the subscriber doesn't register successfully on EZLoop® IP-PBX Enterprise SIP Server, the IP Address will not be displayed.



3.3 Management

User can execute EZLoop® IP-PBX Enterprise SIP Server system configuration and management under this category.



3.3.1 Network

Enter **Management** → **Network** to configure WAN and LAN IP.



■ WAN

- **Mode:** Select EZLoop® IP-PBX Enterprise SIP Server WAN port network mode to be Fixed IP, DHCP or PPPoE.
- **IP Address/NETMASK/GATEWAY:** If user has set EZLoop® IP-PBX Enterprise SIP Server to be fixed IP mode. User need to input IP address/Subnet Mask/ Default Gateway.

- **DNS:** Input DNS address.
- **PPPoE ID:** If you choose the Mode to PPPoE, you should also input the PPPoE ID here for authentication.
- **PPPoE PWD:** If you choose the Mode to PPPoE, you should also input the PPPoE password here for authentication.
- **Mac:** Mac address of EZLoop® IP-PBX Enterprise SIP Server WAN port. The Mac address cannot be modified.

■ LAN

- **IP Address:** Input IP address for LAN port of EZLoop® IP-PBX Enterprise SIP Server.
- **NETMASK:** Input Subnet Mask for LAN port of EZLoop® IP-PBX Enterprise SIP Server.
- **Mac:** Mac address of EZLoop® IP-PBX Enterprise SIP Server LAN port. The Mac address cannot be modified.

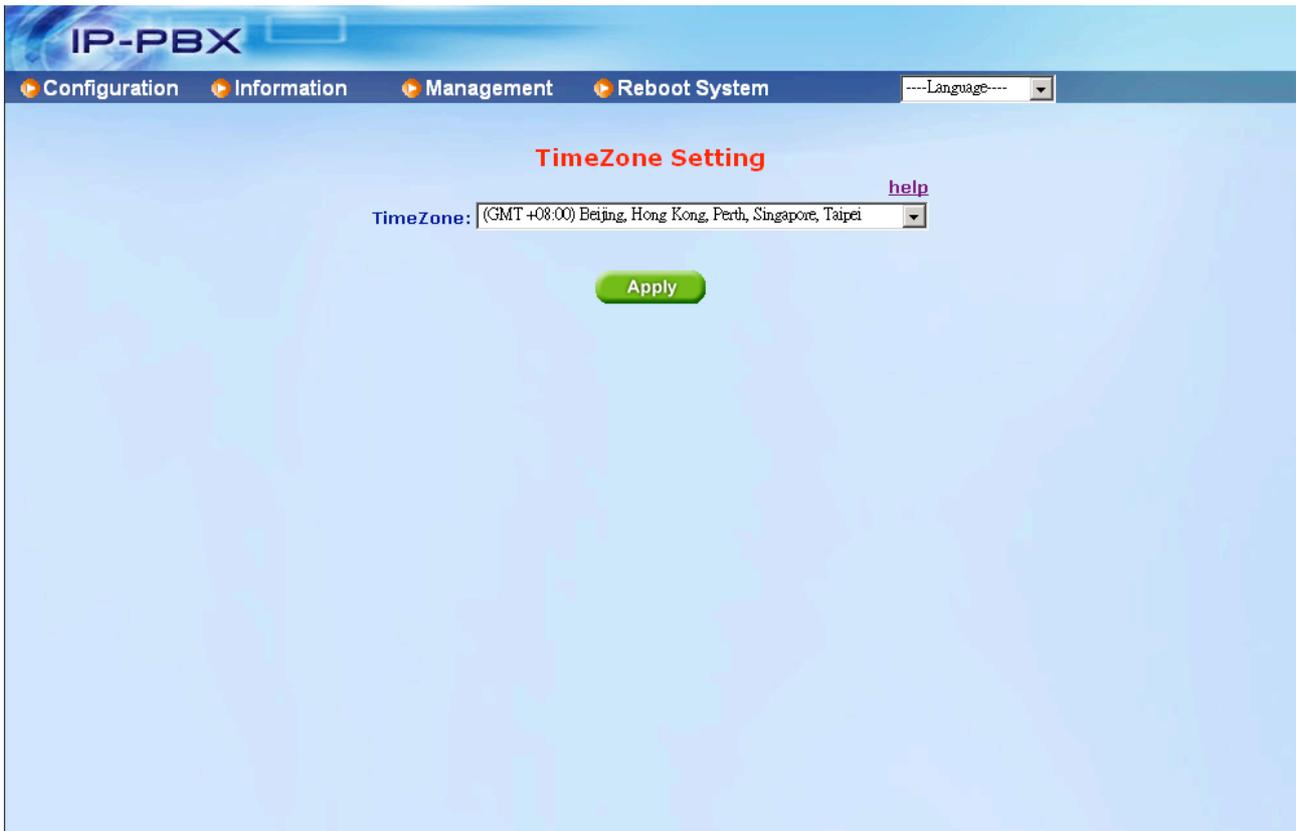
Press Apply to save configuration, or press Cancel to quit configuration.

3.3.2 TimeZone

Enter Management → **TimeZone** to select correct Time Zone for EZLoop® IP-PBX Enterprise SIP Server, this time will affect CDR and voice mail time display.

Note:

- CDR Function can only work in local area network. Please prepare the CDR server under LAN.



3.3.3 SMTP Setting

EZLoop® IP-PBX Enterprise SIP Server can support Voice Mail to e-mail. Before activate this feature, you should give the EZLoop® IP-PBX Enterprise SIP Server an e-mail account and set the SMTP for EZLoop® IP-PBX Enterprise SIP Server, so that the EZLoop® IP-PBX Enterprise SIP Server has the ability to send the leaved message to the subscriber's mail box.



The screenshot shows the 'SMTP Setting' configuration page in the IP-PBX web interface. The page features a blue header with the 'IP-PBX' logo and navigation tabs for 'Configuration', 'Information', 'Management', and 'Reboot System'. A language dropdown menu is located in the top right corner. The main content area is titled 'SMTP Setting' and contains a form with the following fields:

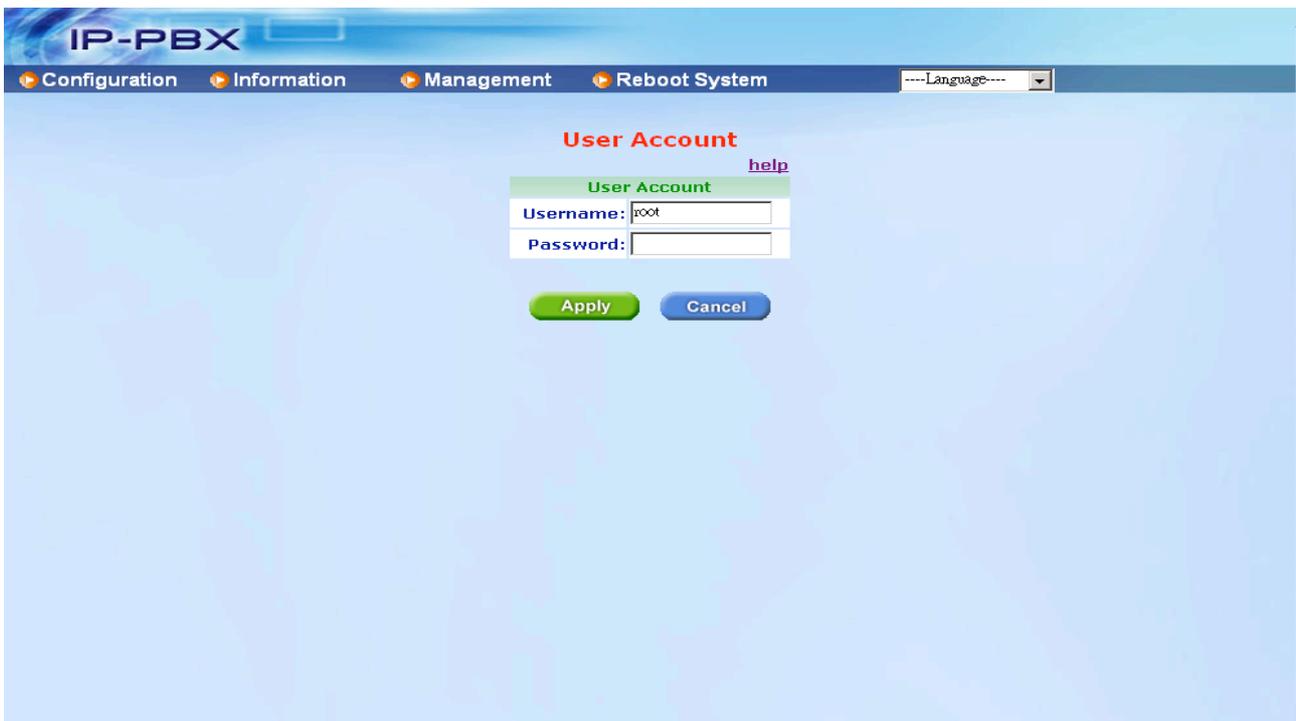
SMTP Setting	
Mail address:	<input type="text"/>
SMTP Server:	<input type="text"/>
Account:	<input type="text"/>
Password:	<input type="text"/>

Below the form are two buttons: 'Apply' (green) and 'Cancel' (blue).

- **Mail Address:** Input the mail address here for EZLoop® IP-PBX Enterprise SIP Server.
- **SMTP server:** Input the SMTP server address.
- **Account:** If your SMTP server needs the user account for Authentication, please input user account here.
- **Password:** If your SMTP server needs the password for Authentication, please input password here.

3.3.4 User Account

Enter **Management** → **User Account** User can set login User name and Password here. System only one set of user.



3.3.5 Firmware Upload

Enter **Management** → **Firmware Upload** → **Choose the Firmware options (Application, Kernel and File System)** → **Press Browse** and select firmware file → **Press Apply** to start firmware upload.

Note:

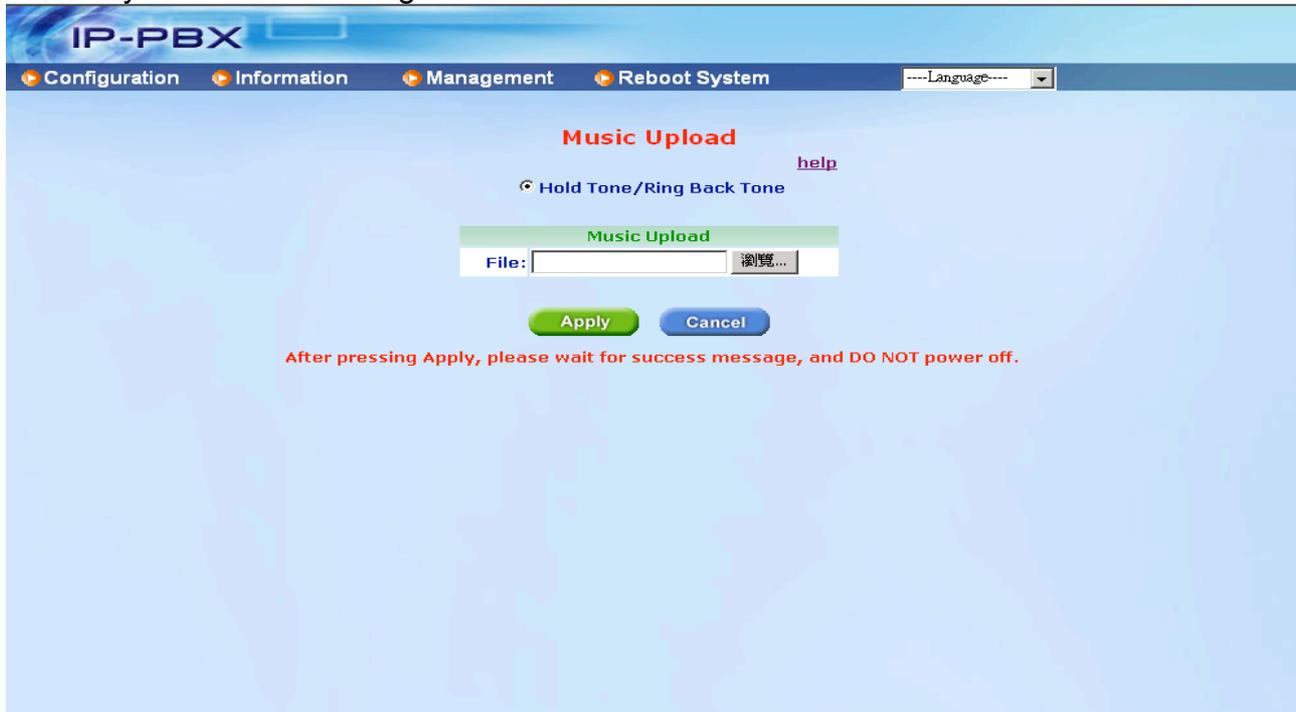
- Normally, you just need to upgrade the Application but in some situation you may need to also upgrade the Kernel or File System. For more information, please refer to the release note of EZLoop® IP-PBX Enterprise SIP Server.
- After pressing Apply, please wait for success message, and **DO NOT** power off.
- After upload succeed, on screen will show success message. Please reboot system to renew system firmware.



3.3.6 Music Upload

User can customize Ring Back Tone (Transferring Tone) by upload new wave file on EZLoop® IP-PBX Enterprise SIP Server. Please record wave file format as **PCM, Channel Mode: Mono, Frequency: 8K, Bit Rate: 16 bit**.

Enter **Management** → **Music Upload** → **Press Browse...** → **select wave file** → **Press Apply** to upload special Ring Back Tone. After Upload is finished, press Reboot to reboot system to renew Ring Back Tone.



3.3.7 Import Setting

If there is EZLoop® IP-PBX Enterprise SIP Server setting file exported from EZLoop® IP-PBX Enterprise SIP Server, user can import this file and doesn't need to re-configure for EZLoop® IP-PBX Enterprise SIP Server.

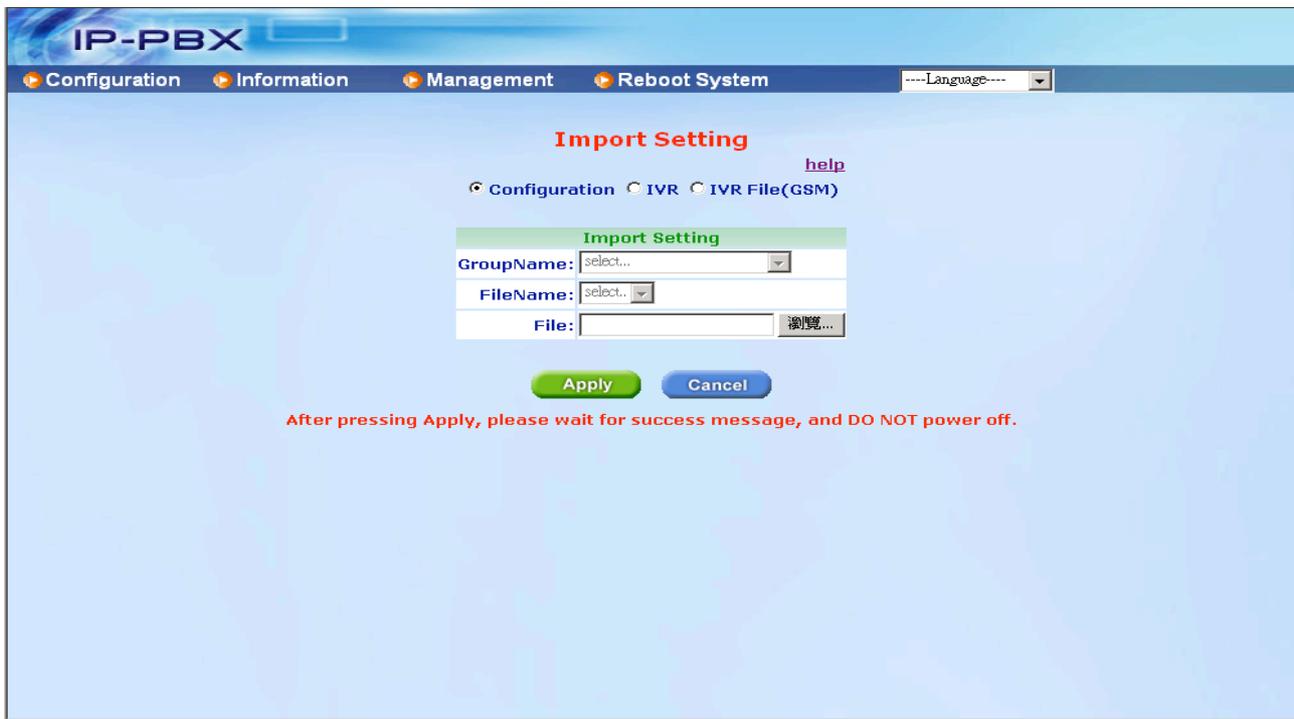
Enter **Management** → **Import Setting** → **Choose the Import options (Configuration, IVR or IVR File (GSM))** → **Press Browse and select setting file** → **Press Apply** to Import Setting file.

