Dynamix IP PBX-100

User's Manual

Version: ippbxUM. 100

CH1. Overview

The Dynamix IP PBX-100 is the next generation al-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, cal transfer, cal 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 case.

Customers can select different suite and optional products to meet their request. To Integrate with Dynamix DW can provide PSTN access function, LAN IP Phone and Dynamix DW can provide extensions. With flexible and full functionality, Dynamix IP PBX-100 can give a compete transition from traditional PABX to the new generation IP-PBX.

1.1 Specifications

> **Protocol**

- SIP (Session Initiation Protocol)

> **Call Features**

- Authentication
- Automated Attendant
- Cal Transfer (CPE based)
- Blind Transfer (CPE based)
- Cal Forward on Busy (CPE based)
- Cal Forward on No Answer (CPE based)
- Cal Forward Unconditional (CPE based)
- Cal Hold/Retrieval (CPE based)
- Cal Routing
- Cal Waiting (CPE based)
- Caller ID
- Do Not Disturb (CPE based)
- Flexible Extension Logic
- Music On Hold
- Music On Transfer
- Cal Pickup (Global Cal Pickup)
- Three-way Conference (IP PHONE)
- Time and Date
- Trunking (Dynamix DW)
- VoIP Gateways (Dynamix DW)
- Voice Mail to e-mail

> **Codecs**

- G.711 (A-Law& //-Law)

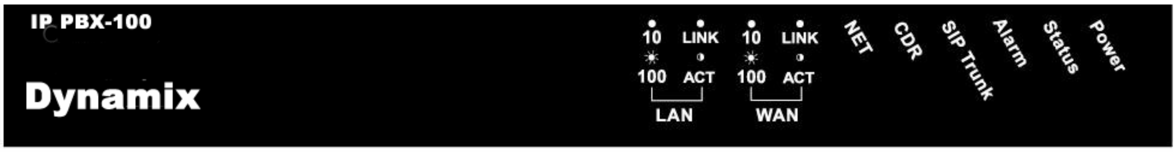
- G.729

> **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 and DHCP mode

1.2 Hardware Overview

1.2.1 Front Panel and LED Indication



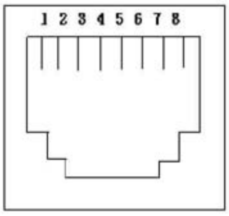
Power: Light on when IP PBX-100 is powered on.

- **Status:** Light on when system is ready.
- **Alarm:** Light Flash when system has problems.
- **SIP Trunk:** Light on when IP PBX-100 successfully registered to all of the enabled SIP Trunks; Light flash when IP PBX-100 failed to register to one of the enabled SIP Trunks; Light off when there is no SIP Trunk has enabled.
- **CDR:** IP PBX-100 can output Call Detail Records to external computer. User has to execute CDR program on computer, when IP PBX-100 is ready to connect with CDR server and output data, this indication will light on.

  Note:
  - This Function is still not released in version app_100.app.
  - It will be released in next version.
  - CDR Function can only work in local area network. Pease prepare the CDR server under LAN.
- **NET:** Display Network status. If WAN port of IP PBX-100 is under Fixed IP mode, LCD will light on. If WAN port is under DHCP mode, and IP PBX-100 succeeds in getting IP, LED will be flashing. If WAN port is under DHCP mode, and IP PBX-100 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.
  - 1/1: 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.
  - 1/1: Light on when network rate is 100 Mb/s, and light off when network rate is 10 Mb/s.

1.2.2 Back Pane

- **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 ePBX-1

2.1 Step 1

Connect **LAN** port of IP PBX-100 with PC via crossover cable or connect with Switch/ Hub via straight through cable.

2.2 Step 2

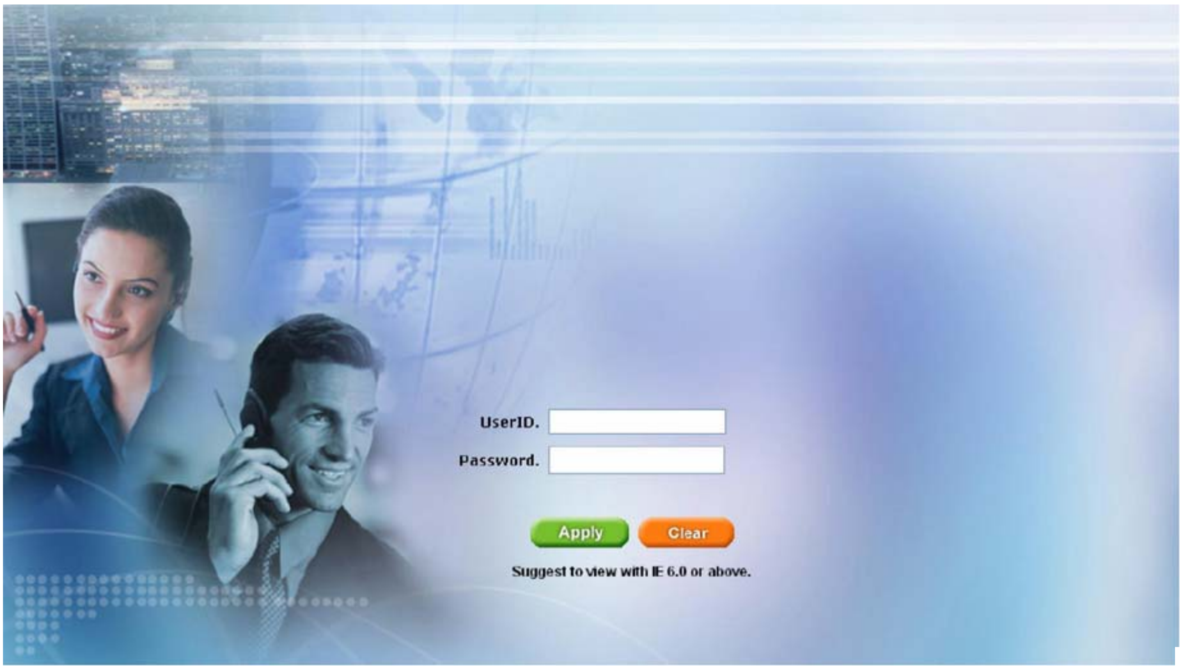
Prepare one computer, and change the IP address to be **192.168.123.12x with subnet mask 255.255.255.**

2.3 Step 3

Open browser and link to default LAN IP address of IP PBX-100 **"192.168.123.123".**

2.4 Step 4

Login IP PBX-100 with default user ID: **"root",** and **no password.** After login IP PBX-100, user can start to configure basic and essential configurations.



2.5 Step 5: To configure basic and essential configurations

To make IP PBX-100 work have to make some basic and essential configurations, which including 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.

WAN

Mode: Select IP PBX-100 WAN port network mode to be Fixed IP or DHCP.

IP Address/NETMASK/GATEWAY: If user has set IP PBX-100 to be fixed IP mode. User need to input IP address/Subnet Mask/ Default Gateway.

DNS: Input DNS address.

Mac: Mac address of IP PBX-100 WAN port. The Mac address cannot be modified.

• **LAN**

IP Address: Input IP address for LAN port of IP PBX-100.

NETMASK: Input Subnet Mask for LAN port of IP PBX-100.

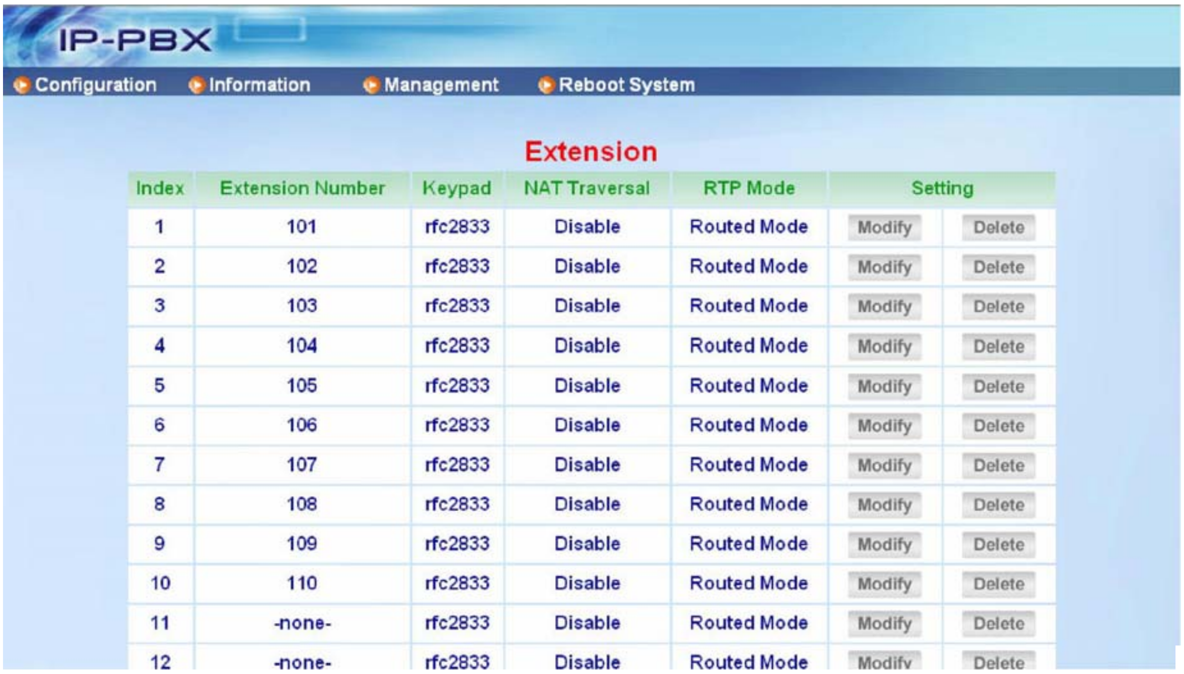
Mac: Mac address of IP PBX-100 LAN port. The Mac address cannot be modified.

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

2.5.2 Extension Configuration

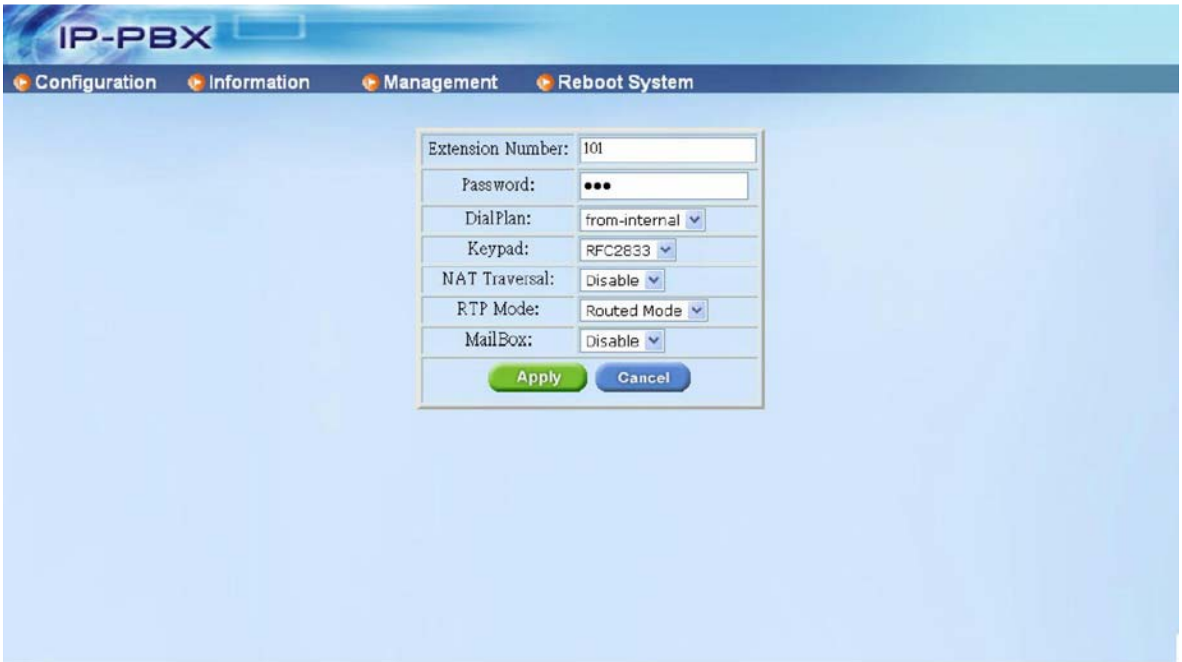
User has to set Extension account for other device to register on IP PBX-100.

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

After press Modify can input detail setting for Extension.



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 IP PBX-100.

Dial Plan: Define the dialing pan for Extension.

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 Extension is behind NAT but register to IP PBX-100 which is under

Public IP, you should enable NAT Traversal otherwise the registration should be fail.

RTP Mode: User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode.

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

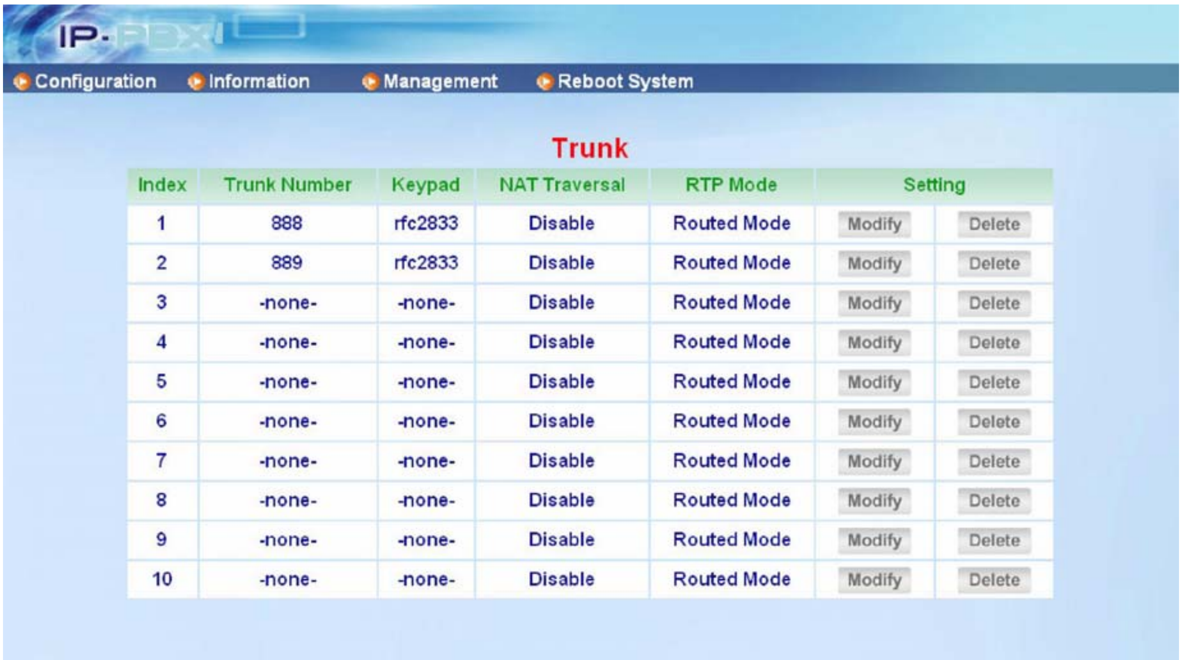
Note:

- For more information about Extension setting, please refer to **3.Ful Web Configurations**

2.5.3 Trunk Configuration

User has to set Trunk account for Trunk (FXO device, e.g. Dynamix DW) to register to IP PBX-100 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 1 sets Trunks. User can press **Modify** to add new Trunk or modify



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

configured Trunk data. Press **Delete** will delete the specified Trunk **■PBXi**

After press Modify can input detail setting for Trunk **IP-PBXi**

- **Trunk Number:** Assign the number of Trunk. This number is also the register name for Trunk device.
- **Password:** Assign the register password for device to register on IP PBX-100.

