SIPPBX 6200A
User's Manual

Version: SIPPBXUM.100

CH1. Overview

SIPPBX 6200A IP-PBX is a IP based **IP-PBX** which including legend digital PABX telephony services, Auto Attendant, Voice Mail, Music Ring Back Tone, Conference and Announcement features together. It also works on both IPv6 and IPv4 network address simultaneously, which makes migration from IPv4 to IPv6 network smoothly.

SIPPBX 6200A is not only an IP-PBX, but also an efficient communication tool to help your business and management more efficient. With flexible and full functionality, Welltech SIPPBX 6200A can give a complete transition from traditional PABX to the new generation IP-PBX.

1.1 Specifications

- ➢ **Protocol**
  - ◼ SIP RFC 3261 Compliance/ Asterisk Compatible
  - ◼ Support IPv4/IPv6 Dual Stack
  - ◼ Network (IPv4): Support Fixed IP, DHCP, And PPPoE mode
  - ◼ Network (IPv6): Support Fixed And Autoconfig
- ➢ **IVR**
  - ◼ Web Based Auto Attendant Call Rule
  - ◼ Scheduled And Fixed Greeting
  - ◼ Support Branch Office
  - ◼ System Prompt Recording By Phone Set
- ➢ **Voice Mail**
  - ◼ Voice Mail To e-Mail (IPV4 Only)
  - ◼ Voice Mail System
  - ◼ Message Waiting Indication
  - ◼ Personal Greeting
- ➢ **Toll Restriction**
  - ◼ Provided Different Level Toll Restriction Service
  - ◼ Support Call Duration Restriction
  - ◼ Support Personal Password
  - ◼ Support Outgoing Call Routing Password
- ➢ **Call Features**
  - ◼ Flexible Extension Logic
  - ◼ CPE Based Call Transfer (Consultant, Blind)
  - ◼ Server Based Call Transfer (Consultant, Blind)
  - ◼ Call Forward (Busy, No Answer, Unconditional, Unavailable)
  - ◼ Call Hold/Retrieval
  - ◼ Call Routing
  - ◼ Call Waiting

- Call Pickup (Global, Specific)
- Call Park
- Call Camp-On (Call Back on Busy)
- CLIR (Caller Line Identification Restriction)
- DND (Do Not Disturb)
- Dial Group (Ring All, Sequential Ring, Dynamic Ring)
- Speed Dial
- Music Ring Back Tone
- Music On Hold
- Music On Transfer
- Built-in CDR Report
- Call Monitor
- Broadcasting Service (6 rooms, 16 members per group)
- Meetme Conference (6 rooms, 8 members per room)
- Busy Lamp Field (RFC 4235)
- Asterisk TAPI

➢ **Codecs**
- G.711 (A-Law & µ-Law)
- G.729
- G.723 Pass-Thru
- GSM
- H.263 Pass-Thru
- MPEG4 Pass-Thru

➢ **Technical Features**
- Support Subscriber Registration for IPv4 And IPv6
- Support Call Routing Between IPv4 And IPv6
- Support RTP Proxy Between IPv4 And IPv6
- Support T.38 FAX (IPV4 Only)
- Support DDNS (IPV4 Only)
- Built-in NTP Client/Server
- Built-in DHCP Server (IPV4 Only)
- Built-in Simple Firewall
- Subscriber NAT Transversal
- Behind NAT Support (IPV4 Only)
- Voice Codec Transform (G.711/ G.729/ GSM)
- Auto Provision (Welltech Proprietary)
- Multiple Language Support
- Management: Web Browser Management
- HTTP Firmware Upgrade
- Export/Import Configuration

- Network Interface: 1WAN 1LAN
- DTMF: in-band, RFC2833, SIP-Info

> **Capacity**
- 200 Concurrent Registers
- 50 Concurrent Calls

> **Dimension**
- 19 Inch Rack Mount

1.2 Hardware Overview
1.2.1 The Front Panel

The SIPPBX 6200A LEDs, which inform you about network activities, are located on the front panel.
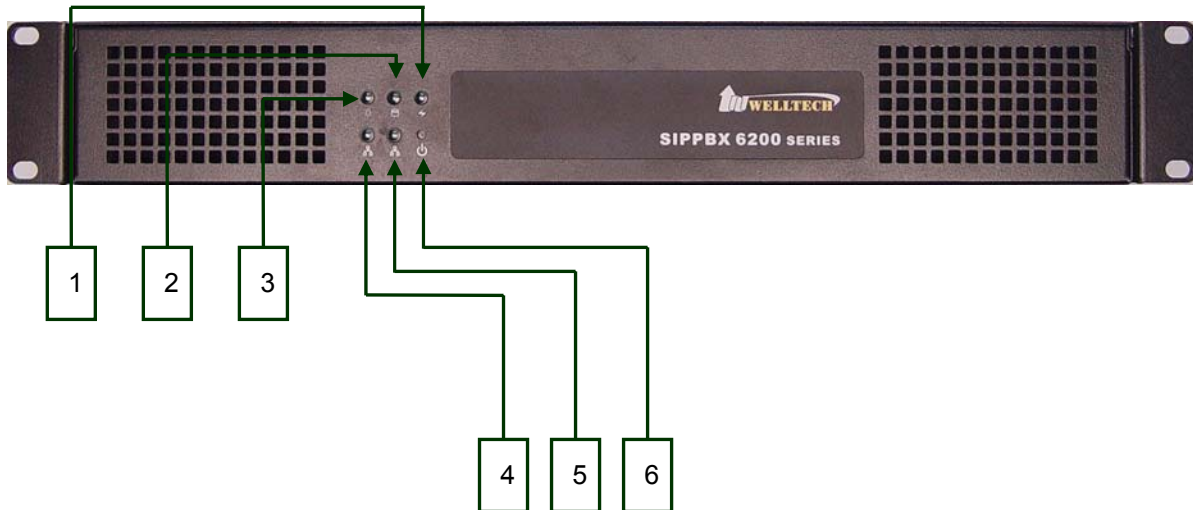
Functions:

- ➢ **1: System Status LED (not used)**
- ➢ **2: H/D LED**
- ➢ **3: Power LED**
- ➢ **4: Network Interface LED (not used)**
- ➢ **5: Network Interface LED (not used)**
- ➢ **6: Power Switch**

1.2.2 The Back Panel

The SIPPBX 6200A ports are located on the back panel.
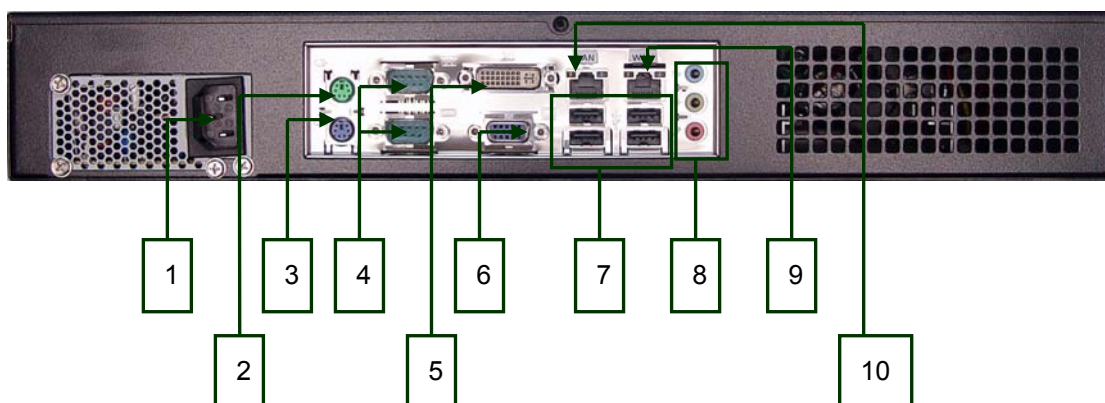
Functions:

- ➢ **1: AC Power Outlet**
- ➢ **2: Mouse (not used)**
- ➢ **3: Keyboard**

- ➢ **4: RS232 Console Port**
- ➢ **5: DVI (not used)**
- ➢ **6: VGA**
- ➢ **7: USB (not used)**
- ➢ **8: Sound (not used)**
- ➢ **9: WAN Interface**
- ➢ **10: LAN Interface**

CH2. Start to configure SIPPBX 6200A

2.1 Connection SIPPBX 6200A

Step 1: Connect LAN port of SIPPBX 6200A with PC via crossover cable or connect with Switch/ Hub via straight through cable.

Step 2: Prepare one computer, and change the IP address to be 192.168.123.12x with subnet mask 255.255.255.0.

Step 3: Open browser and link to default LAN IP address of SIPPBX 6200A "192.168.123.123" with default port number 10087, i.e. http://192.168.123.123:10087

Step 4: Login SIPPBX 6200A with default user ID/Password: "root/root". After login SIPPBX 6200A, user can start to configure basic and essential configurations.
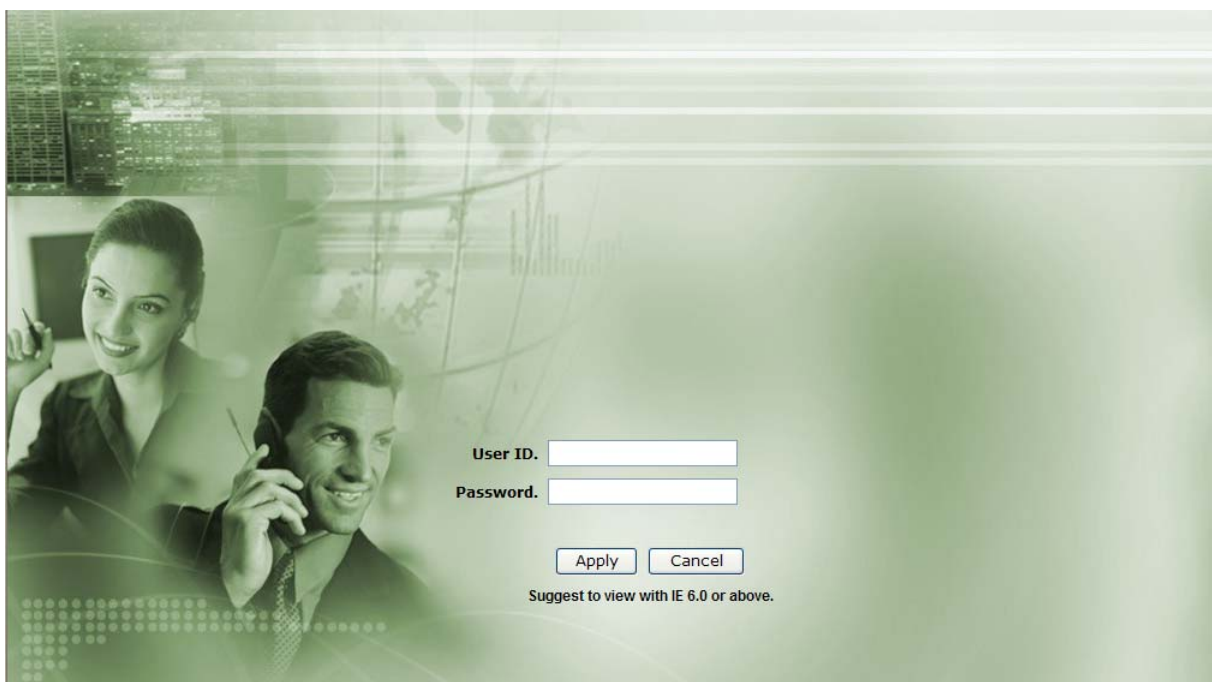
Step 5: To configure basic and essential configurations

To make SIPPBX 6200A work have to set some basic and essential configurations, those include Network, Extension (FXS and IP Phone devices), and Trunk (FXO devices).

2.1.1 Network Configuration

To change your Network Setting, click **Network**, and then click the **Network Setting** table. The screen appears as shown.

Figure Network: Network Setting

The following table describes the table in this screen

Table   Network: Network Setting

Label	Description
WAN	
IP Support	Select use IP mode, provide ipv4, ipv6, ipv4/ipv6.
Mode	Select SIPPBX 6200A WAN port network mode to be Fixed IP, DHCP or PPPoE.
IP Address	Enter the IP Address. If user has set SIPPBX 6200A to be fixed IP mode.
Subnet Mask	Enter the Subnet Mask Address. If user has set SIPPBX 6200A to be fixed IP mode.
Default Gateway	Enter the Default Gateway Address. If user has set SIPPBX 6200A to be fixed IP mode.

Primary DNS	Enter the IP address for Primary DNS. The default is 168.95.1.1.
Secondary DNS	Enter the IP address for Secondary DNS. The default is null.
Default Gateway Backup	Select Enable option, if any connection problem occurred on primary default gateway connections, all the traffic will be guided and switched to the secondary default gateway for proper operation. The default is Disable.
Secondary Default Gateway	Enter the Secondary Default Gateway. If you choose the Default Gateway Backup to Enable.
Check Point	Enter the Check Point IP address. SIPPBX 6200A use ping this ip address, to check have any connection problem occurred on primary default gateway connections. If you choose the Default Gateway Backup to Enable.
Current use Default Gateway	This field will display now use the Defaut Gateway IP address.
PPPoE ID	Enter the PPPoE ID. If you choose the Mode to PPPoE.
PPPoE PWD	Enter the PPPoe Password If you choose the Mode to PPPoE.
IPV6 Mode	Select Manual option, you can enter IP Address/ Prefix Length/ Gateway.
IPV6 IP Address	Display the IPV6 IP Address. You can enter the IPV6 IP Address, if you choose the IPV6 mode to Manual.
IPV6 Prefix Length	Display the IPV6 Prefix Length. You can enter the IPV6 IP Address, if you choose the IPV6 mode to Manual.
IPV6 IP Gateway	Display the IPV6 Default Gateway Address. You can enter the IPV6 Default Gateway Address, if you choose the IPV6 mode to Manual.
MAC	This field shows the MAC address. The Mac address cannot be modified.
LAN (IPV4)	
IP Address	Enter the IP Address. The default is 192.168.123.123.
Subnet Mask	Enter the Subnet Mask Address. The default is 255.255.255.0.
MAC	This field shows the MAC address. The Mac address cannot be modified.
Network Routing Table (IPV4)	
Select	Select this check box, then modify or delete it.
Destination	This field shows the IP address.
Network	This field shows the Subnet Mask address.
Gateway	This field shows the Default Gateway address.
Add	Click on the Add button, then display Network Router screen.

Modify	IP address can be modify by click on the checkbox next to the IP address and click on the Modify button.
Delete	IP address can be deleted by click on the checkbox next to the IP address and click on the Delete button.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

2.1.2 Extension Configuration

User has to set Extension account for other device to register on SIPPBX 6200A.

To change your Extension, click **Configuration**, and then click the **Extension** table. The screen appears as shown.

Figure Configuration: Extension

The following table describes the table in this screen

Table   Configuration: Extension

Label	Description
Select	Select this check box, then modify or delete it.
Extension Number	This field shows the Extension Number No information
Comment	This field shows the Comment information.
Type	This field shows the Type information.
NAT Traversal	This field shows the NAT Traversal information.
Dial Plan	This field shows the Dial Plan information.
Add	Click on the Add button, then display Extension Setting screen.
Modify	A extension can be modify by click on the checkbox next to the extension and click on the Modify button.
Delete	A extension can be deleted by click on the checkbox next to the A extension and click on the Delete button.
Quick Add	Click on the Add button, then display Extension Setting screen.
Quick Delete	Click on the Add button, then display Extension Setting screen.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.

Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Extension Setting

The following table describes the table in this screen

Table Configuration: Extension Setting

Label	Description
Extension Number	Assign the number of Extension. This number is also the register name for device.
Subscriber Type	Chose one option to Subscriber Type. Provide drop-down options: Normal, Autoconfig. Normal: You must enter Password. Autoconfig: You must enter MAC Address.
Password	Select Normal option to Subscriber Type. Assign the register password for device to register on SIPPBX 6200A.
Subscriber MAC	Select Autoconfig option to Subscriber Type. You must enter the CPE Device MAC Address.
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: CH3.1.8 Outgoing Routing.
Keypad	User can select Keypad type to be RFC2833, In-band, SIP-Info and Auto. You can choose Auto to auto select the Keypad type. Choose RFC2833, Inband or SIP-Info here will force the Extension use RFC2833, Inband or SIP-Info only and the setting should be also match the Keypad setting of Extension device. **Note:** Now SIPPBX 6200A could not support G729 with Inband Keypad type. If SIPPBX 6200A detect the caller or callee not

	support RFC2833 DTMF type. Then SIPPBX 6200A will force the Codec to G711 to make sure the DTMF detection is correctly.
NAT Traversal	If the Extension device is behind a device performing NAT, such as firewall or router, and need to register to SIPPBX 6200A on public network, then user has to enable this function. Enable NAT Traversal to force SIPPBX 6200A to ignore the contact information for the Extension and use the address from which the packets are being received.
Fixed Trunk ID	User can define a Fixed Trunk for a certain extension. When such extension makes an outgoing call via routing table, SIPPBX 6200A will check "Fixed Outgoing Call Rule" first. If "Fixed Outgoing Call Rule" is enabled, then SIPPBX 6200A will confirm the Fix Trunk ID for the calling party. That means the outbound call will be routed by Fixed Trunk ID, if you define the Fixed Trunk ID for the calling party and you also enable "Fixed Outgoing Call Rule". **Note:** For more information about Fixed Outgoing Call Rule, please refer to the user manual: CH3.1.8 Outgoing Routing.
Absolute Timeout	Specific the timeout value for the outgoing calls. Please also go to Outgoing Call Rule page to enable the Route Timeout function.
BLF	Enable BLF function for extensions.
Forward CallerID	By default, the "from header of SIP invite" will contain the caller's line number when forward function is activated. But this may make some errors occurred for some SIP Trunk services. So we add this function in the "Extension Setting" page, to let user modify the line number of SIP Invite's from header, from calling party's number to the called party's number.
Unconditional FWD	Enable Unconditional forward function for extensions.
No Answer FWD	Enable No Answer forward function for extensions.
Busy FWD	Enable Busy forward function for extensions.
Unavailable FWD	Enable Unavailable forward function for extensions.
Comment	You can input a 20 bytes note for each extension here.
Mail Box	User can select to disable or enable mail box function. If this function is enabled, user could 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. You can also login the mail box system by dialing to *98.