After Import finished, on screen will show related information. **Please reboot system to renew system configuration.**

Note:

- After pressing Apply, please wait for success message, and DO NOT power off.
- After Import succeed, on screen will show related information. Please reboot system to renew settings.
- You can choose the Configuration or IVR option and import the file to restore the configuration setting, and you can also choose the IVR File option and import a specific IVR files to EZLoop® IP-PBX Enterprise SIP Server.
- If you choose the option to IVR File(GSM). Before import the IVR File to EZLoop® IP-PBX Enterprise SIP Server, you should prepare the gsm file by yourself first. You should choose the Group and select file name, then you can import a specific file with gsm format to EZLoop® IP-PBX Enterprise SIP Server to instead the old one.
- For example, the IVR file of DAY Greeting is in [Group Auto Attendant Sounds Files], and the file name is [greeting-day.gsm], so you should record the IVR file

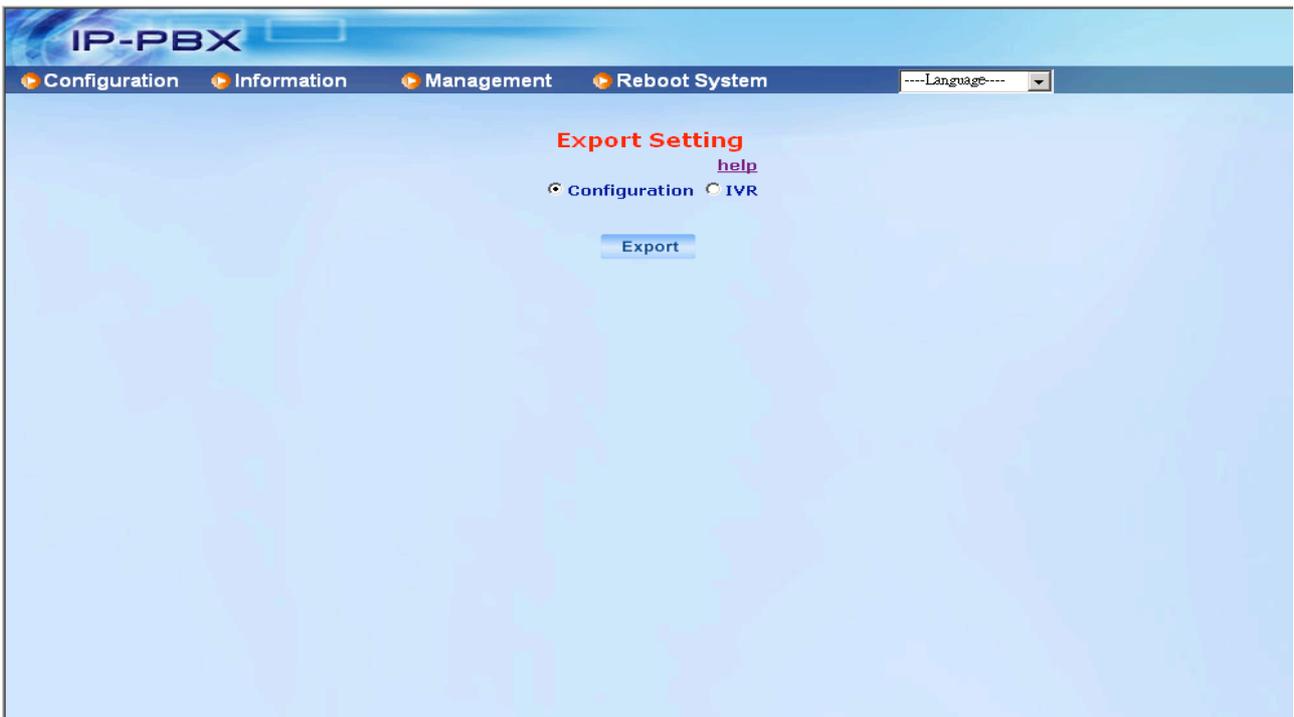
- by your pc and switch the format and file name to greeting-day.gsm. And you should choose the correct group and file name to instead the old one.
- For more information about the detail IVR files, please refer to user manual: **4.1.3 How to record the other announcements**



3.3.8 Export Setting

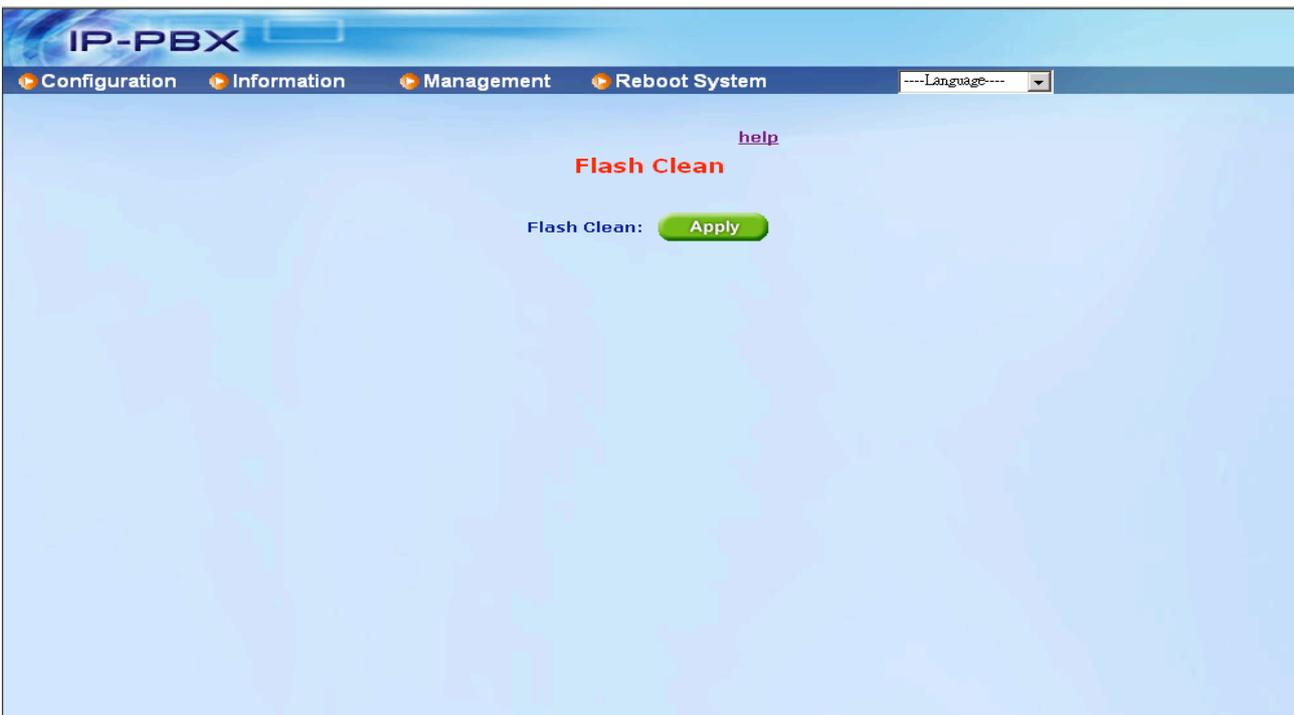
User can export configuration and voice wave files. If there is more than one EZLoop® IP-PBX Enterprise SIP Server need to be configured, user can export configuration of one EZLoop® IP-PBX Enterprise SIP Server, and then import this setting file for the other EZLoop® IP-PBX Enterprise SIP Servers, so that user doesn't need to re-configure for each EZLoop® IP-PBX Enterprise SIP Server.

Enter **Management** → **Export Setting** → **Choose the Export options (Configuration or IVR)** → **Press Export**, wait for system to collect setting-select directory to save setting file.



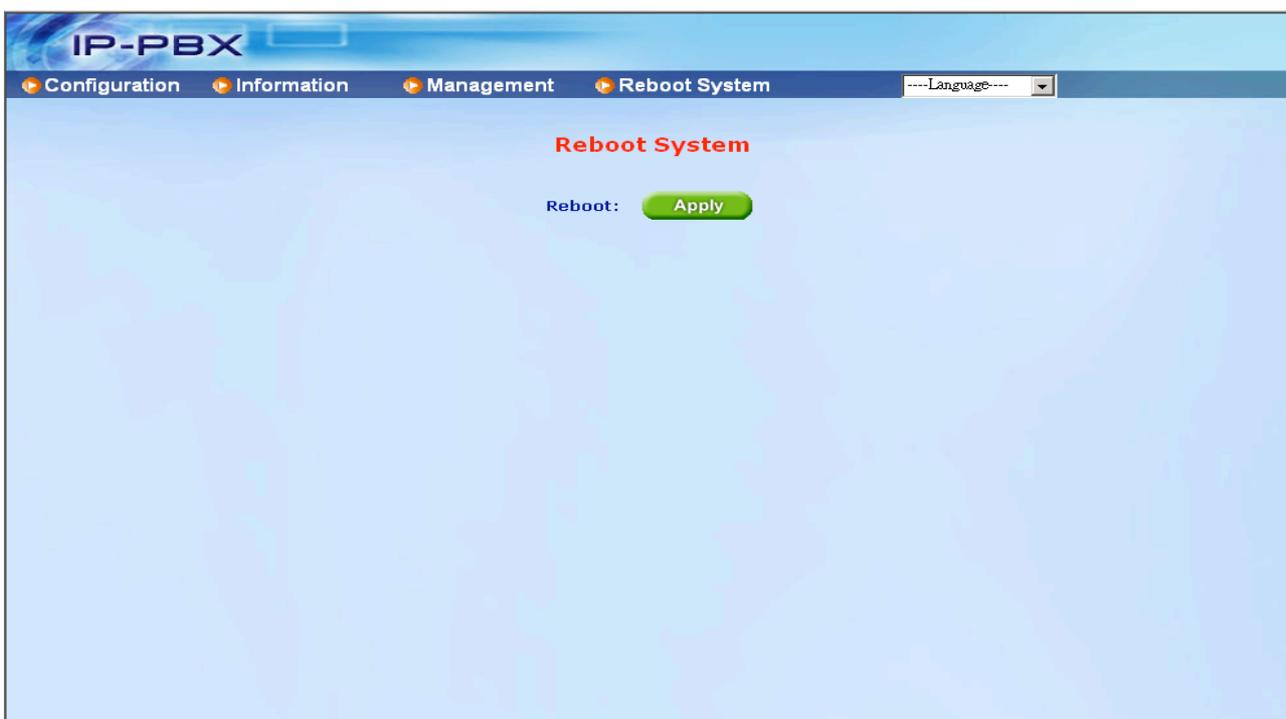
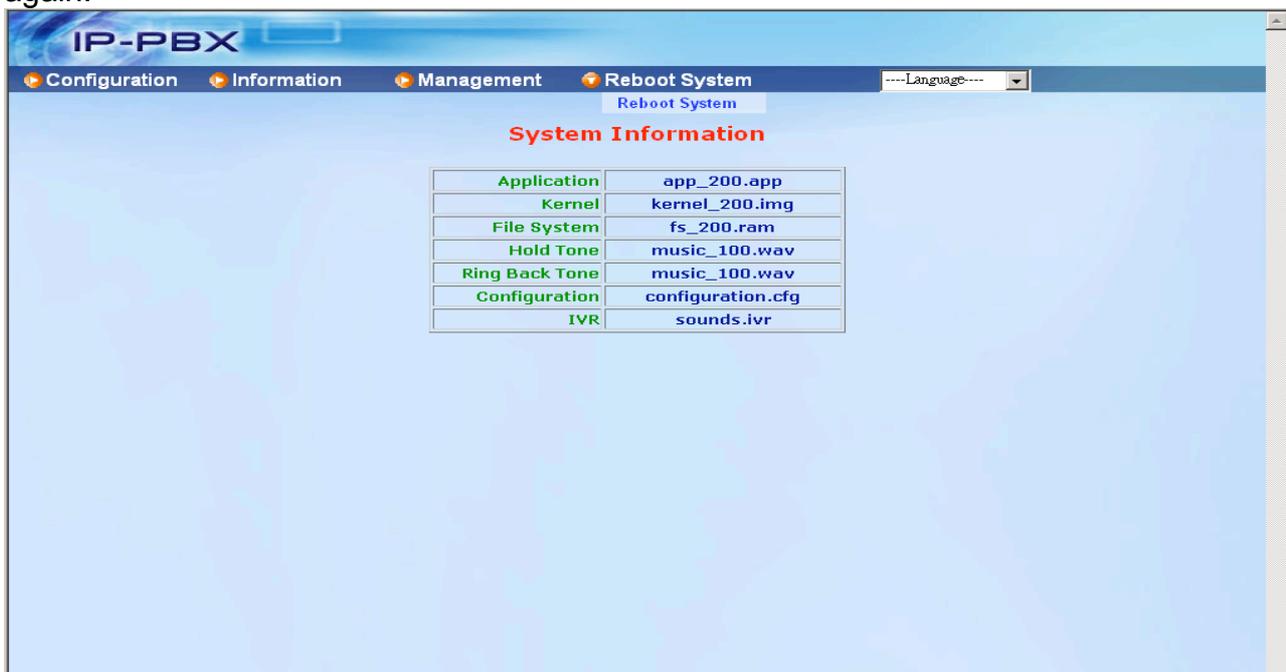
3.3.9 Flash Clean

Enter Management → Flash Clean → Press Apply will clean all configurations of EZLoop® IP-PBX Enterprise SIP Server and reset to factory default value, including Network setting and Web login account. After Flash Clean, EZLoop® IP-PBX Enterprise SIP Server will auto reboot.



3.4 Reboot System

Press Apply to reboot system. Please wait for a few minutes and reload web page again.



CH4. Application Setting

4.1 Customize Automated Attendant

4.1.1 Record Greeting

Use any Extension phone set to dial in ****111** will hear beep, then user can start to record, after recording press **#** then hang up phone set. Greeting will renew immediately after recording.

4.1.2 Enable Automated Attendant

User has to Enable Trunk (e.g. EZLoop® PSTN Gateway FX04/06) hotline function and point to destination number ****999** (Number of Automated Attendant for EZLoop® IP-PBX Enterprise SIP Server). Once system has incoming call from PSTN, it will automatically connect to Automated Attendant.

Note:

- All of the Extensions can also dial to ****999** to reach Automated Attendant directly.
- For more information, please refer to the CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.

4.1.3 How to record the other announcements

User can record the greeting by dialing to ****111**, so that the caller will hear the new greeting if user call to ****999**. When the user called to an Extension, which is on the phone, he will also hear an announcement of Extension is busy. How to record a new busy announcement? The procedure is just like recording new greeting, dialing to Record Code for recording. And user can also dial to Playback Code to listen the announcement. You can also record the sound files with GSM format by your PC or other equipment, and choose the specific file name in the page of Import Setting, import the GSM file to instead the old one. For example, you can record the greeting announcement by you PC as a WAV file. And use some tools to switch it to GSM format, name it as greeting-day.gsm. then you can go to "Import Setting" page, choose **IVR File(GSM)** → Choose GroupName to Auto Attendant Sounds Files → Choose FileName to greeting-day.gsm → Upload new file. Below is the detail for announcement.