- **Host:** Setting the Trunk to a Dynamic address or a specified IP or FQDN.
- **DialPlan:** Define the dialing pan for Trunk.
- **DTMF:** User can select DTMF type to be RFC2833, In-band, or SIP-Info.
- **NAT Traversal:** If the Extension is behind NAT but register to IP PBX-100 which is under Public IP, you should enable NAT Traversal otherwise the registration should be fail.
- **RTP Mode:** User can choose for two type of RTP mode, one is Routed Mode another is Direct Mode.
- **Port:** You can use this to define the SIP signal port if you want to listen on a nonstandard SIP signal port. (The default SIP signal port is 5060)
- **External Server Address:** This field will allow you to set the domain in the SIP From URI.
- **Maximum Channels:** This will limit the maximum channels for this Trunk.
- **Outbound Caller ID:** Set the Caller ID if the cal is going via Trunk

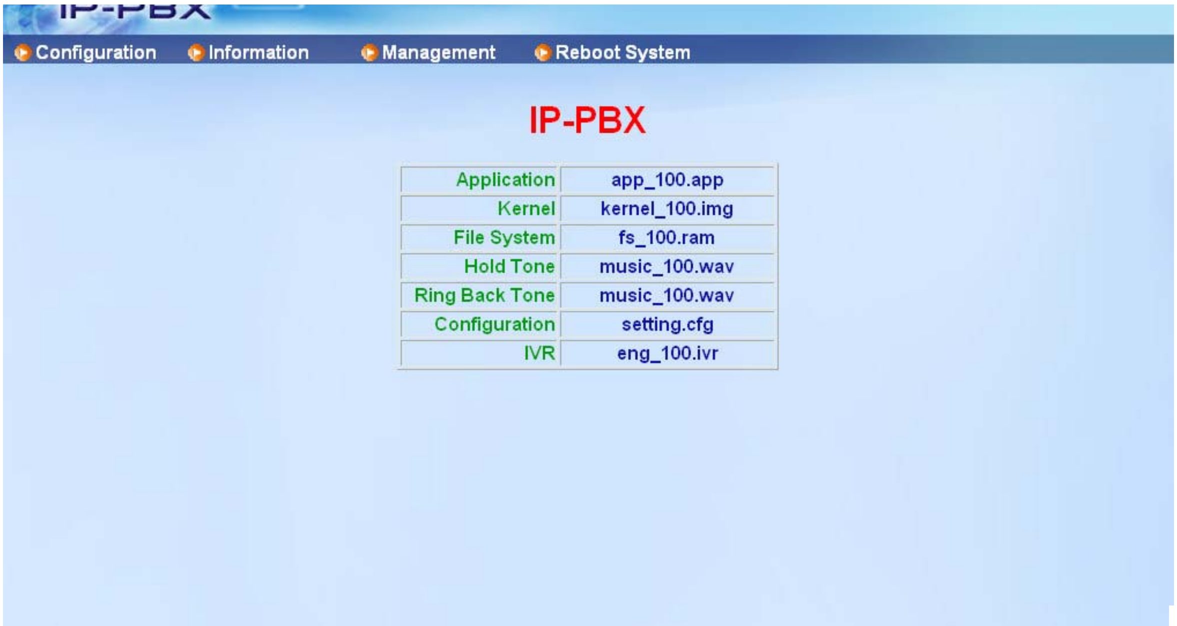
Press Apply to save configuration, or press Cancel to quit configuration.

Note:

- For more information about Trunk setting, please refer to **3.Ful WEB Configurations.**

CH3. Full Web Configurations

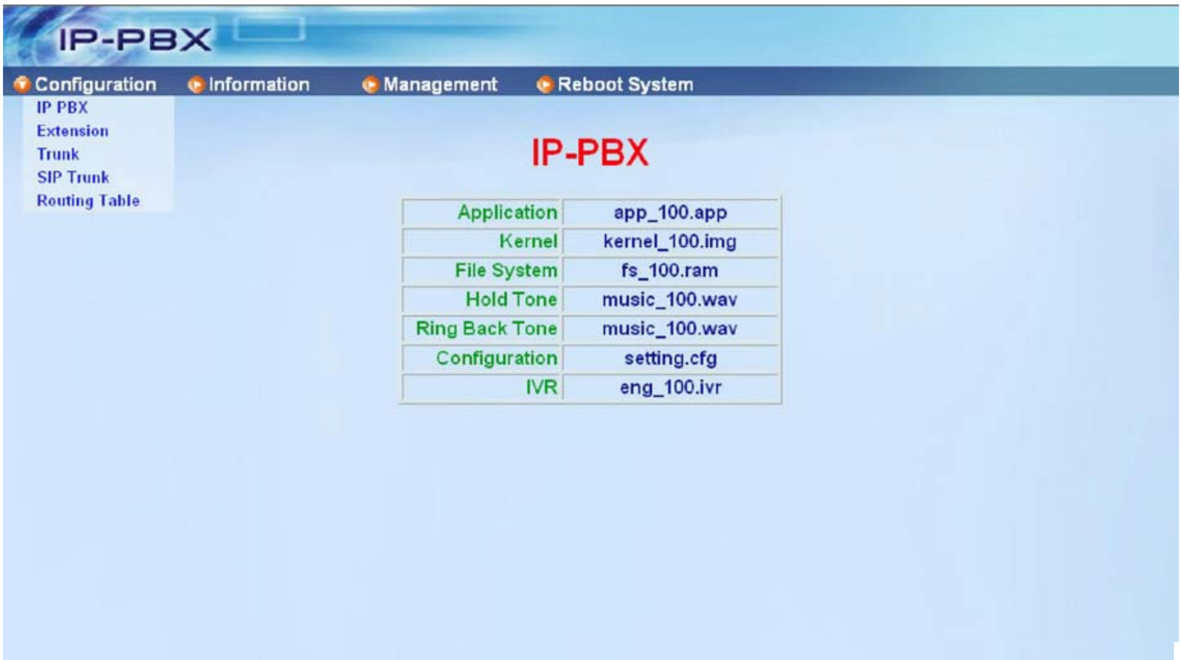
After Login IP PBX-100 will see screen as below, and there are four main categories, user can click on each category to extend detail items.



- Configuration: Include al telephony configuration of IP PBX-100.
  - IP PBX
  - Extension
  - Trunk
  - SIP Trunk
- • Routing Table Information: To show related information.
- • Subscriber Management: Include al system management of IP PBX-100.
  - Network
  - Time Zone
  - User Account
  - Firmware Upload
  - Music Upload
  - Import Setting
  - Export Setting
- Reboot System: To reboot system of IP PBX-100.

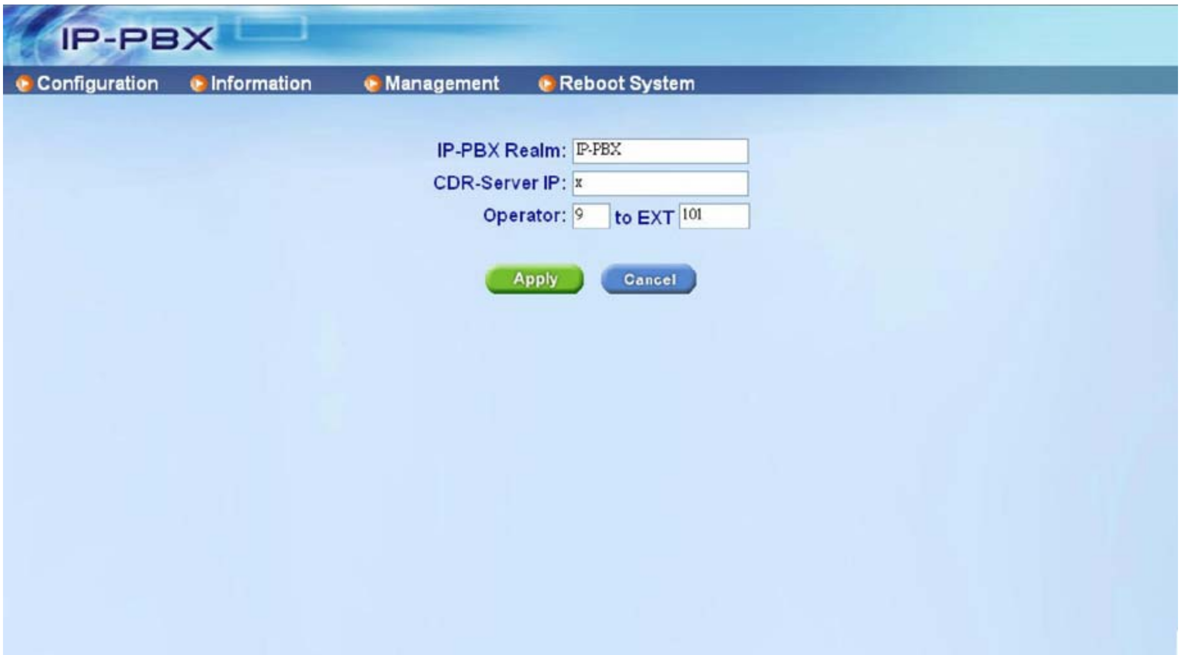
3.1 Configuration

User can set IP PBX-100 telephony configuration under Configuration category.



3.1.1 IPPBX

Enter **Configuration** -> **IP PBX** to configure PBX data.



IP-PBX Realm: Configure Realm of IP PBX-100. This parameter is essential when there is more than one IP PBX-100, and user wants to have inter-case between IP PBX-100s. Pease refer to SIP Trunk configuration.

CDR-Server IP: Configure CDR server IP address for IP PBX-100 to output CDR.

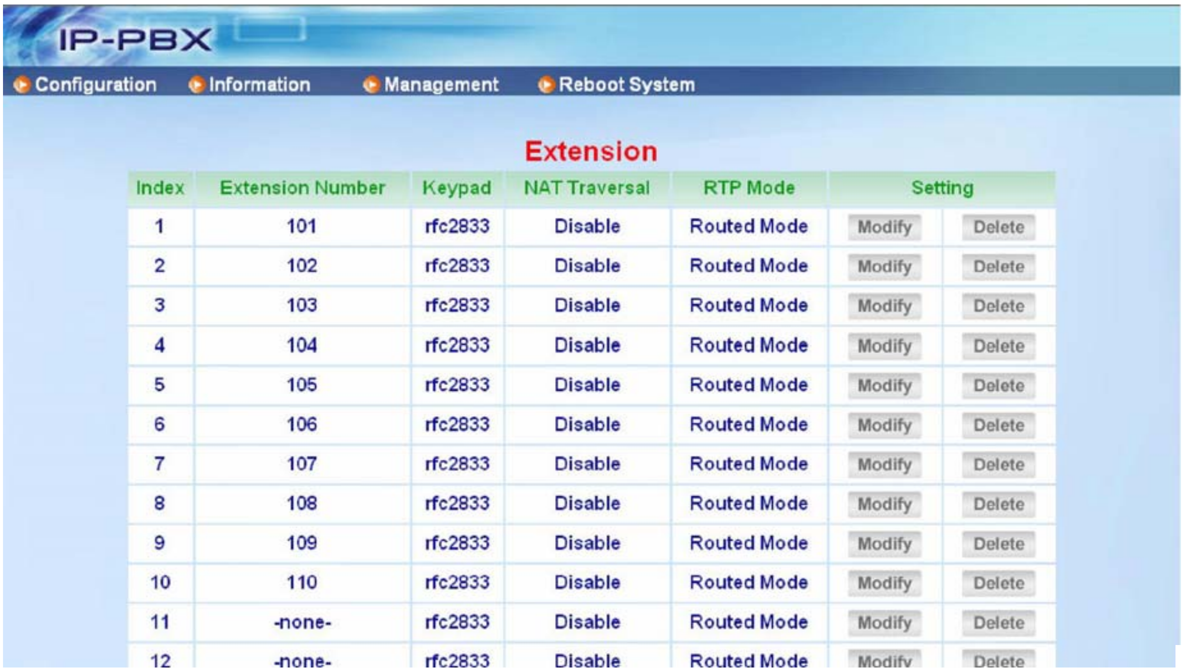
Operator: Assign operator access code. When care dial in ePBX, press this assigned

code can reach operator.
to EXT: Assign operator 's extension number. When caller press operator's access code, ePBX will transfer this cal to the assigned Extension.

3.1.2 Extension

User has to set Extension account for other device to register on IP PBX-100.

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

After press Modify can input detail setting for Extension.

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 IP PBX-100.

DialPlan: Define the dialing pan 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 case for this extension. In phase 1, there is only one dial pan called [from-internal].

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 IP PBX-100 on public network, then user has to enable this function. Enable NAT Traversal to force IP PBX-100 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 oftheIP PBX-100.

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 IP PBX-100 if the IP PBX-100 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 cal, system will send this voice mail to the specified e-mail address. **Note:**

- There is no setting of SMTP in IP PBX-100 now.
- But there is a build-in software called "Mail-IP" in IP PBX-100, it is an automatically send mail software.
- We will implement the SMTP in the next version.
- If the IP PBX-100 got a new message, it will send the message to the user by email then delete the message from IP PBX-100 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, e.g. Dynamix DW) to register to IP PBX-100 or set some necessary configuration for SIP trunk (For more application, please go to). Enter **Configuration Trunk** to configure Trunk data.

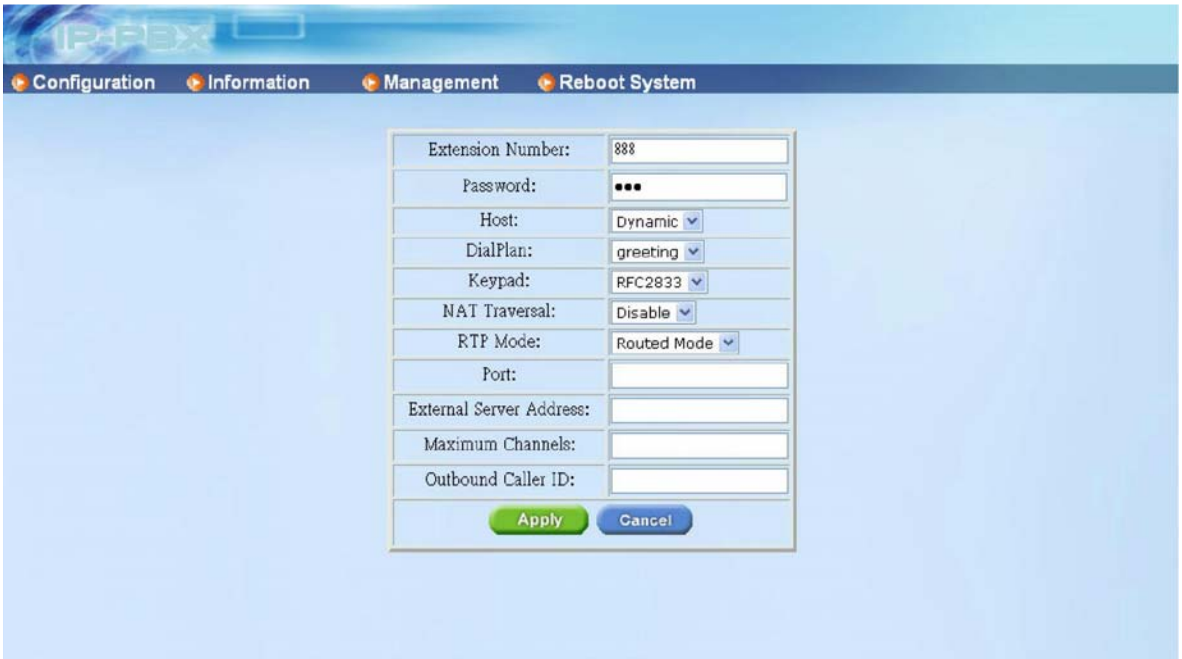
On screen will show 1 sets Trunks. User can press **Modify** to add new Trunk or modify

Trunk

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

configured Trunk data. Press **Delete** will delete the specified Trunk.

After press Modify can input detail setting for Trunk



Example 1: Set Trunk for FXO gateway

Example 2: Set Trunk ID for SIP Trunk

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 IP PBX-100.

Host: Setting the Host to Dynamic will require the trunk to register the IP PBX-100 so that the IP PBX-100 knows how to reach the trunk. You can also set the Host to an IP address or FQDN if you set the Host to Address. There will be a field called [Address] appear when you choose Host to Address. This limits only where you pace case to, as the user is allowed to place case from anywhere.

DialPlan: Define the dialing pan 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 case for this extension. In phase 1, there is only one DiaPan called [greeting].

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 IP PBX-100 on public network, then user has to enable this function. Enable NAT Traversal to force IP PBX-100 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 oftheIP PBX-100.

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 IP PBX-100 if the IP PBX-100 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. (The default SIP signal port is 5060)

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 case could go via this Trunk. Default is no limited.

Outbound Caller ID: Some service provider will require the correct registered caller ID if it got an incoming cal. Default the IP PBX-100 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 ePBX's IP address, but some ITSP may reject this cal 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 cal is going via the ePBX to the Destination (Trunk) to avoid such security issue.
- If you set a Dynamix DW 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 configure Extension Number, Password, Host: Dial Plan, 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 Appendix A: Application between Dynamix DW CPE device and ePBX-1.**

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

3.1.4 SIP Trunk

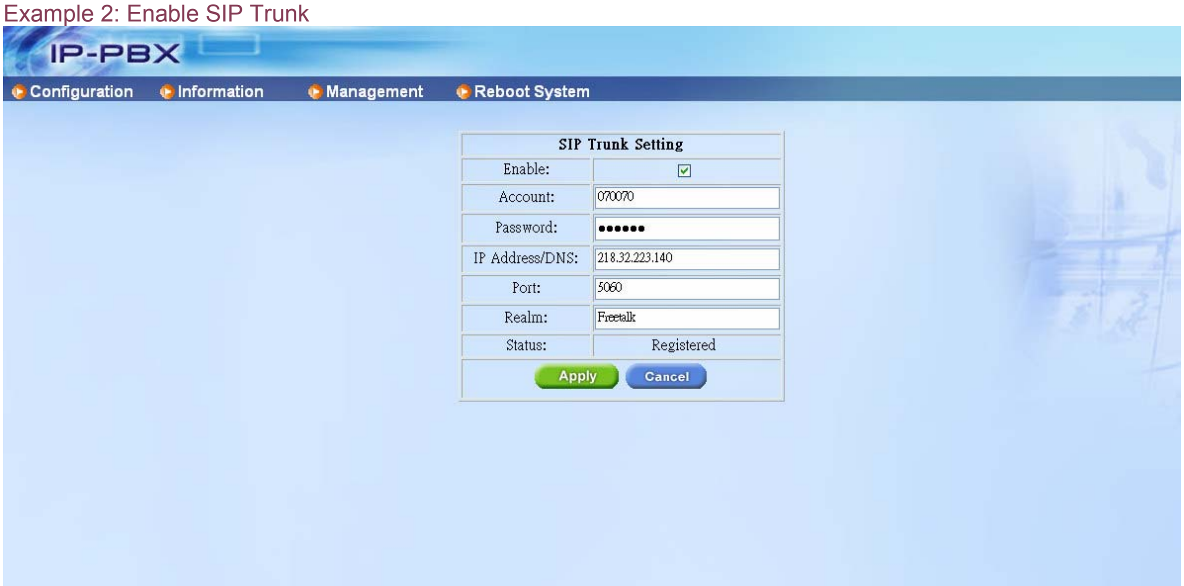
SIP Trunk is for IP PBX-100 **to register to other systems only,** such as ITSP or another IP PBX-100.

