VOI-9300

SIP IP PBX

User Manual

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1. Introduction

1.1. Overview

The VOI-9300 is an embedded Voice over IP (VoIP) PBX Server with Session Initiation Protocol (SIP) to provide IP extension phone connections for global virtual office of small-to-medium business (SMB) companys. Equipped with 4 x FXO ports, Ethernet LAN and WAN ports plus Life Line features, VOI-9300 integrates the telephony network and the data network into a manageable converged network to provide an efficient and economical PBX for global long distance voice communications.

VOI-9300 IP PBX works with various IP phones (Desktop, WiFi, Bluetooth, and DECT), VoIP gateways, and analog telephone adapters (ATA) to route calls among client phones, analog phones, and PSTN network. Call features such as conferencing, auto attendant, and voicemail can be seamlessly enabled for all phone devices. In addition, it also provides Internet access to all LAN devices through VPN NAT router.

VOI-9300 IP PBX provides call control and media relay services to SIP clients, and it performs many primary functions, such as SIP Registrar, SIP Outbound Proxy with media relay, SIP Gateways (FXO), SIP PBX for extension calls, Auto Attendant Interactive Voice Response (IVR), and Find-Me Conferencing.

VOI-9300 IP PBX has a built-in suite of PBX applications for supplemental services. This lowers down the total cost of a converged network enabled by VOI-9300 IP PBX than building separated infrastructures for legacy telephony network and data network. In addition, with a web-browsable interface to the data network configuration and voice service provisioning, VOI-9300 brings the manageability of both networks together to facilitate administration locally and/or remotely.

Note that VOI-9300 requires an IP address, a subnet mask, and its gateway Router IP address for its own use to connect to Internet. These three are available from your Internet service provider. VOI-9300 may enable PPPoE or DHCP features to automatically get an assigned dynamic IP from the ITSP. Please refer to Web Configurations for detailed information.

2. Features

The VOI-9300 IP PBX is equipped with RJ45 & RJ11 connectors and is featuring as the following:

- SIP Server supports 50 user registrations and 20 concurrent calls
- SIP v1 (RFC2543), v2 (RFC3261) with MD5 authentication (RFC2069 and RFC 2617)
- RJ45 x 2 for Ethernet WAN and LAN ports + RJ11 x 4 for FXO ports + Life Line FXS port
- Supports ITU-T G.711a, G.711u, GSM/MS-GSM, G.729A/B, VAD and CNG for Speech Codec
- Configurations by Web Browser
- Embedded NAT/DHCP Server
- PPPoE/DHCP Client for Dynamic IP plus NAT, VPN, DNS, and DDNS Clients
- Support STUN server and DMZ functions for NAT Traversal
- Support VPN function
- Support Call features; Call Forward/Waiting/Transfer/Hold, and Voice Conference Room
- Support E.164 ENUM Dial Number via SIP server
- Incoming Call Pickup for Group users
- Incoming Call Ringing for Group users
- Number Bonding and Call restrictions.
- Find Me function
- Extension Pickup for Attendant
- Bill Rate Table with Voice Mail
- Interactive Voice Recording (IVR) Settings by XML
- Programmable Prompt messages
- On-Line Subscriber Status
- Remote Firmware Upgraded by HTTP Web Interface
- Auto Provision Settings
- Out-Band DTMF (RFC 2833) / In-Band DTMF / Send DTMF SIP Info

3. Standard Compliance

The VOI-9300 IP PBX supports for the following standards

VoIP Protocols:	IETF RFC3261 and RFC 2543 for SIP
SIP Authentication:	IETF RFC2069 and RFC 2617 for MD5
Speech Codec:	ITU-T G.711a, G.711u, GSM/MS-GSM, G.729A/B, VAD and CNG
Echo Cancellation:	ITU-T G.165/168

4. Packing Content

Inside the package you should find:

(1) One VOI-9300 IP PBX

(2) One AC100~240V to 12VDC/1A Power Adaptor

(3) One Cat 5 Ethernet Cable

(4) One User Manual CD

5. LED Indicators & Interface Connectors

LED Indicators

On the front panel of VOI-9300, there are 12 LED indicators as the following table