E-Mail Address	This field will appear when you enable Mail Box function and you can input the E-Mail Address here for voice mail to E-mail. Note: Please remember set the SMTP in the page of click Management, and then click SMTP Setting to activate the Voice Mail to E-mail. If the SIPPBX 6200A got a new message, it will send the message to the user by email immediately. If you are using SIPPBX 6200A and u just hope the SIPPBX 6200A to save voice mail to it and not send the email. You just need to input "x" to E-Mail Address.
Save VM to Local	If you select Enable to Save VM To Local, when you have Voice Mail message, will backup to Local folder.
VM Login Password	SIPPBX 6200A has a built-in voice mail system. And user can login voice mail system by dialing to *98, then input the mailbox number and password for voice mail. User can define the Voice Mail box login password here. Another way to login the voice mail system is dial to *98+extension number. For example, dial to *98101 can login EXT101's voice mail box, and caller can just input password to access voice mail.
Voice Mail Count	Display the exact count of New Messages and Old Messages.
Delete MailBox Content	User can delete all of the voice mails and personal greeting by mark the "Delete MailBox Content" and then press Apply.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

2.1.3 Trunk Configuration

User has to set Trunk account for Trunk (FXO device, e.g. WG3804A) to register to SIPPBX 6200A or set some necessary configuration for SIP trunk.

To change your Trunk, click **Configuration**, and then click the **Trunk** table. The screen appears as shown.

Figure Configuration: Trunk

The following table describes the table in this screen

Table Configuration: Trunk

Label	Description
Select	Select this check box, then modify or delete it.
Trunk Number	This field shows the Trunk Number information
Comment	This field shows the Comment information
NAT Traversal	This field shows the NAT Traversal information
Maximum Channels	This field shows the Maximum Channels information
Add	Click on the Add button, then display Trunk Setting screen.
Modify	A Trunk can be modify by click on the checkbox next to the Trunk and click on the Modify button.
Delete	A Trunk can be deleted by click on the checkbox next to the A Trunk and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page

	number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Example 1: Set Trunk for FXO gateway

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Trunk Setting

The following table describes the table in this screen

Table   Configuration: Trunk Setting

Label	Description
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 SIPPBX 6200A.
Host	Setting the Host to Dynamic will require the trunk to register the SIPPBX 6200A so that the SIPPBX 6200A know 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] appeared 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.

- 20 -

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 Trunk. 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: CH3.1.8 Outgoing Routing.
Keypad	User can select Keypad type to be RFC2833, In-band, or SIP-Info and Auto. You can choose Auto to auto select the Keypad type. Choose RFC2833, Inband or SIP-Info here will force the Extension use RFC2833, Inband or SIP-Info only and the setting should be also match the Keypad setting of Trunk device.
NAT Traversal	If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to SIPPBX 6200A on public network, then user has to enable this function. Enable NAT Traversal to force SIPPBX 6200A to ignore the contact information for the Trunk and use the address from which the packets are being received.
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 SIPPBX 6200A 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 SIPPBX 6200A'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 SIPPBX 6200A to the Destination (Trunk) to avoid such security issue.
	● If you set a Welltech 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.
Caller ID Display Name	When inbound call is coming from Trunk, such as 888. The caller ID Name will be the "PSTN number" or "888". Specify this will use the current setting instead.
Comment	You can input a 10 byte note for each Trunk here.
CLIR Support	CLIR means "Caller Line Identification Restriction". It is a proper noun. It is a feature to hide the caller's number. For example, ext 101 call to ext 102. But 101 won't like to show the caller ID to 102. So 101 can activate this feature to hide the caller ID. When 102 got a call from 101, the LCD of 102 should display "Anonymous".
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

CH3. Full Web Configurations

After Login SIPPBX 6200A will see screen as below, and there are four main categories, user can click on each category to extend detail items.
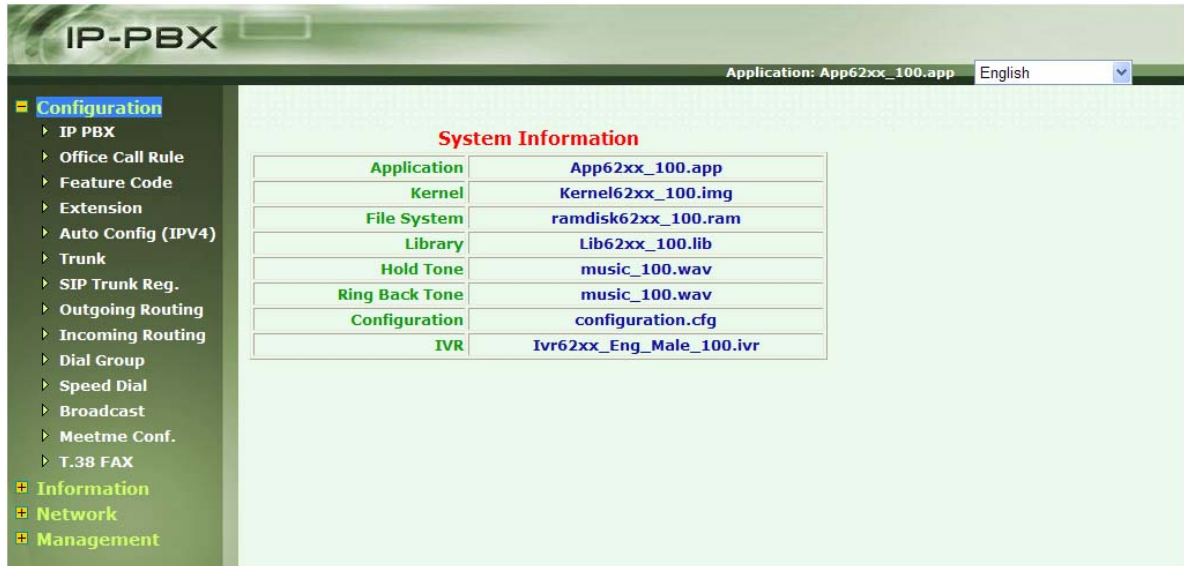
- **Configuration**: Include all telephony configurations of SIPPBX 6200A.
  - IP PBX
  - Office Call Rule
  - Feature Code
  - Extension
  - Auto Config (IPv4)
  - Trunk
  - SIP Trunk Reg.
  - Outgoing Routing
  - Incoming Routing
  - Dial Group
  - Speed Dial
  - Broadcast
  - Meetme Conf.
  - T.38 FAX
- **Information**: To show related information.
  - Subscriber Info.
  - Call Monitor
  - CDR
  - System Info.
- **Network**: To show related information.
  - Network

- DHCP Srv. (Pv4)
- DDNS Srv. (Pv4)

■ Management: Include all system management of SIPPBX 6200A.

- Time Setting
- SMTP Setting
- VM Setting
- Security
- Firmware Upload
- Music Upload
- Import Setting
- Export Setting
- Reset To Default
- Reboot System: To reboot system of SIPPBX 6200A.
- Power Off System

3.1 Configuration

The Network screens can help you configuration IP PBX, Office Call Rule, Feature Code, Extension, Auto config (IPV4), Trunk, SIP Trunk Reg., Outgoing Routing, Dial Group, Speed Dial, Broadcast, Meetme Conf., T.38 FAX.

3.1.1 IP PBX

To change your IP PBX Setting, click **Configuration**, and then click the **IP PBX** table. The screen appears as shown.

Figure Configuration: IP PBX

The following table describes the table in this screen

Table   Configuration: IP PBX

Label	Description
SIP Setting	
IP-PBX Realm	Enter the IP-PBX Realm of SIPPBX 6200A. This parameter is essential when there is more than one SIPPBX 6200A, and user wants to have inter-calls between SIPPBX 6200A. Please refer to SIP Trunk configuration.
IP-PBX User Agent	Enter the IP-PBX User Agent. IP-PBX User Agent takes as its argument a string specifying the value for the user agent field in the SIP header. The default value is IP-PBX.
Proxy Port	Enter the Proxy Port. These optional parameters allow you to control the port on which you wish the SIPPBX 6200A to accept SIP connections. Default is 5060.
RTP Port Start	Enter the RTP Port Start. The voice media will use RTP as the transport protocol. You can define the RTP port range that SIPPBX 6200A opened. Default start port is 10000.

RTP Port End	Enter the RTP Port End. The voice media will use RTP as the transport protocol. You can define the RTP port range that SIPPBX 6200A opened. Default end port is 20000. Note: Default RTP port range is 10000 to 20000 and default proxy port is 5060. If your SIPPBX 6200A 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 SIPPBX 6200A.
Max Expire Time	This sets the maximum amount of time, in seconds. This is used for the registration expire time. If this value less than the expired time from the client, and then click the SIPPBX 6200A will reply a certain expire time which is defined in "Default Expire Time" to client.
Default Expire Time	This sets the default SIP registration expiration time, in seconds. A client will normally define this value when it initially registers, so the default value you set here will be used only if the client does not specify a timeout when it registers. If you are registering to another SIP Trunk, this is the registration timeout that it will send to the far end.
Codec Priority	
Codec Priority	Codec negotiation is attempted in the order in which the Codec Priority is defined. Default is G729 with first priority, G711u with second priority, G711A with third priority and GSM is fourth priority. That means the SIPPBX 6200A can only recognize these four Codec and it will force the Codec with the specified priority and forward to another subscriber. Now, SIPPBX 6200A can support G729, G711U, G711A, GSM and G723 Pass-Thru.
PBX Setting	
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.
Music RBT	If this is call between extensions. Enabling this option will provide music to the calling party until the call is answered.
Music RBT (After AA)	If this is call comes from Auto Attendant. Enabling this option will provide music to the calling party until the call is answered.
Call Monitor Refresh	SIPPBX 6200A have call monitor function. The call situation will

	be refreshed by the refresh time. Default is 30 seconds and user can change it here.
RTP Timeout	It terminates a call if no RTP data received within the time specified.
Video Support	This field will enable video call with H263 pass-through or MPEG4 pass-through.
Video Format	Chose one option to Video Fomat. Provide drop-down options: H263 pass-through or MPEG4 pass-through.
SRVlookup Support	Enable or disable SRV lookup. DNS SRV is a way of setting up a logical, resolvable address where you can be reached. This allows calls to be forwarded to different locations without the need to change the logical address, but your DNS Server must support it also. If not sure, disable it. This option is Disable turned on by default!!!
Invalid Number Support	Normally, a busy tone will be heard if caller dial to a non-exist callee. Enable this option, the caller should hear an announcement to notify him that callee is not exist.
Behind NAT	
Behind NAT	If your SIPPBX 6200A is behind NAT, we strongly suggest you to enable Behind NAT to avoid some unexpected issue, such as "one way voice".
External IP	If you input External IP, SIPPBX 6200A will take that IP address as its argument. If SIPPBX 6200A 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.
Secondary External IP	This should cooperate with Default Gateway Backup in Network page. When Default Gateway Backup is enabled, the SIPPBX 6200A will auto switch the default gateway to secondary one if primary default gateway is broken. The External IP is not functional when Backup Default Gateway is chosen. So you must input the Secondary External IP for Backup Default Gateway.
External Host	External Host takes a fully qualified domain name as its argument. If SIPPBX 6200A is behind NAT, the SIP header will normally use the private IP address assigned to the server. If you set this option, SIPPBX 6200A 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 choose one for External IP or External Host for "Behind NAT".
Local Net	Local Net is used to tell SIPPBX 6200A which IP addresses are considered local. If one of caller or callee is not under Local Net, SIPPBX 6200A 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
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.2 Office Call Rule

You can define a business time to forward incoming call to company announcement or a certain destination.

6200A provide 12 kinds office call rule (**998 to **988). User can setup a FXO gateway and hotline to **999 (for office 1) or **988 (for office 12) to reach auto attendant.

Now user can make SIPPBX 6200A to decide the destination when it got an invite with called number as **999 to **988. When SIPPBX 6200A got an invite with **999, SIPPBX 6200A will confirm the current time and forward this call to AA, Ext, Group or Outbound. If you choose the destination to EXT, Group or Outbound, please remember to input the destination number into the following field.

When you set the destination to AA, please refer to CH4.1.3 How to record the other system prompts for the greeting recording.

To change your Office Call Rule, click **Configuration**, and then click the **Office Call Rule** table. The screen appears as shown.

Figure Configuration: Office Call Rule

Office Call Rule				
Select	**Representative No.**	**Comment**	**Operator**	**Call Rule**
☐	**999	Office 1	9	Scheduled Call Rule
☐	**998	Office 2	9	Scheduled Call Rule
☐	**997	Office 3	9	Scheduled Call Rule
☐	**996	Office 4	9	Scheduled Call Rule
☐	**995	Office 5	9	Scheduled Call Rule
☐	**994	Office 6	9	Scheduled Call Rule
☐	**993	Office 7	9	Scheduled Call Rule
☐	**992	Office 8	9	Scheduled Call Rule
☐	**991	Office 9	9	Scheduled Call Rule
☐	**990	Office 10	9	Scheduled Call Rule
☐	**989	Office 11	9	Scheduled Call Rule
☐	**988	Office 12	9	Scheduled Call Rule
		Modify		

The following table describes the table in this screen

Table   Configuration: Office Call Rule

Label	Description
Select	Select this check box, then modify it.
Representative No.	This field shows the Representative No information
Comment	This field shows the Comment information.
Operator	This field shows the Operator information.
Call Rule	This field shows the Call Rule information.
Modify	A Representative no can be modify by click on the checkbox next to the Representative no and click on the Modify button.

Click **Modify**. The screen appears as shown.

Figure Configuration: Office 1 Call Rule

The following table describes the table in this screen

Table Configuration: Extension Setting

Label	Description
General Setting	
Representative No	Enter the Representative No.
Comment	You can input a 20 byte note for each Comment information..
Call Rule	
Rule	Chose one option to the Call Rule. Provide drop-down options: Scheduled Call Rule, Fixed to Working Time Call Rule, Fixed to Non-working Time Call Rule, Fixed to Special Time Call Rule.
Working Day	Chose one option to the Working Day.
Working Time AM	Chose one option to the Working Time AM. To: Chose one option to the Rule. Provide drop-down options: AA, EXT, Group, Outbound. If you select "EXT or Group and Outbound", will auto dispaly "-- " field. You must input the "EXT or Group and Outbound" data in this field.
Lunch Break	Chose one option to the Lunch Break. To: Chose one option to the Rule. Provide drop-down options:

	AA, EXT, Group, Outbound. If you select "EXT or Group and Outbound", will auto dispaly "-- " field. You must input the "EXT or Group and Outbound" data in this field.
Working Time PM	Chose one option to the Working Time PM.
Non-working Time	Chose one option to the Non-working Time.
Operator setting	
Operator	Configure the Operator number and the destination to Extension or Call Group.
OP to EXT	If you set Operator to EXT, you can set extension number here.
OP Ext End To	When you set Operator as an Extension, you can define the final destination to IVR system or Extension's function (i.e. voice mail) if Operation does not answer.
AA setting	
Play AA Greeting	You can define the times of greeting announcement, when caller entered Auto Attendant system.
AA Greeting End & Interval Timer	By default, the caller will hear greeting message 3 times when he reach the auto attendant. There will be an 3 seconds interval between these greeting messages. Now users can change the intervals here.
AA Greeting End To (Working Time)	Decide the destination after greeting announcement finished on working time.
AA Greeting End To (Lunch Break)	Decide the destination after greeting announcement finished on Lunch Break time.
AA Greeting End To (non-Working Time)	Decide the destination after greeting announcement finished on non-Working time.
AA Greeting End To (Special Time)	Decide the destination after greeting announcement finished on Special time.
Import Greeting	File Name: Chose one option Greeting file. Provide drop-down options: greeting-day.wav (Working Time Greeting)…, greeting-noon.wav (Lunch Break Greeting). File: Type in the location of the file you want to upload in this field or click on the Browse.. button to find it. Import: Click on the Import button to upload greeting file.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.3 Feature Code

To change your Feature Code, click **Configuration**, and then click the **Feature Code** table. The screen appears as shown.

Figure Configuration: Feature Code

Feature Code	
System Prompt Recording:	*50
System Prompt Recording PWD:	000
System Prompt Recording Prefix:	**
System Prompt Listen Prefix:	***
DND Activated:	*78
DND Deactivated:	*79
UCF Activated:	*72
UCF Deactivated:	*73
BF Activated:	*90
BF Deactivated:	*91
NAF Activated:	*92
NAF Deactivated:	*93
UAF Activated:	*94
UAF Deactivated:	*95
OUCF Start:	*87
OUCF Activated:	*88
OUCF Deactivated:	*89
CF Deactivated:	*96
Voice Mail Box Login:	*98
Camp-On Activated:	*66
CLIR(per call) Prefix:	*67
CLIR Activated:	*31
CLIR Deactivated:	*32
Ext Pwd Activated:	*80
Ext Pwd Deactivated:	*81

Apply   Cancel

The following table describes the table in this screen

Table   Configuration: Feature Code

Label	Description
System Prompt Recording	User could dial an access code for system prompt recording, such as **111 for greeting-day.wav. Before dialing to **111, user should dial to the feature code of "System Prompt Recording" to start recording. Default feature code for System Prompt Recording is [*50]. So the recording procedure should be "Dial to [*50]→ Input password (which defined in [System Prompt Recording PWD])→ dial to access code (i.e. **111)→ Start recording". Add the feature for recording will avoid an unknown user incautious to record the system prompt.
System Prompt Recording PWD	Before recording System Prompt, user may need to input password. Here you can specify the password for System

	Prompt Recording. Default is 000. That means password is not necessary if this field is empty.
System Prompt Recording Prefix	The prefix is for access code of System Prompt Recording. Default is **. For example, the access code for [greeting-day.wav] is **111. So the System Prompt Recording Prefix is **. If you change the Prefix to *1, that means the access code for [greeting-day.wav] should be *1111.
	Note: Previously, you can just dial to the access code, such as **111, for announcement recording. But we change this procedure due to the security issue.
	For example, the record procedure of greeting message will be: "Dial to [*50]→ Input password [000]→ dial to access code [**111]→ Start to record greeting-day.wav". For more information about announcement recording, please refer to user manual: **CH4.1.3 How to record the other system prompts**
System Prompt Listen Prefix	The prefix is for access code of System Prompt listening. Default is ***. For example, the access code for [greeting-day.wav] listening is ***111. So the System Prompt Recording Prefix is ***. If you change the Prefix to *11, that means the access code for [greeting-day.wav] listening should be *11111.
DND Activated	The code to activate DND. Default is *78.
DND Deactivated	The code to deactivate DND. Default is *79.
UCF Activated	The code to activate Unconditional Forward. Default is *72. For example, dialing to *72101 will forward all the call to 101.
UCF Deactivated	The code to deactivate Unconditional Forward. Default is *73.
BF Activated	The code to activate Busy Forward. Default is *90. For example, dialing to *90101 will forward call to 101 if you are on the phone.
BF Deactivated	The code to deactivate Busy Forward. Default is *91.
NAF Activated	The code to activate No Answer Forward. Default is *92. For example, dialing to *92101 will forward call to 101 if you are not answering.
NAF Deactivated	The code to deactivate No Answer Forward. Default is *93.
UAF Activated	The code to activate Unavailable Forward. Default is *94. For example, dialing to *94101 will forward call to 101 if your phone is not registering.
UAF Deactivated	The code to deactivate Unavailable Forward. Default is *95.