On screen of SIP Trunk will show al 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.

Enter **Configuration SIP Trunk-Add New** to configure IP PBX-100 register to ITSR ITSP will provide related account information for IP PBX-100 to register. Pease input the data here.

Example 1: Disable SIP Trunk

Example 2: Enable SIP Trunk



Enable: Check to enable this SIP Trunk.

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 IP PBX-100.

Note:

- When a cal was sent form IP PBX-100 to a remote SIP-Trunk, the SIP-Trunk may attempt to authenticate the "cal". So IP PBX-100 should reply the correct Account ID and Password. How does the IP PBX-100 know which ID and Password it should send? When the cal is going to SIP-Trunk via ePBX, the SIP-Trunk may reply a 407 code which will contain a parameter called "Realm" for authentication, IP PBX-100 will re-send the cal 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 IP PBX-100, you may hope those IP PBX-100 could cal to each other. You should set the Extension to let those IP PBX-100 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 Appendix A: Application between Dynamix DW CPE device and IP PBX-100.**

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

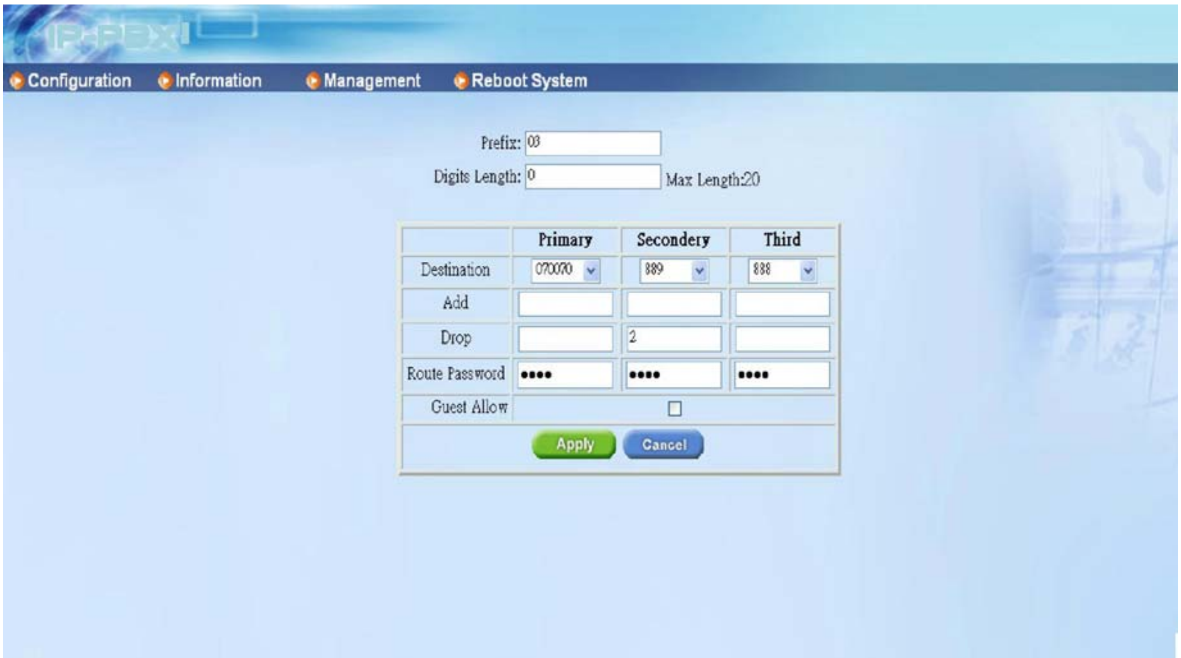
Routing Table is to set routing rule of IP PBX-100. There are two directions to set rules: Outgoing Cal Rule means from subscriber (Extension or Trunk registered on IP PBX-100) to cal out. Incoming Cal Rule means cal from other non-subscriber device to IP PBX-100.

Enter **Configuration ■* Routing Table-select direction and press Add New** to set routing table.

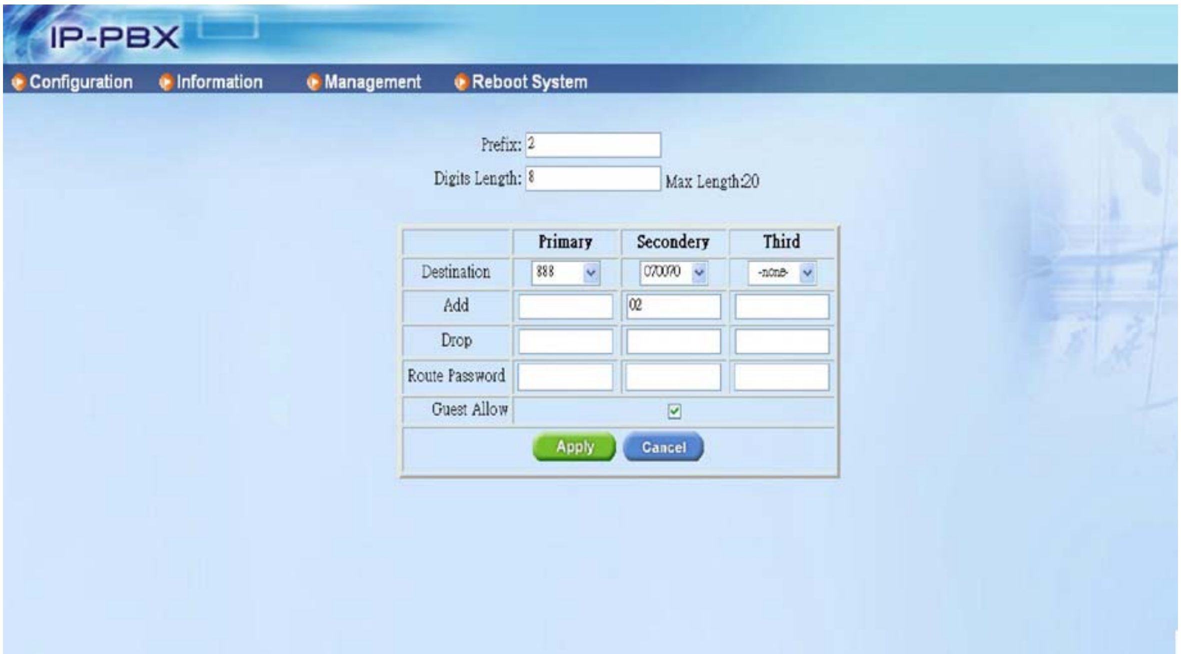
3.1.5.1 Outgoing Call Rule

Outgoing Call Rule means from subscriber (Extension or Trunk registered on IP PBX-100) to call out.

Example 1: Routing record with prefix 3 and no limit for Digits Length. Enable Route Password and Drop function.



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



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, pease set this parameter as 0. The maximum length is 20.

Note:

- **If you set the Digits Length as a specific value, such as 1, the dialed number should ful match to 1,** 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 IP PBX-100 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 page of Trunk 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 IP PBX-100, IP PBX-100 will skip this priority and route cal to next priority immediately without trying. If the Trunk was setting to Address in the Host field, but the Address is not reachable, IP PBX-100 will try three times then route cal to next priority.

Drop: To drop **specified length of number.** For example, you set 2 here and the called number is 03123, the IP PBX-100 will drop 03 then send 123 as outgoing number. **Add:** To add assigned number. For example, you set 02 here and the called number is 03123, the IP PBX-100 will add 02 then send 0203123 as the called number.

Note:

- If you set both of the Drop and Add, IP PBX-100 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 cal matches the routing rule.

002912345678–912345678(Drop 3 digits) –0912345678(Add 0)

Finally, IP PBX-100 will send 0912345678 to Trunk ID 888.

Route Password: Set password here so the IP PBX-100 will request password before sending the cal 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 IP PBX-100 is **999) first then send cal 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 Appendix A: Application between Dynamix DW CPE device and IP PBX-100.

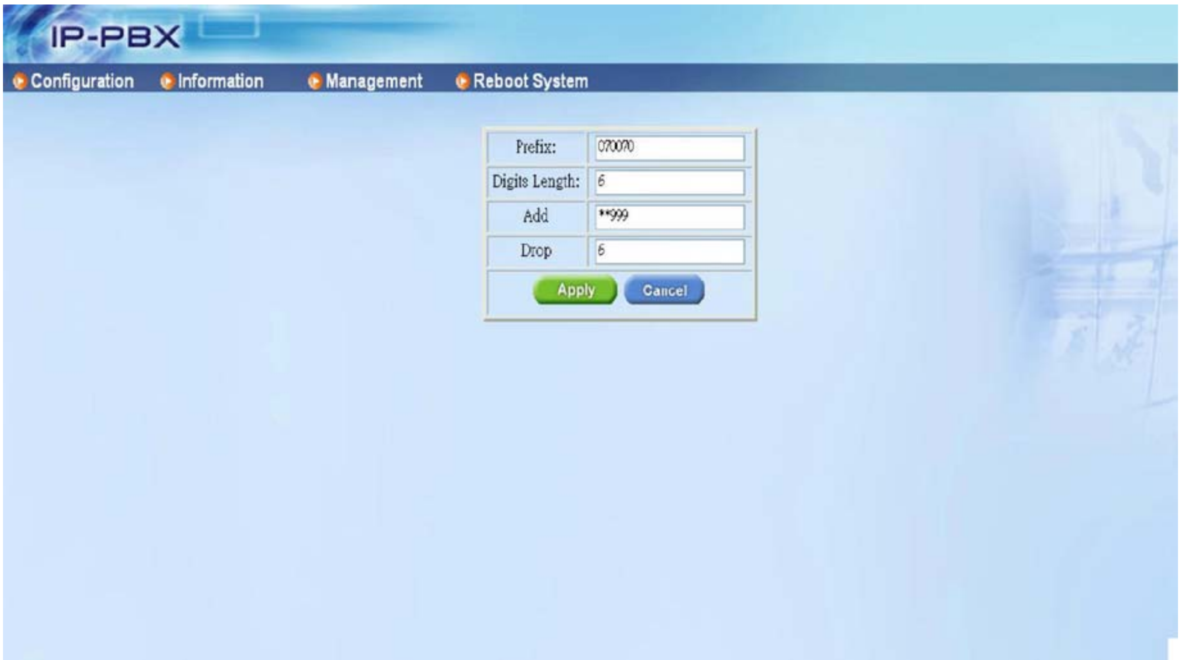
Press Apply to save configuration, or press Cancel to quit configuration.

3.1.5.2 Incoming Call Rule

Incoming Cal Rule means cal from other non-subscriber device to IP PBX-100.

For example, you set the IP PBX-100 to register an ISTP as a SIP Trunk, so your IP PBX-100 could be as an "Extension" of ITSP. The other subscriber of ITSP could cal to IP PBX-100 by the registered line number, when the IP PBX-100 got an incoming cal which is not its own subscriber, what will the IP PBX-100 do? The IP PBX-100 will perform the following action based on the "Incoming Cal Rule".

The above example means the ePBX-10 got a called number 7070, which was not sending from a "Extension", ePBX-10 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.



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 1, the dialed number should ful match to 1.** or you can set the Digits Length to 0 to ignore the digits length.

- If the called number 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.

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 IP PBX-100 will drop 070070 then send 101 as called number.

Add: To add assigned number. For example, you set **999 here and you do not set Drop. If the called number is 070070101, the IP PBX-100 will add **999 then send **999070070101 as the called number.

Note:

- If you set both of the Drop and Add, IP PBX-100 will Drop first then Add. For example, the IP PBX-100 got a called number 070070, which was not sending from a "Extension", IP PBX-100 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.

3.2 Information

User can check some information of IP PBX-100 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 IP PBX-100, the IP Address will show up, on the other hand, if the subscriber doesn't register successfully on IP PBX-100, the IP Address will not be displayed.

3.3 Management

User can execute IP PBX-100 system configuration and management under this category.

3.3.1 Network

Enter **Management -> Network** to configure WAN and LAN IP.

■ **WAN**

- **Mode:** Select IP PBX-100 WAN port network mode to be Fixed IP or DHCP.
- **IP Address/NETMASK/GATEWAY:** If user has set IP PBX-100 to be fixed IP mode. User need to input IP address/Subnet Mask/ Default Gateway.
- **DNS:** Input DNS address.
- Mac: Mac address of IP PBX-100 WAN port. The Mac address cannot be modified.

■ **LAN**

- **IP Address:** Input IP address for LAN port of IP PBX-100.
- **NETMASK:** Input Subnet Mask for LAN port of IP PBX-100.
- Mac: Mac address of IP PBX-100 LAN port. The Mac address cannot be modified.

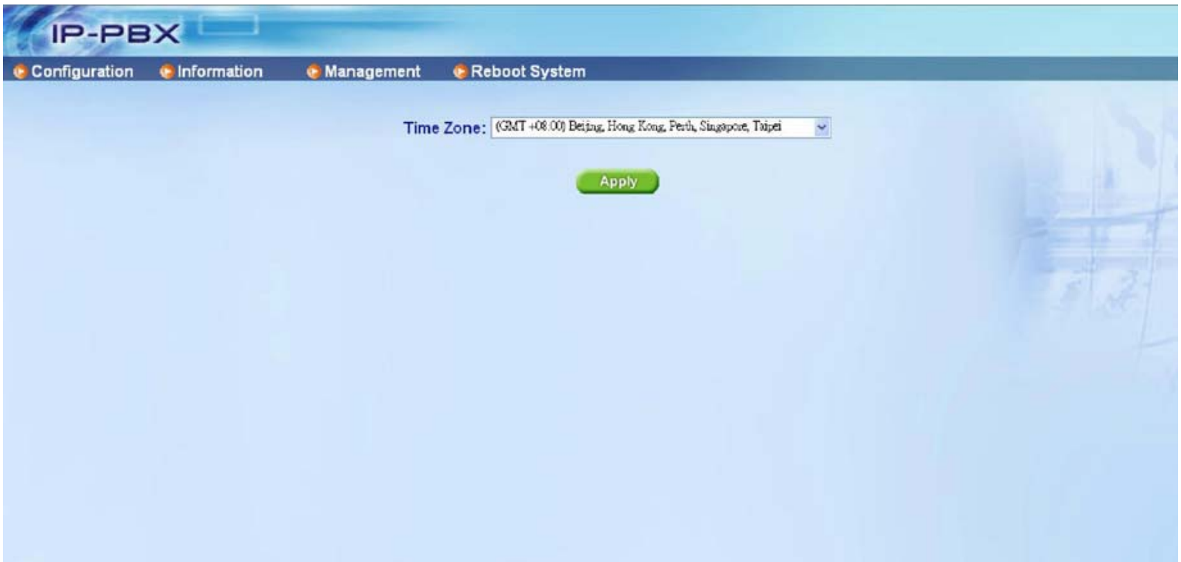
Press Apply to save configuration, or press Cancel to quit configuration.

3.3.2 Time Zone

Enter **Management** -> **Time Zone** to select correct Time Zone for IP PBX-100, this time will affect CDR and voice mail time display.