GroupName: Digits Sounds Files			
Record Code	Playback Code	File Name	Default Announcement
0000	*0000	0.gsm	zero
0001	*0001	1.gsm	one
0002	*0002	2.gsm	two
0003	*0003	3.gsm	three
0004	*0004	4.gsm	four
0005	*0005	5.gsm	five
0006	*0006	6.gsm	six
0007	*0007	7.gsm	seven
0008	*0008	8.gsm	eight
0009	*0009	9.gsm	nine

0010	*0010	10.gsm	ten
0011	*0011	11.gsm	eleven (It is reserved, not functional now)
0012	*0012	12.gsm	twelve (It is reserved, not functional now)
0013	*0013	13.gsm	thirteen (It is reserved, not functional now)
0014	*0014	14.gsm	fourteen (It is reserved, not functional now)
0015	*0015	15.gsm	fifteen (It is reserved, not functional now)
0016	*0016	16.gsm	sixteen (It is reserved, not functional now)
0017	*0017	17.gsm	seventeen (It is reserved, not functional now)
0018	*0018	18.gsm	eighteen (It is reserved, not functional now)
0019	*0019	19.gsm	nineteen (It is reserved, not functional now)
0020	*0020	20.gsm	twenty (It is reserved, not functional now)
0030	*0030	30.gsm	thirty (It is reserved, not functional now)
0040	*0040	40.gsm	forty (It is reserved, not functional now)
0050	*0050	50.gsm	fifty (It is reserved, not functional now)
0060	*0060	60.gsm	sixty (It is reserved, not functional now)
0070	*0070	70.gsm	seventy (It is reserved, not functional now)
0080	*0080	80.gsm	eighty (It is reserved, not functional now)
0090	*0090	90.gsm	ninety (It is reserved, not functional now)

GroupName: Time & Date Sounds Files

Record Code	Playback Code	File Name	Default Announcement
0101	*0101	h-1.gsm	first (It is reserved, not functional now)
0102	*0102	h-2.gsm	second (It is reserved, not functional now)
0103	*0103	h-3.gsm	third (It is reserved, not functional now)
0104	*0104	h-4.gsm	fourth (It is reserved, not functional now)
0105	*0105	h-5.gsm	fifth (It is reserved, not functional now)
0106	*0106	h-6.gsm	sixth (It is reserved, not functional now)
0107	*0107	h-7.gsm	seventh (It is reserved, not functional now)
0108	*0108	h-8.gsm	eighth (It is reserved, not functional now)
0109	*0109	h-9.gsm	ninth (It is reserved, not functional now)
0110	*0110	h-10.gsm	tenth (It is reserved, not functional now)
0111	*0111	h-11.gsm	eleventh (It is reserved, not functional now)
0112	*0112	h-12.gsm	twelfth (It is reserved, not functional now)
0113	*0113	h-13.gsm	thirteenth (It is reserved, not functional now)
0114	*0114	h-14.gsm	fourteenth (It is reserved, not functional now)
0115	*0115	h-15.gsm	fifteenth (It is reserved, not functional now)
0116	*0116	h-16.gsm	sixteenth (It is reserved, not functional now)
0117	*0117	h-17.gsm	seventeenth (It is reserved, not functional now)
0118	*0118	h-18.gsm	eighteenth (It is reserved, not functional now)
0119	*0119	h-19.gsm	nineteenth (It is reserved, not functional now)
0120	*0120	h-20.gsm	twentieth (It is reserved, not functional now)
0130	*0130	h-30.gsm	thirtieth (It is reserved, not functional now)
0131	*0131	at.gsm	at (It is reserved, not functional now)
0132	*0132	a-m.gsm	AM (It is reserved, not functional now)
0133	*0133	p-m.gsm	PM (It is reserved, not functional now)
0134	*0134	hundred.gsm	hundred (It is reserved, not functional now)

0135	*0135	thousands.gsm	thousand (It is reserved, not functional now)
0136	*0136	million.gsm	million (It is reserved, not functional now)
0137	*0137	minus.gsm	minus (It is reserved, not functional now)
0201	*0201	day-0.gsm	Sunday (It is reserved, not functional now)
0202	*0202	day-1.gsm	Monday (It is reserved, not functional now)
0203	*0203	day-2.gsm	Tuesday (It is reserved, not functional now)
0204	*0204	day-3.gsm	Wednesday (It is reserved, not functional now)
0205	*0205	day-4.gsm	Thursday (It is reserved, not functional now)
0206	*0206	day-5.gsm	Friday (It is reserved, not functional now)
0207	*0207	day-6.gsm	Saturday (It is reserved, not functional now)
0208	*0208	dollars.gsm	dollars (It is reserved, not functional now)
0209	*0209	mon-0gsm	January (It is reserved, not functional now)
0210	*0210	mon-1.gsm	February (It is reserved, not functional now)
0211	*0211	mon-2.gsm	March (It is reserved, not functional now)
0212	*0212	mon-3.gsm	April (It is reserved, not functional now)
0213	*0213	mon-4.gsm	May (It is reserved, not functional now)
0214	*0214	mon-5.gsm	June (It is reserved, not functional now)
0215	*0215	mon-6.gsm	July (It is reserved, not functional now)
0216	*0216	mon-7.gsm	August (It is reserved, not functional now)
0217	*0217	mon-8.gsm	September (It is reserved, not functional now)
0218	*0218	mon-9.gsm	October (It is reserved, not functional now)
0219	*0219	mon-10.gsm	November (It is reserved, not functional now)
0220	*0220	mon-11.gsm	December (It is reserved, not functional now)
0221	*0221	oh.gsm	oh (spoken in a way meaning "zero") (It is reserved, not functional now)
0222	*0222	oclock.gsm	o'clock (It is reserved, not functional now)
0223	*0223	pound.gsm	pound (It is reserved, not functional now)
0224	*0224	star.gsm	star (It is reserved, not functional now)
0225	*0225	today.gsm	today (It is reserved, not functional now)
0226	*0226	tomorrow.gsm	tomorrow (It is reserved, not functional now)
0227	*0227	yesterday.gsm	yesterday (It is reserved, not functional now)

GroupName: Auto Attendant Sounds Files

Record Code	Playback Code	File Name	Default Announcement
111	*111	greeting-day.gsm	Please dial the extension number or press 9 for operator.
112	*112	greeting-noon.gsm	Please dial the extension number, thank you
113	*113	greeting-night.gsm	Please dial the extension number, thank you
114	*114	greeting-holiday.gsm	Please dial the extension number, thank you (It is reserved, not functional now)
115	*115	greeting-temporary.gsm	Please dial the extension number, thank you
116	*116	noanswer.gsm	I am sorry, the extension number you dialed is no answer, please dial another extension number or press 9 for operator.
117	*117	busy.gsm	I am sorry, the extension number you dialed is busy, please dial another extension number or press 9 for operator.
118	*118	goodbyeivr.gsm	goodbye

119	*119	unavailable.gsm	I am sorry, the extension number you dialed is unavailable, please dial another extension number or press 9 for operator.
120	*120	invalid.gsm	I am sorry, that is not a valid extension. Please try again.
121	*121	dnd-act:gsm	Do not disturb active. (It is reserved, not functional now)
122	*122	dnd-deact:gsm	Do not disturb deactivate. (It is reserved, not functional now)
123	*123	unconfwd-act:gsm	Unconditional forward active. (It is reserved, not functional now)
124	*124	unconfwd-deact:gsm	Unconditional forward deactivate. (It is reserved, not functional now)
125	*125	busyfwd-act:gsm	Busy forward active. (It is reserved, not functional now)
126	*126	busyfwd-deact:gsm	Busy forward deactivate. (It is reserved, not functional now)
127	*127	op-noanswer.gsm	The operator is no answer, please call later or dial another extension number
128	*128	op-busy.gsm	The operator is busy, please call later or dial another extension number
129	*129	op-unavailable.gsm	The operator is unavailable, please call later or dial another extension number

GroupName: Voice Mail Sounds Files

Record Code	Playback Code	File Name	Default Announcement
0301	*0301	vm-advopts.gsm	Press 3 for advanced options (It is reserved, not functional now)
0302	*0302	vm-and.gsm	and (It is reserved, not functional now)
0303	*0303	vm-calldiffnum.gsm	Press 2 to enter a different number (It is reserved, not functional now)
0304	*0304	vm-changeto.gsm	Change to which folder? (It is reserved, not functional now)
0305	*0305	vm-cust1.gsm	Folder. Five (It is reserved, not functional now)
0306	*0306	vm-cust2.gsm	Folder. Six (It is reserved, not functional now)
0307	*0307	vm-cust3.gsm	Folder. Seven (It is reserved, not functional now)
0308	*0308	vm-cust4.gsm	Folder. Eight (It is reserved, not functional now)
0309	*0309	vm-cust5.gsm	Folder. Nine (It is reserved, not functional now)
0310	*0310	vm-delete.gsm	Press 7 to delete this message (It is reserved, not functional now)
0311	*0311	vm-deleted.gsm	Message deleted (It is reserved, not functional now)
0312	*0312	vm-dialout.gsm	Please wait while I connect your call (It is reserved, not functional now)
0313	*0313	vm-enter-num-to-call.gsm	Please enter the number you wish to call (It is reserved, not functional now)
0314	*0314	vm-extension.gsm	extension (It is reserved, not functional now)
0315	*0315	vm-Family.gsm	family (It is reserved, not functional now)
0316	*0316	vm-first.gsm	first (It is reserved, not functional now)
0317	*0317	vm-for.gsm	for (It is reserved, not functional now)
0318	*0318	vm-forward.gsm	Press 1 to enter an extension. Press 2 to use

			the directory. (It is reserved, not functional now)
0319	*0319	vm-forwardoptions.gsm	Press 1 to pre-pend the message, or 2 to forward a message without pre-pending. (It is reserved, not functional now)
0320	*0320	vm-Friends.gsm	friend (It is reserved, not functional now)
0321	*0321	vm-from.gsm	from (It is reserved, not functional now)
0322	*0322	vm-from-extension.gsm	Message from extension (It is reserved, not functional now)
0323	*0323	vm-from- phonenumber.gsm	Message from phone number (It is reserved, not functional now)
0324	*0324	vm-goodbye.gsm	goodbye (It is reserved, not functional now)
0325	*0325	vm-helpexit.gsm	Press star for help or pound to exit (It is reserved, not functional now)
0326	*0326	vm-INBOX.gsm	new (It is reserved, not functional now)
0327	*0327	vm-incorrect.gsm	Login incorrect. (It is reserved, not functional now)
0328	*0328	vm-incorrect- mailbox.gsm	Login incorrect. Mailbox (It is reserved, not functional now)
0329	*0329	vm-instructions.gsm	To look into your messages press 1 now. You may quit voicemail at any time by pressing the pound key. (It is reserved, not functional now)
0330	*0330	vm-intro.gsm	Please leave your message after the tone. When done hang up or press the pound key.
0331	*0331	vm-isonphone.gsm	is on the phone
0332	*0332	vm-isunavail.gsm	is unavailable
0333	*0333	vm-last.gsm	last (It is reserved, not functional now)
0334	*0334	vm-leavemsg.gsm	Press 5 to leave a message (It is reserved, not functional now)
0335	*0335	vm-login.gsm	Welcome to Voice Mail. Mailbox (It is reserved, not functional now)
0336	*0336	vm-mailboxfull.gsm	sorry but the user's mail box can't accept more messages (It is reserved, not functional now)
0337	*0337	vm-message.gsm	message (It is reserved, not functional now)
0338	*0338	vm-messages.gsm	messages (It is reserved, not functional now)
0339	*0339	vm-minutes.gsm	minutes (It is reserved, not functional now)
0340	*0340	vm-mismatch.gsm	The passwords you entered and re-entered did not match. Please try again. (It is reserved, not functional now)
0341	*0341	vm-msginstruct.gsm	To hear the next message press 6. To repeat this message press 5. To hear the previous message press 4. To delete or undelete this message press 0. To quit voicemail press pound. (It is reserved, not functional now)
0342	*0342	vm-msgsaved.gsm	Your message has been saved. (It is reserved, not functional now)
0343	*0343	vm-newpassword.gsm	Please enter your new password followed by the pound key. (It is reserved, not functional now)
0344	*0344	vm-newuser.gsm	Welcome to Voice Mail! First, I will guide you through a short setup process. (It is reserved, not functional now)
0345	*0345	vm-next.gsm	Press 6 to play the next message. (It is reserved, not functional now)
0346	*0346	vm-no.gsm	no (It is reserved, not functional now)
0347	*0347	vm-nobodyavail.gsm	Nobody is available to take your call at the moment. (It is reserved, not functional now)
0348	*0348	vm-nobox.gsm	You can not reply to this message because the

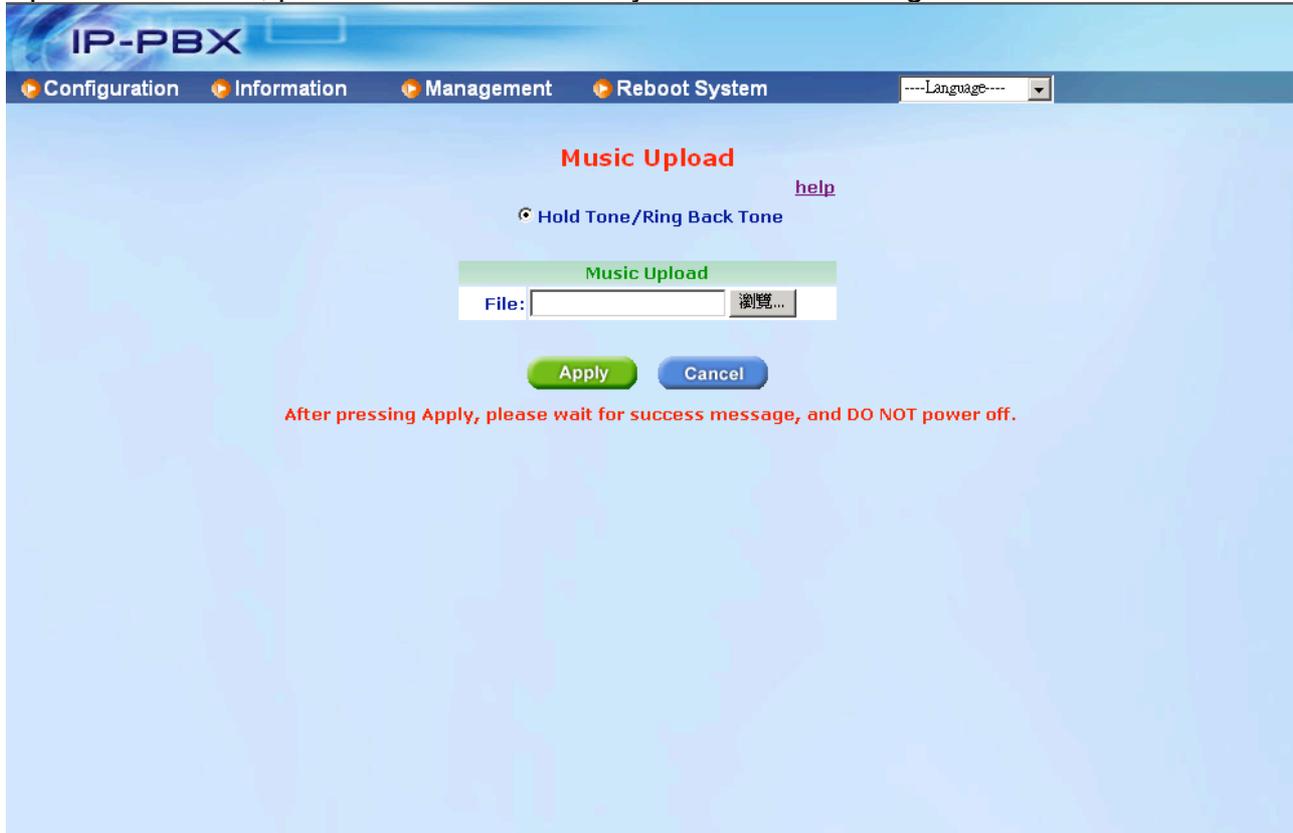
			sender does not have a mailbox (It is reserved, not functional now)
0349	*0349	vm-nomore.gsm	No more messages (It is reserved, not functional now)
0350	*0350	vm-nonumber.gsm	I'm afraid I don't know who sent this message. (It is reserved, not functional now)
0351	*0351	vm-num-i-have.gsm	the number I have is (It is reserved, not functional now)
0352	*0352	vm-Old.gsm	old (It is reserved, not functional now)
0353	*0353	vm-onefor.gsm	Press 1 for (It is reserved, not functional now)
0354	*0354	vm-options.gsm	Press 1 to record your unavailable message. Press 2 to record your busy message. Press 3 to record your name. Press 4 to record your temporary greeting. Press 5 to change your password. Press star to return to the main menu. (It is reserved, not functional now)
0355	*0355	vm-opts.gsm	Press 2 to change folders. Press 3 for advanced options. Press zero for mailbox options. (It is reserved, not functional now)
0356	*0356	vm-passchanged.gsm	Your passwords have been changed. (It is reserved, not functional now)
0357	*0357	vm-password.gsm	password (It is reserved, not functional now)
0358	*0358	vm-press.gsm	press (It is reserved, not functional now)
0359	*0359	vm-prev.gsm	Press 4 for the previous message. (It is reserved, not functional now)
0360	*0360	vm-reachoper.gsm	Press 9 to reach an operator. (It is reserved, not functional now)
0361	*0361	vm-rec-busy.gsm	After the tone say your busy message and then press the pound key. (It is reserved, not functional now)
0362	*0362	vm-received.gsm	received (It is reserved, not functional now)
0363	*0363	vm-rec-name.gsm	After the tone say your name and then press the pound key. (It is reserved, not functional now)
0364	*0364	vm-rec-temp.gsm	After the tone say your temporary message and then press the pound key. (It is reserved, not functional now)
0365	*0365	vm-rec-unv.gsm	After the tone say your unavailable message and then press the pound key. (It is reserved, not functional now)
0366	*0366	vm-reenterpassword.gsm	Please reenter your password followed by the pound key. (It is reserved, not functional now)
0367	*0367	vm-repeat.gsm	Press 5 to repeat the current message. (It is reserved, not functional now)
0368	*0368	vm-review.gsm	Press 1 to accept this recording, press 2 to listen to it, press 3 to re-record your message. (It is reserved, not functional now)
0369	*0369	vm-saved.gsm	saved (It is reserved, not functional now)
0370	*0370	vm-savedto.gsm	save to (It is reserved, not functional now)
0371	*0371	vm-savefolder.gsm	Which folder should I save the message to? (It is reserved, not functional now)
0372	*0372	vm-savemessage.gsm	or 9 to save this message (It is reserved, not functional now)
0373	*0373	vm-saveoper.gsm	Press 1 to accept this recording, otherwise please continue to hold. (It is reserved, not functional now)
0374	*0374	vm-sorry.gsm	I'm sorry I did not understand your response. (It is reserved, not functional now)
0375	*0375	vm-star-cancel.gsm	Press star to cancel (It is reserved, not functional now)