OUCF Start	Provide user use remote control function, then control Unconditional Forward. Default is *87. Note: 1. You can just dial to the PBX, and then PBX will play greeting message. 2. Dial to [*87] to use remote control function; then play greeting message. 3. You can enable or disable OUCF function.
OUCF Activated	The code to activate Unconditional Forward. Default is *88. Note: 1. You can just dial to the PBX, and then PBX will play greeting message. 2. Dial to [*87] to use remote control function; then play greeting message. 3. After, dialing to *880282265699 will forward all the call to 0282265699.
OUCF Deactivated	The code to set disable for Unconditional Forward. Default is *89. Note: 1. You can just dial to the PBX, and then PBX will play greeting message. 2. Dial to [*87] to use remote control function; then play greeting message. 3. After, dialing to [*89] to disable OUCF function.
CF Deactivated	Disable all of the forward function, including Unconditional Forward, Busy Forward, No Answer Forward and Unavailable Forward. Default is *96.
Voice Mail Box Login	For SIPPBX 6200A only. SIPPBX 6200A has the ability to store voice mail within itself, and user can just dial to the feature code to login the voice mail system. The feature code of voice mail system default is *98.
Camp-On Activated	This function means [call back on busy]. For example, you dial to 101 but 101 is on the phone, then you should hear an announcement for called person is busy. You could dial to *66 by default to trigger the SIPPBX 6200A call back to you when 101 is idle. This function will let u talk to called party immediately when called party is not busy. Note: This Function is only workable when voice mail function is disabled.

	When this function is enabled, SIPPBX 6200A will check the status of called party every 20 seconds, at most 15 times. That means the camp-on function may be performed when called party is idled after 20 seconds at most. And 300 (20*15) seconds later, this function will not be workable.
CLIR (per call) Prefix	Default is *67. Add this prefix will hide the caller's number. For example, 101 does not hope to show the caller id to 102. 101 can just dial to "*67102", where the *67 is the prefix for CLIR. When 102 got the incoming call, the LCD of 102 should display "Anonymous". If 101 just dial to "102", then 102 should see the Caller ID as 101.
CLIR Activated	Default is *31. For example. 101 dial to "*31", SIPPBX 6200A should add the CLIR record for 101 into its database. When 101 call to 102, 103...,etc. The LCD of called party should always show "Anonymous".
CLIR Deactivated	Default is *32. Dialing to *32 will remove the CLIR record from the database of SIPPBX 6200A.
Ext Pwd Activated	Ext Pwd means extension password. This is a personal password for external outgoing call. For example, ext 101 dial to *80+123, The Ext Pwd is activated. From now on, the ext 101 must input the password 123 for the external outgoing calls which are defined in the [Outgoing Routing] page..
Ext Pwd Deactivated	This is a feature code to disable extension password (Ext Pwd).
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.4 Extension

User has to set Extension account for other device to register on SIPPBX 6200A.

To change your Extension, click **Configuration**, and then click the **Extension** table. The screen appears as shown.

Figure Configuration: Extension

Select	Extension Number ▾	Comment	Type	NAT Traversal	Dial Plan
☐	101	EasonTest	Normal	Enable	ext+allroute
☐	105	none	Normal	Disable	ext+allroute
☐	107	none	Normal	Disable	ext+allroute
☐	108	none	Normal	Disable	ext+allroute
☐	109	none	Normal	Disable	ext+allroute
☐	201	none	Normal	Disable	ext+allroute
☐	202	none	Normal	Disable	ext+allroute
☐	203	none	Normal	Disable	ext+allroute
☐	204	none	Normal	Disable	ext+allroute
☐	206	none	Normal	Disable	ext+allroute
☐	207	none	Normal	Disable	ext+allroute
☐	208	none	Normal	Disable	ext+allroute

Add　Modify　Delete　Quick Add　Quick Delete

Total Count: 199　Number Search:＿＿＿＿＿　Total Page: 17　Page: 1　Next

The following table describes the table in this screen

Table　Configuration: Extension

Label	Description
Select	Select this check box, then modify or delete it.
Extension Number	This field shows the Extension Number No information
Comment	This field shows the Comment information.
Type	This field shows the Type information.
NAT Traversal	This field shows the NAT Traversal information.
Dial Plan	This field shows the Dial Plan information.
Add	Click on the Add button, then display Extension Setting screen.
Modify	A extension can be modify by click on the checkbox next to the extension and click on the Modify button.
Delete	A extension can be deleted by click on the checkbox next to the A extension and click on the Delete button.
Quick Add	Click on the Add button, then display Extension Setting screen.
Quick Delete	Click on the Add button, then display Extension Setting screen.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.

Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Extension Setting

The following table describes the table in this screen

Table   Configuration: Extension Setting

Label	Description
Extension Number	Assign the number of Extension. This number is also the register name for device.
Subscriber Type	Chose one option to Subscriber Type. Provide drop-down options: Normal, Autoconfig. Normal: You must enter Password. Autoconfig: You must enter MAC Address.
Password	Select Normal option to Subscriber Type. Assign the register password for device to register on SIPPBX 6200A.
Subscriber MAC	Select Autoconfig option to Subscriber Type. You must enter the CPE Device MAC Address.
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: CH3.1.8 Outgoing Routing.
Keypad	User can select Keypad type to be RFC2833, In-band, SIP-Info and Auto. You can choose Auto to auto select the Keypad type. Choose RFC2833, Inband or SIP-Info here will force the Extension use RFC2833, Inband or SIP-Info only and the setting should be also match the Keypad setting of Extension device. **Note:** Now SIPPBX 6200A could not support G729 with Inband Keypad type. If SIPPBX 6200A detect the caller or callee not

	support RFC2833 DTMF type. Then SIPPBX 6200A will force the Codec to G711 to make sure the DTMF detection is correctly.
NAT Traversal	If the Extension device is behind a device performing NAT, such as firewall or router, and need to register to SIPPBX 6200A on public network, then user has to enable this function. Enable NAT Traversal to force SIPPBX 6200A to ignore the contact information for the Extension and use the address from which the packets are being received.
Fixed Trunk ID	User can define a Fixed Trunk for a certain extension. When such extension makes an outgoing call via routing table, SIPPBX 6200A will check "Fixed Outgoing Call Rule" first. If "Fixed Outgoing Call Rule" is enabled, then SIPPBX 6200A will confirm the Fix Trunk ID for the calling party. That means the outbound call will be routed by Fixed Trunk ID, if you define the Fixed Trunk ID for the calling party and you also enable "Fixed Outgoing Call Rule". **Note:** For more information about Fixed Outgoing Call Rule, please refer to the user manual: CH3.1.8 Outgoing Routing.
Absolute Timeout	Specific the timeout value for the outgoing calls. Please also go to Outgoing Call Rule page to enable the Route Timeout function.
BLF	Enable BLF function for extensions.
Forward CallerID	By default, the "from header of SIP invite" will contain the caller's line number when forward function is activated. But this may make some errors occurred for some SIP Trunk services. So we add this function in the "Extension Setting" page, to let user modify the line number of SIP Invite's from header, from calling party's number to the called party's number.
Unconditional FWD	Enable Unconditional forward function for extensions.
No Answer FWD	Enable No Answer forward function for extensions.
Busy FWD	Enable Busy forward function for extensions.
Unavailable FWD	Enable Unavailable forward function for extensions.
Comment	You can input a 20 bytes note for each extension here.
Mail Box	User can select to disable or enable mail box function. If this function is enabled, user could 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. You can also login the mail box system by dialing to *98.

E-Mail Address	This field will appear when you enable Mail Box function and you can input the E-Mail Address here for voice mail to E-mail. Note: Please remember set the SMTP in the page of click Management, and then click SMTP Setting to activate the Voice Mail to E-mail. If the SIPPBX 6200A got a new message, it will send the message to the user by email immediately. If you are using SIPPBX 6200A and u just hope the SIPPBX 6200A to save voice mail to it and not send the email. You just need to input "x" to E-Mail Address.
Save VM to Local	If you select Enable to Save VM To Local, when you have Voice Mail message, will backup to Local folder.
VM Login Password	SIPPBX 6200A has a built-in voice mail system. And user can login voice mail system by dialing to *98, then input the mailbox number and password for voice mail. User can define the Voice Mail box login password here. Another way to login the voice mail system is dial to *98+extension number. For example, dial to *98101 can login EXT101's voice mail box, and caller can just input password to access voice mail.
Voice Mail Count	Display the exact count of New Messages and Old Messages.
Delete MailBox Content	User can delete all of the voice mails and personal greeting by mark the "Delete MailBox Content" and then press Apply.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Click **Quick Add**. The screen appears as shown.

Figure Configuration: Extension Setting

The following table describes the table in this screen

Table   Configuration: Extension Setting

Label	Description
Quick Add	Enter the number of the Qucik Add. If you enter 30, then extension number enter 330, will auto create 330 ~ 359 extension number.
Extension Number	Assign the start number of Extension. This number is also the register name for device.
Subscriber Type	Chose one option to Subscriber Type. Provide drop-down options: Normal, Autoconfig. Normal: You must enter Password.
Password	Select Normal option to Subscriber Type. Assign the register password for device to register on SIPPBX 6200A, the password will automatically increase by 1 digit.
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: CH3.1.8 Outgoing Routing.
NAT Traversal	If the Extension device is behind a device performing NAT, such as firewall or router, and need to register to SIPPBX 6200A on public network, then user has to enable this function. Enable NAT Traversal to force SIPPBX 6200A to ignore the contact information for the Extension and use the address from which the packets are being received.
BLF	Enable BLF function for extensions.
Mail Box	User can select to disable or enable mail box function. If this function is enabled, user could 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. You can also login the mail box system by dialing to *98, if you are using an SIPPBX 6200A.
Save VM to Local	If you select Enable to Save VM To Local, when you have Voice Mail message, will backup to Local folder.
VM Login Password	SIPPBX 6200A has a built-in voice mail system. And user can login voice mail system by dialing to *98, then input the mailbox number and password for voice mail. The password will automatically increase by 1 digit. User can define the Voice Mail box login password here. Another way to login the voice mail system is dial to *98+extension number. For example, dial to *98101 can login EXT 101's voice mail box, and caller can

	just input password to access voice mail.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Click **Quick Delete**. The screen appears as shown.

Figure Configuration: Extension Setting

The following table describes the table in this screen

Table   Configuration: Extension Setting

Label	Description
Quick Delete	Enter the number of the Qucik Delete. If you enter 30, then extension number enter 330, will auto delete 330 ~ 359 extension number.
Extension Number	Enter the will delete start number of Extension Number.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.5 Auto COnfig (IPv4)

To change your Auto Config (IPV4), click **Configuration**, and then click the **Auto Config (IPV4)** table. The screen appears as shown.

Figure Configuration: Auto Config-Device

Auto Config- Device

Select	Model	Call Waiting	Broadcasting	End of Digit	Current Version
☐	LP600n	Disable	Disable	#	lp600n.106_2.bin

Add Modify Delete

Auto Config- Office

Select	Office	Device Refresh	Firmware Update	Time Zone	Daylight Saving
☐	welltech	60	99:99	GMT +3.5	Disable

Add Modify Delete

The following table describes the table in this screen

Table   Configuration: Auto Config-Device

Label	Description
Auto Config-Device	
Select	Select this check box, then modify or delete it.
Model	This field shows the Model information
Call Waiting	This field shows the Call Waiting information
Broadcasting	This field shows the Broadcasting information
End of Digit	This field shows the End of Digit information
Current Version	This field shows the Current Version information
Add	Click on the Add button, then display Config-Device screen.
Modify	A Model can be modify by click on the checkbox next to the A Model and click on the Modify button.
Delete	A Model can be deleted by click on the checkbox next to the A Model and click on the Delete button.
Auto Config-Office	
Select	Select this check box, then modify or delete it.
Office	This field shows the Office information
Device Refresh	This field shows the Device Refresh information
Firmware Update Time	This field shows the Firmware Update Time information
Time Zone	This field shows the Time Zone information
Daylight Saving	This field shows the Daylight Saving information
Add	Click on the Add button, then display Config-Office Setting screen.

Modify	A office can be modify by click on the checkbox next to the office and click on the Modify button.
Delete	A office can be deleted by click on the checkbox next to the A office and click on the Delete button.

To change your Auto Config-Device, click **Configuration**, and then click the **Auto Config-Device** table. The screen appears as shown.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Auto Config-Device

The following table describes the table in this screen

Table   Configuration: Auto Config-Device

Label	Description
Device Model	
Model Name	Chose one option to Model name. This function is welltech proprietary, now only support lp600n.
Current Version	This field shows the S/W Version.
Firmware Upload [Edit]	Upload the S/W file for the IP-Phone model.
Device Features	
Call Waiting	Select Enable option to use Call waiting.
Broadcasting	Select Enable option to use Broadcasting

End of Digit	Chose one option to End of Digit. Provide drop-down options: none, *, #.
Login	
Administrator Login	Enter the Administrator Login account/password
Supervisor Login	Enter the Supervisor Login Login account/password
User Login	Enter the User Login account/password
Codec Priority	
Codec Priority 1	Chose one option to Codec Priority.
Codec Priority 2	Chose one option to Codec Priority.
Codec Priority 3	Chose one option to Codec Priority.
Codec Priority 4	Chose one option to Codec Priority.
Codec Priority 5	Chose one option to Codec Priority.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Click Firmware Updte **Edit**. The screen appears as shown.

Figure Configuration: Firmware Upload-xxxx

The following table describes the table in this screen

Table Configuration: Firmware Upload-xxxx

Label	Description
Firmware Upload	
File	Type in the location of the file you want to upload in this field or click on the Browse.. button to find it.
Browse	Click **Browse**... to find the file you want to upload.
Apply	Click on the Apply button to begin the upload process. This process may take up to two minute.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

To change your Auto Config-Office, click **Configuration**, and then click the **Auto Config-Office** table. The screen appears as shown.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Auto Config-Office

The following table describes the table in this screen

Table   Configuration: Auto Config-Office

Label	Description
Basic Setting	
Office	Enter the Office name.
Phone Book Refresh (min)	Enter the Phone Book Refresh time (min).
Device Config Refresh (min)	Enter the Device Config Refresh time (min).
Firmware Update Time	Enter the Firmware Update time. Default 99:99 for update ASAP.
Time Setting	
Remote NTP Server	Enter the Remote NTP Server IP Address.
NTP Refresh Interval (sec)	Enter the NTP Refresh Interval time (sec).
Time Zone	Select the current country's time zone.
Daylight Saving	Select the Enable option. SIPPBX 6200A is in a site whose country use Summer Time.
Basic Setting	
Dial Rule [Edit]	Click on the Edit button to Dialing Rule-xxx screen.
Phone Book [Edit]	Click on the Edit button to Phone Book-xxx screen.

Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Click Dial Rule **Edit**. The screen appears as shown.

Figure Configuration: Dialing Rule-xxxx

The following table describes the table in this screen

Table　Configuration: Dialing Rule- xxxx

Label	Description
Dialed Prefix	Enter the Dialed Prefix.
Max Digits	Enter the Max Digits.
Apply	Click on the Apply button to save your customized settings.
Cancel	Click on the Cancel button to begin configuration this screen afresh.
Index	Select this Index no.
Dialed Prefix	This field shows the Dialed Prefix information
Max Digits	This field shows the Max Digits information
Delete	Click on the Delete button to delete this record data.

Click Phone Book **Edit**. The screen appears as shown.

Figure Configuration: Phone Book- xxxx

The following table describes the table in this screen

Table　Configuration: Phone Book- xxxx

Label	Description

Name	Enter the Name.
Tel No	Enter the Tel No.
Apply	Click on the Apply button to save your customized settings.
Cancel	Click on the Cancel button to begin configuration this screen afresh.
Index	Select this Record no.
Name	This field shows the Name information
Tel No	This field shows the Tel No information
Delete	Click on the Delete button to delete this record data.

3.1.6 Trunk

User has to set Trunk account for Trunk (FXO device, e.g. WG3804A) to register to SIPPBX 6200A or set some necessary configuration for SIP trunk (For more application, please go to…….).

To change your Trunk, click **Configuration**, and then click the **Trunk** table. The screen appears as shown.

Figure Configuration: Trunk

The following table describes the table in this screen

Table   Configuration: Trunk

Label	Description
Select	Select this check box, then modify or delete it.
Trunk Number	This field shows the Trunk Number information
Comment	This field shows the Comment information
NAT Traversal	This field shows the NAT Traversal information
Maximum Channels	This field shows the Maximum Channels information
Add	Click on the Add button, then display Trunk Setting screen.
Modify	A Trunk Number can be modify by click on the checkbox next to the Trunk Number and click on the Modify button.
Delete	A Trunk Number can be deleted by click on the checkbox next to the A Trunk Number and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.

Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Example 1: Set Trunk for FXO gateway

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Trunk Setting

The following table describes the table in this screen

Table   Configuration: Trunk Setting

Label	Description
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 SIPPBX 6200A.
Host	Setting the Host to Dynamic will require the trunk to register the SIPPBX 6200A so that the SIPPBX 6200A know 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] appeared 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 Trunk. 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: CH3.1.8 Outgoing Routing.
Keypad	User can select Keypad type to be RFC2833, In-band, or SIP-Info and Auto. You can choose Auto to auto select the Keypad type. Choose RFC2833, Inband or SIP-Info here will force the Extension use RFC2833, Inband or SIP-Info only and the setting should be also match the Keypad setting of Trunk device.
NAT Traversal	If the Trunk device is behind a device performing NAT, such as firewall or router, and need to register to SIPPBX 6200A on public network, then user has to enable this function. Enable NAT Traversal to force SIPPBX 6200A to ignore the contact information for the Trunk and use the address from which the packets are being received.
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 SIPPBX 6200A 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 SIPPBX 6200A'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 SIPPBX 6200A to the Destination (Trunk) to avoid such security issue. ● If you set a Welltech 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.
Caller ID Display Name	When inbound call is coming from Trunk, such as 888. The caller ID Name will be the "PSTN number" or "888". Specify this will use the current setting instead.
Comment	You can input a 20 byte note for each Trunk here.
CLIR Support	CLIR means "Caller Line Identification Restriction". It is a proper noun. It is a feature to hide the caller's number. For example, ext 101 call to ext 102. But 101 won't like to show the caller ID to 102. So 101 can activate this feature to hide the caller ID. When 102 got a call from 101, the LCD of 102 should display "Anonymous".
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Example 2: Set Trunk ID for SIP Trunk

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Trunk Setting

Trunk Setting

Trunk Number:	0702069798
Password:	••••••••••
Host:	Pre-define
Address:	192.168.32.253
Dial Plan:	from-pstn
Keypad:	Auto
NAT Traversal:	Disable
Port:	5060
External Server Address:	192.168.32.253
Maximum Channels:	
Outbound Caller ID:	0702069798
Caller ID Display Name:	
Comment:	Welltech Ext #798
CLIR Support:	⊙ Disable ○ Enable

Apply   Cancel

3.1.7 SIP Trunk Reg.

SIP Trunk is for SIPPBX 6200A to register to other systems only, such as ITSP or another SIPPBX 6200A.

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.

To change your SIP Trunk Setting, click **Configuration**, and then click the **SIP Trunk Setting** table. The screen appears as shown.

Figure Configuration: SIP Trunk Setting

The following table describes the table in this screen

Table   Configuration: SIP Trunk Setting

Label	Description
Select	Select this check box, then modify or delete it.
Line Number	This field shows the Line Number information
Account	This field shows the Account information
IP Address/DNS	This field shows the IP Address/DNS information
Port	This field shows the Port information
Status	This field shows the Status information
Add	Click on the Add button, then display SIP Trunk Setting screen.
Modify	A Line Number can be modify by click on the checkbox next to the Line Number and click on the Modify button.
Delete	A Line Number can be deleted by click on the checkbox next to the A Line Number and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page

	number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Example 1: Disable SIP Trunk

In **SIP Trunk Setting** table. Click **Add**. The screen appears as shown.

Add New to configure SIPPBX 6200A register to ITSP. ITSP will provide related account information for SIPPBX 6200A to register. Please input the data here.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: SIP Trunk Setting

The following table describes the table in this screen

Table   Configuration: SIP Trunk Setting

Label	Description
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.
SIP Domain	You can change the SIP domain here if necessary. Some SIP platform will confirm SIP domain which locate in the From header. Modify this field will let SIPPBX 6200A register to SIP Trunk successfully.
Realm	Realm of ITSP or another SIPPBX 6200A. Note: ● When a call was sent form SIPPBX 6200A to a remote SIP-Trunk, the SIP-Trunk may attempt to authenticate the

	"call". So SIPPBX 6200A should reply the correct Account ID and Password. How does the SIPPBX 6200A know which ID and Password it should send? When the call is going to SIP-Trunk via SIPPBX 6200A, the SIP-Trunk may reply a 407 code, which will contain a parameter called "Realm" for authentication, SIPPBX 6200A 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 SIPPBX 6200A, you may hope those SIPPBX 6200A could call to each other. You should set the Extension to let those SIPPBX 6200A can register to each other, and you should also confirm the [Realm] in the page of Configuration→ IP PBX.
Status	Once SIP Trunk is configured and enabled, here will show the registration status.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Example 2: Enable SIP Trunk

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: SIP Trunk Setting

3.1.8 Outgoing Routing

Outgoing Call Rule means from subscriber (Extension or Trunk registered on SIPPBX 6200A) to call out.

To change your Outgoing Call Routing, click **Configuration**, and then click the **Outgoing Routing** table. The screen appears as shown.

Figure Configuration: Outgoing Call Routing

The following table describes the table in this screen

Table   Configuration: Outgoing Call Routing

Label	Description
Select	Select this check box, then modify or delete it.
Prefix	This field shows the Prefix information
Digits Length	This field shows the Digits Length information
Destination	This field shows the Destination information
Add	Click Add to Add Outgoing Routing Setting Screen.
Modify	Click Modify to Modify Outgoing Routing Setting screen.
Delete	Click Delete to select list.
Add	Click on the Add button, then display Outgoing Routing screen.
Modify	A prefix can be modify by click on the checkbox next to the prefix and click on the Modify button.
Delete	A prefix can be deleted by click on the checkbox next to the A prefix and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page

	data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.
Total Count	This field shows Total Count information.

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

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Outgoing Call Routing

The following table describes the table in this screen

Table   Configuration: Outgoing Call Routing

Label	Description
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.
Template Copy	It will copy settings for below Basic Setting and Routed

	Features based on the other Outgoing Routing record.
Basic Setting	
Primary/ Secondary/ Third/ Fourth/ Fifth/ Sixth/ Seventh/ Eighth	User can set 8 priorities for each routing rule, if SIPPBX 6200A fails to route to primary destination three times, it will try to route to secondary or third destination and so on.
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 SIPPBX 6200A, SIPPBX 6200A 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, SIPPBX 6200A 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 SIPPBX 6200A 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 SIPPBX 6200A will drop 03 then send 123 as outgoing number. Note: If you set both of the Drop and Add, SIPPBX 6200A will Drop first then Add.
Route Password	Set password here so the SIPPBX 6200A will request password before sending the call to Trunk.
Routed Features	
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 SIPPBX 6200A 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.
Fixed Outgoing Call Rule	SIPPBX 6200A will confirm the Fixed Trunk ID if you enable Fixed Outgoing Call Rule. That means the SIPPBX 6200A will route the call to a fixed Trunk if you enable this feature. For example, you set Fixed Trunk ID for extension 101 to 888. And

	you enable Fixed Outgoing call Rule for a certain route record, such as 8 with length 10. When 101 call a number 8xxxxxxxxx, the SIPPBX 6200A will route this call to 888 no matter you set what's the Primary Destination for such route record.
	Note: For more information about Fixed Trunk ID, please refer to the user manual: CH3.1.4 Extension.
Route Timeout	Enable this field will make this route record initial Timeout function.
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.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

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

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Outgoing Call Routing

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, SIPPBX 6200A will send 0912345678 to Trunk ID 888.

3.1.9 Incoming Call Rule

Incoming Call Rule means call from other non-subscriber device to SIPPBX 6200A.

For example, you set the SIPPBX 6200A to register an ISTP as a SIP Trunk, so your SIPPBX 6200A could be as an "Extension" of ITSP. The other subscriber of ITSP could call to SIPPBX 6200A by the registered line number, when the SIPPBX 6200A got an incoming call, which is not its own subscriber, what will the SIPPBX 6200A do? The SIPPBX 6200A will perform the following example based on the "Incoming Call Rule".

The following example means the SIPPBX 6200A got a called number 070070, which was sending from a non-subscriber of SIPPBX 6200A, SIPPBX 6200A 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.

To change your Incoming Routing, click **Configuration**, and then click the **Incoming Routing** table. The screen appears as shown.

Figure Configuration: Incoming Routing

The following table describes the table in this screen

Table Configuration: Incoming Routing

Label	Description
Select	Select this check box, then modify or delete it.
Prefix	This field shows the Prefix information.
Digits Length	This field shows the Digits Length information.
Add	This field shows the Add information.
Drop	This field shows the Drop information.
Add	Click on the Add button, then display Incoming Routing Setting screen.
Modify	A prefix can be modify by click on the checkbox next to the prefix and click on the Modify button.
Delete	A prefix can be deleted by click on the checkbox next to the A prefix and click on the Delete button.
Total Count	This field shows Total Count information.

Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Incoming Routing Setting

Incoming Routing Setting

Prefix: 070070
Digits Length: 6
Add: **999
Drop: 6

Apply   Cancel

The following table describes the table in this screen

Table   Configuration: Incoming Routing Setting

Label	Description
Prefix	Enter the prefix number for routing rule.
Digits Length	Enter the Digits Length for dialed number, if user doesn't want to limit the length, please set this parameter as 0. The maximum length is 20. ● 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	Enter the Add assigned number. For example, you set **999 here and you do not set Drop. If the called number is 070070101, the SIPPBX 6200A will add **999 then send **999070070101 as the called number.

Drop	Enter the drop specified length of number. For example, you set 6 here and you do not set Add. If the called number is 070070101, the SIPPBX 6200A will drop 070070 then send 101 as called number. If you set both of the Drop and Add, SIPPBX 6200A will Drop first then Add. For example, the SIPPBX 6200A got a called number 070070, which was sending from a non-subscriber of SIPPBX 6200A, SIPPBX 6200A 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.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.10 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.

To change your Dial Group, click **Configuration**, and then click the **Dial Group** table. The screen appears as shown.

Figure Configuration: Dial Group

The following table describes the table in this screen

Table Configuration: Dial Group

Label	Description
Select	Select this check box, then modify or delete it.
Group Number	This field shows the Group Number.
Telephone Number	This field shows the Telephone Number.
Ring Type	This field shows the Ring Type.
Add	Click on the Add button, then display Dial Group Setting screen.
Modify	A Group Number can be modify by click on the checkbox next to the Group Number and click on the Modify button.
Delete	A Group Number can be deleted by click on the checkbox next to the A Group Number and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.

Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Dial Group Setting

The following table describes the table in this screen

Table Configuration: Dial Group Setting

Label	Description
Group Number	Enter the Group Number. The Group number should NOT be the same with Extension, Trunk Number, and Speed Dial Number. All of the Numbers should be unique for SIPPBX 6200A system.
Telephone Number	Enter the Telephone Number to join this Dial Group. You can input multiple members by a "comma", such as [101,102,103]
Ring Type	You can choose the Ring type for Dial Group; Ring All, Sequential Ring and Random Ring.
Dial Group End To	You can decide the final destination if no one of group answered. You can select IVR, EXT Voice Mail or a Number (extension or PSTN number…, etc).
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.11 Speed Dial

Speed Dial Table is used to set the Speed Dial function; you can just input a Speed Dial Number and set the destination number to Telephone Number field. Subscriber can just dial to the Speed Dial number and SIPPBX 6200A will switch the call to Telephone Number then call out.

To change your Speed Dial Table, click **Configuration**, and then click the **Speed Dial** table. The screen appears as shown.

Figure Configuration: Speed Dial Table

Speed Dial Table

Select	Speed Dial Number ▾	Telephone Number
☐	*1111	101
☐	*400	0282265699
☐	1	1
☐	3	3
☐	4	4
☐	4	4
☐	5	5

Add Modify Delete

Total Count: 7 Number Search: ____ Total Page: 1 Page: 1

The following table describes the table in this screen

Table   Configuration: Speed Dial Table

Label	Description
Select	Select this check box, then modify or delete it.
Speed Dial Number	This field shows the Speed Dial Number.
Telephone Number	This field shows the Telephone Number.
Add	Click on the Add button, then display Speed Dial screen.
Modify	A Speed Dial Number can be modify by click on the checkbox next to the Speed Dial Number and click on the Modify button.
Delete	A Speed Dial Number can be deleted by click on the checkbox next to the A Speed Dial Number and click on the Delete button.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will

	auto display the Next or Previous Information.

Example:

User set SpeedDial number as *400 and Telephone Number as 0282265699. When caller called *400, SIPPBX 6200A will call to 0282265699 as destination number.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Speed Dial

The following table describes the table in this screen

Table   Configuration: Speed Dial

Label	Description
Speed Dial Number	Enter the Speed Dial Number. The SpeedDial Number should not be the same with Extension, Trunk Number, and Group Number. All of the Numbers should be unique for SIPPBX 6200A system.
Telephone Number	Enter the Telephone Number.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.12 Broadcast

To change your Broadcast, click **Configuration**, and then click the **Broadcast** table. The screen appears as shown.

Figure Configuration: Broadcast

	Broadcast	
Select	**Broadcast Number ▾**	**Telephone Number**
☐	*11	101,102,103,104,105,106,107,108
☐	*12	101,102,103,104,105,106,107,108
☐	*13	101,102,103,104,105,106,107,108
☐	*14	101,102,103,104,105,106,107,108
☐	*15	101,102,103,104,105,106,107,108
☐	*17	101,102,103,104,105,106,107,108, ...

Add Modify Delete

The following table describes the table in this screen

Table   Configuration: Broadcast

Label	Description
Select	Select this check box, then modify or delete it.
Broadcast Number	This field shows the Broadcast Number.
Telephone Number	This field shows the Telephone Number.
Add	Click on the Add button, then display Broadcast Setting screen.
Modify	A Broadcst Number can be modify by click on the checkbox next to the Broadcst Number and click on the Modify button.
Delete	A Broadcst Number can be deleted by click on the checkbox next to the A Broadcst Number and click on the Delete button.

Example:

User set Broadcast number as *11. When caller called *11, SIPPBX 6200A will call to [101, 102, 103, 104, 105, 106, 107, 108] number.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Broadcast Setting

Broadcast Setting	
Broadcast Number:	*11
Telephone Number:	101,102,103,104,105,106,107,108 (1-16 members)

Apply Cancel

The following table describes the table in this screen

Table   Configuration: Broadcast Setting

Label	Description
Broadcast Number	Enter the Broadcast Number.
Telephone Number	Enter the Telephone Number. Members
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.1.13 Meetme Conf.

To change your Meetme Conference, click **Configuration**, and then click the **Meetme Conference** table. The screen appears as shown.

Figure Configuration: Meetme Conference

The following table describes the table in this screen

Table   Configuration: Meetme Conference

Label	Description
Select	Select this check box, then modify or delete it.
Room Number	This field shows the Room Number.
Current Members	This field shows the Current Members.
Add	Click on the Add button, then display Meetme Conference Setting screen.
Modify	A Room Number can be modify by click on the checkbox next to the Room Number and click on the Modify button.
Delete	A Room Number can be deleted by click on the checkbox next to the A Room Number and click on the Delete button.

Example:

User set Room number as *21. When caller called *21, will Login to *21 Room.

Click **Add/Modify**. The screen appears as shown.

Figure Configuration: Meetme Conference Setting
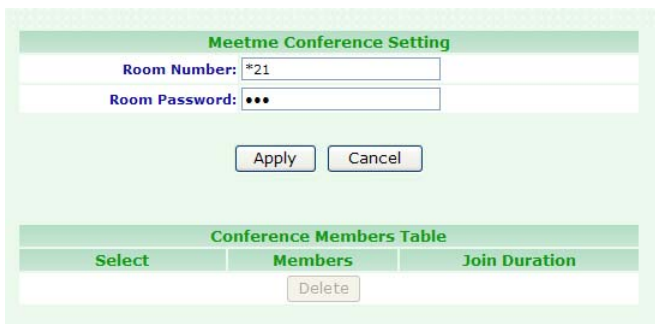
The following table describes the table in this screen

Table   Configuration: Meetme Conference Setting

Label	Description
Meetme Conference Setting	
Room Number	Enter the login Room Number.
Room Number	Enter the login Room Password.
Apply	Click on the Apply button to save your customized settings.
Cancel	Click on the Cancel button to begin configuration this screen afresh.
Conference Members Table	
Select	Select this check box, then delete it.
Member	This field shows the Member phone number.
Join Duration	This field shows the Join Duration.
Delete	A Member can expelled from the Conference room by click on the checkbox next to the A Member and click on the Delete button.

3.1.14 T.38 FAX

To change your T.38 FAX, click **Configuration**, and then click the **T.38 FAX** table. The screen appears as shown.

Figure Configuration: T.38 FAX

The following table describes the table in this screen

Table   Configuration: T.38 FAX

Label	Description
Mode	Select Enable option if your will use T.38 FAX function. The default is Enable.
T.38 Start Port	Enter the Start Port Number. The default is 4000.
T.38 End Port	Enter the End Port Number. The default is 49999.
T.38 Redundancy	Enter the Redundancy count. The default is 2.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.2 Information

The Information screens can help you view Subscriber Info., Call Monitor, CDR, and System Information.

3.2.1 Subscriber Info

You can check Phone Number, IP Address, for Extension and Trunk here. If subscriber registered on SIPPBX 6200A, the IP Address will show up, on the other hand, if the subscriber doesn't register successfully on SIPPBX 6200A, the IP Address will not be displayed.

Click **Information**, and then click the **Subscriber Info** table. The screen appears as shown.

Figure Information: Subscriber Info

The following table describes the table in this screen

Table   Information: Subscriber Info

Label	Description
Phone number	This field shows Phone Number and status.
	🔵: Idle,
	🔴: Answered,
	🔵🔴 (blinking): Ringing,

	◔: Un-register.
Type	This field shows Type information.
IP Address	This field shows IP Address information.
DND	This field shows Do Not Disturb information.
CLR	This field shows Caller Line Identification Restriction Support information.
Ext Pwd	This field shows Extension Password information.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

3.2.2 Call Monitor

Click **Information**, and then click the **Call Monitor** table. The screen appears as shown.

Figure Information: Call Monitor

The following table describes the table in this screen

Table Information: Call Monitor

Label	Description
State	This field shows State information and status. ◔: Idle, ●: Answered ◔● (blinking): Ringing.
Caller Number	This field shows Caller Number information.
Called Number	This field shows Called Number information.
Start Time	This field shows Call Date information.
Elapsed Time	This field shows Elapsed Time information.
Total Count	This field shows Total Count information.
Number Search	Enter the search number, then click enter key. The screen will display match search data.

Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will auto display the Next or Previous Information.

3.2.3 CDR

Click **Information**, and then click the **CDR** table. The screen appears as shown.