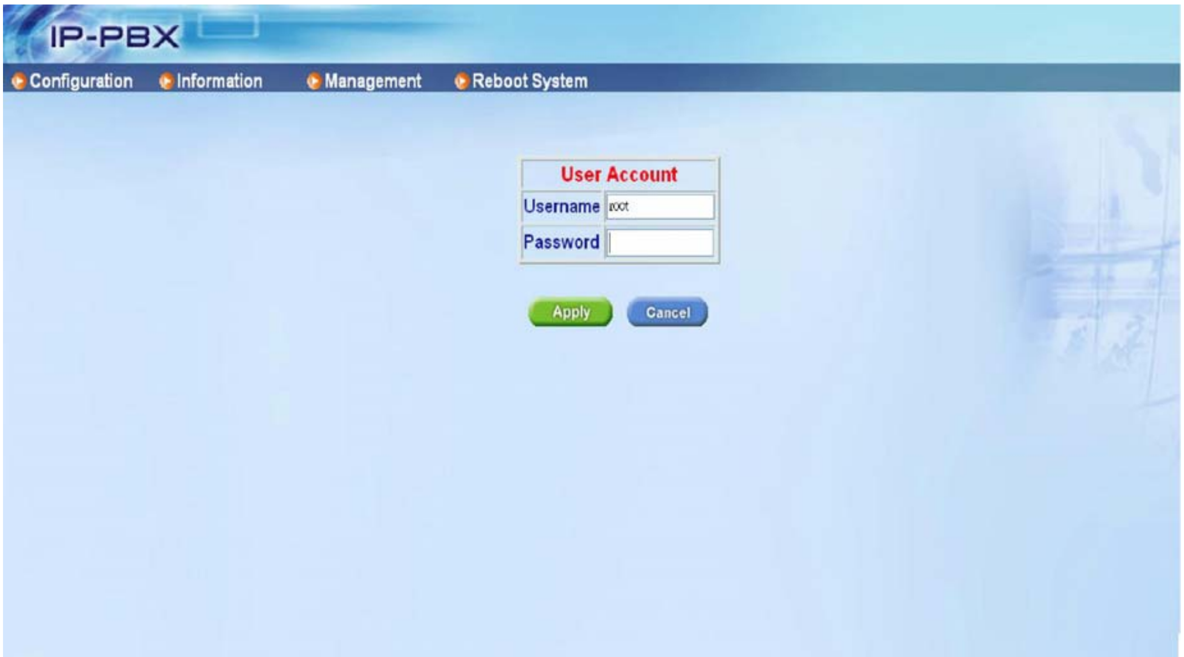
Note:

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



3.3.3 User Account

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

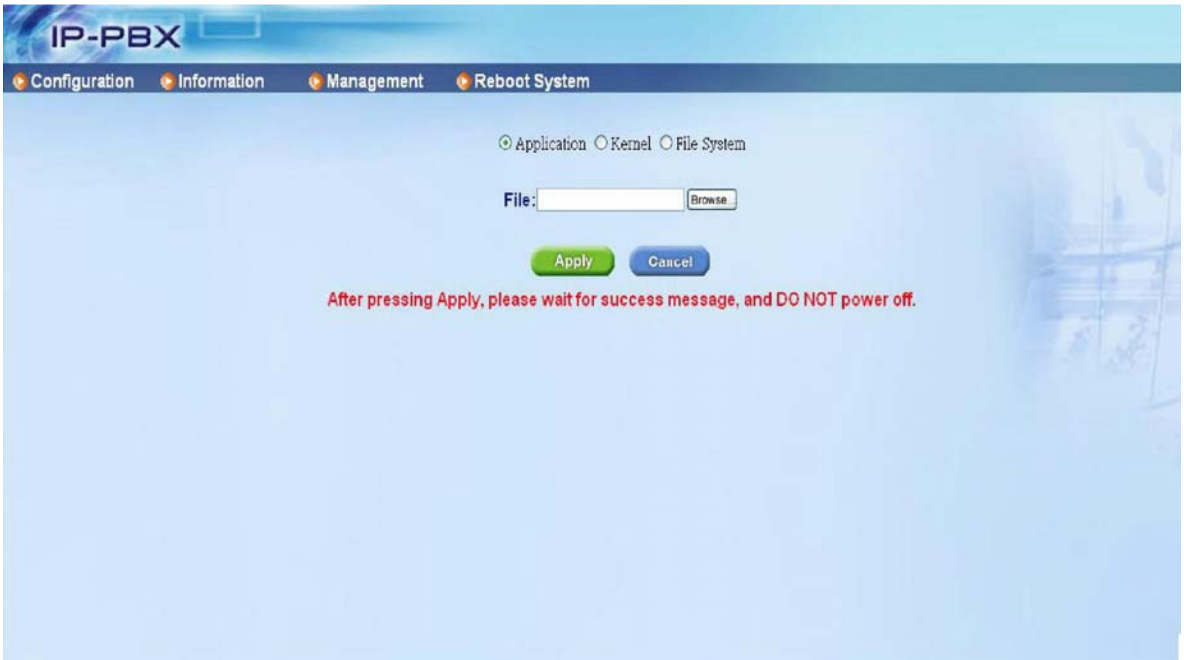


3.3.4 Firmware Upload

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:

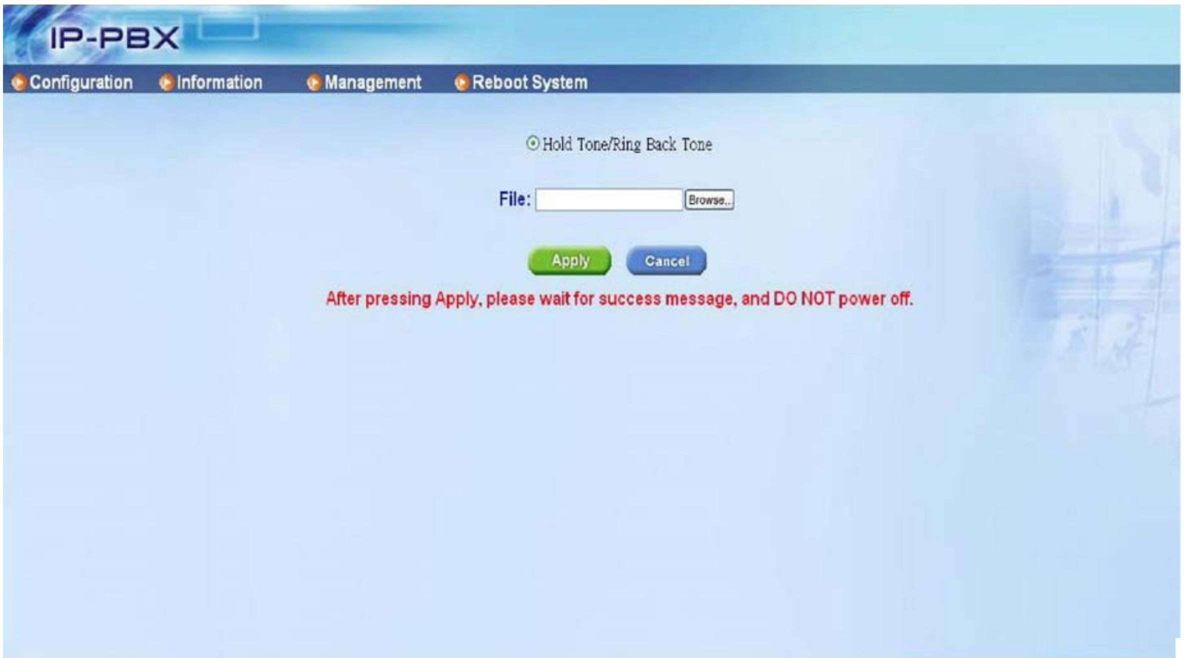
Normally, you just need to upgrade the Application but in some situation you may need to also upgrade the Kernel or Fie System. For more information, please refer to the release note of IP PBX-100.After pressing Apply, please wait for success message, and DO NOT power off. After upload succeed, on screen will show success message. Pease reboot system to renew system firmware.



3.3.5 Music Upload

User can customize Ring Back Tone (Transferring Tone) by upload new wave file on IP PBX-100. Pease 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.



3.3.6 Import Setting

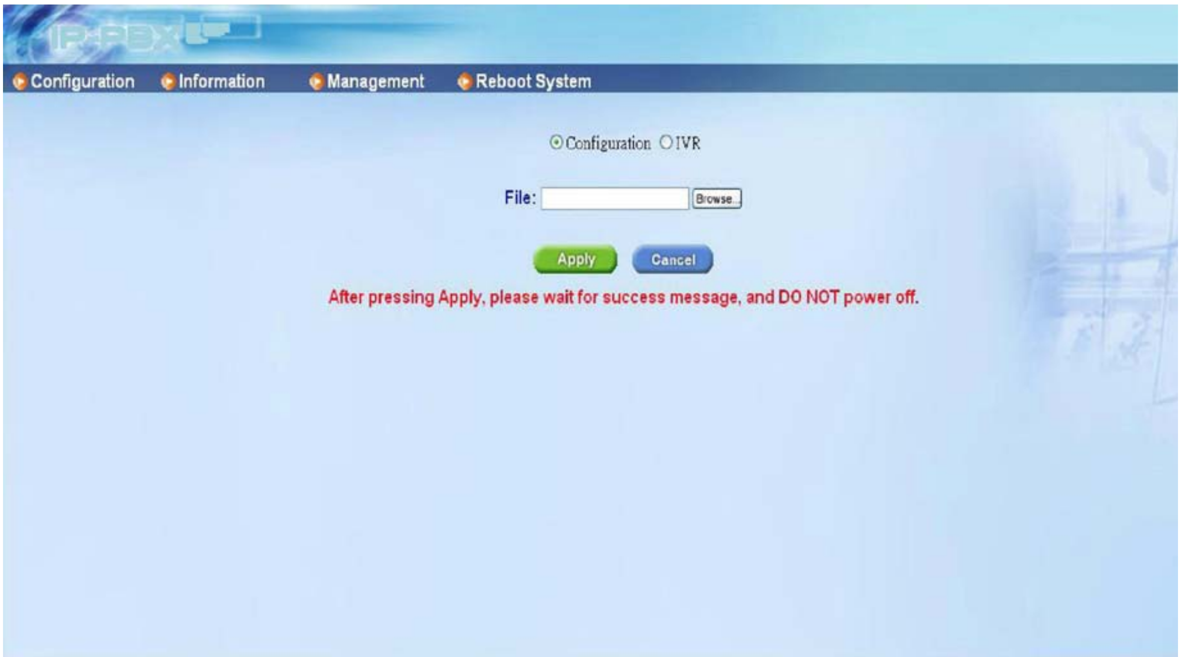
If there is IP PBX-100 setting file exported from IP PBX-100, user can import this file and doesn't need to re-configure al parameters for IP PBX-100.

Enter **Management ■* Import Setting ■* Choose the Import options (Configuration or IVR) ■* 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:

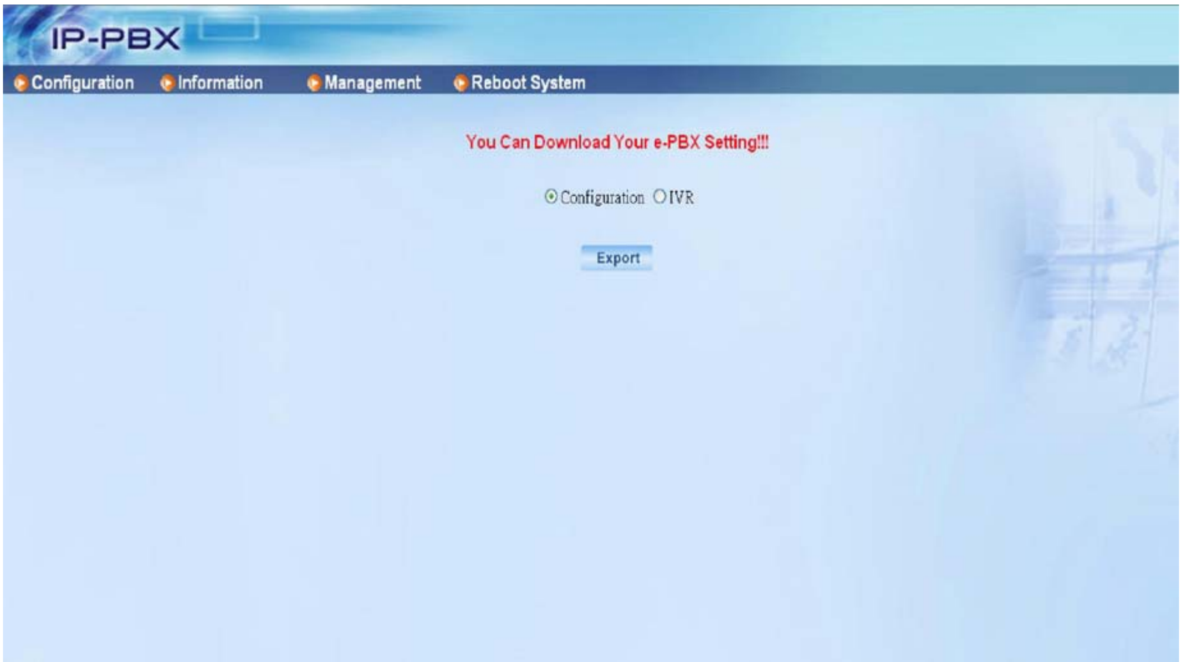
- After pressing Apply, please wait for success message, and DO NOT power off.
- After Import succeed, on screen will show related information. Pease reboot system to renew settings.



3.3.7 Export Setting

User can export configuration and voice wave files. If there is more than one IP PBX-100 need to be configured, user can export configuration of one IP PBX-100, and then import this setting file for the other IP PBX-100s, so that user doesn't need to re-configure al parameters for each IP PBX-100.

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.



3.4 Reboot System

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

4. 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. Dynamix DW) hotline function and point to destination number **999 (Number of Automated Attendant for IP PBX-100). Once system has incoming cal from PSTN, it will automatically connect to Automated Attendant.

Note:

- Al of the Extensions can also dial to **999 to reach Automated Attendant directly.
- For more information, please refer to the Appendix A: Application between Dynamix DW CPE device and IP PBX-100.

4.1.3 How to record the other announcements

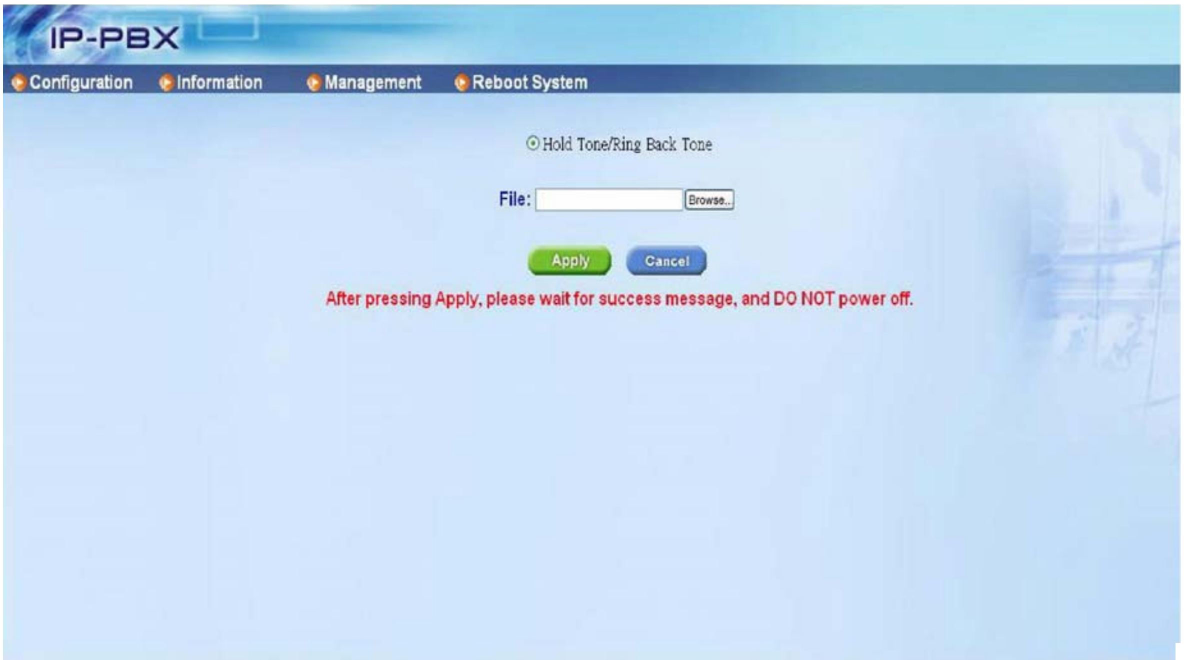
User can record the greeting by dialing to **111, so that the caller will hear the new greeting if user cal to **999. When the user called to an Extension, which is on the phone, he will also hear an announcement of Extension is busy. How to record a new busy announcement? The procedure is just like recording new greeting, dialing to a special code for recording. Below is the special code for announcement.

Code	**File Name**	**Default Announcement**
****111**	greeting-day.gsm	Please dial the extension number or press 9 for operator.
****112**	greeting-noon.gsm	Please dial the extension number, thank you (It is reserved, not useful now)
****113**	greeting-night.gsm	Please dial the extension number, thank you (It is reserved, not useful now)
****114**	greeting-holiday.gsm	Please dial the extension number, thank you (It is reserved, not useful now)
****115**	greeting-temporary.gsm	Please dial the extension number, thank you (It is reserved, not useful now)
****116**	noanswer.gsm	The extension number you dialed is no answer, please dial to the other extension or

		press 9 for operator.
**117	busy.gsm	I am sorry, the extension number you dialed is busy, please dial to the other extension or press 9 for operator.
**118	goodbyeivr.gsm	goodbye
**119	unavailable.gsm	I am sorry, the extension number you dialed is unavailable, please dial to the other extension or press 9 for operator.
**120	invalid.gsm	I am sorry, that is not a valid extension. Please try again.
**121	vm-theperson.gsm	The person at extension
**122	vm-isonphone.gsm	is on the phone
**123	vm-isunavail.gsm	is unavailable
**124	vm-intro.gsm	Please leave your message after the tone. When done hang up or press the pound key.
**125	op-noanswer.gsm	The operator is no answer, please call later or dial another extension number
**126	op-busy.gsm	The operator is busy, please call later or dial another extension number
**127	op-unavailable.gsm	The operator is unavailable, please call later or dial another extension number

4.2 Customize Ring Back Tone (Transferring Tone)

User can customize Ring Back Tone by upload new wave file on IP PBX-100. Pease 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.