0376	*0376	vm-starmain.gsm	Press star to return the main menu (It is reserved, not functional now)
0377	*0377	vm-tempgreeting2.gsm	Press 1 to record your temporary greeting. Or press 2 to erase your temporary greeting. (It is reserved, not functional now)
0378	*0378	vm-tempgreeting.gsm	Press 1 to record your temporary greeting. (It is reserved, not functional now)
0379	*0379	vm-tempremoved.gsm	Your temporary greeting has been removed. (It is reserved, not functional now)
0380	*0380	vm-then-pound.gsm	Then press pound (It is reserved, not functional now)
0381	*0381	vm-theperson.gsm	The person at extension
0382	*0382	vm-tocallback.gsm	Press 2 to call the person who sent this message. (It is reserved, not functional now)
0383	*0383	vm-tocallnum.gsm	Press 1 to call this number. (It is reserved, not functional now)
0384	*0384	vm-tocancel.gsm	or press pound to cancel. (It is reserved, not functional now)
0385	*0385	vm-tocancelmsg.gsm	Press star to cancel this message (It is reserved, not functional now)
0386	*0386	vm-toenternumber.gsm	Press 1 to enter a number (It is reserved, not functional now)
0387	*0387	vm-toforward.gsm	Press 8 to forward the message to another user. (It is reserved, not functional now)
0388	*0388	vm-tohearenv.gsm	Press 3 to hear the message envelope. (It is reserved, not functional now)
0389	*0389	vm-tomakecall.gsm	Press 4 to place an outgoing call (It is reserved, not functional now)
0390	*0390	vm-tooshort.gsm	Your message is too short (It is reserved, not functional now)
0391	*0391	vm-toreply.gsm	Press 1 to send a reply (It is reserved, not functional now)
0392	*0392	vm-torerecord.gsm	Press 3 to record your message (It is reserved, not functional now)
0393	*0393	vm-undelete.gsm	Press 7 to undelete this message. (It is reserved, not functional now)
0394	*0394	vm-undeleted.gsm	Message undeleted. (It is reserved, not functional now)
0395	*0395	vm-unknow-caller.gsm	from an unknown caller (It is reserved, not functional now)
0396	*0396	vm-whichbox.gsm	To leave a message, please enter a mailbox number. (It is reserved, not functional now)
0397	*0397	vm-Work.gsm	work (It is reserved, not functional now)
0398	*0398	vm-youhave.gsm	you have (It is reserved, not functional now)
GroupName: General Sounds Files			
Record Code	Playback Code	File Name	Default Announcement
0400	*0400	beep.gsm	(This is a beep tone)
0401	*0401	hours.gsm	hours (It is reserved, not functional now)
0402	*0402	minutes.gsm	minutes (It is reserved, not functional now)
0403	*0403	auth-incorrect.gsm	Password incorrect. Please enter your password followed by the pound key.
0404	*0404	auth-thankyou.gsm	thank you
0405	*0405	pbx-invalid.gsm	I am sorry, that is not a valid extension. Please try again.
0406	*0406	pbx-invalidpark.gsm	I am sorry, there is no call parked on that extension. Please try again. (It is reserved, not functional now)

0407	*0407	pax-transfer.gsm	Transfer.
0408	*0408	privacy-incorrect.gsm	I'm sorry, that number is not valid. (It is reserved, not functional now)
0409	*0409	privacy-prompt.gsm	Please enter your ten-digit phone number, starting with the area code. (It is reserved, not functional now)
0410	*0410	privacy-thankyou.gsm	Thank you.
0411	*0411	privacy-unident.gsm	The party you are trying to reach does not accept unidentified calls. (It is reserved, not functional now)
0412	*0412	ss-noservice.gsm	The number you have dialed is not in service. Please check the number and try again. (It is reserved, not functional now)
0413	*0413	transfer.gsm	Please hold while I try that extension
GroupName: Agent Sounds Files			
Record Code	Playback Code	File Name	Default Announcement
0445	*0445	agent-pass.gsm	Please enter password followed by the pound key

4.2 Customize Ring Back Tone (Transferring Tone)

User can customize Ring Back Tone by upload new wave file on EZLoop® IP-PBX Enterprise SIP Server. Please record wave file format as **PCM, Channel Mode: Mono, Frequency: 8K, Bit Rate: 16 bit**. Enter **Management** → **Music Upload** → **Press Browse...** → **select wave file** → **Press Apply** to upload special Ring Back Tone. After Upload is finished, press Reboot to reboot system to renew Ring Back Tone.



The screenshot displays the IP-PBX management web interface. At the top, there is a navigation bar with the following menu items: Configuration, Information, Management, and Reboot System. A language selection dropdown is also present. The main content area is titled "Music Upload" and includes a "help" link. Below the title, there is a radio button option for "Hold Tone/Ring Back Tone". A "Music Upload" section contains a "File:" label, an empty text input field, and a "浏览..." (Browse...) button. Below the input field are two buttons: "Apply" and "Cancel". A red warning message at the bottom of the section reads: "After pressing Apply, please wait for success message, and DO NOT power off."

4.3 Call Features

4.3.1 Authentication

When EZLoop® IP-PBX Enterprise SIP Server gets a Registration or Invite (incoming call) from a remote location, it will reply Authentication for security issue.

4.3.2 Automated Attendant

The EZLoop® IP-PBX Enterprise SIP Server supports Automated Attendant; you can record the default greeting and the other announcements by Extension. For more information, please refer to [4.1.3 How to record the other announcements.](#)

4.3.3 Call Transfer

The EZLoop® IP-PBX Enterprise SIP Server supports “server transfer” now; you can enable the Hot-key Tran function in IP PBX and Trunk page. After enable Hot-key Tran, you can press *9 for Call Transfer. You can also perform the Client based Call transfer by subscriber device and the transfer function of the subscriber device should follow SIP standard.

4.3.4 Blind Transfer (Client based)

The EZLoop® IP-PBX Enterprise SIP Server supports “server blind transfer” now; you can enable the Hot-key Tran function in IP PBX and Trunk page. After enable Hot-key Tran, you can press *0 for Blind Transfer. You can also perform the Client based Call transfer by subscriber device and the transfer function of the subscriber device should follow SIP standard.

4.3.5 Call Forward on Busy (Client based)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server forward” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by LCD or “WEB” interface.

4.3.6 Call Forward on No Answer (Client based)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server forward” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by LCD or “WEB” interface.

4.3.7 Call Forward Unconditional (Client based)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server forward” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by LCD or “WEB” interface.

4.3.8 Call Hold/Retrieval (Client based)

Normally, the call hold and call retrieval is done by Client, the EZLoop® IP-PBX Enterprise SIP Server just relay the SIP signal for such function.

4.3.9 Call Routing

In the **Configuration → Routing Table**, you can set the Routing record for a specified Prefix.

4.3.10 Call Waiting (Client based)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server Call Waiting” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by “CLI”.

4.3.11 Caller ID

The EZLoop® IP-PBX Enterprise SIP Server will relay the caller ID from caller to callee.

4.3.12 Do Not Disturb (Client based)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server DND” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by DND button.

4.3.13 Flexible Extension Logic

You can set the digits length of subscriber to 30 digits.

4.3.14 Music On Hold

The EZLoop® IP-PBX Enterprise SIP Server will play music if the user is under Hold status.

4.3.15 Music On Transfer

The EZLoop® IP-PBX Enterprise SIP Server will play music if the user is under Transfer status.

4.3.16 Call Pickup (Global Call Pickup)

The EZLoop® IP-PBX Enterprise SIP Server can support Global Call Pickup. For example: Ext-A is ringing, Ext-B can press *8 for call pickup.

4.3.17 Three-way Conference (LP388)

The EZLoop® IP-PBX Enterprise SIP Server does not support “server Conference” now, this feature should be done by client. For example, if the client is LP388, you can enable this feature by Conf button.

4.3.18 Time and Date

You can select correct Time Zone for EZLoop® IP-PBX Enterprise SIP Server; this time will affect CDR and voice mail time display.

4.3.19 Trunking (EZLoop® PSTN Gateway FX04/06)

You can install a FXO gateway as a Trunk. The FXO gateway can connect with a PSTN line so that your Extension can dial to PSTN via FXO gateway. For more info, please refer to: [CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.](#)

4.3.20 VoIP Gateways (EZLoop® PSTN Gateway FX04/06;)

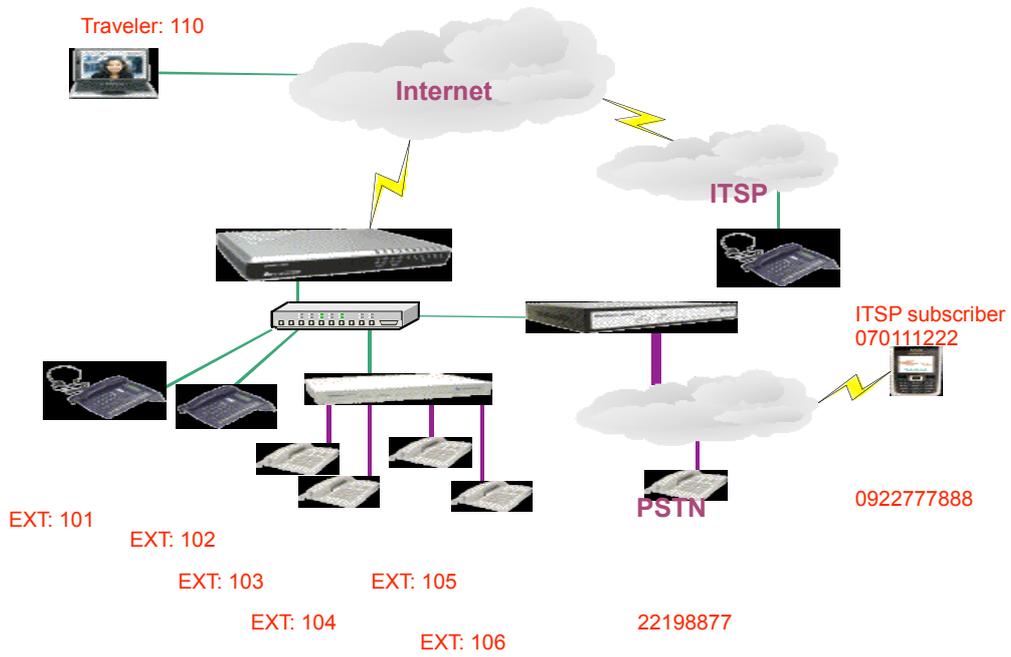
You can install a FXO gateway as a Trunk. The FXO gateway can connect with a PSTN line so that your Extension can dial to PSTN via FXO gateway. You can also install a FXS gateway as an Extension. For more info, please refer to: [CH5. Appendix-](#)

Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.

4.3.21 Voice Mail to e-mail

EZLoop® IP-PBX Enterprise SIP Server does not have enough Flash Rom to store voice mail within itself, but it can send voice mail by e-mail. You should configure the SMTP setting to perform Voice Mail to e-mail. If the EZLoop® IP-PBX Enterprise SIP Server got a new message, it will send the message to user by email immediately.

CH5. Appendix-Application between CPE device and EZLoop® IP-PBX Enterprise SIP Server.



VOICE YOUR NET

Installation Wizard

Advanced Config

- Network Configuration
- SIP Configuration
- System Configuration
- Number Configuration
- Media Configuration
- Device Management

System Status

Reboot

Network Configuration Behind NAT SNTP

Static IP Configuration

Network Connection Mode	Static IP (Set IP address manually)
IP Address	192.168.13.13
Subnet Mask	255.255.248.0
Default Gateway	192.168.8.254
Primary DNS Address	168.95.192.1
Secondary DNS Address	168.95.1.1
HTTP Port for WEB Management (1024~65535)	80

OK CANCEL

4A FXS Gateway Configuration Menu

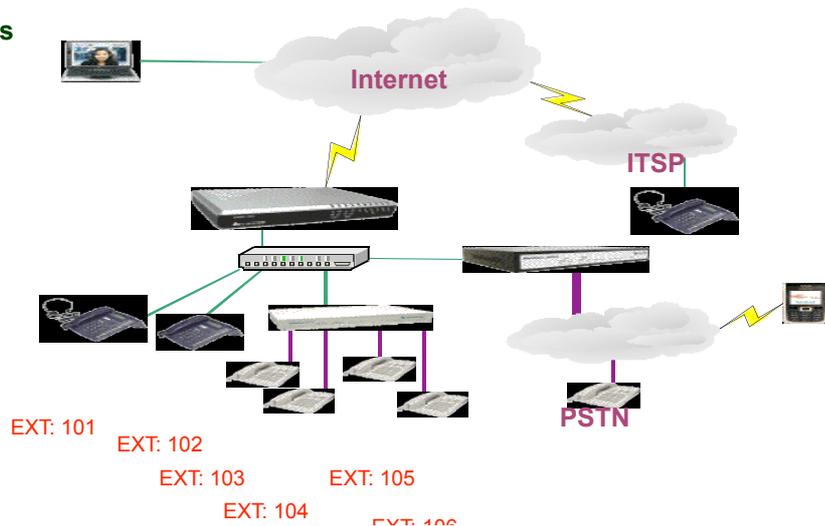
- Network Interface
- SIP Information
- System Configuration
- PPPoE Configuration
- Voice Setting
- Phone Pattern
- Support Function
- Prefix Configuration
- Phone Book
- DSCP Configuration
- Password
- ROM Configuration
- Flash Clean
- Commit Data
- Reboot System

Network Interface

IP Address:	192 168 13 67
Subnet Mask:	255 255 248 0
Default routing gateway:	192 168 8 254
HTTP Port:	80
DHCP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	168 95 195 12
GMT:	8
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	210 59 163 198
Primary DNS Server:	168 95 192 1
Secondary DNS Server:	168 95 1 1

OK

Step2: Configure Extensions



- In EZLoop® IP-PBX Enterprise SIP Server, prepare Extension accounts for Client device. (There are 10 default Extensions, from 101 to 110, so we use the default setting for this Example) For more information about Extension page, please go to user's manual [CH3- Full Web Configurations](#).

Index	Extension Number	Keypad	NAT Traversal	RTP Mode	Setting
1	101	rfc2833	Disable	Routed Mode	Modify Delete
2	102	rfc2833	Disable	Routed Mode	Modify Delete
3	103	rfc2833	Disable	Routed Mode	Modify Delete
4	104	rfc2833	Disable	Routed Mode	Modify Delete
5	105	rfc2833	Disable	Routed Mode	Modify Delete
6	106	rfc2833	Disable	Routed Mode	Modify Delete
7	107	rfc2833	Disable	Routed Mode	Modify Delete
8	108	rfc2833	Disable	Routed Mode	Modify Delete
9	109	rfc2833	Disable	Routed Mode	Modify Delete
10	110	rfc2833	Disable	Routed Mode	Modify Delete
11	none	rfc2833	Disable	Routed Mode	Modify Delete
12	none	rfc2833	Disable	Routed Mode	Modify Delete
13	none	rfc2833	Disable	Routed Mode	Modify Delete
14	none	rfc2833	Disable	Routed Mode	Modifv Delete

VOICE YOUR NET

Installation Wizard

Advanced Config

Network Configuration

SIP Configuration

System Configuration

Number Configuration

Media Configuration

Device Management

System Status

Reboot

SIP Main Configuration

SIP Advanced Configuration

SIP Main Configuration

Primary Proxy Address: 192.168.13.95 Port: 5060

Secondary Proxy Address: x Port: 5060

Outbound Proxy Address: x Port: 5060

Phone Number: 101

Registration Account Name: 101

Registration Account Password: •••

OK CANCEL

- This Example, we have 1 WG3504 register to EZLoop® IP-PBX Enterprise SIP Server with extension 103 to 106. You can set the SIP Information of 3504 by its COM port or you can also login its WEB interface for configuration as below. Set the 3504 to proxy mode, and also set the

Primary Proxy IP Address and Line Number..., etc. After configure SIP information, please press OK → Commit Data →.