Figure Information: CDR

The following table describes the table in this screen

Table   Information: CDR

Label	Description
Start Date (YYYY-MM-DD)	Enter the search start date.
End Date (YYYY-MM-DD)	Enter the search end date.
Caller Number	Enter the search caller number.
Called Number	Enter the search called number.
CDR Display	Click CDR to search your customized date, then display in below.
Export CDR	Click Export CDR to save CDR report to CDR_xxxx.csv file.
CDR Display	
Index	This field shows Index information.
Call Date	This field shows Call Date information.
Caller Number	This field shows Caller Number information.
Called Number	This field shows Called Number information.
Elapsed Time	This field shows Elapsed Time information.
Bill Time	This field shows Bill Time information.
State	This field shows State information.
Total Count	This field shows Total Count information.
Total Page	This field shows Total Page information.
Page	This field shows Page Number information. You can Enter page number, then click enter key. The screen will display this page data.
Next/Prev	Click on the Next/Prev to Next/Previous Page. The system will

	auto display the Next or Previous Information.

Figure Information: CDR_2010XXXX_2010XXXX.csv

The following table describes the table in this screen

Table Information: CDR_2010xxxx_2010xxxx.csv

Label	Description
Call Date	This field shows Call Date information.
Caller Number	This field shows Caller Number information.
Called Number	This field shows Called Number information.
Elapsed Time	This field shows Elapsed Time information.
Bill Time	This field shows Bill Time information.
State	This field shows State information.

3.2.4 System Info

Click **Information**, and then click the **System Info** table. The screen appears as shown.

Figure Information: System Info

System Information	
Application	App62xx_100.app
Kernel	Kernel62xx_100.img
File System	ramdisk62xx_100.ram
Library	Lib62xx_100.lib
Hold Tone	music_100.wav
Ring Back Tone	music_100.wav
Configuration	configuration.cfg
IVR	Ivr62xx_Eng_Male_100.ivr

The following table describes the table in this screen

Table   Information: System Info

Label	Description
Application	This field shows Application information.
Kernel	This field shows Kernel information.
File System	This field shows File System information.
Library	This field shows Library information.
Hold Tone	This field shows Hold Tone information.
Ring Back Tone	This field shows Ring Back Tone information.
Configuration	This field shows Configuration information.
IVR	This field shows IVR information.

3.3 Network

The Network screens can help you configuration Network, DHCP Srv.(IPV4), DDNS Srv. (IPV4).

3.3.1 Network

To change your Network Setting, click **Network**, and then click the **Network Setting** table. The screen appears as shown.

Figure Network: Network Setting

The following table describes the table in this screen

Table Network: Network Setting

Label	Description
WAN	
IP Support	Select use IP mode, provide ipv4, ipv6, ipv4/ipv6.
Mode	Select SIPPBX 6200A WAN port network mode to be Fixed IP, DHCP or PPPoE.
IP Address	Enter the IP Address. If user has set SIPPBX 6200A to be fixed IP mode.
Subnet Mask	Enter the Subnet Mask Address. If user has set SIPPBX 6200A to be fixed IP mode.
Default Gateway	Enter the Default Gateway Address. If user has set SIPPBX 6200A to be fixed IP mode.
Primary DNS	Enter the IP address for Primary DNS. The default is 168.95.1.1.
Secondary DNS	Enter the IP address for Secondary DNS. The default is null.
Default Gateway Backup	Select Enable option, if any connection problem occurred on primary default gateway connections, all the traffic will be guided and switched to the secondary default gateway for proper operation. The default is Disable.
Secondary Default Gateway	Enter the Secondary Default Gateway. If you choose the Default Gateway Backup to Enable.
Check Point	Enter the Check Point IP address. SIPPBX 6200A use ping this ip address, to check have any connection problem occurred on primary default gateway connections. If you choose the Default Gateway Backup to Enable.
Current use Default Gateway	This field will display now use the Defaut Gateway IP address.
PPPoE ID	Enter the PPPoE ID. If you choose the Mode to PPPoE.
PPPoE PWD	Enter the PPPoe Password If you choose the Mode to PPPoE.
IPV6 Mode	Select Manual option, you can enter IP Address/ Prefix Length/ Gateway.
IPV6 IP Address	Enter the IPV6 IP Address.
IPV6 Prefix Length	Enter the IPV6 Prefix Lentgh. The default is 64.
IPV6 IP Gateway	Enter the D IPV6 efault Gateway Address.
MAC	This field shows the MAC address. The Mac address cannot be modified.
LAN (IPV4)	
IP Address	Enter the IP Address. The default is 192.168.123.123.
Subnet Mask	Enter the Subnet Mask Address. The default is 255.255.255.0.
MAC	This field shows the MAC address. The Mac address cannot be

	modified.
Network Routing Table (IPV4)	
Select	Select this check box, then modify or delete it.
Destination	This field shows the IP address.
Network	This field shows the Subnet Mask address.
Gateway	This field shows the Default Gateway address.
Add	Click on the Add button, then display Network Router screen.
Modify	IP address can be modify by click on the checkbox next to the IP address and click on the Modify button.
Delete	IP address can be deleted by click on the checkbox next to the IP address and click on the Delete button.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Click **Add/Modify**. The screen appears as shown.

You can add or modify a network routing record. Input subnet as Destination, subnet mask as Net mask, and gateway as Gateway.

Figure Management: Network Routing

The following table describes the table in this screen

Table   Management: Network Routing

Label	Description
IP Address	Enter the IP Address.
Subnet Mask	Enter the Subnet Mask Address.
Default Gateway	Enter the Default Gateway Address.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.3.2 DHCP Srv. (IPv4)

To change your DHCP Srv (IPV4) Settings, click **Network**, and then click the **DHCP Srv. (IPV4)** table. The screen appears as shown.

Figure Network: DHCP Srv. (IPV4)

The following table describes the table in this screen

Table   Network: DHCP Srv. (IPV4)
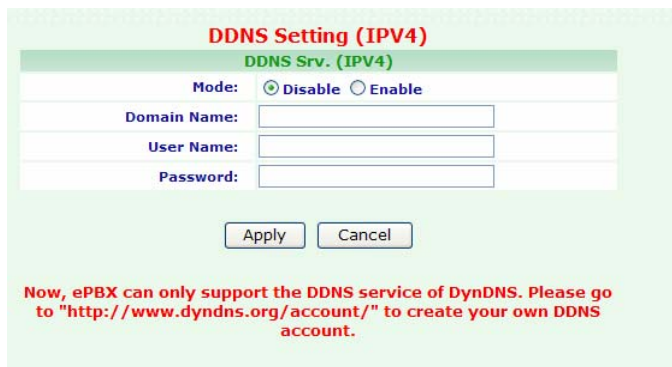
Label	Description
DHCP Server	When select Enable the SIPPBX 6200A provides TCP/IP configuration for the clients, If disable, DHCP Service id disabled and your must have another DHCP Server on your LAN, or else the computers must be manually configured. The default is Enable.
DHCP Interface	6200A has one WAN and one LAN. User can chose the DHCP interface (WAN or LAN) to assign IP settings for DHCP client.
Client Range Start IP	This field specifies the first of the contiguous addresses in the IP address Pool.
Client Range End IP	This field specifies the End of the contiguous addresses in the IP address Pool.
Subnet Mask	Enter the IP address for Subnet Mask. The default is 255.255.255.0.
Primary DNS	Enter the IP address for Primary DNS. The default is 168.95.1.1.
Secondary DNS	Enter the IP address for Secondary DNS. The default is null.
SIP Provision	Enter the IP address for SIP Provision. The default is 192.168.123.123
Apply	Click on the Apply button to save your customized settings and exit this screen.

Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.3.3 DDNS Srv. (IPv4)

To change your DDNS Setting (IPV4), click **Network**, and then click the **DDNS Srv. (IPV4)** table. The screen appears as shown.

Figure Network: DDNS Setting (IPV4)

The following table describes the table in this screen

Table   Network: DDNS Setting (IPV4)

Label	Description
Mode	Select the Enable option if your will use DDNS function. The default is Disable.
Domain Name	Enter the IP address or URL for your DDNS server.
User Name	Enter your account.
Password	Enter the password assigned to you.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.4 Management

The management screens can help you configuration Time, SMTP, VM, Security, Firmware Upload, Music Upload, Import, Export, Rest to Default, Reboot system.

3.4.1 Time Setting

To select correct Time Setting for SIPPBX 6200A, this time will affect CDR and voice mail time display. And you can also check the system current time here.

To change your Time Setting, click **Management**, and then click the **Time Setting** table. The screen appears as shown.

Figure Management: Time Setting

The following table describes the table in this screen

Table   Management: Time Setting

Label	Description
Time Setting	
NTP Client	Select Enable option if your will use NTP client. The default is Enable.
Remote NTP Server	Enter the IP address or URL for your time server. Check with your SIP/Network administrator if you are unsure of this information. The default is 168.95.195.12.

Time Zone	Choose the Time Zone of your location. This will set the time different between your time zone and Greenwich Mean Tie (GMT). The default is (GMT+08:00) Beijing, Hong Kong, Perth, Singapore, Taipei.
Current Time	This field display the time of your SIPPBX 6200A.
NTP Server	
Local NTP Server	Select the Enable or Disable option if you will use Local NTP Server. Enable this option will make SIPPBX 6200A be a NTP server and send the SIPPBX 6200A's local time to the NTP client. The default is Enable.
Apply	Click on the Apply button to save your customized settings and exit this screen.

3.4.2 SMTP Setting

SIPPBX 6200A can support Voice Mail to e-mail. Before activate this feature, you should give the SIPPBX 6200A an e-mail account and set the SMTP for SIPPBX 6200A, so that the SIPPBX 6200A has the ability to send the leaved message to the subscriber's mail box. To change your SMTP Setting, click **Management**, and then click the **SMTP Setting** table. The screen appears as shown.

Figure Management: SMTP Setting

The following table describes the table in this screen

Table   Management: SMTP Setting

Label	Description
Mail Address	Enter the IP address or URL for SIPPBX 6200A.
SMTP server	Enter the IP address or URL for your SMTP server. Check with your Email administrator if you are unsure of this information.
Account	Enter your account.
Password	Enter the password assigned to you.
SMTP Server Auth.	Select the Enable option if your will use SMTP Server Auth.. The default is Disable
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.4.3 VM Setting

User can set the configurations related with Voice Mail.

To change your VM Setting, click **Management**, and then click the **VM Setting** table. The screen appears as shown.

Figure Management: VM Setting

The following table describes the table in this screen

Table   Management: VM Setting

Label	Description
General Setting	
Maximum Length	Enter the maximum length to Voice Mail recording message. It means the maximum length per voice mail.
Maximum Conut	Enter the maximum count to Voice Mail recording message. It means the maximum voice mails per mailbox.
Voice mail to e-Mail Setting	
e-Mail Subject	Specifies the email subject of voicemail notification email messages.
e-Mail Encode	Defines the character set for voicemail messages.
e-Mail Body	Specifies the email body of voicemail notification email messages. Below are the variables to let user input into e-Mail Subject and e-Mail Body. ● VM_NAME: The receiver's name. ● VM_DUR: The total time of message.

	● VM_MSGNUM: The id of current message. ● VM_MAILBOX: The receiver's mailbox number. ● VM_CIDNUM: The sender's number. ● VM_DATE: The date of message sent.
Message Envelope Format Setting	
Envelope Function	Select the Enable if your will use envelope playback before message playback.
Envelope Format	Choose the Envelope Format of your location. Your SIPPBX 6200A may be located in different country, or may have different message announcements for user's introductory message when they enter the voicemail system. And you may also need to modify the envelope Format. Thie default is 'vm-received' Q 'digits/at' IMp, where 'vm-received' and 'digits/at' are the sound files of SIPPBX 6200A and the others are the SIPPBX 6200A supported variable. Below lists the supported variable of SIPPBX 6200A. filename: filename of a sound file (such as 'vm-received', 'digits/at'…etc) A or a: Day of week (Saturday, Sunday, ...) B or b or h : Month name (January, February, ...) d or e: numeric day of month (first, second, ..., thirty-first) Y: Year I: Hour, 12 hour clock H: Hour, 24 hour clock (single digit hours preceded by "oh") K: Hour, 24 hour clock (single digit hours NOT preceded by "oh") M: Minute, with 00 pronounced as "o'clock" N: Minute, with 00 pronounced as "hundred" (US military time) P or p: AM or PM Q: "today", "yesterday" or AbdY q: "" (for today), "yesterday", weekday, or AbdY R: 24 hour time, including minute
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.4.4 Security

To change your Security Settings, click **Management,** and then click the **Security** table. The screen appears as shown.

Figure Management: Security

The following table describes the table in this screen

Table   Management: Security

Label	Description
HTTP	
WAN Access	Select Enable option if your will login SIPPBX 6200A Web page. The default is Enable.
Http Port	Enter the port number to You can change the Http port for SIPPBX 6200A. Default is 10087. Note: After Changing the Http port, you should reboot your SIPPBX 6200A manually to activate Http Port setting.
User ID	Enter the User ID. Default is root.
Password	Enter the Password. Default is root.
Telnet	
WAN Access	Select Enable option if your will Telnet to SIPPBX 6200A. The default is Enable.
Http Port	Enter the port number to You can change the Telnet port for SIPPBX 6200A. Default is 23.
User ID	Enter the User ID. Default is root.
Password	Enter the Password. Default is root.
Ping (IPV4)	

WAN Access	The SIPPBX 6200A will not respond to any incoming Ping request when Disable is select.
Apply	Click on the Apply button to save your customized settings and exit this screen.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

3.4.5 Firmware Upload

Click **Management**, and then click the **Firmware Upload** table. The screen appears as shown.

Figure Management: Firmware Upload

Firmware Upload
Application Kernel File System Library

Firmware Upload
File: [   ] [瀏覽...]

[Apply] [Cancel]

After pressing Apply, please wait for success
message, and DO NOT power off.
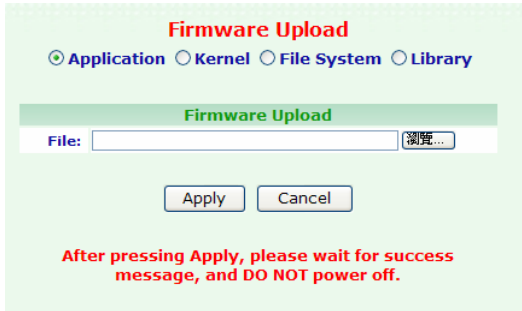
The following table describes the table in this screen

Table   Management: Firmware Upload

Label	Description
Appliation, Kernel, File System, Library	Select one option to upload SIPPBX 6200A Firmware from your computer. The default is Application.
Firmware Upload	
File Name	Choose the file name of your IVR file (wave).
File	Type in the location of the file you want to upload in this field or click on the Browse.. button to find it.
Browse	Click **Browse**... to find the file you want to upload.
Apply	Click on the Apply button to begin the upload process. This process may take up to two minute.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

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 SIPPBX 6200A.
- 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.

3.4.6 Music Upload

You can customize Ring Back Tone (Transferring Tone) by upload new wave file on SIPPBX 6200A. Please record wave file format as PCM, Channel Mode: Mono, Frequency: 8K, Bit Rate: 16 bit. And the file size must less than 10M.

Click **Management**, and then click the **Music Upload** table. The screen appears as shown.

Figure Management: Music Upload

The following table describes the table in this screen

Table   Management: Music Upload

Label	Description
Hold Tone/Ring Back Tone	Select this option to upload SIPPBX 6200A Hold Tone/Ring Back Tone file from your computer. The default is Application.
Music Upload	
File Name	Choose the file name of your IVR file (wave).
File	Type in the location of the file you want to upload in this field or click on the Browse.. button to find it.
Browse	Click **Browse**... to find the file you want to upload.
Apply	Click on the Apply button to begin the upload process. This process may take up to two minute.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

Note:

● After Upload succeed, on screen will show related information. Please reboot system to renew settings.

3.4.7 Import Setting

If there is SIPPBX 6200A setting file exported from SIPPBX 6200A, user can import this file and doesn't need to re-configure for SIPPBX 6200A.

Click **Management,** and then click the **Import Setting** table. The screen appears as shown.

Figure Management: Import Setting

The following table describes the table in this screen

Table Management: Import Setting

Label	Description
Configuration, IVR Package, IVR File (wave)	Select one option to upload SIPPBX 6200A current configuration from your computer.
Import Setting	
Group Name	Choose the Group Name of your IVR Group.
File Name	Choose the file name of your IVR file (wave).
File	Type in the location of the file you want to upload in this field or click on the Browse.. button to find it.
Browse	Click **Browse**... to find the file you want to upload.
Apply	Click on the Apply button to begin the upload process. This process may take up to two minute.
Cancel	Click on the Cancel button to begin configuration this screen afresh.

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 SIPPBX 6200A.

- If you choose the option to IVR File (WAV). Before import the IVR File to SIPPBX 6200A, you should prepare the wav file by yourself first. You should choose the Group and select file name, then you can import a specific file with wav format to SIPPBX 6200A to instead the old one.
- For more information about the detail IVR files, please refer to user manual: CH4.1.3 How to record the other system prompts

3.4.8 Export Setting

You can export configuration and voice wave files. If there is more than one SIPPBX 6200A need to be configured, user can export configuration of one SIPPBX 6200A, and then import this setting file for the other SIPPBX 6200As, so that user doesn't need to re-configure for each SIPPBX 6200A.

Click **Management**, and then the **Export Setting** table. The screen appears as shown.

Figure Management: Export Setting

The following table describes the table in this screen

Table   Management: Export Setting

Label	Description
Configuration	Select one option to save SIPPBX 6200A current configuration to your computer. The default is Configuration. Default file Name: configuration.cfg
IVR Package	Select this option to save SIPPBX 6200A IVR voice wave file to your computer. Default file Name: sounds.pak
Export	Click Export to expore save your customized settings and exit this screen.

3.4.9 Rest To Default

Click **Management**, and then the **Rest To Default** table.

Click on the Apply button to clears all user-entered configuration information and returned the SIPPBX 6200A to factory defaults.

Figure Management: Rest To Default

3.4.10 Reboot System

Click **Management**, and then the **Reboot System** table.

Click on the Apply button to have the SIPPBX 6200A reboot. This does not affect the SIPPBX 6200A configuration.

Figure Management: Reboot System

3.4.11 Power Off System

Click **Management**, and then the **Power Off System** table.

Click on the Apply button to have the SIPPBX 6200A shutdown. This does not affect the SIPPBX 6200A configuration.

Figure Management: Power Off System

CH4. Application Setting
4.1 Customize System prompt
4.1.1 Record Greeting

Use any Extension phone to enter the recording process. The feature code of System Prompt Recording is [*50]. The record procedure of greeting message will be: "Dial to [*50]→ Input password [000]→ dial to access code [**111]→ Start to record greeting-day.wav". Greeting will renew immediately after recording.