4.3 Call Features

4.3.1 Authentication

When IP PBX-100 gets a Registration or Invite (incoming cal) from a remote location, it will reply Authentication for security issue.

4.3.2 Automated Attendant

The IP PBX-100 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 (Client based)

The IP PBX-100 does not support "server transfer" now, so you should do it by subscriber device and the transfer function of the subscriber device should follow the SIP standard.

4.3.4 Blind Transfer (Client based)

The IP PBX-100 does not support "server blind transfer" now, so you should do it by subscriber device and the blind transfer function of the subscriber device should follow the SIP standard.

4.3.5 Call Forward on Busy (Client based)

The IP PBX-100 does not support "server forward" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature by LCD or "WEB" interface. For more information about IP PHONE, please go to: http://www.dynamix.ua

4.3.6 Call Forward on No Answer (Client based)

The IP PBX-100 does not support "server forward" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature by LCD or "WEB" interface. For more information about IP PHONE, please go to: http://www.dynamix.ua

4.3.7 Call Forward Unconditional (Client based)

The IP PBX-100 does not support "server forward" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature by LCD or "WEB" interface. For more information about IP PHONE, please go to: http://www.dynamix.ua

4.3.8 Call Hold/Retrieval (Client based)

Normally, the cal hold and cal retrieval is done by Client, the IP PBX-100 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.

4.3.1 Call Waiting (Client based)

The IP PBX-100 does not support "server Cal Waiting" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature by "CLI". For more information about IP PHONE, please go to:
http://www.dynamix.ua

4.3.11 Caller ID

The IP PBX-100 will relay the caller ID from caller to caller.

4.3.12 Do Not Disturb (Client based)

The IP PBX-100 does not support "server DND" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature by DND button. For more information about IP PHONE, please go to:
http://www.dynamix.ua

4.3.13 Flexible Extension Logic

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

4.3.14 Music On Hold

The IP PBX-100 will pay music if the user is under Hold status.

4.3.15 Music On Transfer

The IP PBX-100 will pay music if the user is under Transfer status.

4.3.16 Call Pickup (Global Call Pickup)

The IP PBX-100 can support Global Cal Pickup. For example: Ext-A is ringing, Ext-B can press *8 for cal pickup.

4.3.17 Three-way Conference (IP PHONE)

The IP PBX-100 does not support "server Conference" now, this feature should be done by client. For example, if the client is Dynamix DW IP PHONE, you can enable this feature

by Conf button. For more information about IP PHONE, please go to:
http://www.dynamix.ua

4.3.18 Time and Date

You can select correct Time Zone for IP PBX-100; this time will affect CDR and voice mail time display.

4.3.19 Trunking

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: Appendix A: Application between Dynamix DW CPE device and IP PBX-100.

4.3.20 VoIP Gateways

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: Appendix A: Application between Dynamix DW CPE device and IP PBX-100.

4.3.21 Voice Mail to e-mail

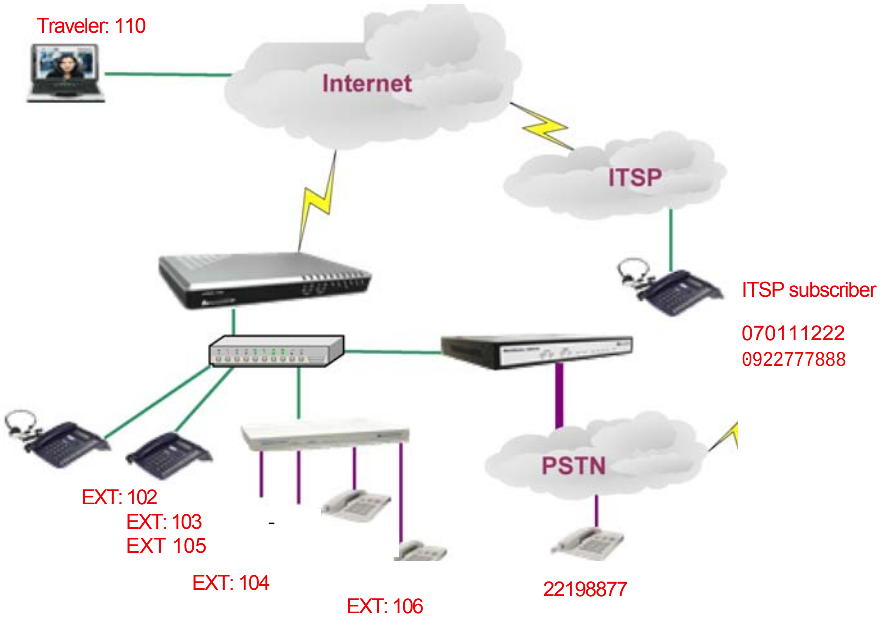
IP PBX-100 does not have enough Flash Rom to store voice mail within itself, but it can send voice mail by e-mail. There is no setting of SMTP in IP PBX-100 now. But there is a build-in software called "Mail-IP" in IP PBX-100, it is an automatically send mail software. If the IP PBX-100 got a new message, it will send the message to user by email then delete the message from IP PBX-100 immediately.

Appendix-A Application between Dynamix DW CPE device and IP PBX-100.

1) Example Architecture with One IP PBX-100

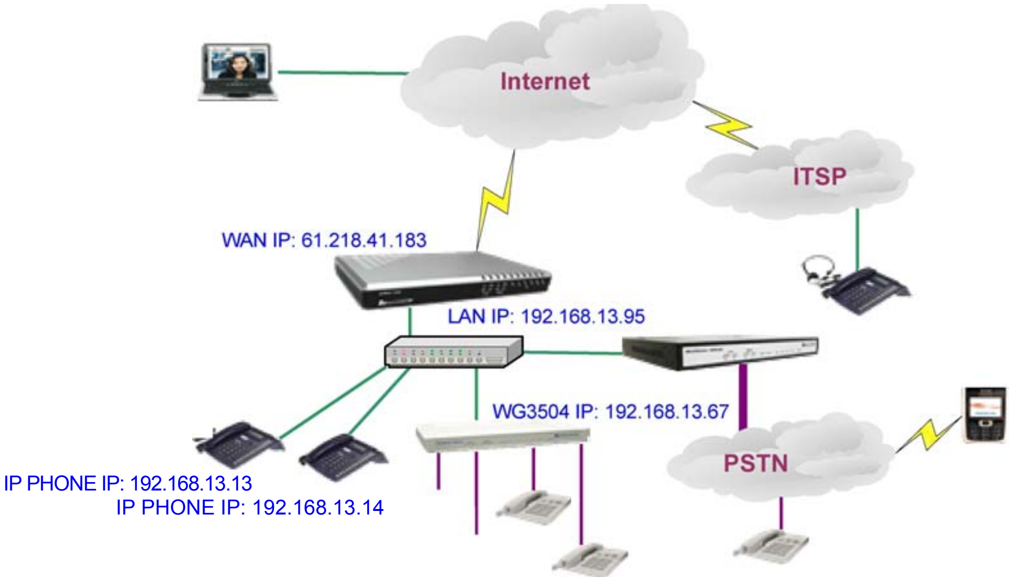
Test Case

- Extensions register to IP PBX-100 with number 101 to 106. And Extensions can talk to each other.
- The cal will be forward to Mailbox if the extension 101 is busy or no answer.
- The Trunk (Dynamix DW) can also register to IP PBX-100 (registered number 888).
- IP PBX-100 can register to ITSP as a SIP-Trunk.
- Al of the Extensions can cal out to local PSTN via Dynamix DW.
- User in PSTN side should be able to contact with Extensions via Dynamix DW.
- User in ITSP side should be able to contact with Extensions
- User in ITSP side can cal out to local PSTN via Dynamix DW.
- Traveler can cal back to EXT



Extensions register to IP PBX-100 with number 11 to 16. And Extensions can talk to each other.

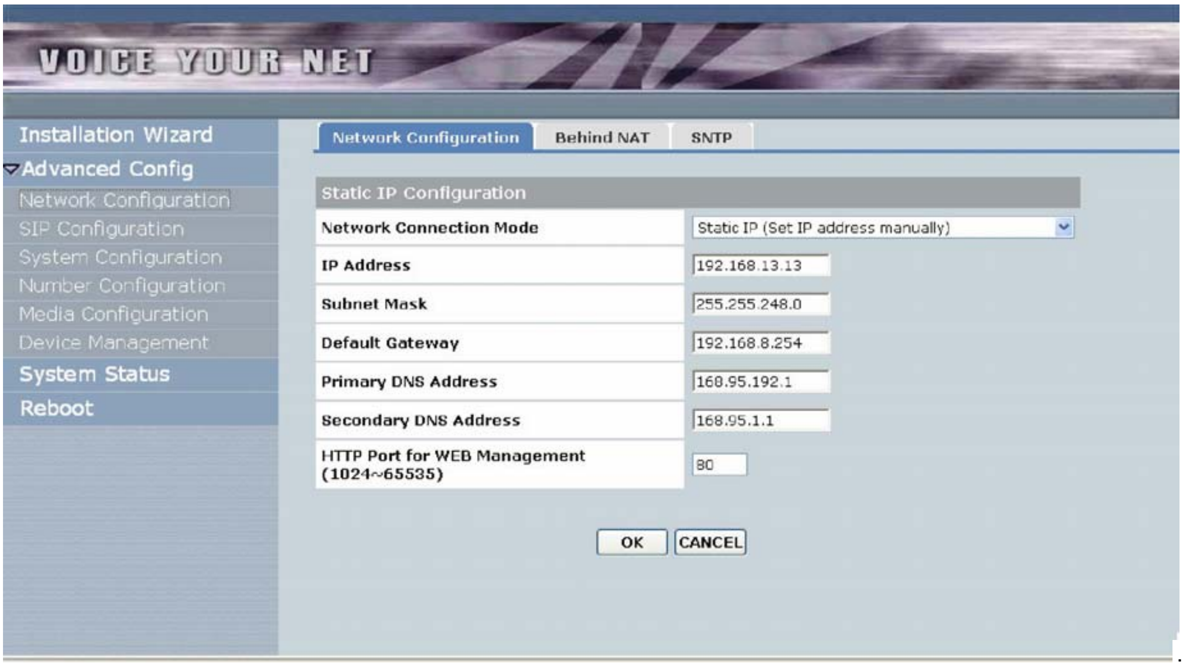
Step1: Setup Network for IP PBX-100, IP PHONE, DW.



Set PBX-100 with WAN [IP_ADDRESS: 61.218.41.183, NETMASK: 255.255.255.240, Gateway: 61.218.41.177, DNS: 168.95.1.1], LAN [IP_ADDRESS: 192.168.13.95, NETMASK: 255.255.248.0]. After setting finish, please press Apply and reboot your IP PBX-100. When you got a new IP PBX-100, you can connect its LAN port to configure Network Setting first. The default LAN IP address is 192.168.123.123. For more info, please refer to user's manual CH2. Start to configure IP PBX-100.

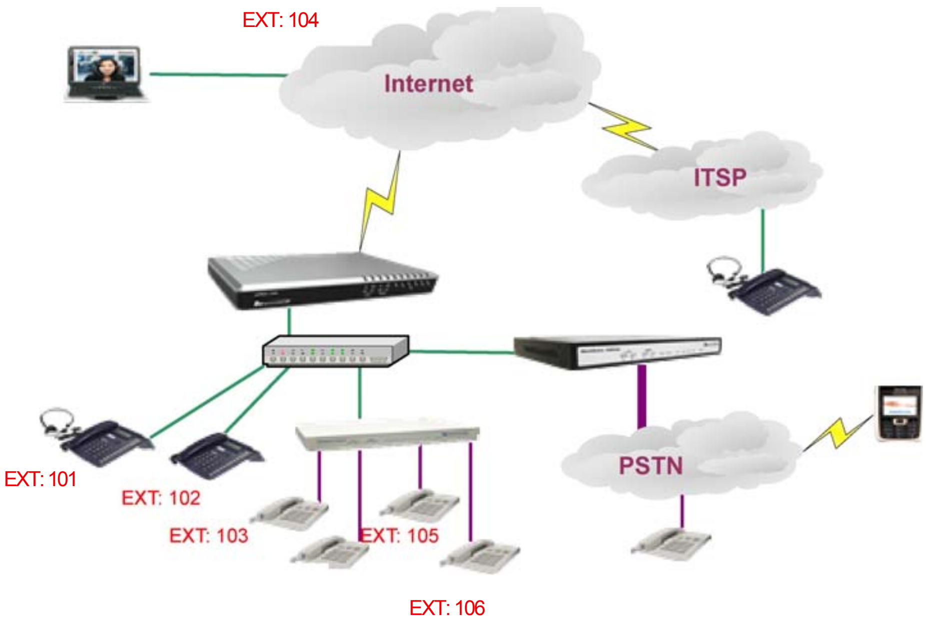
Set IP information for IP PHONE. You can set the IP info of IP PHONE by its LCD, or you can also login its WEB interface by default IP 10.1.1.3. Go to **Advanced Config -> Network Configuration** to setup

network as below, then press OK and reboot your IP PHONE. For more information about IP PHONE, please go to: http://www.dynamix.ua

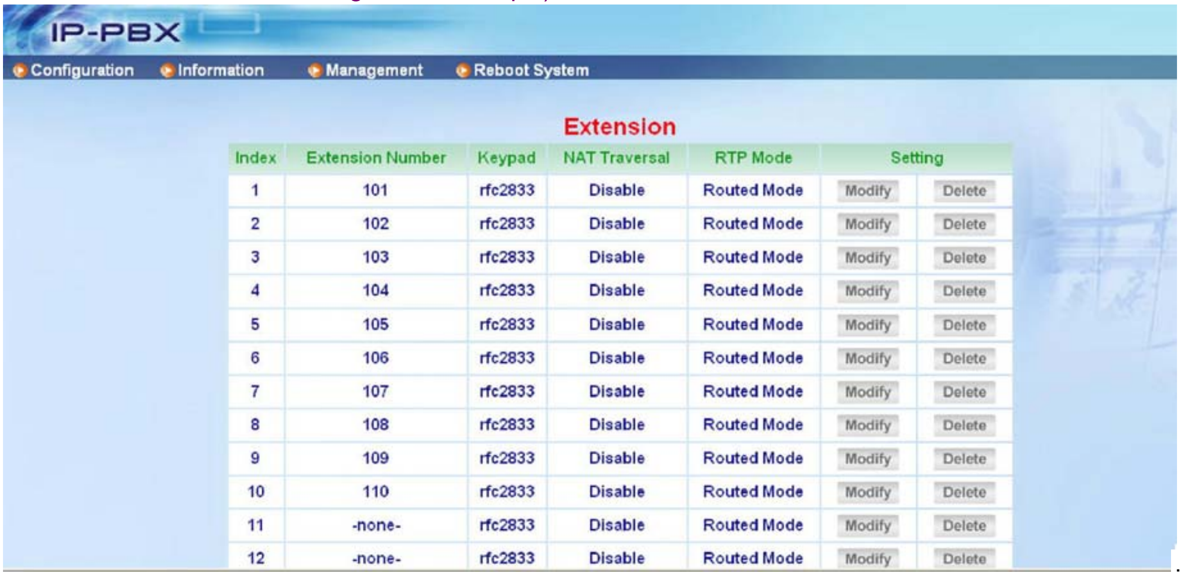


Set IP information for DW. You can set the IP info of DW by its COM port, or you can also login its WEB interface by default IP 10.1.1.3. Go to **Network Interface** page to setup network setting as below. After set the network info, please press OK -* Commit Data -* Reboot System. For more information about DW, please go to: http://www.dynamix.ua

Step2: Configure Extensions

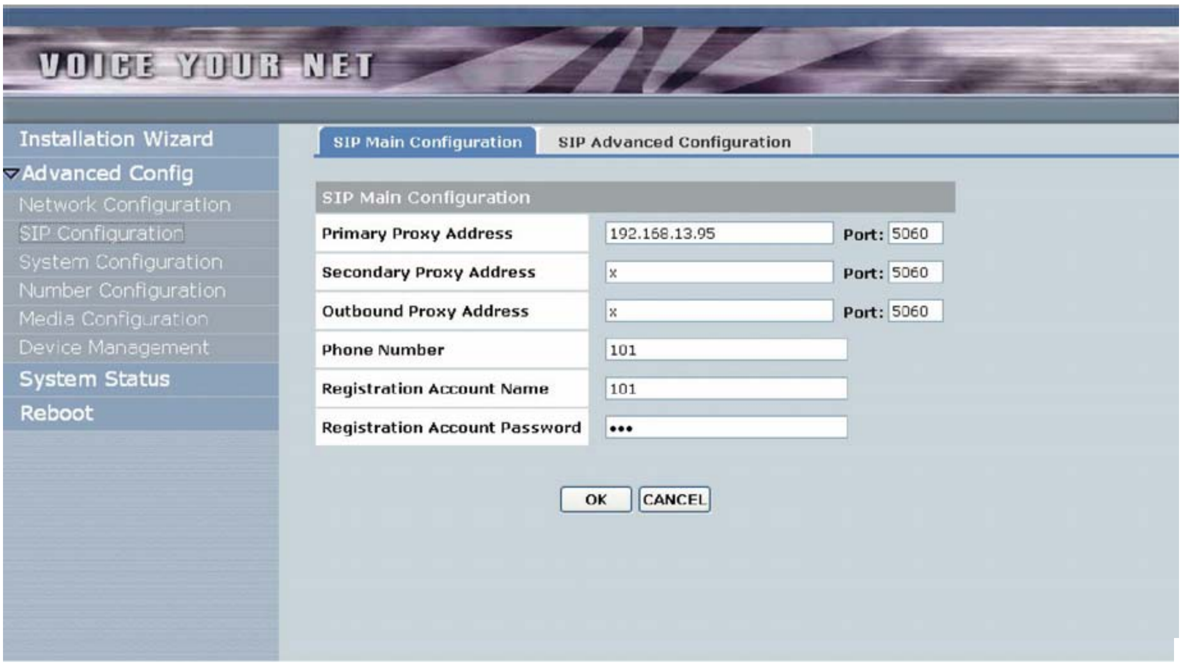


In IP PBX-100, prepare Extension accounts for Client device. (There are 10 default Extensions, from 101 to 110, so we use the default setting for this Example) For more information about Extension



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

This Example, we have 2 IP PHONE register to IP PBX-100 with extension 101 and extension 102. You can set the SIP Configuration of IP PHONE by its LCD, or you can also login its WEB interface. Go to **Advance Config -» SIP Configuration,** setup the Primary proxy Address and Registered Number..., etc. After configuration, please remember to press OK then reboot your IP PHONE. For more information about IP PHONE, please go to: http://www.dynamix.ua



This Example, we have 1 DW register to IP PBX-100 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 ^ Reboot System. For more information about DW, please go to: http://www.dynamix.ua

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

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, al of the Extensions can contact with each other. If the called party is busy or no answer, IP PBX-100 will pay an announcement to indicate the called party's status.