4AEXS Gateway Configuration Menu		SIP Information		
Network Interface	Run Mode:	<input type="radio"/> Peer-2-Peer	<input checked="" type="radio"/> Proxy	<input type="radio"/> Gateway
SIP Information	Primary Proxy IP Address:	<input type="text" value="192.168.13.95"/>		
System Configuration	Primary Proxy port:	<input type="text" value="5060"/>		
PPPoE Configuration	Secondary Proxy IP Address:	<input type="text" value="null"/>		
Voice Setting	Secondary Proxy port:	<input type="text" value="5060"/>		
Phone Pattern	Outbound Proxy:	<input type="text" value="null"/>		
Support Function	Outbound Proxy port:	<input type="text" value="5060"/>		
Prefix Configuration	Prefix String:	<input type="text" value="null"/>		
Phone Book	Line1 Number:	<input type="text" value="103"/>		
DSCP Configuration	Line1 Account:	<input type="text" value="103"/>		
Password	Line1 Password:	<input type="text" value="***"/>		
ROM Configuration	Line2 Number:	<input type="text" value="104"/>		
Flash Clean	Line2 Account:	<input type="text" value="104"/>		
Commit Data	Line2 Password:	<input type="text" value="***"/>		
Reboot System				

- If all of the above settings are correct, you can go to **Information** → **Subscriber** page to confirm the register status.

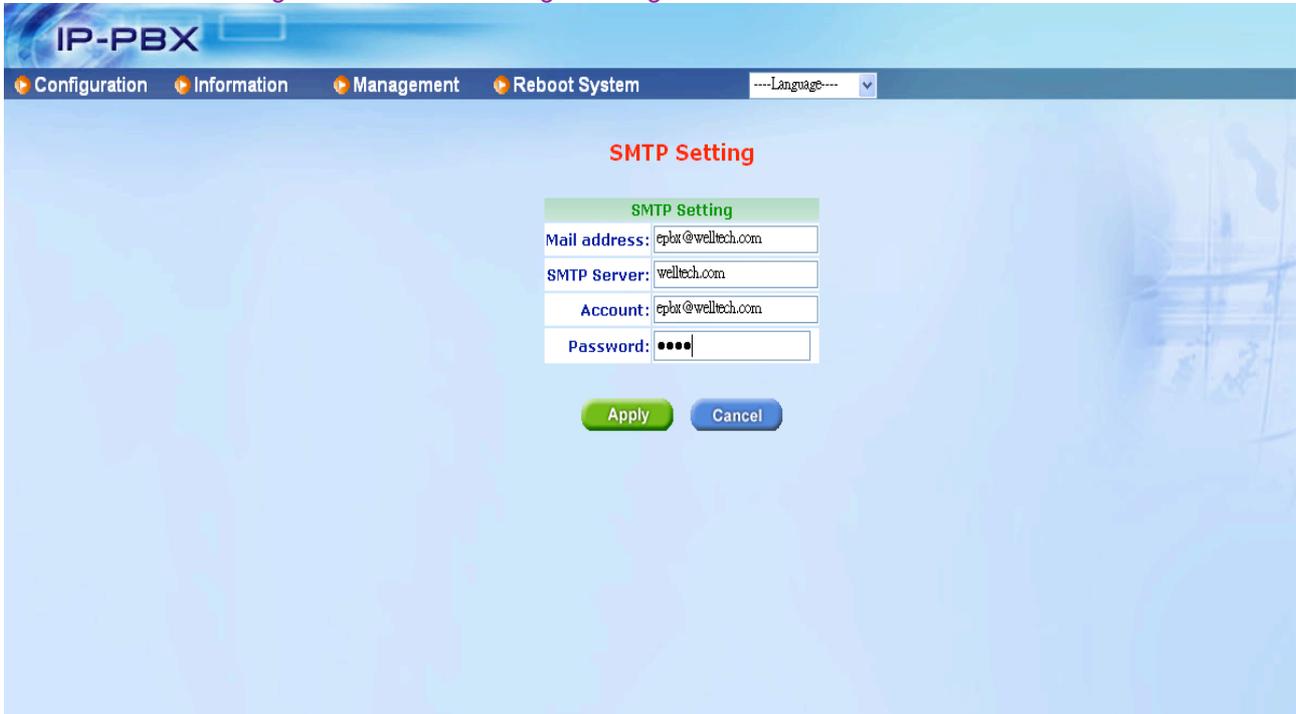
IP-PBX				
Configuration Information Management Reboot System				
Subscriber				
Index	Phone Number	IP Address	NAT Traversal	Mail Address
1	101	192.168.13.14	Disable	-none-
2	102	192.168.13.13	Disable	-none-
3	103	192.168.13.67	Disable	-none-
4	104	192.168.13.67	Disable	-none-
5	105	192.168.13.67	Disable	-none-
6	106	192.168.13.67	Disable	-none-
7	107	-none-	Disable	-none-
8	108	-none-	Disable	-none-
9	109	-none-	Disable	-none-
10	110	-none-	Disable	-none-
11	888	-none-	Disable	-none-
12	889	-none-	Disable	-none-

Now, all of the Extensions can contact with each other. If the called party is busy or no answer, EZLoop® IP-PBX Enterprise SIP Server will play an announcement to indicate the called party's status.

5.2 The call will be forward to MailBox if the extension 101 is busy or no answer.

Step1: Configure SMTP setting

- Go to Management → SMTP Setting to configure the SMTP.



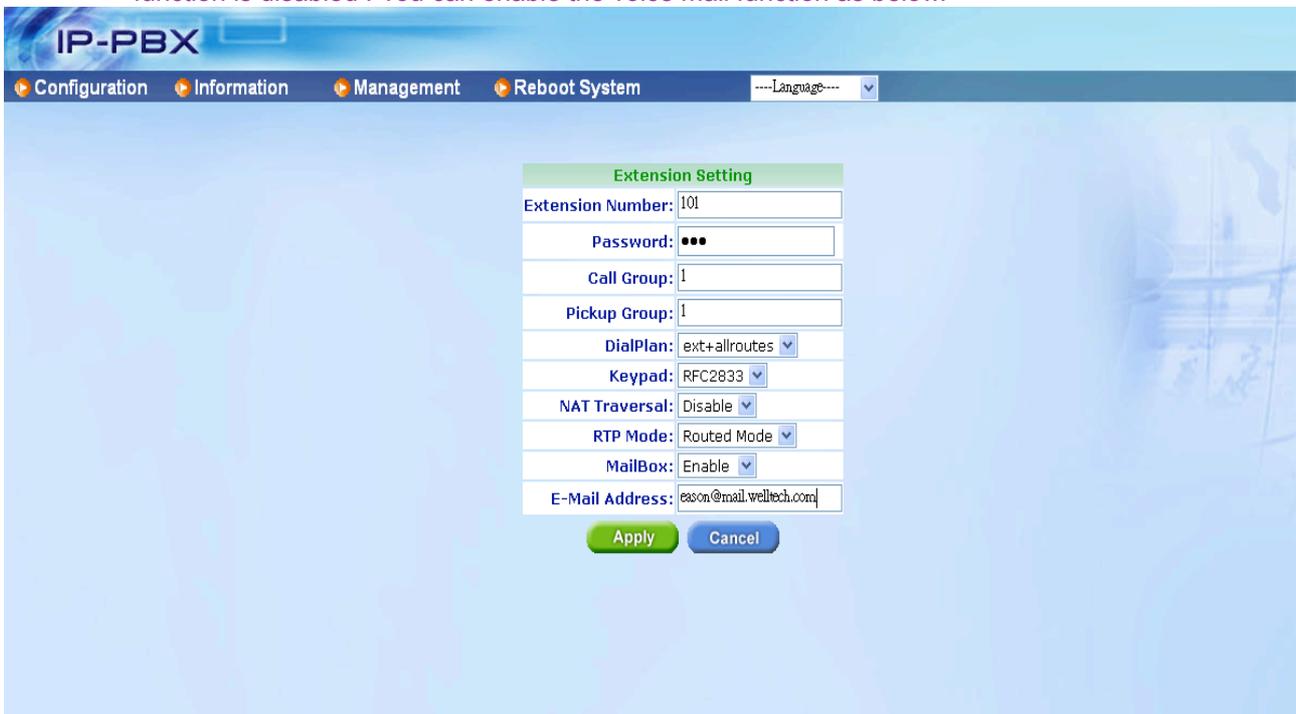
The screenshot shows the IP-PBX Management interface. The top navigation bar includes 'Configuration', 'Information', 'Management', and 'Reboot System'. The main content area is titled 'SMTP Setting' and contains a form with the following fields:

SMTP Setting	
Mail address:	epbx@welltech.com
SMTP Server:	welltech.com
Account:	epbx@welltech.com
Password:	••••

At the bottom of the form are two buttons: 'Apply' and 'Cancel'.

Step2: Enable Voice Mail function

- EZLoop® IP-PBX Enterprise SIP Server has 10 Extensions (101 to 110) and the voice mail function is disabled. You can enable the voice mail function as below.



The screenshot shows the IP-PBX Management interface. The top navigation bar includes 'Configuration', 'Information', 'Management', and 'Reboot System'. The main content area is titled 'Extension Setting' and contains a form with the following fields:

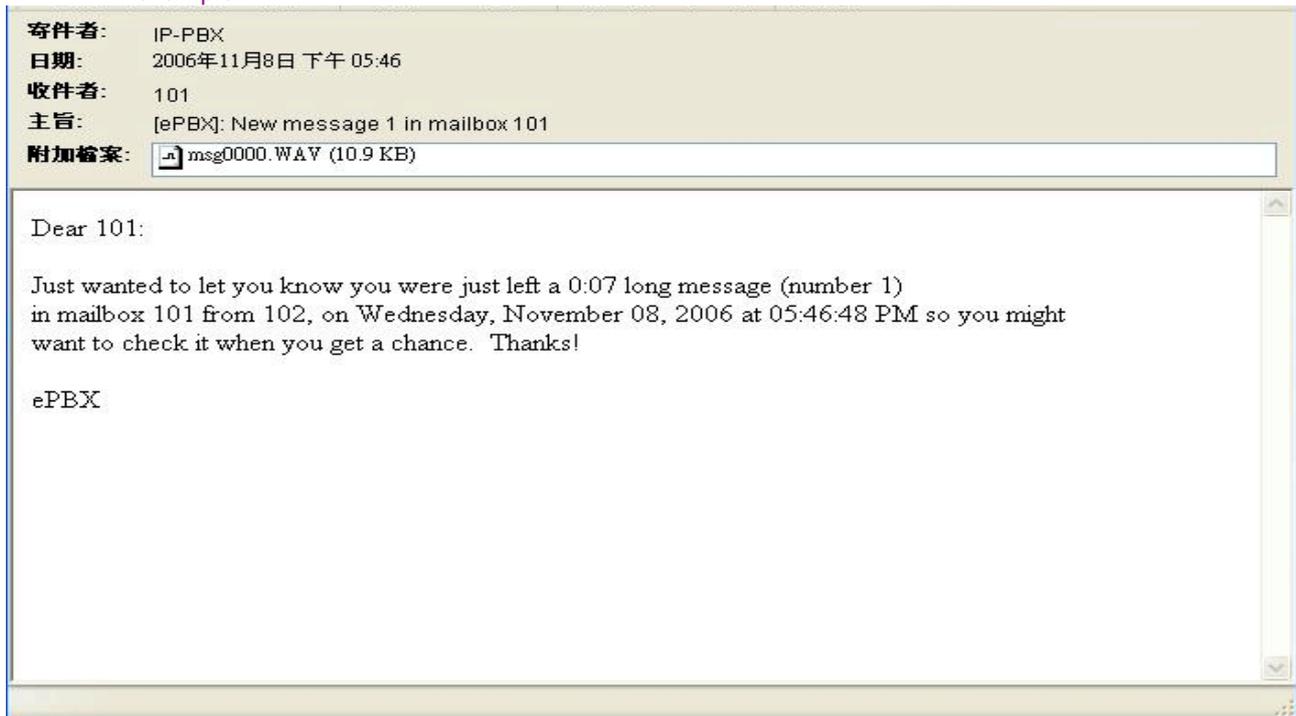
Extension Setting	
Extension Number:	101
Password:	•••
Call Group:	1
Pickup Group:	1
DialPlan:	ext+allroutes
Keypad:	RFC2833
NAT Traversal:	Disable
RTP Mode:	Routed Mode
MailBox:	Enable
E-Mail Address:	person@mail.welltech.com

At the bottom of the form are two buttons: 'Apply' and 'Cancel'.

Step3: Confirm Voice Mail

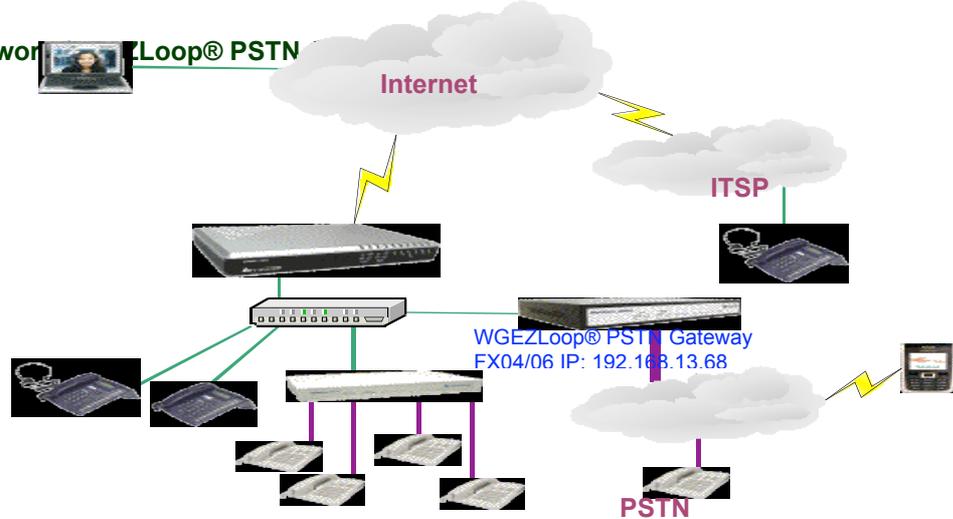
- If 102 call to 101 but 101 is busy, EZLoop® IP-PBX Enterprise SIP Server will play an announcement to indicate the 101 is busy, and 102 can leave message for 101. EZLoop® IP-PBX

Enterprise SIP Server will send voice mail to your mail box with a WAV format. Below is an example.



5.3 The Trunk (EZLoop® PSTN Gateway FX04/06) can also register to EZLoop® IP-PBX Enterprise SIP Server (registered number 888).