Note:

For more information about the feature code of System Prompt Recording, please refer to user manual: CH 3.1.3 Feature Code.

The **111 means the office 1 working time greeting. For the other greeting's access code, please refer to CH4.1.3 How to record the other System Prompts

4.1.2 Enable Automated Attendant

User has to Enable Trunk (e.g. 3804A) hotline function and point to destination number **999 (Number of Automated Attendant for SIPPBX 6200A). 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 Welltech CPE device and SIPPBX 6200A.

The **999 means the office 1 presentation number. For the other presentation number, please refer to CH3.1.2 Office Call Rule.

4.1.3 How to record the other System Prompts

User can record the greeting by "Dial to [*50]→ Input password [000]→ dial to access code [**111]" as default.

After the tone, start recording then press pound key. 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 system prompt? The procedure is just like recording new greeting, dialing to access code for recording. And user can also dial to [*50]→ Input password [000]→ dial to access code [***xxxx]" to listen the system prompt, too.

You can also record the sound files with WAV format by your PC or other equipment, and choose the specific file name in the page of Import Setting, import the WAV file to instead the old one.

For example, you can record the prompt for "no body pick up", by your PC as a WAV file. Name it as noanswer.wav, then you can go to "Import Setting" page, choose IVR File (WAV)

→ Choose Group Name to Auto Attendant Sounds Files → Choose File Name to noanswer.wav → Upload new file. Below is the detail for system prompts.

Group Name: Digits Sounds Files			
Record Code	**Playback Code**	**File Name**	**Default Announcement**
**0000	***0000	0.wav	zero
**0001	***0001	1.wav	one
**0002	***0002	2.wav	two
**0003	***0003	3.wav	three
**0004	***0004	4.wav	four
**0005	***0005	5.wav	five
**0006	***0006	6.wav	six
**0007	***0007	7.wav	seven
**0008	***0008	8.wav	eight
**0009	***0009	9.wav	nine
**0010	***0010	10.wav	ten
**0011	***0011	11.wav	eleven
**0012	***0012	12.wav	twelve
**0013	***0013	13.wav	thirteen
**0014	***0014	14.wav	fourteen
**0015	***0015	15.wav	fifteen
**0016	***0016	16.wav	sixteen
**0017	***0017	17.wav	seventeen
**0018	***0018	18.wav	eighteen
**0019	***0019	19.wav	nineteen
**0020	***0020	20.wav	twenty
**0030	***0030	30.wav	thirty
**0040	***0040	40.wav	forty
**0050	***0050	50.wav	fifty
**0060	***0060	60.wav	sixty
**0070	***0070	70.wav	seventy
**0080	***0080	80.wav	eighty
**0090	***0090	90.wav	ninety
Group Name: Time & Date Sounds Files			
Record Code	**Playback Code**	**File Name**	**Default Announcement**

**0101	***0101	h-1.wav	first
**0102	***0102	h-2.wav	second
**0103	***0103	h-3.wav	third
**0104	***0104	h-4.wav	fourth
**0105	***0105	h-5.wav	fifth
**0106	***0106	h-6.wav	sixth
**0107	***0107	h-7.wav	seventh
**0108	***0108	h-8.wav	eighth
**0109	***0109	h-9.wav	ninth
**0110	***0110	h-10.wav	tenth
**0111	***0111	h-11.wav	eleventh
**0112	***0112	h-12.wav	twelfth
**0113	***0113	h-13.wav	thirteenth
**0114	***0114	h-14.wav	fourteenth
**0115	***0115	h-15.wav	fifteenth
**0116	***0116	h-16.wav	sixteenth
**0117	***0117	h-17.wav	seventeenth
**0118	***0118	h-18.wav	eighteenth
**0119	***0119	h-19.wav	nineteenth
**0120	***0120	h-20.wav	twentieth
**0130	***0130	h-30.wav	thirtieth
**0131	***0131	at.wav	at
**0132	***0132	a-m.wav	AM
**0133	***0133	p-m.wav	PM
**0134	***0134	hundred.wav	hundred
**0135	***0135	thousand.wav	thousand
**0136	***0136	million.wav	million
**0137	***0137	minus.wav	minus
**0201	***0201	day-0.wav	Sunday
**0202	***0202	day-1.wav	Monday
**0203	***0203	day-2.wav	Tuesday
**0204	***0204	day-3.wav	Wednesday
**0205	***0205	day-4.wav	Thursday
**0206	***0206	day-5.wav	Friday
**0207	***0207	day-6.wav	Saturday
**0208	***0208	dollars.wav	dollars
**0209	***0209	mon-0.wav	January
**0210	***0210	mon-1.wav	February
**0211	***0211	mon-2.wav	March

**0212	***0212	mon-3.wav	April
**0213	***0213	mon-4.wav	May
**0214	***0214	mon-5.wav	June
**0215	***0215	mon-6.wav	July
**0216	***0216	mon-7.wav	August
**0217	***0217	mon-8.wav	September
**0218	***0218	mon-9.wav	October
**0219	***0219	mon-10.wav	November
**0220	***0220	mon-11.wav	December
**0221	***0221	oh.wav	O (spoken in a way meaning "zero")
**0222	***0222	oclock.wav	a clock
**0223	***0223	pound.wav	pound
**0224	***0224	star.wav	star
**0225	***0225	today.wav	today
**0226	***0226	tomorrow.wav	tomorrow
**0227	***0227	yesterday.wav	yesterday
**0228	***0228	year.wav	year
**0229	***0229	date.wav	date

Group Name: Auto Attendant Sounds Files

Record Code	Playback Code	File Name	Default Announcement
**111	***111	greeting-day.wav	Please dial the extension number, or press, 9, for the operator
**112	***112	greeting-noon.wav	Please dial the extension number. Thank you.
**113	***113	greeting-night.wav	Please dial the extension number. Thank you.
**114	***114	greeting-holiday.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**115	***115	greeting-temporary.wav	Please dial the extension number. Thank you.
**116	***116	noanswer.wav	I am sorry, the extension number you dialed, is not answering. Please dial another extension number. Thank you.
**117	***117	busy.wav	I am sorry. the extension number you dialed is busy. Please dial

			another extension number. Thank you.
**118	***118	goodbyeivr.wav	goodbye
**119	***119	unavailable.wav	I am sorry, the extension number you dialed is un available, please dial another extension number. Thank you.
**120	***120	invalid.wav	I am sorry, that's not a valid extension. Please try again.
**121	***121	dnd-act.wav	Do not disturb, activated.
**122	***122	dnd-deact.wav	Do not disturb, dee-activated.
**123	***123	unconfwd-act.wav	Unconditional forward, activated.
**124	***124	unconfwd-deact.wav	Unconditional forward, dee-activated.
**125	***125	busyfwd-act.wav	Busy forward, activated.
**126	***126	busyfwd-deact.wav	Busy forward, dee-activated.
**127	***127	op-noanswer.wav	The operator is not answering. Please call later, or dial another extension number.
**128	***128	op-busy.wav	The operator is busy. Please call later, or dial another extension number.
**129	***129	op-unavailable.wav	The operator is unavailable. Please call later, or dial another extension number.
**130	***130	noanswerfwd-act.wav	No answer forward, activated.
**131	***131	noanswerfwd-deact.wav	No answer forward, dee-activated.
**132	***132	allfwd-deact.wav	Call forward, dee-activated.
**133	***133	transferop.wav	Transfer-ring to operator. Please hold.
**134	***134	unavailablefwd-act.wav	The "Unavailable Forward" is activated.
**135	***135	unavailablefwd-deact.wav	The "Unavailable Forward" is deactivated.
**136	***136	greeting-day2.wav	Please dial the extension number, or press, 9, for the operator
**137	***137	greeting-noon2.wav	Please dial the extension number. Thank you.
**138	***138	greeting-night2.wav	Please dial the extension number.

			Thank you.
**139	***139	greeting-holiday2.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**140	***140	greeting-temporary2.wav	Please dial the extension number. Thank you.
**141	***141	greeting-day3.wav	Please dial the extension number, or press, 9, for the operator
**142	***142	greeting-noon3.wav	Please dial the extension number. Thank you.
**143	***143	greeting-night3.wav	Please dial the extension number. Thank you.
**144	***144	greeting-holiday3.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**145	***145	greeting-temporary3.wav	Please dial the extension number. Thank you.
**146	***146	greeting-day4.wav	Please dial the extension number, or press, 9, for the operator
**147	***147	greeting-noon4.wav	Please dial the extension number. Thank you.
**148	***148	greeting-night4.wav	Please dial the extension number. Thank you.
**149	***149	greeting-holiday4.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**150	***150	greeting-temporary4.wav	Please dial the extension number. Thank you.
**151	***151	greeting-day5.wav	Please dial the extension number, or press, 9, for the operator
**152	***152	greeting-noon5.wav	Please dial the extension number. Thank you.
**153	***153	greeting-night5.wav	Please dial the extension number. Thank you.
**154	***154	greeting-holiday5.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**155	***155	greeting-temporary5.wav	Please dial the extension number. Thank you.

**156	***156	greeting-day6.wav	Please dial the extension number, or press, 9, for the operator
**157	***157	greeting-noon6.wav	Please dial the extension number. Thank you.
**158	***158	greeting-night6.wav	Please dial the extension number. Thank you.
**159	***159	greeting-holiday6.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**160	***160	greeting-temporary6.wav	Please dial the extension number. Thank you.
**161	***161	greeting-day7.wav	Please dial the extension number, or press, 9, for the operator
**162	***162	greeting-noon7.wav	Please dial the extension number. Thank you.
**163	***163	greeting-night7.wav	Please dial the extension number. Thank you.
**164	***164	greeting-holiday7.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**165	***165	greeting-temporary7.wav	Please dial the extension number. Thank you.
**166	***166	greeting-day8.wav	Please dial the extension number, or press, 9, for the operator
**167	***167	greeting-noon8.wav	Please dial the extension number. Thank you.
**168	***168	greeting-night8.wav	Please dial the extension number. Thank you.
**169	***169	greeting-holiday8.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**170	***170	greeting-temporary8.wav	Please dial the extension number. Thank you.
**171	***171	greeting-day9.wav	Please dial the extension number, or press, 9, for the operator
**172	***172	greeting-noon9.wav	Please dial the extension number. Thank you.
**173	***173	greeting-night9.wav	Please dial the extension number. Thank you.

**174	***174	greeting-holiday9.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**175	***175	greeting-temporary9.wav	Please dial the extension number. Thank you.
**176	***176	greeting-day10.wav	Please dial the extension number, or press, 9, for the operator
**177	***177	greeting-noon10.wav	Please dial the extension number. Thank you.
**178	***178	greeting-night10.wav	Please dial the extension number. Thank you.
**179	***179	greeting-holiday10.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**180	***180	greeting-temporary10.wav	Please dial the extension number. Thank you.
**181	***181	greeting-day11.wav	Please dial the extension number, or press, 9, for the operator
**182	***182	greeting-noon11.wav	Please dial the extension number. Thank you.
**183	***183	greeting-night11.wav	Please dial the extension number. Thank you.
**184	***184	greeting-holiday11.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**185	***185	greeting-temporary11.wav	Please dial the extension number. Thank you.
**186	***186	greeting-day12.wav	Please dial the extension number, or press, 9, for the operator
**187	***187	greeting-noon12.wav	Please dial the extension number. Thank you.
**188	***188	greeting-night12.wav	Please dial the extension number. Thank you.
**189	***189	greeting-holiday12.wav	Please dial the extension number. Thank you. (It is reserved, not functional now)
**190	***190	greeting-temporary12.wav	Please dial the extension number. Thank you.

Group Name: Voice Mail Sounds Files			
Record Code	Playback Code	File Name	Default Announcement
**0301	***0301	vm-advopts.wav	Press, 3, for advanced options.
**0302	***0302	vm-and.wav	and
**0303	***0303	vm-calldiffnum.wav	Press, 2, to enter a different number.
**0304	***0304	vm-changeto.wav	Change to which folder?
**0305	***0305	vm-cust1.wav	Folder. Five
**0306	***0306	vm-cust2.wav	Folder. Six
**0307	***0307	vm-cust3.wav	Folder. Seven
**0308	***0308	vm-cust4.wav	Folder. Eight
**0309	***0309	vm-cust5.wav	Folder. Nine
**0310	***0310	vm-delete.wav	Press, 7, to delete this message.
**0311	***0311	vm-deleted.wav	Message deleted.
**0312	***0312	vm-dialout.wav	Please wait, while I connect your call.
**0313	***0313	vm-enter-num-to-call.wav	Please enter the number, you wish to call.
**0314	***0314	vm-extension.wav	extension
**0315	***0315	vm-Family.wav	family
**0316	***0316	vm-first.wav	first
**0317	***0317	vm-for.wav	for
**0318	***0318	vm-forward.wav	Press, 1, to enter an extension. Press, 2, to use the directory.
**0319	***0319	vm-forwardoptions.wav	Press, 1, to pre-pend the message, or, 2, to forward a message without pre-pending.
**0320	***0320	vm-Friends.wav	friend
**0321	***0321	vm-from.wav	from
**0322	***0322	vm-from-extension.wav	Message from extension,
**0323	***0323	vm-from-phonenumber.wav	Message from phone number,
**0324	***0324	vm-goodbye.wav	goodbye
**0325	***0325	vm-helpexit.wav	Press, star for help, or pound to exit.
**0326	***0326	vm-INBOX.wav	new
**0327	***0327	vm-incorrect.wav	Login incorrect.
**0328	***0328	vm-incorrect-mailbox.wav	Login incorrect. Mailbox
**0329	***0329	vm-instructions.wav	To check your messages, press, 1, now. You may quit voicemail, at any

			time by pressing the, pound key.
**0330	***0330	vm-intro.wav	Please leave your message after the tone. When done, hang up or press the pound key.
**0331	***0331	vm-isonphone.wav	is on the phone.
**0332	***0332	vm-isunavail.wav	is unavailable.
**0333	***0333	vm-last.wav	last
**0334	***0334	vm-leavemsg.wav	Press, 5, to leave a message.
**0335	***0335	vm-login.wav	Welcome to Voice Mail. Mailbox,
**0336	***0336	vm-mailboxfull.wav	Sorry, but the user's mail box, can not accept anymore messages.
**0337	***0337	vm-message.wav	message
**0338	***0338	vm-messages.wav	messages
**0339	***0339	vm-minutes.wav	minutes
**0340	***0340	vm-mismatch.wav	The passwords you entered and re-entered did not match. Please try again.
**0341	***0341	vm-msginstruct.wav	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 voice mail, press, pound.
**0342	***0342	vm-msgsaved.wav	Your message has been saved.
**0343	***0343	vm-newpassword.wav	Please enter your new password, followed by the pound key.
**0344	***0344	vm-newuser.wav	Welcome to Voice Mail! First, I will guide you through a short setup process.
**0345	***0345	vm-next.wav	Press, 6, to play the next message.
**0346	***0346	vm-no.wav	no
**0347	***0347	vm-nobodyavail.wav	nobody is available to take your call at this moment.
**0348	***0348	vm-nobox.wav	You can not reply to this message, because the sender, does not have a mailbox.
**0349	***0349	vm-nomore.wav	No more messages.
**0350	***0350	vm-nonumber.wav	I'm afraid I don't know who sent this message.

**0351	***0351	vm-num-i-have.wav	the number I have is,
**0352	***0352	vm-Old.wav	old
**0353	***0353	vm-onefor.wav	Press, 1, for
**0354	***0354	vm-options.wav	Press, 1, to record your un available message. Press, 2, to record your busy message. Press, 3, to record your name. Press, 4, to record your temporary greeting. Press, star to return to the main menu.
**0355	***0355	vm-opts.wav	Press, 2, to change folders. Press, 3, for advanced options. Press, zero, for mailbox options.
**0356	***0356	vm-passchanged.wav	Your passwords have been changed.
**0357	***0357	vm-password.wav	password
**0358	***0358	vm-press.wav	press
**0359	***0359	vm-prev.wav	Press, 4, for the previous message.
**0360	***0360	vm-reachoper.wav	Press, 9, to reach the operator.
**0361	***0361	vm-rec-busy.wav	After the tone, say your busy message, then press the pound key.
**0362	***0362	vm-received.wav	received
**0363	***0363	vm-rec-name.wav	After the tone, say your name, then press the pound key.
**0364	***0364	vm-rec-temp.wav	After the tone, say your temporary message, then press the pound key.
**0365	***0365	vm-rec-unv.wav	After the tone, say your un available message, then press the pound key.
**0366	***0366	vm-reenterpassword.wav	Please re enter your password followed by the pound key.
**0367	***0367	vm-repeat.wav	Press, 5, to repeat the current message.
**0368	***0368	vm-review.wav	Press, 1, to accept this recording. press, 2, to listen to it. press, 3, to re-record your message.
**0369	***0369	vm-saved.wav	saved
**0370	***0370	vm-savedto.wav	save to
**0371	***0371	vm-savefolder.wav	Which folder should I save the message to?
**0372	***0372	vm-savemessage.wav	or, 9, to save this message.
**0373	***0373	vm-saveoper.wav	Press, 1,to accept this recording,

			otherwise please continue to hold.
**0374	***0374	vm-sorry.wav	I am sorry. I did not understand your response.
**0375	***0375	vm-star-cancel.wav	Press, star to cancel.
**0376	***0376	vm-starmain.wav	Press, star to return to the main menu.
**0377	***0377	vm-tempgreeting2.wav	Press, 1, to record your temporary greeting, or press, 2, to erase your temporary greeting
**0378	***0378	vm-tempgreeting.wav	Press, 1, to record your temporary greeting.
**0379	***0379	vm-tempremoved.wav	Your temporary greeting has been removed.
**0380	***0380	vm-then-pound.wav	Then press, pound.
**0381	***0381	vm-theperson.wav	The person at extension,
**0382	***0382	vm-tocallback.wav	Press, 2, to call the person, who sent this message.
**0383	***0383	vm-tocallnum.wav	Press, 1, to call this number.
**0384	***0384	vm-tocancel.wav	or press, pound, to cancel.
**0385	***0385	vm-tocancelmsg.wav	Press, star to cancel this message.
**0386	***0386	vm-toenternumber.wav	Press, 1, to enter a number
**0387	***0387	vm-toforward.wav	Press, 8, to forward the message to another user.
**0388	***0388	vm-tohearenv.wav	Press, 3, to hear the message envelope.
**0389	***0389	vm-tomakecall.wav	Press, 4, to place an out-going call.
**0390	***0390	vm-tooshort.wav	Your message is too short
**0391	***0391	vm-toreply.wav	Press, 1, to send a reply.
**0392	***0392	vm-torerecord.wav	Press, 3, to record your message.
**0393	***0393	vm-undelete.wav	Press, 7, to undelete this message.
**0394	***0394	vm-undeleted.wav	Message undeleted.
**0395	***0395	vm-unknow-caller.wav	from an unknown caller
**0396	***0396	vm-whichbox.wav	To leave a message, please enter a mailbox number.
**0397	***0397	vm-Work.wav	work
**0398	***0398	vm-youhave.wav	you have,
Group Name: General Sounds Files			
Record	**Playback**	**File Name**	**Default Announcement**