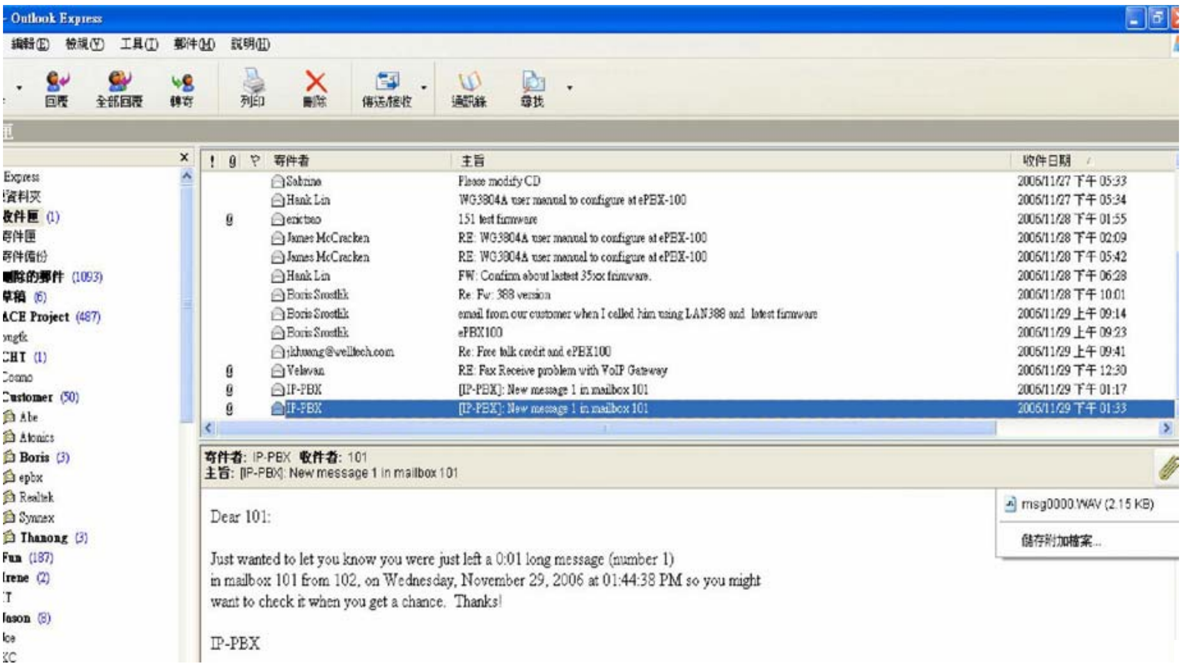
The call will be forward to MailBox if the extension 11 is busy or no answer.

Step1: Enable Voice Mail function

- IP PBX-100 has 10 Extensions (101 to 110) and the voice mail function default disable. You can enable the voice mail function as below.

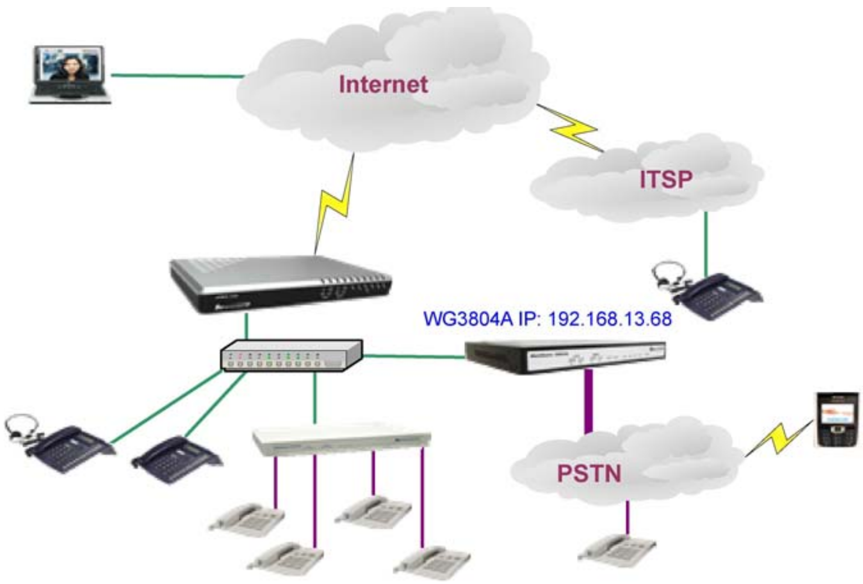
Step2: Confirm Voice Mail

- If 102 cal to 101 but 101 is busy, IP PBX-100 will pay an announcement to indicate the 101 is busy and 102 can eave message for 101. IP PBX-100 will send voice mail to your mail box with a WAV format. Below is an example.

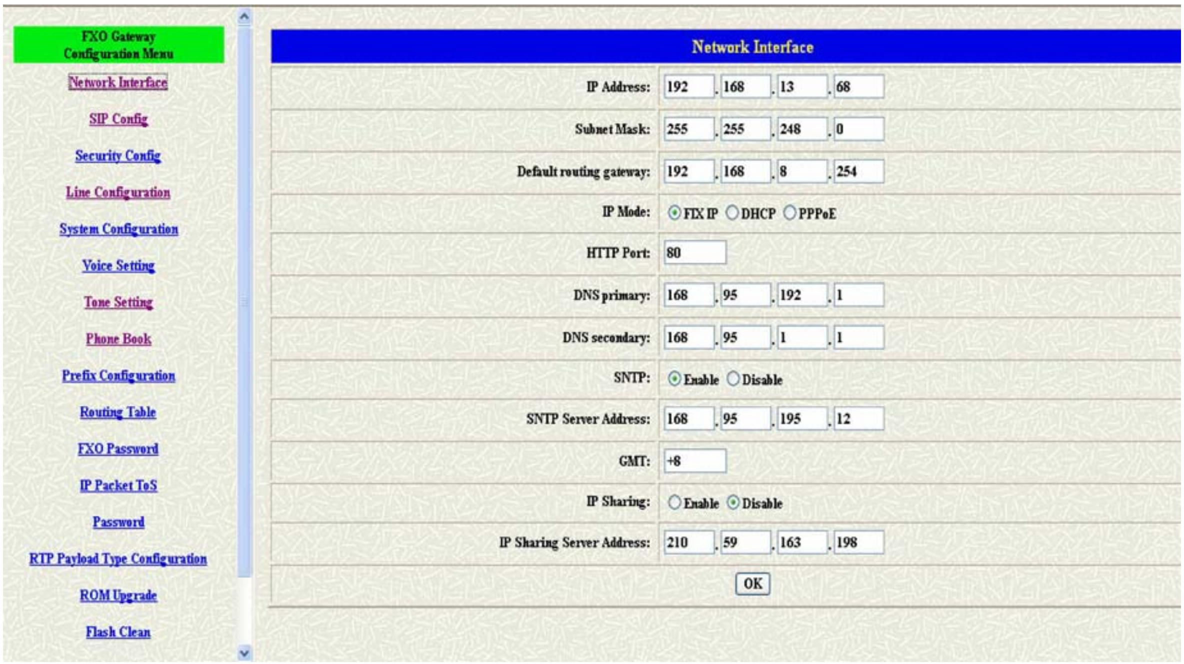


The Trunk can also register to ePBX-1 (registered number 888).

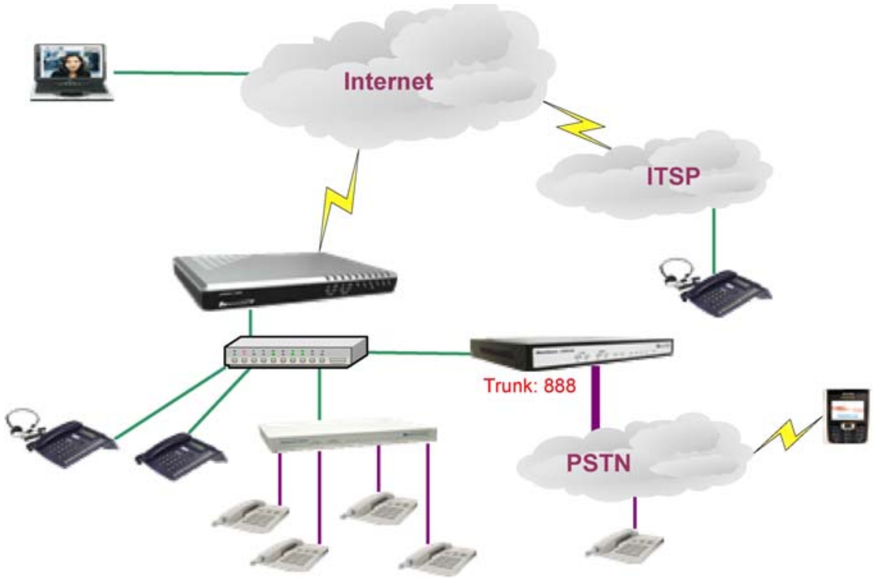
Step1: Setup Network for 384A



Set IP information for Dynamix DW. You can set the IP info of Dynamix DW by its COM port, or you can also login its WEB interface by default IP 10.1.1.3. Go to **Network Interface** page to setup network setting as below. After set the network info, please press OK -* Commit Data -* Reboot System. For more information about Dynamix DW, please go to: http://www.Dynamix.ua



Step2: Prepare Trunk number



• In IP PBX-100, prepare Trunk accounts for Dynamix DW. (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.

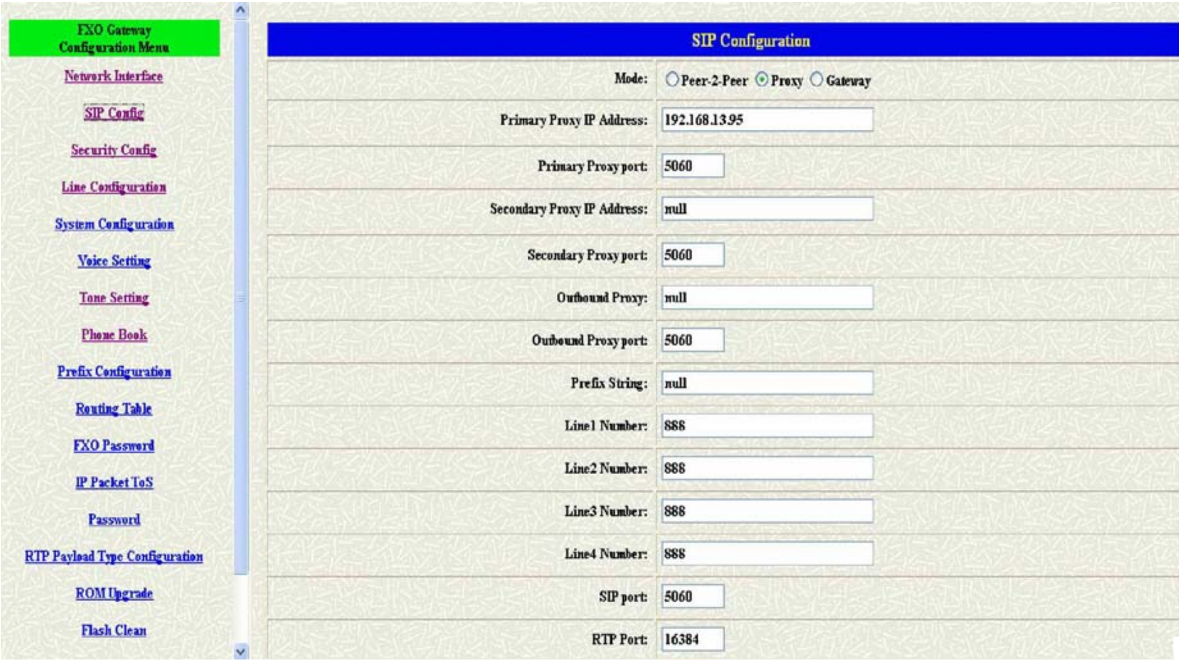
Configuration Information Management Reboot System

Trunk

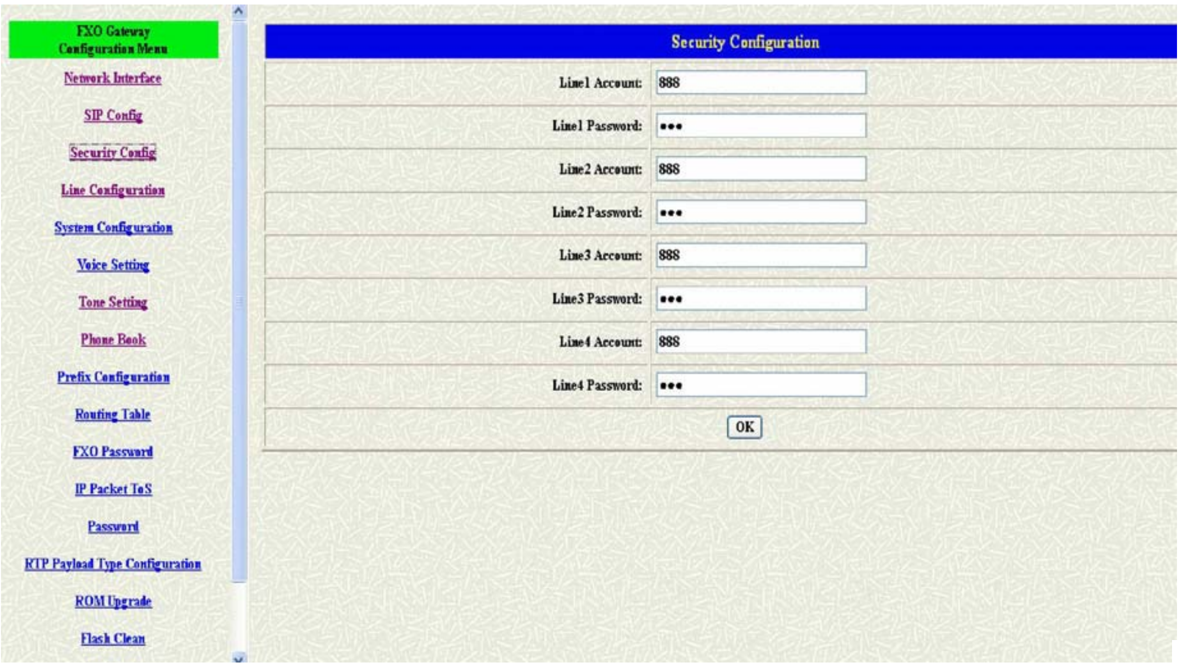
Index	Trunk Number	Keypad	NAT Traversal	RTP Mode	Setting	
1	888	rfc2833	Disable	Routed Mode	Modify	Delete
2	889	rfc2833	Disable	Routed Mode	Modify	Delete
3	-none-	-none-	Disable	Routed Mode	Modify	Delete
4	-none-	-none-	Disable	Routed Mode	Modify	Delete
5	-none-	-none-	Disable	Routed Mode	Modify	Delete
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

Step3: Setup Trunk

• This Example, we have 1 Dynamix DW register to IP PBX-100 with Trunk number 888. You can set the SIP Information of Dynamix DW by its COM port or you can also login its WEB interface for configuration as below. Go to **SIP Config** page to set the Dynamix DW as Proxy mode (or Gateway mode), Primary Proxy IP Address and line number (If you set the Dynamix DW to Proxy mode, you should set line number for al of the line1 to line4. If you set the Dynamix DW to Gateway mode, you can just only set line1 number). Go to **Security Config** page to input the registered account (If you set the Dynamix DW o Proxy mode, you should set Account for al of the line 1 to line4. If you set the Dynamix DW to Gateway mode, you can just only set line Account). After configure, please press OK -* Commit Data ^ Reboot System.