Step1: Setup Network

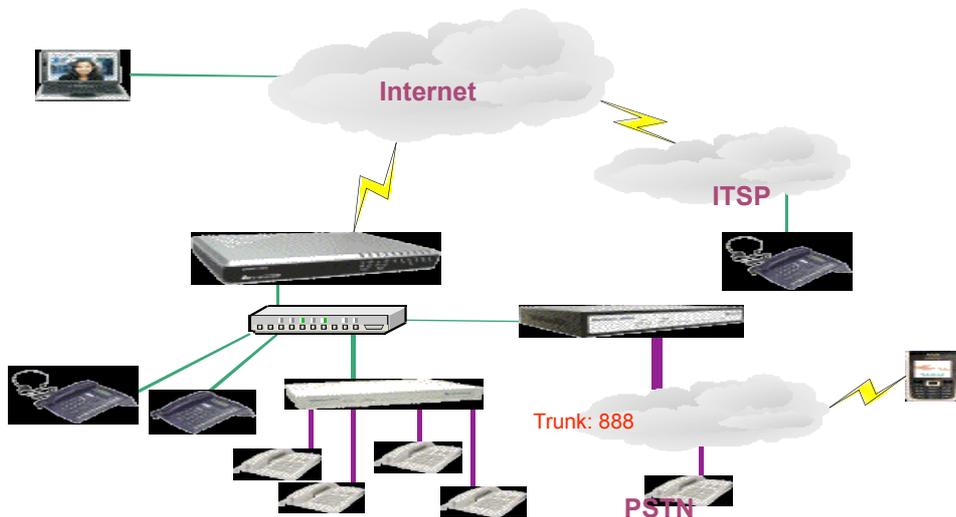


FXO Gateway Configuration Menu

- [Network Interface](#)
- [SIP Config](#)
- [Security Config](#)
- [Line Configuration](#)
- [System Configuration](#)
- [Voice Setting](#)
- [Tone Setting](#)
- [Phone Book](#)
- [Prefix Configuration](#)
- [Routing Table](#)
- [FXO Password](#)
- [IP Packet ToS](#)
- [Password](#)
- [RTP Payload Type Configuration](#)
- [ROM Upgrade](#)
- [Flash Clean](#)

Network Interface	
IP Address:	192 . 168 . 13 . 68
Subnet Mask:	255 . 255 . 248 . 0
Default routing gateway:	192 . 168 . 8 . 254
IP Mode:	<input checked="" type="radio"/> FIX IP <input type="radio"/> DHCP <input type="radio"/> PPPoE
HTTP Port:	80
DNS primary:	168 . 95 . 192 . 1
DNS secondary:	168 . 95 . 1 . 1
SNTP:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	+8
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
IP Sharing Server Address:	210 . 59 . 163 . 198
<input type="button" value="OK"/>	

Step2: Prepare Trunk number for EZLoop® PSTN Gateway FX04/06



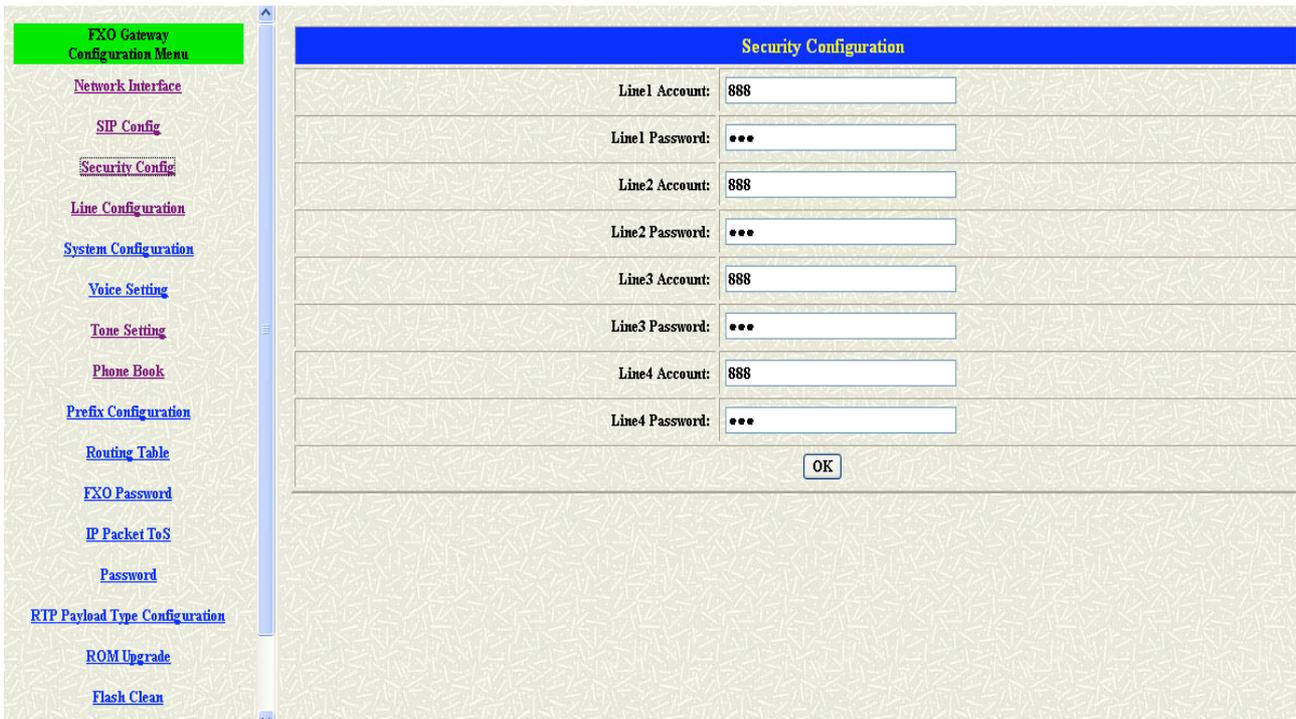
- In EZLoop® IP-PBX Enterprise SIP Server, prepare Trunk accounts for EZLoop® PSTN Gateway FX04/06. (There are 2 default Trunks, 888 and 889, so we use the default setting for this Example) For more information about Trunk page, please go to user's manual [CH3- Full Web Configurations](#).

Index	Trunk Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	888	rfc2833	Disable	Routed Mode	Modify	Delete
2	889	rfc2833	Disable	Routed Mode	Modify	Delete
3	none	none	Disable	Routed Mode	Modify	Delete
4	none	none	Disable	Routed Mode	Modify	Delete
5	none	none	Disable	Routed Mode	Modify	Delete
6	none	none	Disable	Routed Mode	Modify	Delete
7	none	none	Disable	Routed Mode	Modify	Delete
8	none	none	Disable	Routed Mode	Modify	Delete
9	none	none	Disable	Routed Mode	Modify	Delete
10	none	none	Disable	Routed Mode	Modify	Delete
11	none	none	Disable	Routed Mode	Modify	Delete

Step3: Setup Trunk for EZLoop® PSTN Gateway FX04/06

This Example, we have 1 register to EZLoop® IP-PBX Enterprise SIP Server with Trunk number 888. You can set the SIP Information of EZLoop® PSTN Gateway FX04/06 by its COM port or you can also login its WEB interface for configuration as below. Go to **SIP Config** page to set the EZLoop® PSTN Gateway FX04/06 as Proxy mode (or Gateway mode), Primary Proxy IP Address and line number (If you set the EZLoop® PSTN Gateway FX04/06 to Proxy mode, you should set line number for all of the line1 to line4. If you set the EZLoop® PSTN Gateway FX04/06 to Gateway mode, you can just only set line1 number). Go to **Security Config** page to input the registered account (If you set the EZLoop® PSTN Gateway FX04/06 to Proxy mode, you should set Account for all of the line1 to line4. If you set the EZLoop® PSTN Gateway FX04/06 to Gateway mode, you can just only set line1 Account). After configure, please press OK → Commit Data → Reboot System.

Set EZLoop® PSTN Gateway FX04/06 to Proxy Mode, and also set the line number for line1 to line4.



Set Account and Password for EZLoop® PSTN Gateway FX04/06

- If all of the above settings are correct, you can go to **Information** → **Subscriber** page to confirm the register status.



- There are also some other necessary configuration of EZLoop® PSTN Gateway FX04/06 to compatible with EZLoop® IP-PBX Enterprise SIP Server. But these settings do not exist in WEB interface, only exist in command line. Below is an example for command line.

Configuration of FXO

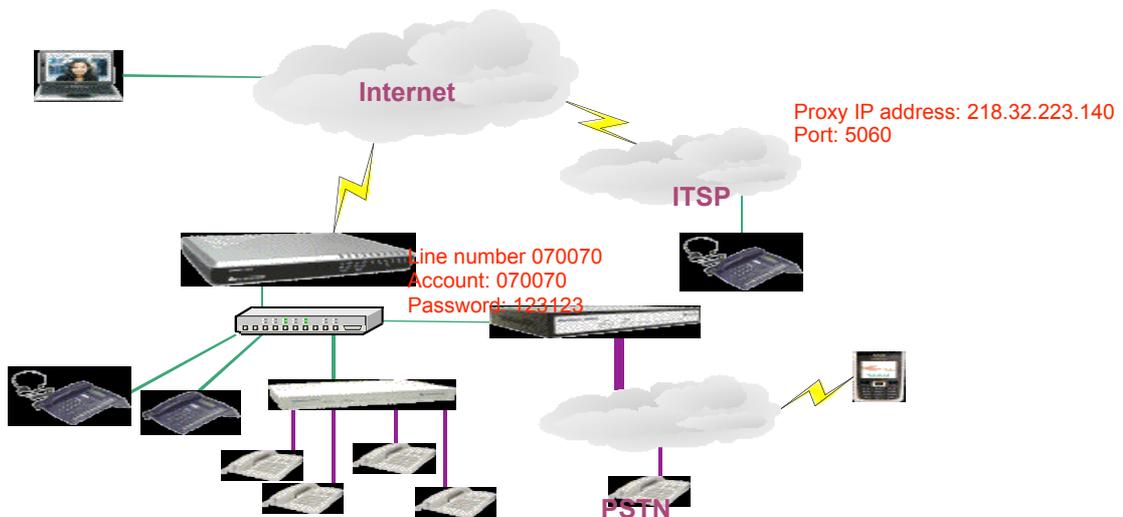
```
usr/config$ ifaddr -ip 192.168.13.68 -mask 255.255.248.0 -gate 192.168.13.254
(set IP address for EZLoop® PSTN Gateway FX04/06)
usr/config$ sip -px 192.168.13.95 (set EZLoop® PSTN Gateway FX04/06 to register EZLoop® IP-PBX
Enterprise SIP Server)
usr/config$ sip -line1 888 -line2 888 -line3 888 -line4 888 (set line number)
usr/config$ security -line 1 -name 888 -pwd 888
usr/config$ security -line 2 -name 888 -pwd 888
usr/config$ security -line 3 -name 888 -pwd 888
usr/config$ security -line 4 -name 888 -pwd 888 (set ID and Password)
usr/config$ sysconf -silence 0 (disable CNG function)
```

5.4 EZLoop® IP-PBX Enterprise SIP Server can register to ITSP as a SIP-Trunk.

EZLoop® IP-PBX Enterprise SIP Server can register to another ITSP as a SIP trunk. So that the Subscriber of ITSP can contact with EZLoop® IP-PBX Enterprise SIP Server and can call to ITSP.

Step1: Obtain register account

- We got an account from ITSP with “Line number 070070, Account: 070070, Password: 123123”. And the proxy address of ITSP is 218.32.223.140, port 5060. Maybe the ITSP also need to provide “**Realm**”, so you should also input Realm for the SIP Trunk, otherwise the call from EZLoop® IP-PBX Enterprise SIP Server to ITSP may be rejected. For more information about “**Realm**”, please contact with your ITSP.



Step2: Set EZLoop® IP-PBX Enterprise SIP Server to register ITSP.

- Input the necessary information in SIP Trunk Reg. page. In this example, our “**Realm**” is empty due to our ITSP does not need Authentication for incoming call. For more information about SIP Trunk, please go to user’s manual [CH3- Full Web Configurations](#).

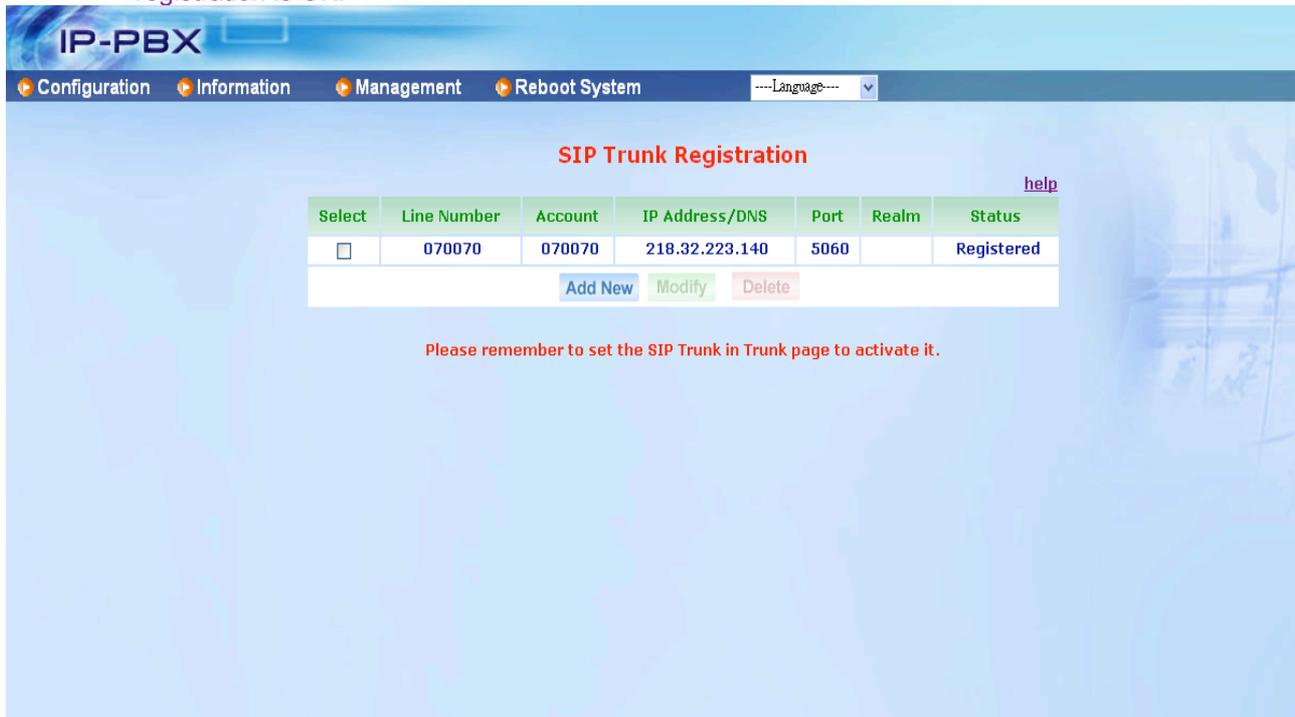
The screenshot shows the IP-PBX web interface. The top navigation bar includes Configuration, Information, Management, and Reboot System. The main content area displays the SIP Trunk Setting page with the following fields:

SIP Trunk Setting	
Enable:	<input checked="" type="checkbox"/>
Line Number:	070070
Account:	070070
Password:	••••••
IP Address/DNS:	218.32.223.140
Port:	5060
Realm:	
Status:	Registered

At the bottom of the form are two buttons: Apply and Cancel.

Step3: Confirm the register status of SIP Trunk

- Please confirm the register status. If the Status shows Registered, which means SIP Trunk registration is OK.



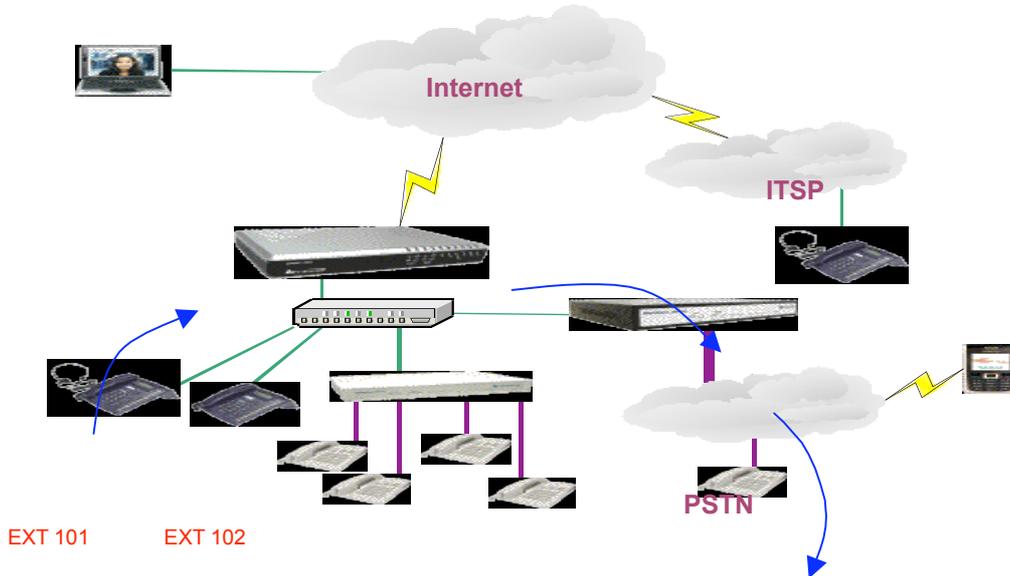
The screenshot shows the IP-PBX management interface. At the top, there is a navigation bar with 'IP-PBX' on the left and menu items: 'Configuration', 'Information', 'Management', and 'Reboot System'. A language dropdown menu is set to '---Language---'. The main content area is titled 'SIP Trunk Registration' with a 'help' link. Below the title is a table with the following data:

Select	Line Number	Account	IP Address/DNS	Port	Realm	Status
<input type="checkbox"/>	070070	070070	218.32.223.140	5060		Registered

Below the table are three buttons: 'Add New', 'Modify', and 'Delete'. At the bottom of the page, a red text message reads: 'Please remember to set the SIP Trunk in Trunk page to activate it.'