Code	Code		
**0400	***0400	beep.wav	(This is a beep tone)
**0401	***0401	hours.wav	hours
**0402	***0402	minutes.wav	minutes
**0403	***0403	auth-incorrect.wav	Password incorrect. Please enter your password followed by the pound key.
**0404	***0404	auth-thankyou.wav	thank you
**0405	***0405	pbx-invalid.wav	I am sorry, that is not a valid extension. Please try again.
**0406	***0406	pbx-invalidpark.wav	I am sorry, there is no call parked on that extension. Please try again. (It is reserved, not functional now)
**0407	***0407	pax-transfer.wav	Transfer.
**0408	***0408	privacy-incorrect.wav	I'm sorry, that number is not valid.
**0409	***0409	privacy-prompt.wav	Please enter your ten-digit phone number, starting with the area code. (It is reserved, not functional now)
**0410	***0410	privacy-thankyou.wav	Thank you.
**0411	***0411	privacy-unident.wav	The party you are trying to reach does not accept unidentified calls. (It is reserved, not functional now)
**0412	***0412	ss-noservice.wav	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.wav	Please hold, while I try that extension.
**0414	***0414	ivrrecord.wav	Please enter the access code. After the tone, start recording, then press the pound key.
**0415	***0415	CB-act.wav	Call back on Busy, activated.
**0416	***0416	clir-act.wav	"Call Line Identification Restriction", activated.
**0417	***0417	clir-deact.wav	"Call Line Identification Restriction", dee-activated.
Group Name: Agent Sounds Files			
Record	**Playback**	**File Name**	**Default Announcement**

Code	Code		
**0445	***0445	agent-pass.wav	Please enter password followed by the pound key

Group Name: Meetme Conf. Files			
Record Code	**Playback Code**	**File Name**	**Default Announcement**
**0470	***0470	conf-adminmenu.wav	Please press 1 to mute or unmute yourself, 2 to lock or unlock the conference, 3 to eject the last user, 4 or 6 to decrease or increase the conference volume, 7 or 9 to decrease or increase your volume, or 8 to exit
**0471	***0471	conf- enteringno.wav	You are entering conference number
**0472	***0472	conf- errormenu.gm	Invalid Choice
**0473	***0473	conf- getchannel.gm	Please enter the channel number followed by the pound key.
**0474	***0474	conf- getconfno.wav	Please enter your conference number followed by the pound key.
**0475	***0475	conf- getpin.wav	Please enter the conference pin number.
**0476	***0476	conf- hasjoin.wav	is now in the conference.
**0477	***0477	conf- hasleft.wav	has left the conference.
**0478	***0478	conf- invalid.wav	That is not a valid conference number. Please try again.
**0479	***0479	conf- invalidpin.wav	That pin is invalid for this conference.
**0480	***0480	conf- kicked.wav	You have been kicked from this conference
**0481	***0481	conf- leaderhasleft.wav	The leader has left the conference.
**0482	***0482	conf- locked.wav	This conference is locked!
**0483	***0483	conf- lockednow.wav	The conference is now locked
**0484	***0484	conf- muted.wav	You are now muted
**0485	***0485	conf- noempty.wav	No empty conferences currently exist.
**0486	***0486	conf- onlyone.wav	There is currently one other participant in the conference.
**0487	***0487	conf- onlyperson.wav	You are currently the only person in

			this conference.
**0488	***0488	conf- otherinparty.wav	other participants in the conference
**0489	***0489	conf- placeintoconf.wav	You will now be placed into the conference.
**0490	***0490	conf- thereare.wav	There are currently
**0491	***0491	conf- unlockednow.wav	The conference is now unlocked
**0492	***0492	conf- unmuted.wav	You are now unmuted
**0493	***0493	conf- usermenu.wav	Please press 1 to mute or unmute yourself, 4 or 6 to decrease or increase the conference volume, 7 or 9 to decrease or increase your volume, or 8 to exit
**0494	***0494	conf- userswilljoin.wav	users will join the conference.
**0495	***0495	conf- userwilljoin.wav	user will join the conference.
**0496	***0496	conf- waitforleader.wav	The conference will begin when the leader arrives.

4.2 Customize Ring Back Tone (Transferring Tone)

User can customize Ring Back Tone by upload new wave file on SIPPBX 6200A. Please record wave file format as PCM, Channel Mode: Mono, Frequency: 8K, Bit Rate: 16 bit. Click **Management**, and then click the **Music Upload**. In Music Upload Table, click Press Browse…, to select wave file, then 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 SIPPBX 6200A got a Registration or Invite (incoming call) from a remote location, it will reply Authentication for security issue.

4.3.2 Automated Attendant

The SIPPBX 6200A supports Automated Attendant; you can record the default greeting and the other announcements by Extension. For more information, please refer to the user manual: CH4.1.3 How to record the other system prompts.

4.3.3 Call Transfer

The SIPPBX 6200A supports "server transfer". User 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

The SIPPBX 6200A supports "server blind transfer". User 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

SIPPBX 6200A can support "server forward". User can dial to *90 to active Call Forward on Busy and *91 to deactivate. For example, extension 101 dial to *90102, there will be an announcement to notify you the call forward is enabled, and someone call to 101 but 101 is on the phone. The call will be routed to 102.

4.3.6 Call Forward on No Answer

SIPPBX 6200A can support "server forward". User can dial to *92 to active Call Forward on No Answer and *93 to deactivate. For example, extension 101 dial to *92102, there will be an announcement to notify you the call forward is enabled, and someone call to 101 but 101 is no answer. The call will be routed to 102.

4.3.7 Call Forward Unconditional

T SIPPBX 6200A can support "server forward". User can dial to *72 to active Unconditionally Forward and *73 to deactivate. For example, extension 101 dial to *72102, there will be an announcement to notify you the call forward is enabled, and someone call to 101, the call will be routed to 102 directly.

4.3.8 Call Forward Unavailable

If subscriber is not registering, the call will be forward to another number when Unavailable Forward is enabled. To activate Unavailable Forward, press *94xxx, where the *94 is the feature code to activate Unavailable Forward and xxx is the destination number. Pressing [*95] for deactivate

For example, extension 101 dial to *94102, there will be an announcement to notify you the Unavailable Forward is enabled, and someone call to 101, but 101 is not registering, the call will be routed to 102.

4.3.9 Call Hold/Retrieval (Client based)

Normally, the call hold and call retrieval is done by Client, the SIPPBX 6200A just relay the SIP signal for such function.

4.3.10 Call Routing

Click **Configuration**, and then click the **Outgoing Routing** table, you can set the Outgoing Call Routing record for a specified Prefix.

Click **Configuration**, and then click the **Incoming Routing** table, you can set the Incoming Call Routing record for a specified Prefix.

4.3.11 Call Waiting (Client based)

The SIPPBX 6200A does not support "server Call Waiting" now. This feature should do by client side.

4.3.12 Caller ID

The SIPPBX 6200A will relay the caller ID from caller to callee.

4.3.13 CLIR (Caller Line Identification Restriction)

CLIR means "Caller Line Identification Restriction". It is a proper noun.

It is a feature to hide the caller's number. For example, ext 101 call to ext 102. But 101 won't like to show the caller ID to 102. So 101 can activate this feature to hide the caller ID. When 102 got a call from 101, the LCD of 102 should display "Anonymous".

SIPPBX 6200A could support two kinds of CLIR.

a) CLIR (per call): for example, 101 won't like to show the caller id for 102. 101 can just dial to "*67102", where the *67 is the feature for CLIR (per call). When 102 got the incoming call, the LCD of 102 should display "Anonymous". If 101 just dial to "102", then 102 should see the Caller ID as 101.

b) CLIR (database type): for example. 101 dial to "*31", SIPPBX 6200A should add the CLIR record for 101 into its database. When 101 call to 102, 103 etc. The LCD of called party should always show "Anonymous". 101 can dial to "*32" to disable this feature.

4.3.14 Do Not Disturb (Client based)

The SIPPBX 6200A can support "server DND". User can dial to *78 to active it and *79 to deactivate. For example, extension 101 dial to *78, there will be an announcement to notify you the DND is enabled, and someone call to 101 then the call will be rejected directly.

4.3.15 Flexible Extension Logic

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

4.3.16 Music On Hold

The SIPPBX 6200A will play music if the user is under Hold status.

4.3.17 Music On Transfer

The SIPPBX 6200A will play music if the user is under Transfer status.

4.3.18 Call Pickup

The SIPPBX 6200A can support Call Pickup. For example: Ext-A is ringing, Ext-B can press *8 for Global or Group pickup. You can also press **8 + ext number for specific call pick up.

4.3.19 Call Park

By default, extension 700 is used to park a call. While in a conversation, press *0 to initiate a blind transfer, and then dial 700. SIPPBX 6200A will now announce the parking extension, most probably 701 or 702. Now hang up - the caller will be left on hold at the announced extension. Walk up to a different phone, dial 701 and the conversation can be continued. If a caller has been parked for a longer time than 45 seconds, then SIPPBX 6200A will again ring the originally dialed extension.

4.3.20 Camp-On (Call Back on Busy)

For example, you dial to 101 but 101 is on the phone, then you should hear an announcement for busy. You could dial to *66 by default to trigger the SIPPBX 6200A call back to you when 101 is idle. This function will let u talk to called party immediately when called party is free.

This Function is only workable when voice mail function of called party is disabled. When this function is enabled, SIPPBX 6200A will check the status of called party every 20 seconds, at most 15 times. That means this function may be performed when called party is idled after 20 seconds at most. And 300 (20*15) seconds later, this function will not be workable.

4.3.21 Meetme Conference

The SIPPBX 6200A can support Meetme Conference. User set Room number as *21. When caller called *21, will Login to *21 Room.

4.3.22 Broadcast

The SIPPBX 6200A can support Broadcast. User set Broadcast number as *11 and Telephone Number as [101, 102, 103, 104, 105, 106, 107, 108]. When caller called *11, SIPPBX 6200A will call to [101, 102, 103, 104, 105, 106, 107, 108] number.

4.3.23 Time and Date

You can select correct Time Zone for SIPPBX 6200A; this time will affect CDR and voice mail time display.

4.3.24 Trunk (WG2680)

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.

4.3.25 VoIP Gateways (WG2680; WG2504)

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.

4.3.26 Voice Mail to e-mail

You should configure the SMTP setting to perform Voice Mail to e-mail. If the SIPPBX 6200A got a new message, it will send the message to user by email immediately.

4.3.27 Access Voice Mail by phone set

The SIPPBX 6200A has a built-in voice mail system. That means SIPPBX 6200A can store voice mail within it. User can just dial to *98 then input mailbox number and password to access voice mail.

4.3.28 Call Monitor

Click **Information,** and then click **Call Monitor,** you can monitor the call status if the call were routed by SIPPBX 6200A.

CH5. Appendix

5.1 Voice Mail System Concept

SIPPBX 6200A has a CF card to store voice mail within itself. Below is an example when user login Voice Mail System.

Press *98 (to enter voice mail system):
1. Input Mailbox number (vm-login.wav)
2. Input password (vm-password.wav)
3. It will announce incorrect message if login incorrect (vm-incorrect-mailbox.wav)

You have (vm-youhave.wav) 1 (digits/1.wav) new (vm-INBOX.wav) and 2 (digits/2.wav) old (vm-Old.wav) messages (vm-messages.wav),
press 1 for (vm-onefor.wav) new (vm-INBOX.wav) messages (vm-messages.wav),
press 2 to change folder, press 3 for Advance options,
press 0 for mailbox option (vm-opts.wav),
press * for help or # for exist (vm-helpexist.wav).

Press 1 for new message: first message received at …….
   3: advance option (vm-advopts.wav)
      1: send reply (vm-toreply.wav) --- When you hear a new message, you can reply a message to the sender's voice mail if the user also enabled the voicemail box function.
      3: hear the message envelope (vm-tohearenv.wav) --- To hear the received time the sender's number.
      5: leave message (vm-leavemeg.wav) --- When you hear an old message, you can leave a message to the sender or another user's voice mail if the user also enabled the voicemail box function.
      * return the main menu (vm-starmain.wav)
   4: previous message (vm-prev. wav)
   5: repeat current message (vm-repeat.wav)
   6: next message (vm-next.wav)
   7: delete or undelete current message (vm-delete.wav) (vm-undelete.wav)
   8: forward this message to another user (vm-toforward.wav)
      1: prepend the message (vm-forwardoptions.wav) --- When you listen a message, you can forward such message to another user, and you can leave your own message before the forwarding message
      2: forward a message without pretending (vm-forwardoptions.wav) --- forward the message to another user directly, without pretending.

9: to save this message (vm-savemessage.wav)

Save to which folder? (vm-savefolder.wav)

    0: for new messages.

    1: for old messages

    2: for Work folder (vm-work.wav)

    3: for Family folder (vm-Family.wav)

    4: for Friends folder (vm-Friends.wav)

    #: to cancel (vm-tocancel.wav)

*: help (vm-helpexist.wav)

#: exist (vm-helpexist.wav)

Press 2 to change folder:

Change to which folder? (vm-changeto.wav)

    0: for new messages.

    1: for old messages

    2: for Work folder (vm-work.wav)

    3: for Family folder (vm-Family.wav)

    4: for Friends folder (vm-Friends.wav)

    #: to cancel (vm-tocancel.wav)

Press 3 for Advance options:

    5: leave message (vm-leavemeg.wav) --- When you hear a message, you can leave a message to the sender or another user's voice mail if the user also enabled the voicemail box function.

    ∗ return the main menu (vm-starmain.wav)

Press 0 of mailbox option:

1: to record your unavailable message. (vm-options.wav) (vm-rec-unv.wav)

    1: accept this recording

    2: listen to it

    3: re-record it

2: to record your busy message. (vm-options.wav) (vm-rec-busy.wav)

    1: accept this recording

    2: listen to it

    3: re-record it

3: to record your name. (vm-options.wav) (vm-rec-name.wav)

    1: accept this recording

    2: listen to it

    3: re-record it

4: to record your temporary greeting. (vm-options.wav) (vm-rec-temp.wav)

    1: accept this recording

    2: listen to it

    3: re-record it

∗ : to return to the main menu.

5.2 System Prompts (Chinese)

語音群組: 數字相關 語音檔			
錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0000	***0000	0.wav	0
**0001	***0001	1.wav	1
**0002	***0002	2.wav	2
**0003	***0003	3.wav	3
**0004	***0004	4.wav	4
**0005	***0005	5.wav	5
**0006	***0006	6.wav	6
**0007	***0007	7.wav	7
**0008	***0008	8.wav	8
**0009	***0009	9.wav	9
**0010	***0010	10.wav	10
**0011	***0011	11.wav	11
**0012	***0012	12.wav	12
**0013	***0013	13.wav	13
**0014	***0014	14.wav	14
**0015	***0015	15.wav	15
**0016	***0016	16.wav	16
**0017	***0017	17.wav	17
**0018	***0018	18.wav	18
**0019	***0019	19.wav	19
**0020	***0020	20.wav	20
**0030	***0030	30.wav	30
**0040	***0040	40.wav	40
**0050	***0050	50.wav	50
**0060	***0060	60.wav	60
**0070	***0070	70.wav	70
**0080	***0080	80.wav	80
**0090	***0090	90.wav	90
語音群組: 時間與日期相關 語音檔			
錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0101	***0101	h-1.wav	1
**0102	***0102	h-2.wav	2

**0103	***0103	h-3.wav	3
**0104	***0104	h-4.wav	4
**0105	***0105	h-5.wav	5
**0106	***0106	h-6.wav	6
**0107	***0107	h-7.wav	7
**0108	***0108	h-8.wav	8
**0109	***0109	h-9.wav	9
**0110	***0110	h-10.wav	10
**0111	***0111	h-11.wav	11
**0112	***0112	h-12.wav	12
**0113	***0113	h-13.wav	13
**0114	***0114	h-14.wav	14
**0115	***0115	h-15.wav	15
**0116	***0116	h-16.wav	16
**0117	***0117	h-17.wav	17
**0118	***0118	h-18.wav	18
**0119	***0119	h-19.wav	19
**0120	***0120	h-20.wav	20
**0130	***0130	h-30.wav	21
**0131	***0131	at.wav	在
**0132	***0132	a-m.wav	上午
**0133	***0133	p-m.wav	下午
**0134	***0134	hundred.wav	百
**0135	***0135	thousands.wav	千
**0136	***0136	million.wav	百萬
**0137	***0137	minus.wav	負
**0201	***0201	day-0.wav	星期日
**0202	***0202	day-1.wav	星期一
**0203	***0203	day-2.wav	星期二
**0204	***0204	day-3.wav	星期三
**0205	***0205	day-4.wav	星期四
**0206	***0206	day-5.wav	星期五
**0207	***0207	day-6.wav	星期六
**0208	***0208	dollars.wav	元
**0209	***0209	mon-0.wav	一月
**0210	***0210	mon-1.wav	二月
**0211	***0211	mon-2.wav	三月
**0212	***0212	mon-3.wav	四月
**0213	***0213	mon-4.wav	五月

**0214	***0214	mon-5.wav	六月
**0215	***0215	mon-6.wav	七月
**0216	***0216	mon-7.wav	八月
**0217	***0217	mon-8.wav	九月
**0218	***0218	mon-9.wav	十月
**0219	***0219	mon-10.wav	十一月
**0220	***0220	mon-11.wav	十二月
**0221	***0221	oh.wav	零 (spoken in a way meaning "zero")
**0222	***0222	oclock.wav	時
**0223	***0223	pound.wav	井字鍵
**0224	***0224	star.wav	米字鍵
**0225	***0225	today.wav	今天
**0226	***0226	tomorrow.wav	明天
**0227	***0227	yesterday.wav	昨天
**0228	***0228	Year.wav	年
**0229	***0229	Date.wav	日