Set Dynamix DW to Proxy Mode, and also set the line number for line1 to line4.



Set Account and Password for Dynamix DW

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

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	-none-	Disable	-none-
11	888	192.168.13.68	Disable	-none-
12	889	-none-	Disable	-none-

- There are also some other necessary configuration of Dynamix DW to compatible with IP PBX-100. 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 Dynamix DW)

usr/config$sip-px 192.168.13.95 (set Dynamix DW to register IP PBX-100)

usr/config$ sip -linel 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)

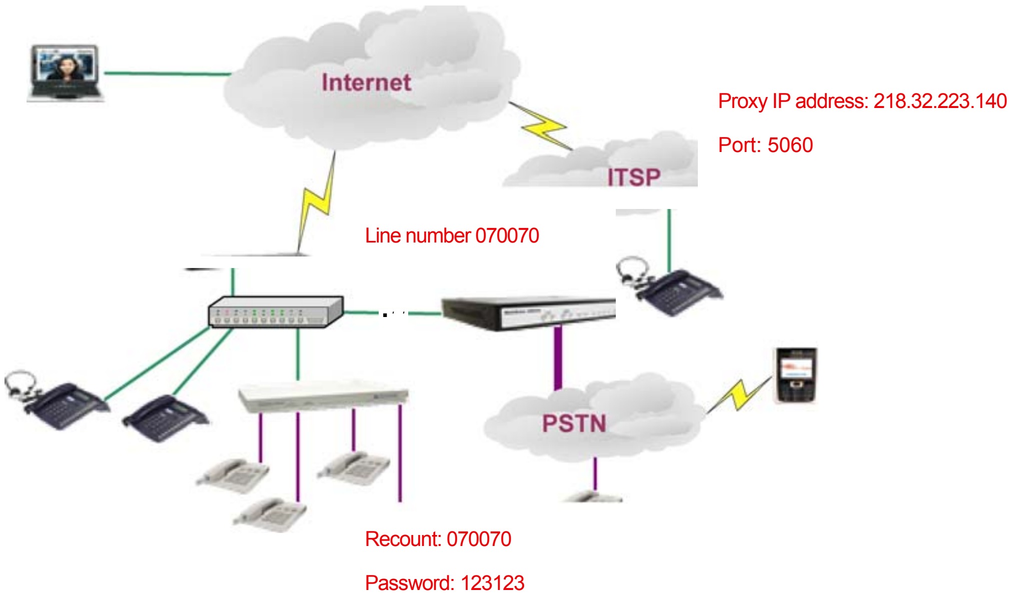
You must disable CNG of Dynamix DW due to the IP PBX-100 does not support CNG, otherwise there will be some voice error occurred. When you disable CNG, please remember to commit and reboot your Dynamix DW. For more information about Dynamix DW, please go to: http://www.dynamix.ua

ePBX-1 can register to ITSP as a SIP-Trunk.

ePBX can register to another ITSP as a SIP trunk. So that the Subscriber of ITSP can contact with IP PBX-100 and IP PBX-100 can cal to ITSP.

Stepl: 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 cal from IP PBX-100 to ITSP may be rejected. For more information about **"Realm",** please contact with your ITSP.

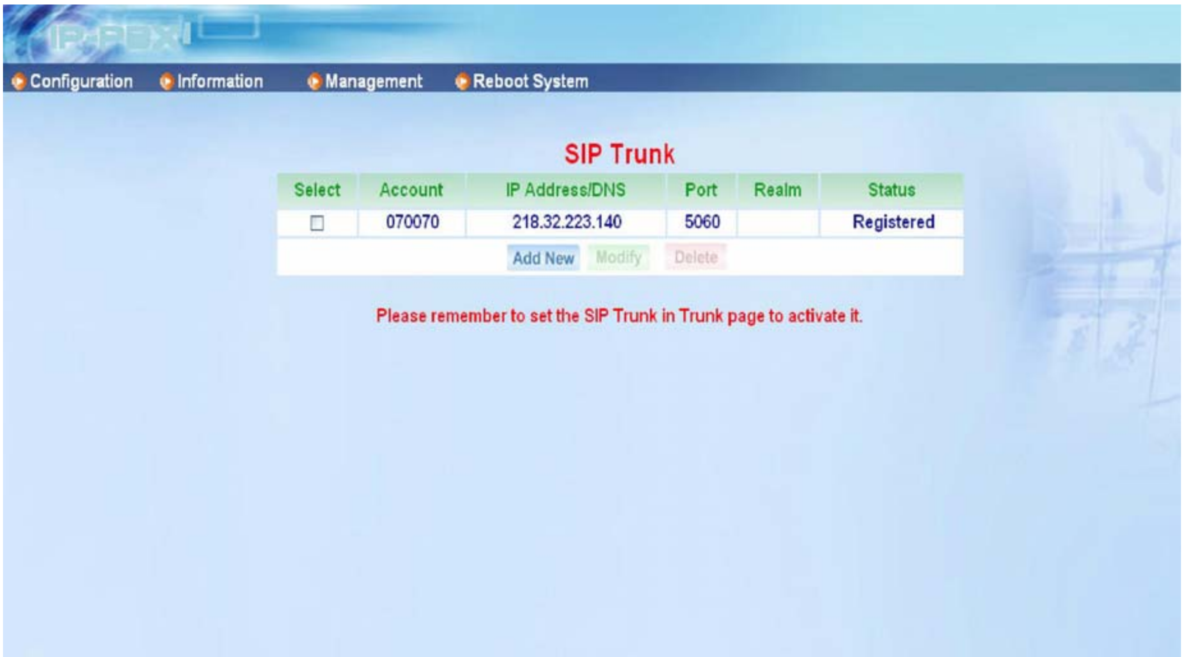


Step2: Set ePBX-1 to register ITSP.

- Input the necessary information in SIP Trunk page. In this example, our "Realm" is empty due to our ITSP does not need Authentication for incoming cal. For more information about SIP Trunk, please go to user's manual CH3- Full Web Configurations.

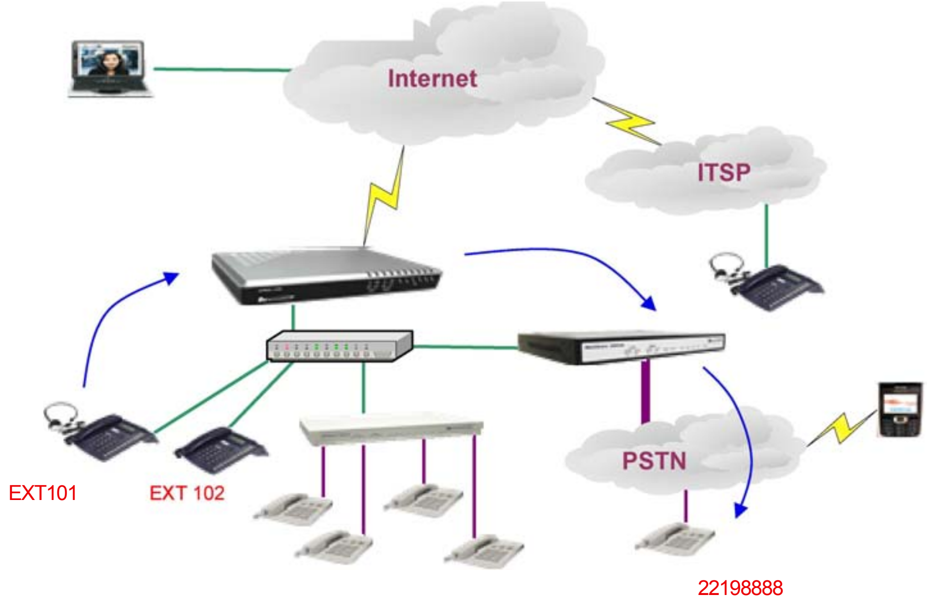
Step3: Confirm the register status of SIP Trunk

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



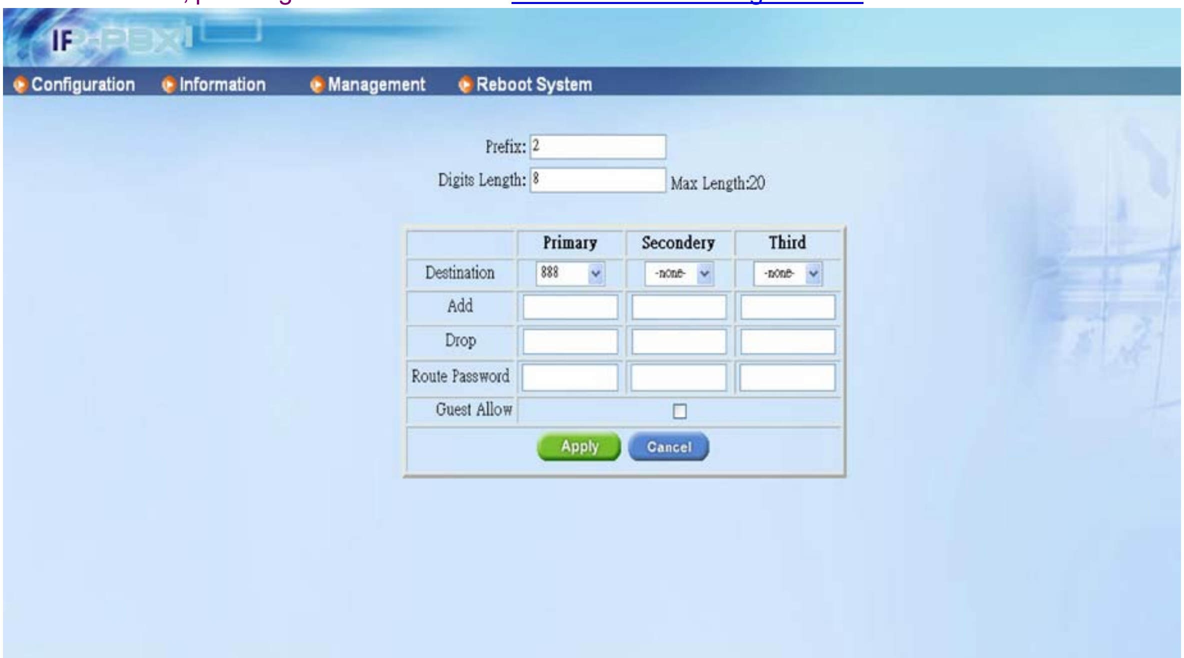
Al of the Extensions can call out to local PSTN.

Now the FXO is registering to IP PBX-100, and we hope the extensions can cal out to local PSTN via the FXO gateway. The Dynamix DW should connect with local PSTN line. We should set the routing table to let the IP PBX-100 route the call to Dynamix DW if the called number is a local PSTN number.



Stepl: Set Prefix route in Routing Table page

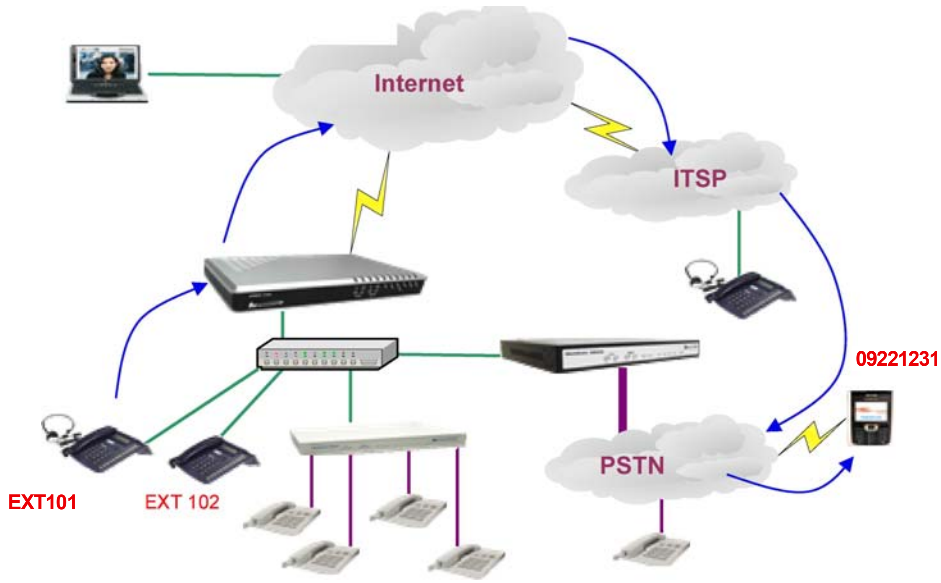
- Pease Go to Routing Table Page to set Prefix route, so that the Extensions can dial to local PSTN 22198888 via Dynamix DW (888). The setting just like below. For more information about Routing Table, please go to user's manual CH3- Full Web Configurations.



Now the Extensions can dial to 22198888 via Dynamix DW.

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

Now we set IP PBX-100 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.



Stepl: Confirm the ePBX-1 register to ITSP successfully

- Pease 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 IP PBX-100 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.

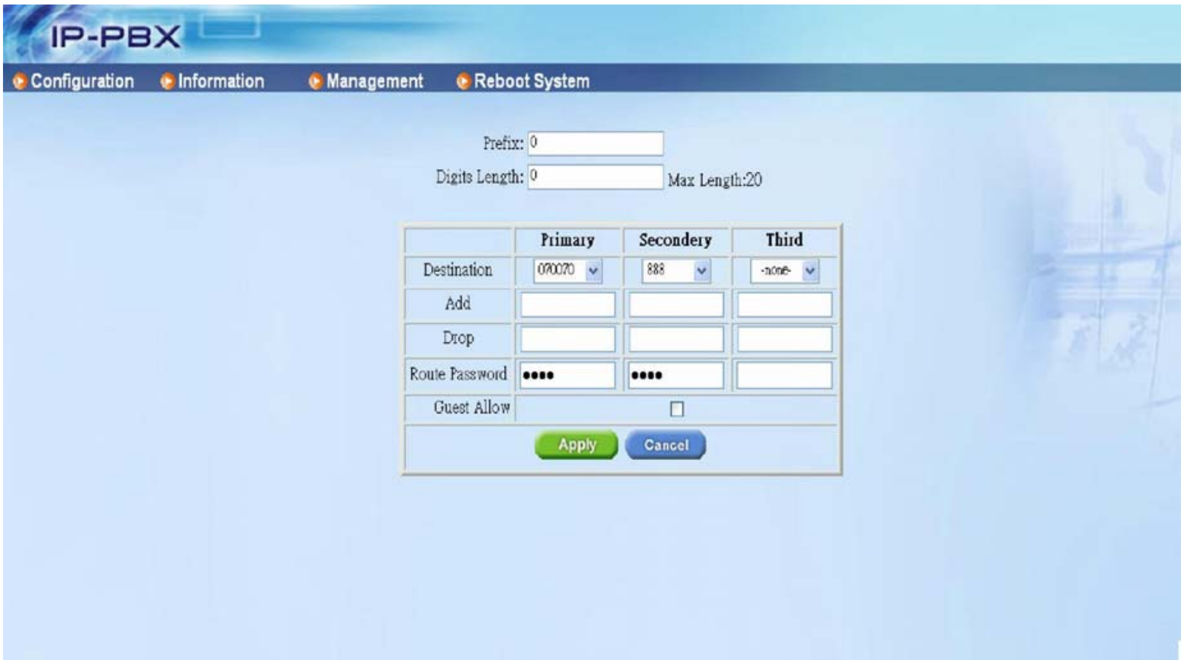
Activate SIP Trunk (ITSP) in Trunk Page.

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

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

Step3: Set Prefix route in Routing Table page

• Pease 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.

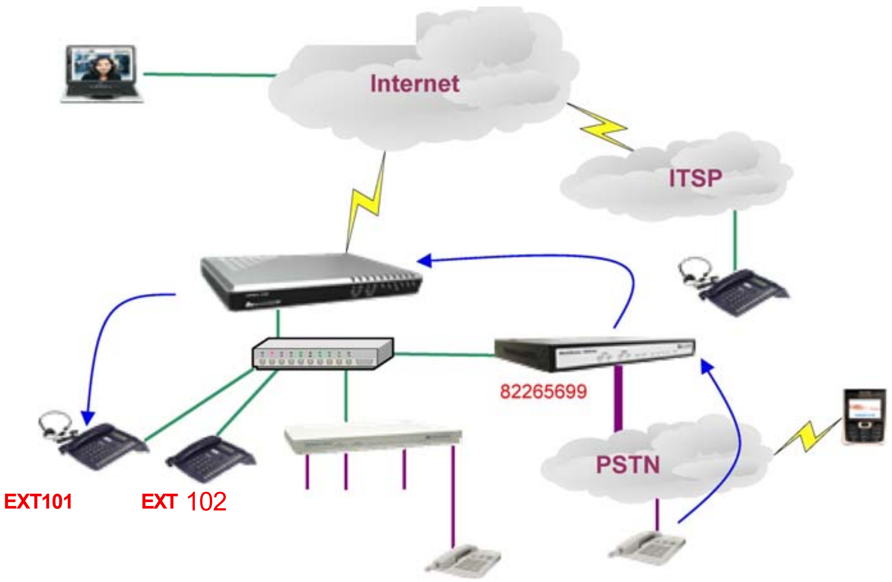


In this example, we set prefix to 0 and there is not limit for digits length (0). The first destination is ITSP and 2nd destination is Dynamix DW (888). And we also set the Routed Password for this prefix route.