5.5 All of the Extensions can call out to local PSTN via EZLoop® PSTN Gateway FX04/06.

Now the FXO is registering to EZLoop® IP-PBX Enterprise SIP Server, and we hope the extensions can call out to local PSTN via the FXO gateway. The EZLoop® PSTN Gateway FX04/06 should connect with local PSTN line. We should set the routing table to let the EZLoop® IP-PBX Enterprise SIP Server route the call to EZLoop® PSTN Gateway FX04/06 if the called number is a local PSTN number.



Step1: Set Prefix route in Routing Table page

22198888

- Please Go to Routing Table Page to set Prefix route, so that the Extensions can dial to local PSTN 22198888 via EZLoop® PSTN Gateway FX04/06 (888). The setting just like below. For more information about Routing Table, please go to user's manual [CH3- Full Web Configurations](#).

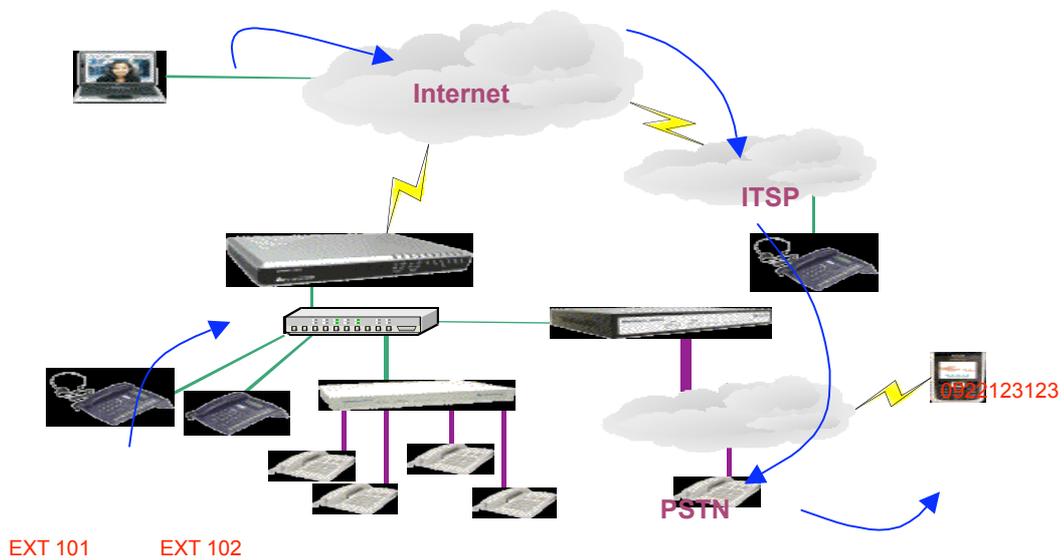
The screenshot shows the 'IP-PBX' web interface. At the top, there is a navigation bar with 'Configuration', 'Information', 'Management', and 'Reboot System' buttons. Below the navigation bar, there is a 'Language' dropdown menu. The main content area is a form for configuring a prefix route. The 'Prefix' field is set to '2', 'Digits Length' is '8', and 'Max Length' is '20'. The 'Destination' dropdown is set to '888', 'Secondary' is 'none', and 'Third' is 'none'. The 'Route Level' is set to 'R4' (checked). There are 'Apply' and 'Cancel' buttons at the bottom of the form.

	Primary	Secondary	Third
Destination:	888	none	none
Add:			
Drop:			
Route Password:			
Guest Allow:	<input type="checkbox"/>		
Route Level:	<input type="checkbox"/> R1 <input type="checkbox"/> R2 <input type="checkbox"/> R3 <input checked="" type="checkbox"/> R4		

Now the Extensions can dial to 22198888 via EZLoop® PSTN Gateway FX04/06.

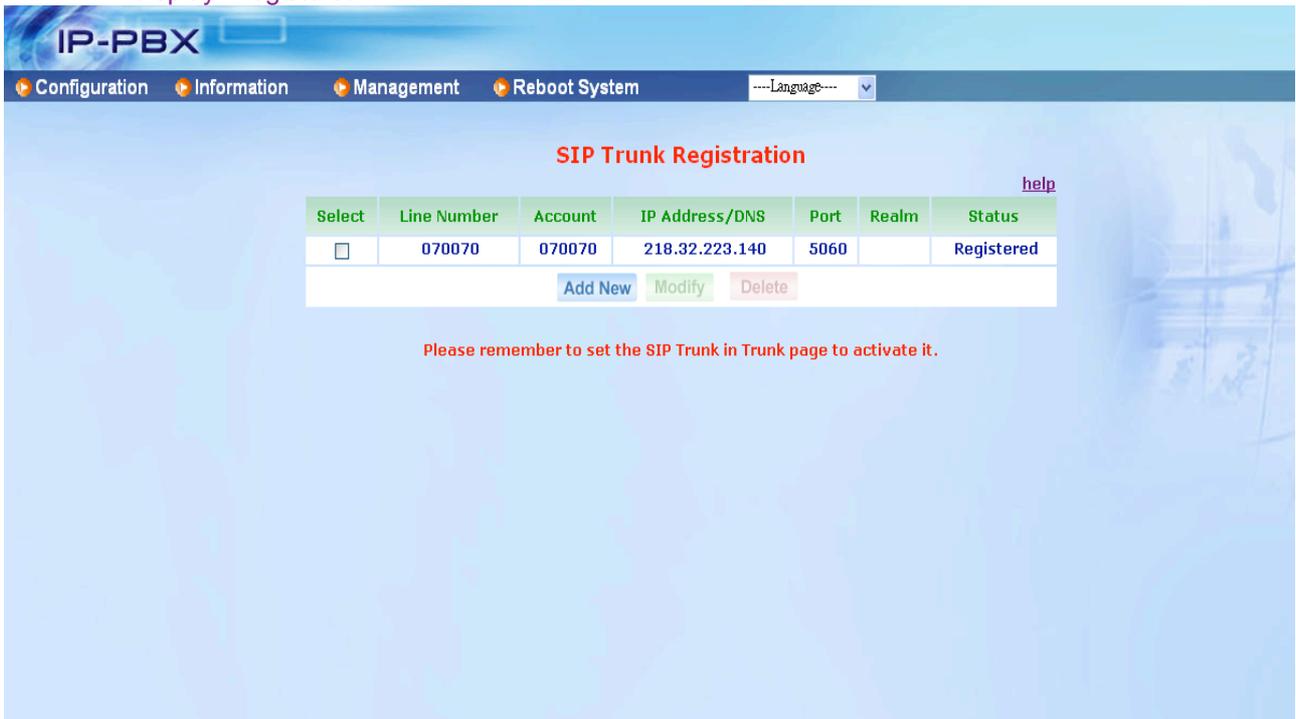
5.6 All of the Extensions can call out to Mobile Phone via ITSP.

Now we set EZLoop® IP-PBX Enterprise SIP Server to register an account 070070 to an ITSP, and we hope the outbound call with mobile phone number should be route to ITSP to reduce the cost.



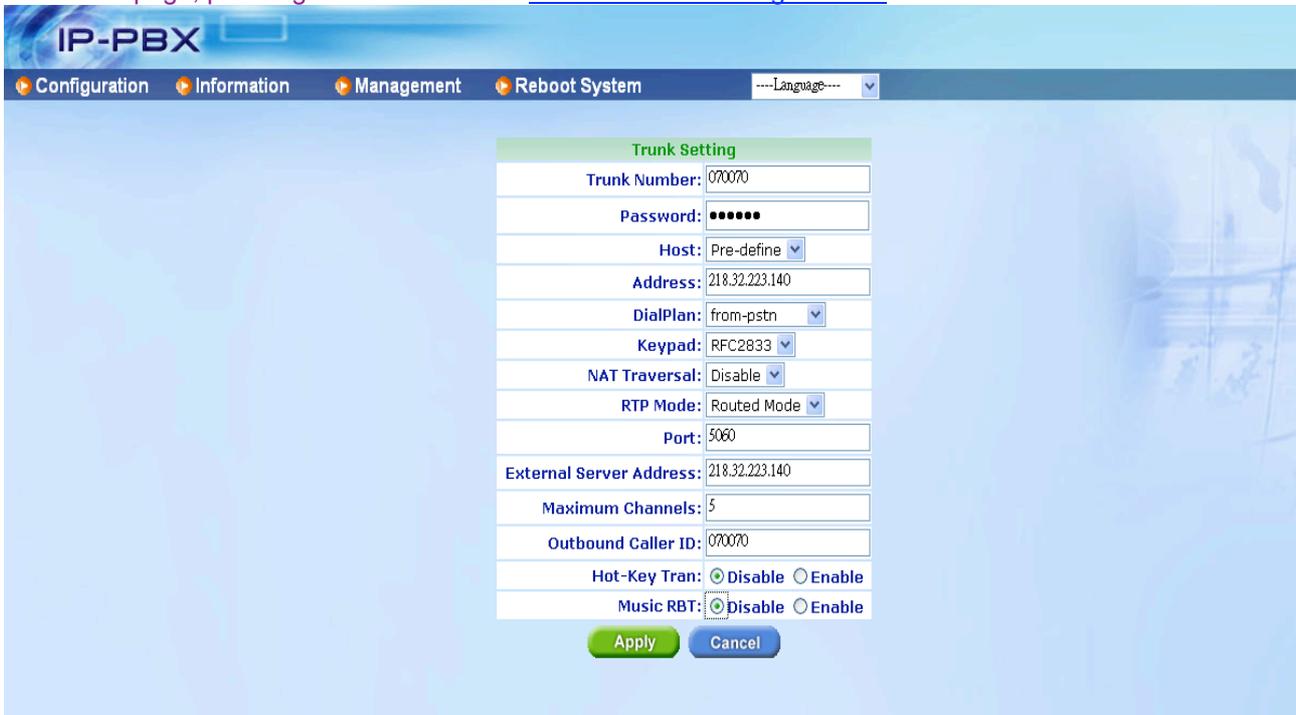
Step1: Confirm the EZLoop® IP-PBX Enterprise SIP Server register to ITSP successfully

- Please Go to SIP Trunk Page to confirm the registered status of SIP Trunk. The Status must display “Registered”



Step2: Set SIP Trunk ID in Trunk page to activate SIP Trunk.

- In SIP Trunk Page, we only set the EZLoop® IP-PBX Enterprise SIP Server to register ITSP. Now, we want to activate SIP Trunk (ITSP), so we should go to Trunk page to add a new Trunk for SIP Trunk (ITSP). For more information about the relationship between SIP Trunk page and Trunk page, please go to user’s manual [CH3- Full Web Configurations](#).



Activate SIP Trunk (ITSP) in Trunk Page.

IP-PBX

Configuration Information Management Reboot System Language

Trunk

[help](#)

Index	Trunk Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	888	rfc2833	Disable	Routed Mode	Modify	Delete
2	889	rfc2833	Disable	Routed Mode	Modify	Delete
3	070070	rfc2833	Disable	Routed Mode	Modify	Delete
4	none	none	Disable	Routed Mode	Modify	Delete
5	none	none	Disable	Routed Mode	Modify	Delete
6	none	none	Disable	Routed Mode	Modify	Delete
7	none	none	Disable	Routed Mode	Modify	Delete
8	none	none	Disable	Routed Mode	Modify	Delete
9	none	none	Disable	Routed Mode	Modify	Delete
10	none	none	Disable	Routed Mode	Modify	Delete
11	none	none	Disable	Routed Mode	Modify	Delete

In Trunk page, you should find there a new record 070070.

Step3: Set Prefix route in Routing Table page

- Please Go to Routing Table Page to set Prefix route, so that the Extensions can dial to Mobile Phone 0922123123 via ITSP. For more information about Routing Table, please go to user's manual [CH3- Full Web Configurations](#).

IP-PBX

Configuration Information Management Reboot System Language

Prefix:

Digits Length: Max Length:20

	Primary	Secondary	Third
Destination:	<input type="text" value="070070"/>	<input type="text" value="888"/>	<input type="text" value="none"/>
Add:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Drop:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Route Password:	<input type="text" value="****"/>	<input type="text" value="****"/>	<input type="text"/>
Guest Allow:	<input type="checkbox"/>		
Route Level:	<input type="checkbox"/> R1 <input type="checkbox"/> R2 <input type="checkbox"/> R3 <input checked="" type="checkbox"/> R4		

In this example, we set prefix to 0 and there is not limit for digits length (0). The first destination is ITSP and 2nd destination is EZLoop® PSTN Gateway FX04/06 (888). And we also set the Routed Password for this prefix route.

IP-PBX

Configuration Information Management Reboot System Language

Outgoing Call Rule [help](#)

Select	Prefix	Digits Length	Primary Dest.	Add	Drop	Guest Allow
			Secondary Dest.			
			Third Dest.			
<input type="checkbox"/>	2	8	888			Disable
<input type="checkbox"/>	0	0	070070 888			Disable

Add New Modify Delete

Incoming Call Rule

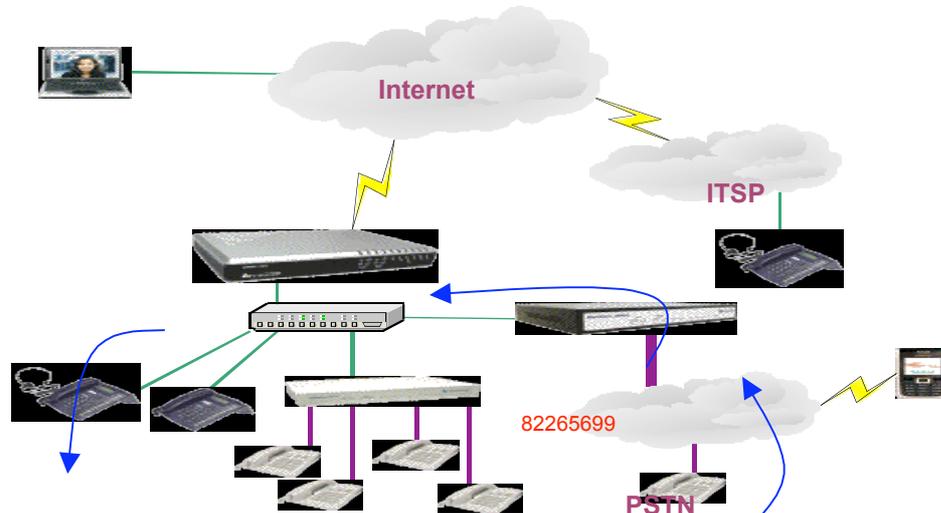
Select	Prefix	Digits Length	Add	Drop

Add New Modify Delete

Confirm the Outgoing Call Rule. Now the Extensions can dial to 0922123123 to reach mobile phone via ITSP. If extension called out with prefix number 0, the EZLoop® IP-PBX Enterprise SIP Server will play an announcement for route password, after input the correct password, then EZLoop® IP-PBX Enterprise SIP Server will dial to destination.

5.7 User in PSTN side should be able to contact with Extensions via EZLoop® PSTN Gateway FX04/06

We hope the EZLoop® IP-PBX Enterprise SIP Server can play as an Auto Attendant, so that the user in PSTN side can contact with the Extensions. In this example, the FXO gateway connect with local PSTN line (82265699), we hope the PSTN caller can dial to 82265699 then contact with Ext 101.



Step1: Set hotline function in your EZLoop® PSTN Gateway FX04/06.

- The default auto attendant number of EZLoop® IP-PBX Enterprise SIP Server is **999. So you should set hotline function of EZLoop® PSTN Gateway FX04/06. When EZLoop® PSTN Gateway FX04/06 got a PSTN incoming call, it should dial to **999 directly. In below picture, we set line1 to line3 hotline to **999 and we set line4 hotline to EXT 102.

The screenshot shows the 'Line Configuration' menu in the EZLoop PSTN Gateway configuration interface. The interface has a sidebar with various configuration options and a main table for line settings.