語音群組: 自動總機相關 語音檔

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**111	***111	greeting-day.wav	您好,請輸入分機號碼,或按 9,由總機為您服務
**112	***112	greeting-noon.wav	您好,請輸入分機號碼,謝謝
**113	***113	greeting-night.wav	您好,請輸入分機號碼,謝謝
**114	***114	greeting-holiday.wav	您好,請輸入分機號碼,謝謝 (It is reserved, not functional now)
**115	***115	greeting-temporary.wav	您好,請輸入分機號碼,謝謝
**116	***116	noanswer.wav	您撥的分機號碼無人接聽,請改撥其他分機號碼,謝謝
**117	***117	busy.wav	對不起,您撥的分機號碼忙線中,請改撥其他分機號碼,謝謝
**118	***118	goodbyeivr.wav	謝謝您,再見
**119	***119	unavailable.wav	對不起,您撥的分機號碼無法接通,請改撥其他分機號碼,謝謝
**120	***120	invalid.wav	對不起,您撥的分機號碼不正確,請重新輸入
**121	***121	dnd-act:gsm	請勿打擾已啓動
**122	***122	dnd-deact:gsm	請勿打擾已解除
**123	***123	unconfwd-act:gsm	無條件轉移已啓動

**124	***124	unconfwd-deact:gsm	無條件轉移已解除
**125	***125	busyfwd-act:gsm	忙線轉移已啓動
**126	***126	busyfwd-deact:gsm	忙線轉移已解除
**127	***127	op-noanswer.wav	總機無人接聽，請稍後再撥或改撥其他分機號碼
**128	***128	op-busy.wav	總機忙線中，請稍後再撥或改撥其他分機號碼
**129	***129	op-unavailable.wav	總機無法接通，請稍後再撥或改撥其他分機號碼
**130	***130	noanswerfwd-act.wav	無人回答轉移已啓動
**131	***131	noanswerfwd-deact.wav	無人回答轉移已解除
**132	***132	allfwd-deact.wav	轉移功能已解除
**133	***133	transferop.wav	轉接總機中，請稍後
**134	***134	unavailablefwd-act.wav	不可利用轉移已啓動
**135	***135	unavailablefwd-deact.wav	不可利用轉移已解除
**136	***136	greeting-day2.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**137	***137	greeting-noon2.wav	您好，請輸入分機號碼，謝謝
**138	***138	greeting-night2.wav	您好，請輸入分機號碼，謝謝
**139	***139	greeting-holiday2.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**140	***140	greeting-temporary2.wav	您好，請輸入分機號碼，謝謝
**141	***141	greeting-day3.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**142	***142	greeting-noon3.wav	您好，請輸入分機號碼，謝謝
**143	***143	greeting-night3.wav	您好，請輸入分機號碼，謝謝
**144	***144	greeting-holiday3.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**145	***145	greeting-temporary3.wav	您好，請輸入分機號碼，謝謝
**146	***146	greeting-day4.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**147	***147	greeting-noon4.wav	您好，請輸入分機號碼，謝謝
**148	***148	greeting-night4.wav	您好，請輸入分機號碼，謝謝
**149	***149	greeting-holiday4.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**150	***150	greeting-temporary4.wav	您好，請輸入分機號碼，謝謝
**151	***151	greeting-day5.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**152	***152	greeting-noon5.wav	您好，請輸入分機號碼，謝謝

**153	***153	greeting-night5.wav	您好，請輸入分機號碼，謝謝
**154	***154	greeting-holiday5.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**155	***155	greeting-temporary5.wav	您好，請輸入分機號碼，謝謝
**156	***156	greeting-day6.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**157	***157	greeting-noon6.wav	您好，請輸入分機號碼，謝謝
**158	***158	greeting-night6.wav	您好，請輸入分機號碼，謝謝
**159	***159	greeting-holiday6.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**160	***160	greeting-temporary6.wav	您好，請輸入分機號碼，謝謝
**161	***161	greeting-day7.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**162	***162	greeting-noon7.wav	您好，請輸入分機號碼，謝謝
**163	***163	greeting-night7.wav	您好，請輸入分機號碼，謝謝
**164	***164	greeting-holiday7.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**165	***165	greeting-temporary7.wav	您好，請輸入分機號碼，謝謝
**166	***166	greeting-day8.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**167	***167	greeting-noon8.wav	您好，請輸入分機號碼，謝謝
**168	***168	greeting-night8.wav	您好，請輸入分機號碼，謝謝
**169	***169	greeting-holiday8.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**170	***170	greeting-temporary8.wav	您好，請輸入分機號碼，謝謝
**171	***171	greeting-day9.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**172	***172	greeting-noon9.wav	您好，請輸入分機號碼，謝謝
**173	***173	greeting-night9.wav	您好，請輸入分機號碼，謝謝
**174	***174	greeting-holiday9.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**175	***175	greeting-temporary9.wav	您好，請輸入分機號碼，謝謝
**176	***176	greeting-day10.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**177	***177	greeting-noon10.wav	您好，請輸入分機號碼，謝謝
**178	***178	greeting-night10.wav	您好，請輸入分機號碼，謝謝
**179	***179	greeting-holiday10.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**180	***180	greeting-temporary10.wav	您好，請輸入分機號碼，謝謝

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**181	***181	greeting-day11.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**182	***182	greeting-noon11.wav	您好，請輸入分機號碼，謝謝
**183	***183	greeting-night11.wav	您好，請輸入分機號碼，謝謝
**184	***184	greeting-holiday11.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**185	***185	greeting-temporary11.wav	您好，請輸入分機號碼，謝謝
**186	***186	greeting-day12.wav	您好，請輸入分機號碼，或按 9，由總機為您服務
**187	***187	greeting-noon12.wav	您好，請輸入分機號碼，謝謝
**188	***188	greeting-night12.wav	您好，請輸入分機號碼，謝謝
**189	***189	greeting-holiday12.wav	您好，請輸入分機號碼，謝謝 (It is reserved, not functional now)
**190	***190	greeting-temporary12.wav	您好，請輸入分機號碼，謝謝

語音群組: 語音信箱相關 語音檔

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0301	***0301	vm-advopts.wav	按 3 至進階選項
**0302	***0302	vm-and.wav	和
**0303	***0303	vm-calldiffnum.wav	撥不同的號碼請按 2
**0304	***0304	vm-changeto.wav	請選擇留言組
**0305	***0305	vm-cust1.wav	資料夾 五
**0306	***0306	vm-cust2.wav	資料夾 六
**0307	***0307	vm-cust3.wav	資料夾 七
**0308	***0308	vm-cust4.wav	資料夾 八
**0309	***0309	vm-cust5.wav	資料夾 九
**0310	***0310	vm-delete.wav	刪除留言請按 7
**0311	***0311	vm-deleted.wav	留言已刪除
**0312	***0312	vm-dialout.wav	請稍後
**0313	***0313	vm-enter-num-to-call.wav	請輸入您想撥的分機號碼
**0314	***0314	vm-extension.wav	分機號碼
**0315	***0315	vm-Family.wav	家人留言資料夾
**0316	***0316	vm-first.wav	第一則
**0317	***0317	vm-for.wav	至
**0318	***0318	vm-forward.wav	按 1 以輸入分機號碼，按 2 使用號碼簿服務
**0319	***0319	vm-forwardoptions.wav	按 1 在留言前加提示語，按 2 直接傳送
**0320	***0320	vm-Friends.wav	朋友留言資料夾
**0321	***0321	vm-from.wav	從

**0322	***0322	vm-from-extension.wav	留言由分機
**0323	***0323	vm-from-phonenumber.wav	留言由號碼
**0324	***0324	vm-goodbye.wav	再見
**0325	***0325	vm-helpexit.wav	尋求協助請按 米 字鍵，退出請按#字鍵
**0326	***0326	vm-INBOX.wav	新
**0327	***0327	vm-incorrect.wav	輸入不正確
**0328	***0328	vm-incorrect-mailbox.wav	輸入不正確，請重新輸入信箱號碼
**0329	***0329	vm-instructions.wav	收聽留言請按 1，退出請按#字鍵
**0330	***0330	vm-intro.wav	請在嗶一聲之後留言，結束請按#字鍵，或直接掛上電話。
**0331	***0331	vm-isonphone.wav	正在忙線中
**0332	***0332	vm-isunavail.wav	無法接通
**0333	***0333	vm-last.wav	上一則
**0334	***0334	vm-leavemsg.wav	留言請按 5
**0335	***0335	vm-login.wav	歡迎進入語音信箱，請輸入信箱號碼
**0336	***0336	vm-mailboxfull.wav	對不起，使用者語音信箱已滿，無法存取新留言
**0337	***0337	vm-message.wav	留言
**0338	***0338	vm-messages.wav	留言
**0339	***0339	vm-minutes.wav	分
**0340	***0340	vm-mismatch.wav	密碼輸入不一致，請重新輸入
**0341	***0341	vm-msginstruct.wav	收聽下一則留言請按 6，重聽本則留言請按 5，收聽上一則留言請按 4，刪除留言請按 0，退出請按#字鍵
**0342	***0342	vm-msgsaved.wav	留言已保存
**0343	***0343	vm-newpassword.wav	請輸入新密碼，結束請按#字鍵
**0344	***0344	vm-newuser.wav	歡迎進入語音信箱，首先，我將為你介紹操作流程
**0345	***0345	vm-next.wav	收聽下一則留言請按 6
**0346	***0346	vm-no.wav	零
**0347	***0347	vm-nobodyavail.wav	此時無人可接聽電話
**0348	***0348	vm-nobox.wav	由於留言者未開啟語音信箱功能，所以你無法回覆此留言
**0349	***0349	vm-nomore.wav	已無其他的留言
**0350	***0350	vm-nonumber.wav	留言者資料無法取得
**0351	***0351	vm-num-i-have.wav	取得號碼為
**0352	***0352	vm-Old.wav	舊
**0353	***0353	vm-onefor.wav	請按 1 至
**0354	***0354	vm-options.wav	錄製主人無法接通之問侯語請按 1，錄製

			主人忙線問侯語請按 2，錄製主人姓名請按 3，錄製主人暫時問後語請按 4，修改密碼請按 5，返回主目錄請按 ＊ 字鍵
**0355	***0355	vm-opts.wav	更改資料夾請按 2，進階選項請按 3，設訂語音信箱選項請按 0
**0356	***0356	vm-passchanged.wav	密碼修改成功
**0357	***0357	vm-password.wav	請輸入密碼
**0358	***0358	vm-press.wav	按
**0359	***0359	vm-prev.wav	收聽上一則留言請按 4
**0360	***0360	vm-reachoper.wav	轉接總機請按 9
**0361	***0361	vm-rec-busy.wav	請在 嗶一聲後錄製主人忙線問侯語,結束請按#字鍵
**0362	***0362	vm-received.wav	收到
**0363	***0363	vm-rec-name.wav	請在 嗶一聲後錄製主人姓名,結束請按#字鍵
**0364	***0364	vm-rec-temp.wav	請在 嗶一聲後錄製主人暫時問後語,結束請按#字鍵
**0365	***0365	vm-rec-unv.wav	請在嗶一聲後錄主人無法接通之問侯語,結束請按#字鍵
**0366	***0366	vm-reenterpassword.wav	請重新輸入密碼結束請按#字鍵
**0367	***0367	vm-repeat.wav	重聽留言請按 5
**0368	***0368	vm-review.wav	儲存錄音內容請按 1，聽取此錄音內容請按 2，重錄請按 3
**0369	***0369	vm-saved.wav	已保存
**0370	***0370	vm-savedto.wav	留言保存至
**0371	***0371	vm-savefolder.wav	請選擇留言保存位置
**0372	***0372	vm-savemessage.wav	保存留言請按 9
**0373	***0373	vm-saveoper.wav	儲存錄音內容請按 1
**0374	***0374	vm-sorry.wav	對不起，此選擇無效
**0375	***0375	vm-star-cancel.wav	取消請按 米 字鍵
**0376	***0376	vm-starmain.wav	回主選單請按米字鍵
**0377	***0377	vm-tempgreeting2.wav	錄製主人暫時問侯語請按 1, 刪除主人暫時問侯語請按 2
**0378	***0378	vm-tempgreeting.wav	錄製主人暫時問侯語請按 1
**0379	***0379	vm-tempremoved.wav	主人暫時問侯語已刪除
**0380	***0380	vm-then-pound.wav	並按#字鍵結束
**0381	***0381	vm-theperson.wav	您撥的分機號碼

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0382	***0382	vm-tocallback.wav	撥號至留言者請按 2
**0383	***0383	vm-tocallnum.wav	撥此號碼請按 1
**0384	***0384	vm-tocancel.wav	或按 # 字鍵取消
**0385	***0385	vm-tocancelmsg.wav	取消此留言請按 * 字鍵
**0386	***0386	vm-toenternumber.wav	輸入號碼請按 1
**0387	***0387	vm-toforward.wav	轉送留言至其他使用者請按 8
**0388	***0388	vm-tohearenv.wav	收聽留言資訊請按 3
**0389	***0389	vm-tomakecall.wav	外撥電話請按 4
**0390	***0390	vm-tooshort.wav	您的留言太簡短
**0391	***0391	vm-toreply.wav	留言回覆請按 1
**0392	***0392	vm-torerecord.wav	錄製您的訊息請按 3
**0393	***0393	vm-undelete.wav	取消留言刪除請按 7
**0394	***0394	vm-undeleted.wav	留言刪除已取消
**0395	***0395	vm-unknow-caller.wav	由未知號碼傳送過來
**0396	***0396	vm-whichbox.wav	如欲留言請輸入信箱號碼
**0397	***0397	vm-Work.wav	工作夥伴留言資料夾
**0398	***0398	vm-youhave.wav	您有

語音群組: 一般 語音檔

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0400	***0400	beep.wav	(This is a beep tone)
**0401	***0401	hours.wav	時
**0402	***0402	minutes.wav	分
**0403	***0403	auth-incorrect.wav	密碼輸入不正確，請重新輸入，結束請按 #字鍵
**0404	***0404	auth-thankyou.wav	謝謝
**0405	***0405	pbx-invalid.wav	對不起，您撥的分機號碼不正確，請重新輸入
**0406	***0406	pbx-invalidpark.wav	對不起，此分機無通話佇留，請重新輸入 (It is reserved, not functional now)
**0407	***0407	pbx-transfer.wav	轉接中
**0408	***0408	privacy-incorrect.wav	對不起，此號碼不正確
**0409	***0409	privacy-prompt.wav	請以區域號碼為開頭，輸入您的 10 位數電話號碼 (It is reserved, not functional now)
**0410	***0410	privacy-thankyou.wav	謝謝
**0411	***0411	privacy-unident.wav	對方號碼無法接受未定義的通話 (It is reserved, not functional now)
**0412	***0412	ss-noservice.wav	您撥的號碼無系統服務，請確認號碼後重

			撥 (It is reserved, not functional now)
**0413	***0413	transfer.wav	轉接中，請稍後
**0414	***0414	ivrrecord.wav	請輸入錄音代碼，在聽到嗶一聲後開始錄音，結束請按#字鍵。
**0415	***0415	CB-act.wav	忙線預約已啓動
**0416	***0416	clir-act.wav	"去電號碼隱藏"，已啓動.
**0417	***0417	clir-deact.wav	"去電號碼隱藏"，已解除.

語音群組: 代理人相關 語音檔

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0445	***0445	agent-pass.wav	請輸入密碼，結束請按#字鍵

語音群組: 會議室相關 語音檔

錄音 前置碼	聽取 前置碼	檔案名稱	系統預設語音
**0470	***0470	conf-adminmenu.wav	請按1靜音或解除靜音, 按2 鎖定或解除鎖定會議室, 按 3 踢出最後一個參加者, 按4或6調整會議室音量, 按7或9調整自己的音量, 或按8離開
**0471	***0471	conf- enteringno.wav	您正進入會議室
**0472	***0472	conf- errormenu.gm	無效的選擇
**0473	***0473	conf- getchannel.gm	請輸入線路號碼，結束請按#字鍵
**0474	***0474	conf- getconfno.wav	請輸入會議室號碼，結束請按#字鍵
**0475	***0475	conf- getpin.wav	請輸入會議室密碼
**0476	***0476	conf- hasjoin.wav	現在已進入會議室
**0477	***0477	conf- hasleft.wav	已離開會議室
**0478	***0478	conf- invalid.wav	會議室號碼錯誤，請輸入正確號碼
**0479	***0479	conf- invalidpin.wav	會議室密碼錯誤
**0480	***0480	conf- kicked.wav	您已被踢出會議室
**0481	***0481	conf- leaderhasleft.wav	會議室主持人已離開
**0482	***0482	conf- locked.wav	該會議室被鎖定
**0483	***0483	conf- lockednow.wav	該會議室現在被鎖定
**0484	***0484	conf- muted.wav	您現在是靜音狀態
**0485	***0485	conf- noempty.wav	目前沒有空的會議室
**0486	***0486	conf- onlyone.wav	目前會議室有一個參加者
**0487	***0487	conf- onlyperson.wav	你目前是會議室唯一的參加者
**0488	***0488	conf- otherinparty.wav	有其他參加者在會議室
**0489	***0489	conf- placeintoconf.wav	您現在已被加入會議室
**0490	***0490	conf- thereare.wav	現在有

**0491	***0491	conf- unlockednow.wav	會議室現在已解除鎖定
**0492	***0492	conf- unmuted.wav	您現在已解除靜音
**0493	***0493	conf- usermenu.wav	請按 1 靜音或解除靜音,按 4 調整會議室音量, 按 7 或 9 調整您自己的音量, 或按 8 離開
**0494	***0494	conf- userswilljoin.wav	有使用者將加入這個會議室
**0495	***0495	conf- userwilljoin.wav	有使用者將加入這個會議室
**0496	***0496	conf- waitforleader.wav	會議室將在主持人加入後開始

會議室現在已解除鎖定

您現在已解除靜音

請按 1 靜音或解除靜音,按 4 調整會議室音量, 按 7 或 9 調整您自己的音量, 或按 8 離開