Confirm the Outgoing Cal Rule. Now the Extensions can dial to 0922123123 to reach mobile phone via ITSR If extension called out with prefix number 0, the IP PBX-100 will pay an announcement for route password, after input the correct password, then IP PBX-100 will dial to destination.

User in PSTN side should be able to contact with Extensions

We hope the IP PBX-100 can pay 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 384A.

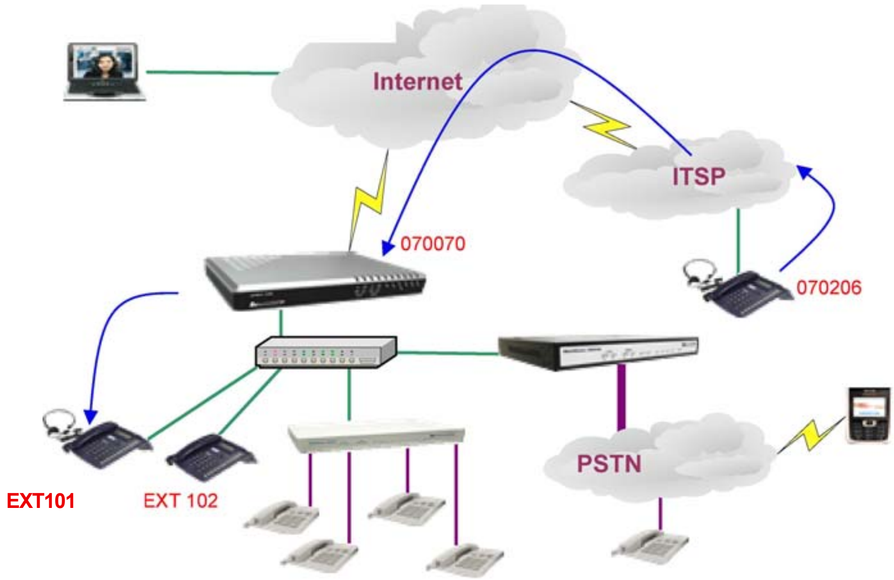
- The default auto attendant number of IP PBX-100 is **999. So you should set hotline function of Dynamix DW. When Dynamix DW got a PSTN incoming cal, it should dial to **999 directly. In below picture, we set line1 to line3 hotline to **999 and we set line4 hotline to EXT 102.

	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

Now, if FXO port1 got a PSTN incoming cal, it will hotline to auto attendant, and caller will hear a greeting then dial extension number. If port4 got a PSTN incoming cal, Dynamix DW will dial to EXT102 directly.

User in ITSP side should be able to contact with Extensions

070206 is a subscriber of ITSP, we hope 070206 can also contact with extension of IP PBX-100.

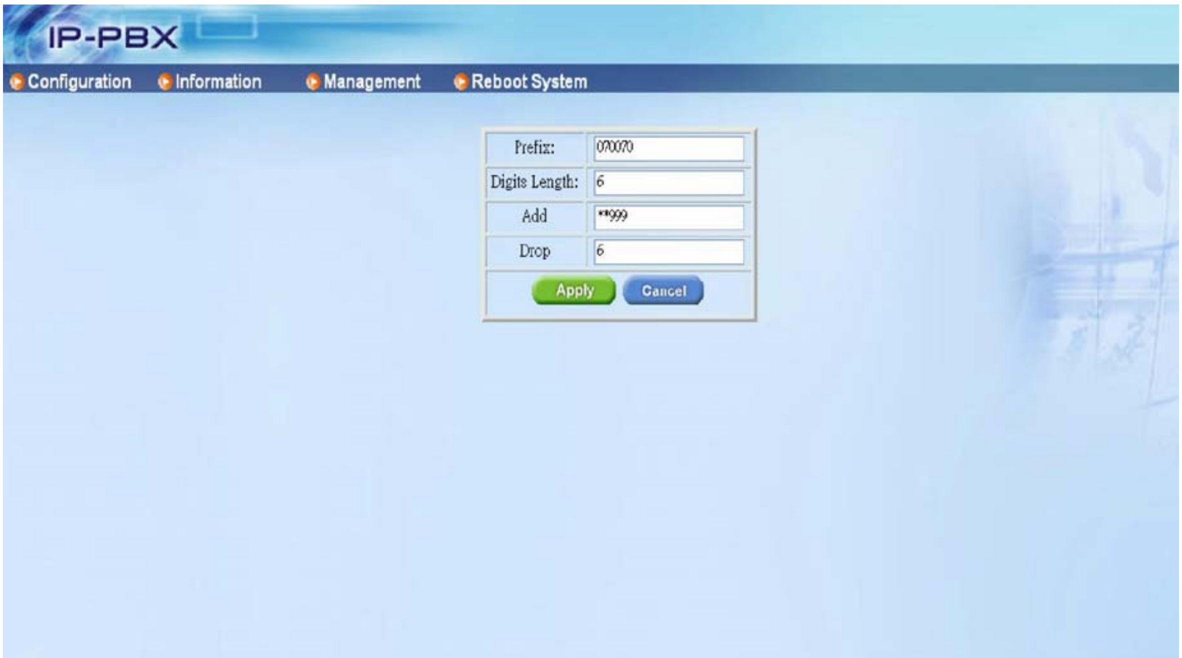


Step1: Confirm the ePBX-1 register to ITSP successfully

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

Step2: Set incoming call rule

- Pease Go to Routing Table page to set incoming cal. If IP PBX-100 got a incoming cal with number 070070, it should pay a greeting so that 070206 can continue to dial EXT 101.

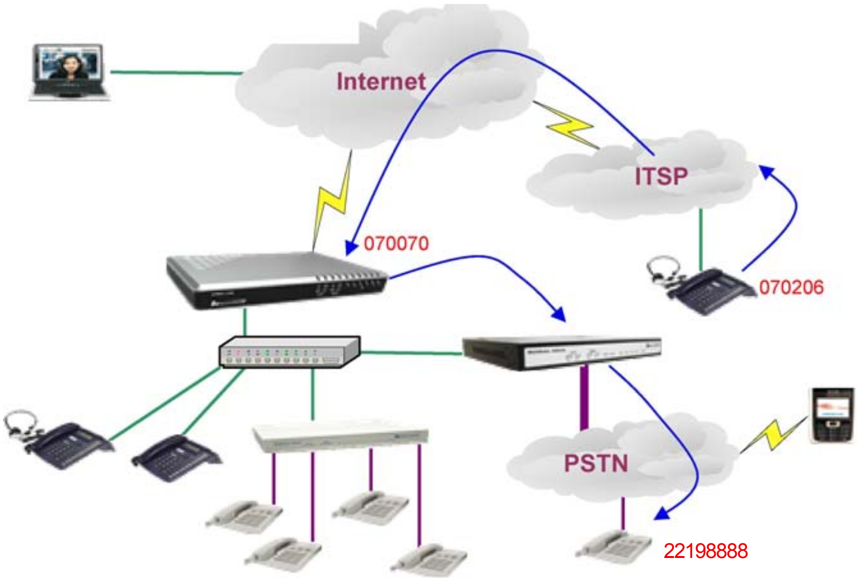


We set the incoming cal rule with Prefix 070070 and Digits Length 6. When IP PBX-100 got a called number with 070070, it will drop 6 digits and add **999.

Now, if IP PBX-100 got a PSTN incoming cal with number 070070, IP PBX-100 will dial to **999 (auto attendant). Then the c a l l e r can continue to dial the Extension.

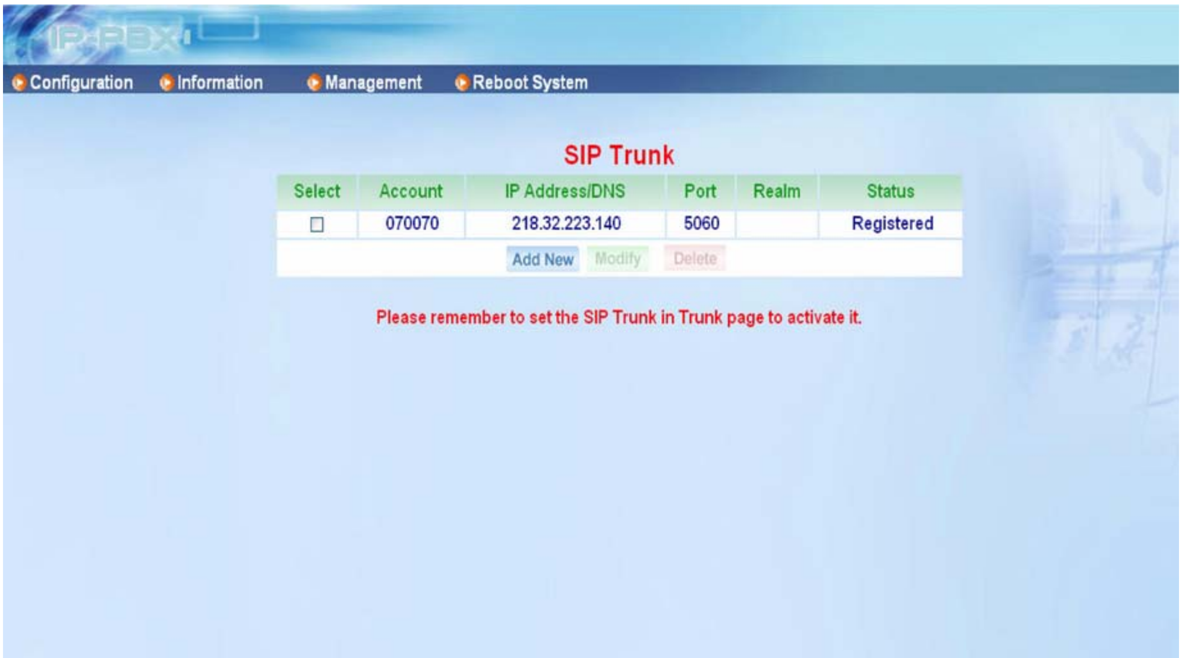
User in ITSP side can call out to local PSTN.

Now, 070206 can reach auto attendant. We hope 070206 can dial to local PSTN 22198888 via Dynamix DW.



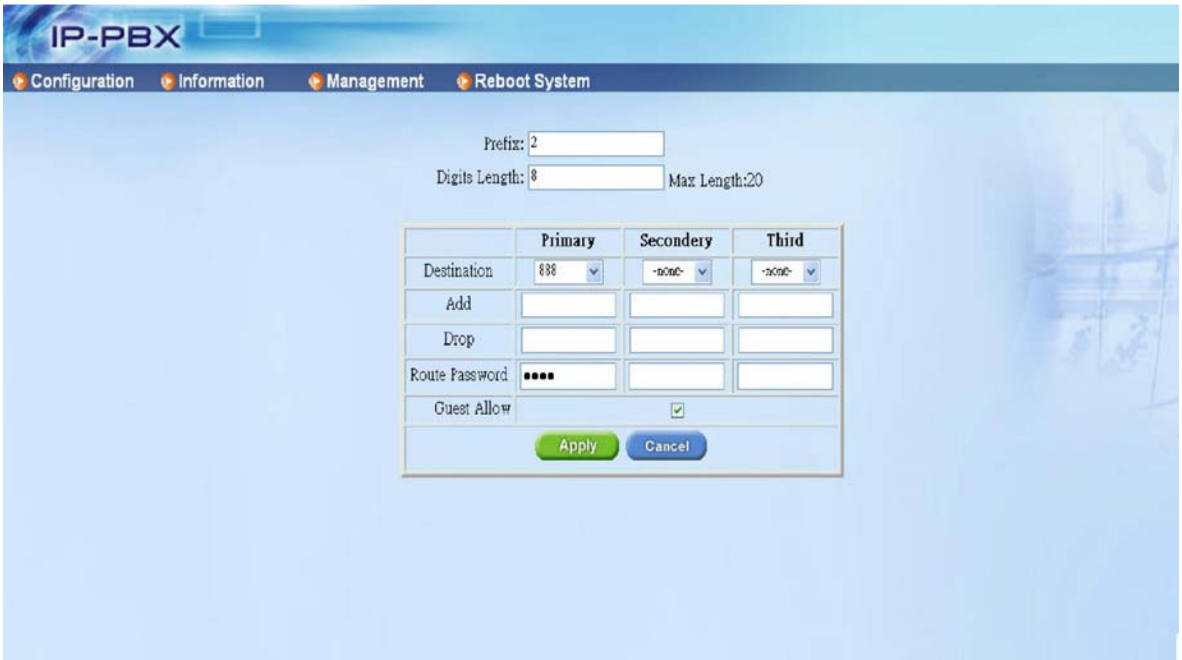
Step1: Confirm the ePBX-1 register to ITSP successfully

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



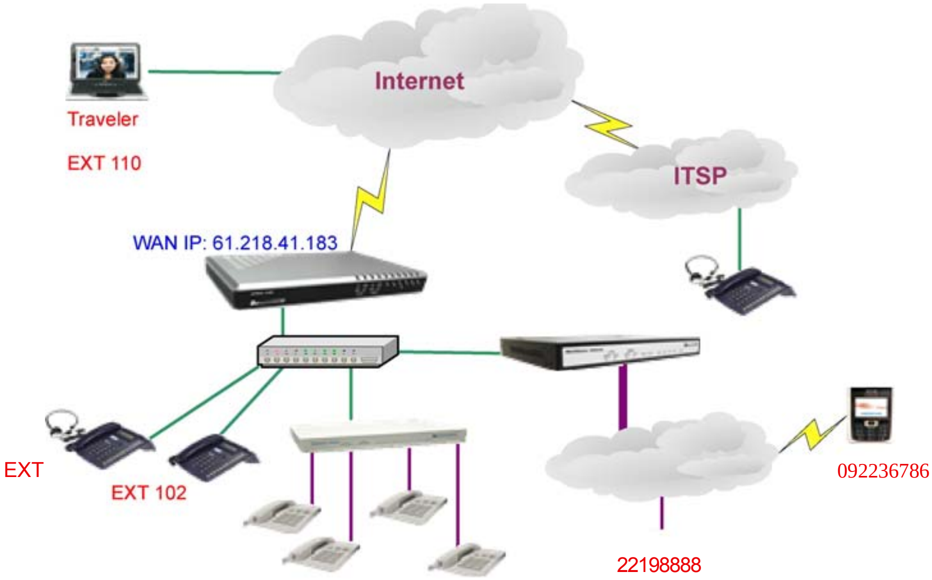
Step2: Enable Guest Alow

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



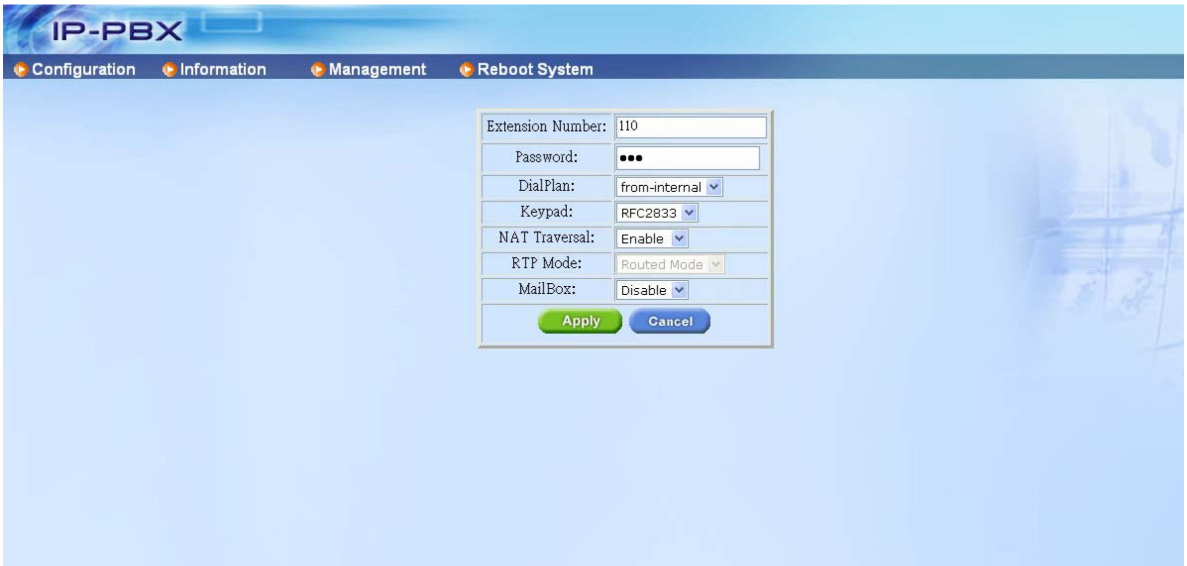
User can reach the auto attendant (**999) now, because we already set incoming cal rule in Routing Table page. Now, they can redial to 22198888 because we enable Guest Allow. For more information about Guest Allow, please go to user's manual CH3- Full Web Configurations.

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 IP PBX-100.



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 IP PBX-100), and she also needs to set the line number/ account/ and password for her Soft Phone. Below is an example

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

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	-none-	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, IP PBX-100 will pay 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.