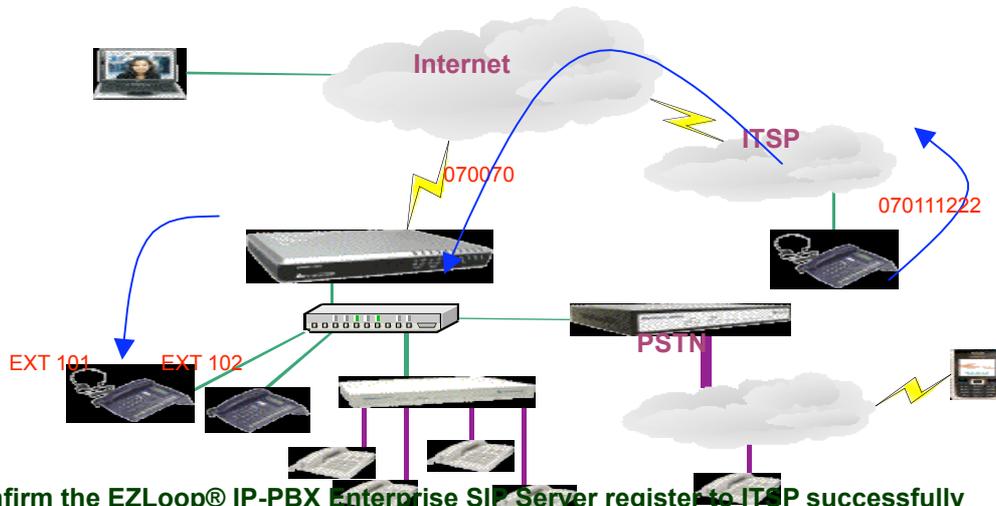
Line Configuration							
Line(LINE):	Type:	Hunting Group:	Hot Line:	Fwd. Type:	Fwd. Number:	Registration:	Status:
Line1(LINE):	FXO	1	**999	Disable	x	Registered	Ready
Line2(LINE):	FXO	2	**999	Disable	x	Registered	Ready
Line3(LINE):	FXO	3	**999	Disable	x	Registered	Ready
Line4(LINE):	FXO	4	102	Disable	x	Registered	Ready

Below the table is an 'OK' button.

Now, if FXO port1 got a PSTN incoming call, it will hotline to auto attendant, and caller will hear a greeting then dial extension number. If port4 got a PSTN incoming call, EZLoop® PSTN Gateway FX04/06 will dial to EXT102 directly.

5.8 User in ITSP side should be able to contact with Extensions

070111222 is a subscriber of ITSP , we hope 0701 11222 can also contact with extension of EZLoop® IP-PBX Enterprise SIP Server.



Step1: Confirm the EZLoop® IP-PBX Enterprise SIP Server register to ITSP successfully

- Please Go to SIP Trunk Page to confirm the registered status of SIP Trunk. The Status must display "Registered"

IP-PBX

Configuration Information Management Reboot System

SIP Trunk Registration

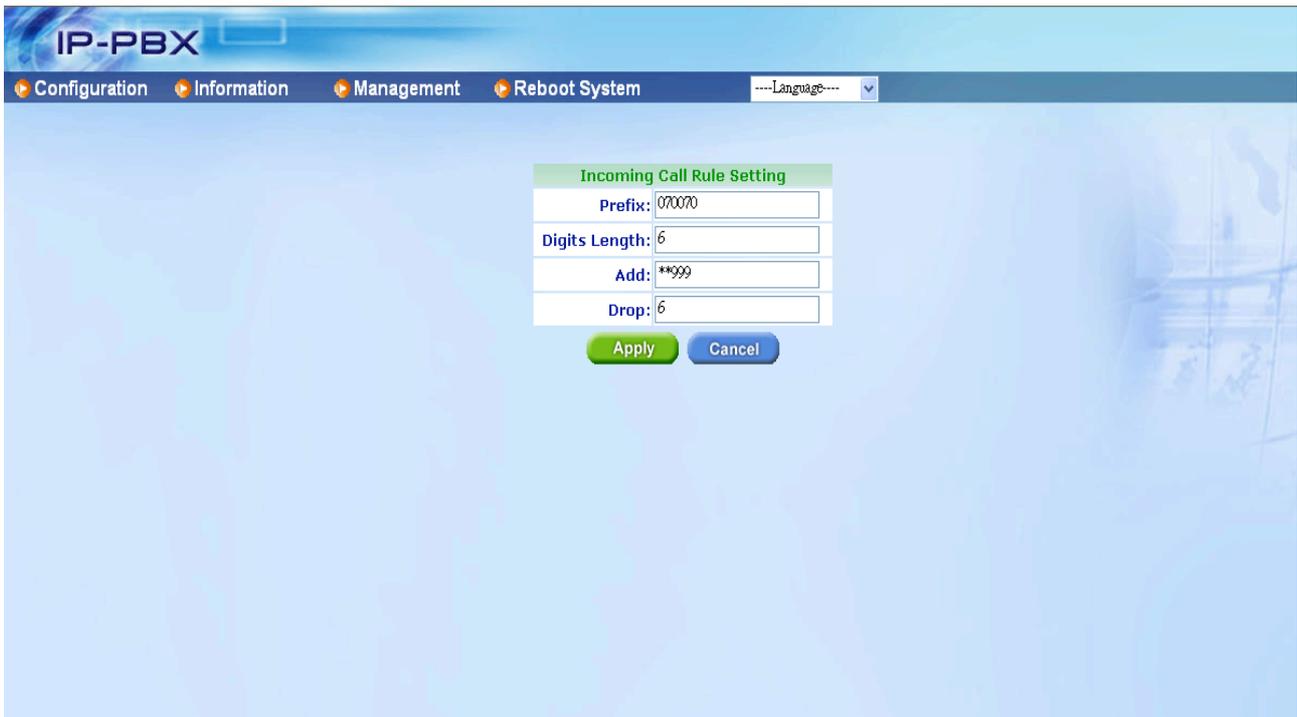
Select	Line Number	Account	IP Address/DNS	Port	Realm	Status
<input type="checkbox"/>	070070	070070	218.32.223.140	5060		Registered

Add New Modify Delete

Please remember to set the SIP Trunk in Trunk page to activate it.

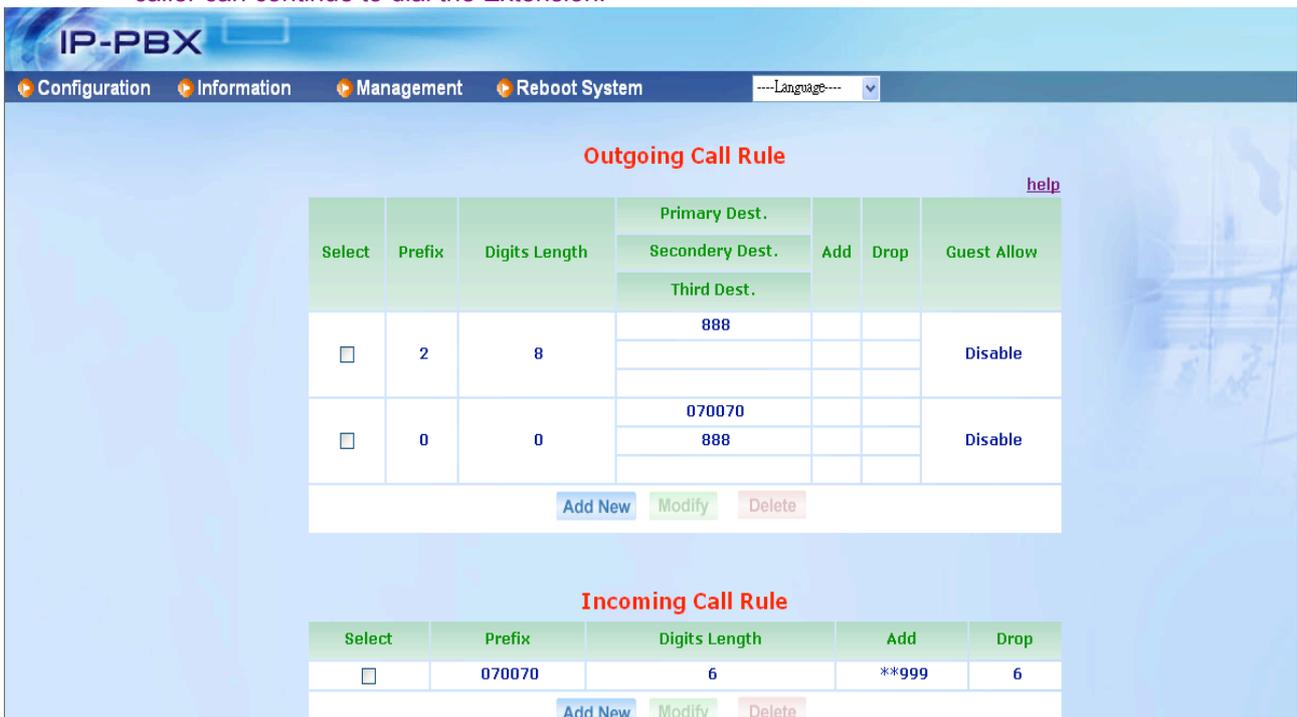
Step2: Set incoming call rule

- Please Go to Routing Table page to set incoming call. If EZLoop® IP-PBX Enterprise SIP Server got an incoming call with number 070070, it should play a greeting so that 070111222 can continue to dial EXT 101.



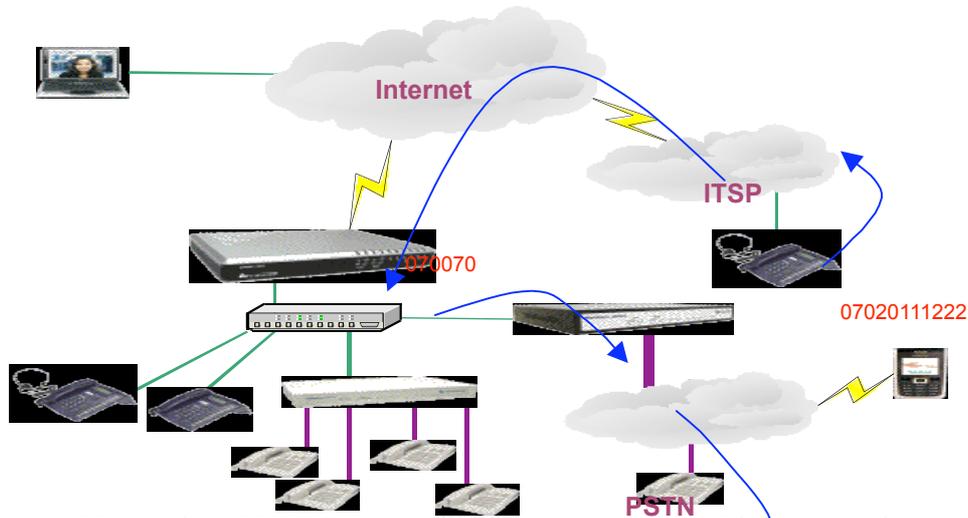
We set the incoming call rule with Prefix 070070 and Digits Length 6. When EZLoop® IP-PBX Enterprise SIP Server got a called number with 070070, it will drop 6 digits and add **999.

- Now, if EZLoop® IP-PBX Enterprise SIP Server got an incoming call from ITSP with number 070070, EZLoop® IP-PBX Enterprise SIP Server will dial to **999 (auto attendant). Then the caller can continue to dial the Extension.



5.9 User in ITSP side can call out to local PSTN via EZLoop® PSTN Gateway FX04/06.

Now, 0701 11222 can reach auto attendant. We hope 0701 11222 can dial to local PSTN 22198888 via EZLoop® PSTN Gateway FX04/06.



Step1: Confirm the EZLoop® IP-PBX Enterprise SIP Server register to ITSP successfully

- Please Go to SIP Trunk Page to confirm the registered status of SIP Trunk. The Status must display "Registered"

22198888

Select	Line Number	Account	IP Address/DNS	Port	Realm	Status
<input type="checkbox"/>	070070	070070	218.32.223.140	5060		Registered

Please remember to set the SIP Trunk in Trunk page to activate it.

Step2: Enable Guest Allow

- In the prefix route of outgoing call rule, we should enable Guest Allow, so that the user can redial destination.

IP-PBX

Configuration Information Management Reboot System ---Language---

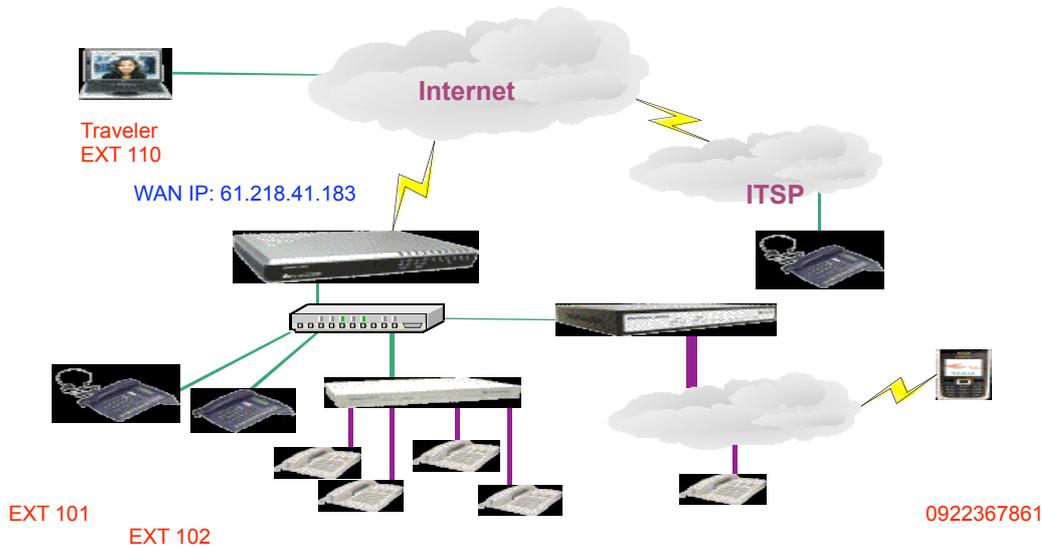
Prefix: 2
 Digits Length: 8 Max Length:20

	Primary	Secondary	Third
Destination:	888	none	none
Add:			
Drop:			
Route Password:			
Guest Allow:	<input checked="" type="checkbox"/>		
Route Level:	<input type="checkbox"/> R1 <input type="checkbox"/> R2 <input type="checkbox"/> R3 <input checked="" type="checkbox"/> R4		

Apply Cancel

User can reach the auto attendant (**999) now, because we already set incoming call rule in Routing Table page. Now, they can redial to 22198888 because we enable Guest Allow. For more information about Guest Allow, please go to user's manual [CH3- Full Web Configurations.](#)

5.10 Traveler can call back to EXT, and Traveler can also call to local PSTN and Mobile phone number.



Step1: Create account for the Traveler

- The Traveler has a business trip and she is using the customer's network. Maybe she is under "Private IP". We should enable "NAT Traversal" for her. So that she can contact with the other Extension and she can also use the routing table of EZLoop® IP-PBX Enterprise SIP Server.

The screenshot shows the 'IP-PBX' web interface. At the top, there are navigation tabs: 'Configuration', 'Information', 'Management', and 'Reboot System'. A language dropdown menu is on the right. The main content area displays the 'Extension Setting' form for extension 110. The form fields are as follows:

Extension Setting	
Extension Number:	110
Password:	***
Call Group:	1
Pickup Group:	1
DialPlan:	ext+allroutes
Keypad:	RFC2833
NAT Traversal:	Enable
RTP Mode:	Routed Mode
MailBox:	Disable

At the bottom of the form, there are two buttons: 'Apply' and 'Cancel'.

Step2: Set register account for Traveler

- The Traveler may use a USB phone or Soft Phone to contact with the other Extensions. In the settings of Soft Phone, she need to set the proxy address to: 61.218.41.183 (it is the WAN IP of EZLoop® IP-PBX Enterprise SIP Server), and she also needs to set the line number/ account/ and password for her Soft Phone. Below is an example.



- If all of the above settings are correct, you can go to **Information** → **Subscriber** page to confirm the register status.

IP-PBX

Configuration Information Management Reboot System

Subscriber

Index	Phone Number	IP Address	NAT Traversal	Mail Address
1	101	192.168.13.14	Disable	eason@mail.welltech.com.tw
2	102	192.168.13.13	Disable	-none-
3	103	192.168.13.67	Disable	-none-
4	104	192.168.13.67	Disable	-none-
5	105	192.168.13.67	Disable	-none-
6	106	192.168.13.67	Disable	-none-
7	107	-none-	Disable	-none-
8	108	-none-	Disable	-none-
9	109	-none-	Disable	-none-
10	110	61.218.41.195	Enable	-none-
11	888	192.168.13.68	Disable	-none-
12	889	-none-	Disable	-none-
13	070070	218.32.223.140	Disable	-none-

Now, the Traveler can contact with other Extension. If the called party is busy or no answer, EZLoop® IP-PBX Enterprise SIP Server will play an announcement to indicate the called party's status. The Traveler can also dial to local PSTN and Mobile phone due to you already set Routing Table.