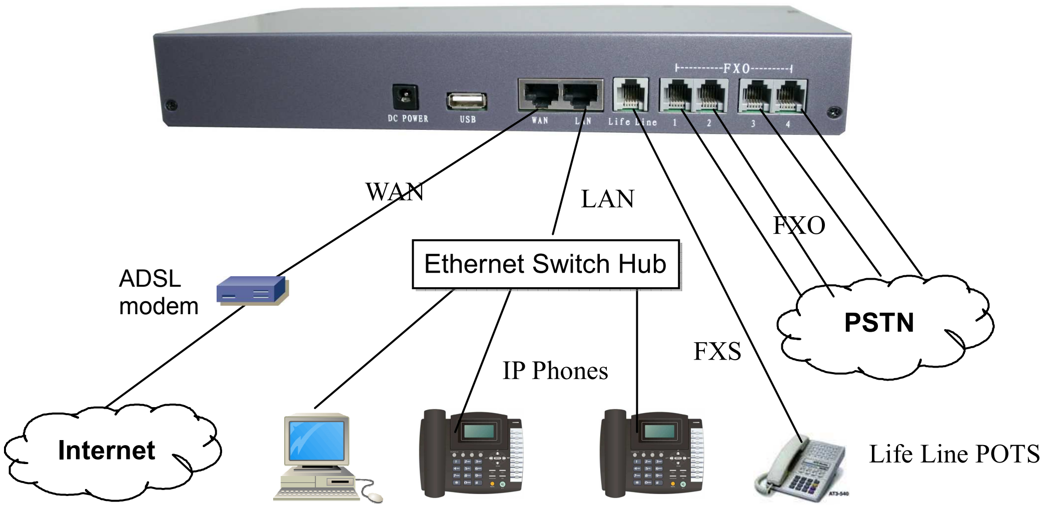
LED	Status	Descriptions
POWER	ON	Power is Normal.
ACTIVE	ON	IP PBX is in Normal Operation
ALARM	ON	IP PBX is at alarm status
VPN	ON	Virtual Private Network function is ON
FXO 1	ON	PSTN Line 1 is enabled and IDLE
	Flashing	PSTN Line 1 is in use
FXO 2	ON	PSTN Line 2 is enabled and IDLE
	Flashing	PSTN Line 2 is in use
FXO 3	ON	PSTN Line 3 is enabled and IDLE
	Flashing	PSTN Line 3 is in use
FXO 4	ON	PSTN Line 4 is enabled and IDLE
	Flashing	PSTN Line 4 is in use
LAN	ON	LAN Port is in connection
	Flashing	LAN Ethernet data activity
10/100M	ON	LAN Ethernet port is in connection at 100Mbps
WAN	ON	WAN Port is in connection
	Flashing	WAN Ethernet data activity
10/100M	ON	WAN Ethernet port is in connection at 100Mbps

Interface Connectors

1. DC Power	12Volt DC / 1A Power Adaptor with 100~240V AC power input
2. FXO ports	4 FXO ports are for connection to PSTN lines, and numbered 1, 2, 3 and 4 from left to right.
3. Life Line port	Life Line FXS port connects to an analog telephone. When power is down, the Life Line will switch to FXO port 1 for PSTN line 1.
4. WAN port	Connect to a broadband ADSL/Cable modem or a WAN router.
5. LAN port	Connection to PC for Web configurations or Laptop, IP Phones, or VoIP Gateways/ATA, etc.
6. USB port	Connect to an external USB drive for backup internal system storage. Click the **Backup** icon in Web configurations and follow instructions to insert the USB connector of an external USB drive.

Note: If PC is directly connected to the LAN port of VOI-9300 for web configurations, please use the enclosed Ethernet CAT5 Cable.

6. Installations



7. Reset to Factory Default

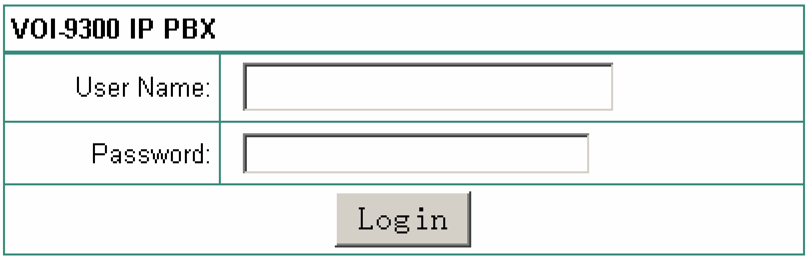
IP PBX VOI-9300 can be reset back to factory default when IP address is not accessible for web configurations. The procedures are as follows:

1. Power off.
2. Press the RESET button and hold continuously, then power on.
3. Hold the RESET button until all the LED indicators start flashing for three times. It may take 15-20 seconds to reset, and the RESET button can then be released after flashing.
4. The LED indicators will begin flashing for 3 times again to activate the IP PBX. Note that the firmware version and IP settings will be reset back to factory defaults and users data base will be cleared.
5. It is suggested that the user data base be exported and backed up to PC before reset.

8. IP PBX Configurations by Web Browser

You may enter the IP address from PC Web browser to configure VOI-9300. For example, enter http://192.168.1.1 from IE web browser to display login page as follows. Note that VOI-9300 support auto-MDIX for LAN port. If a notebook PC is directly connected to the LAN port of VOI-9300 for web configurations, the user may use the Ethernet CAT5 cable included in the accessory.

1). Please enter the default IP address http://192.168.1.1 from PC Web browser. The following Web page shall be displayed on PC. If you have difficulties accessing the Web page from the PC Web browser, the subnet IP of PC might be different from **192.168.1.xxx.** In this case, please refer to Chapter 9 for trouble shooting.



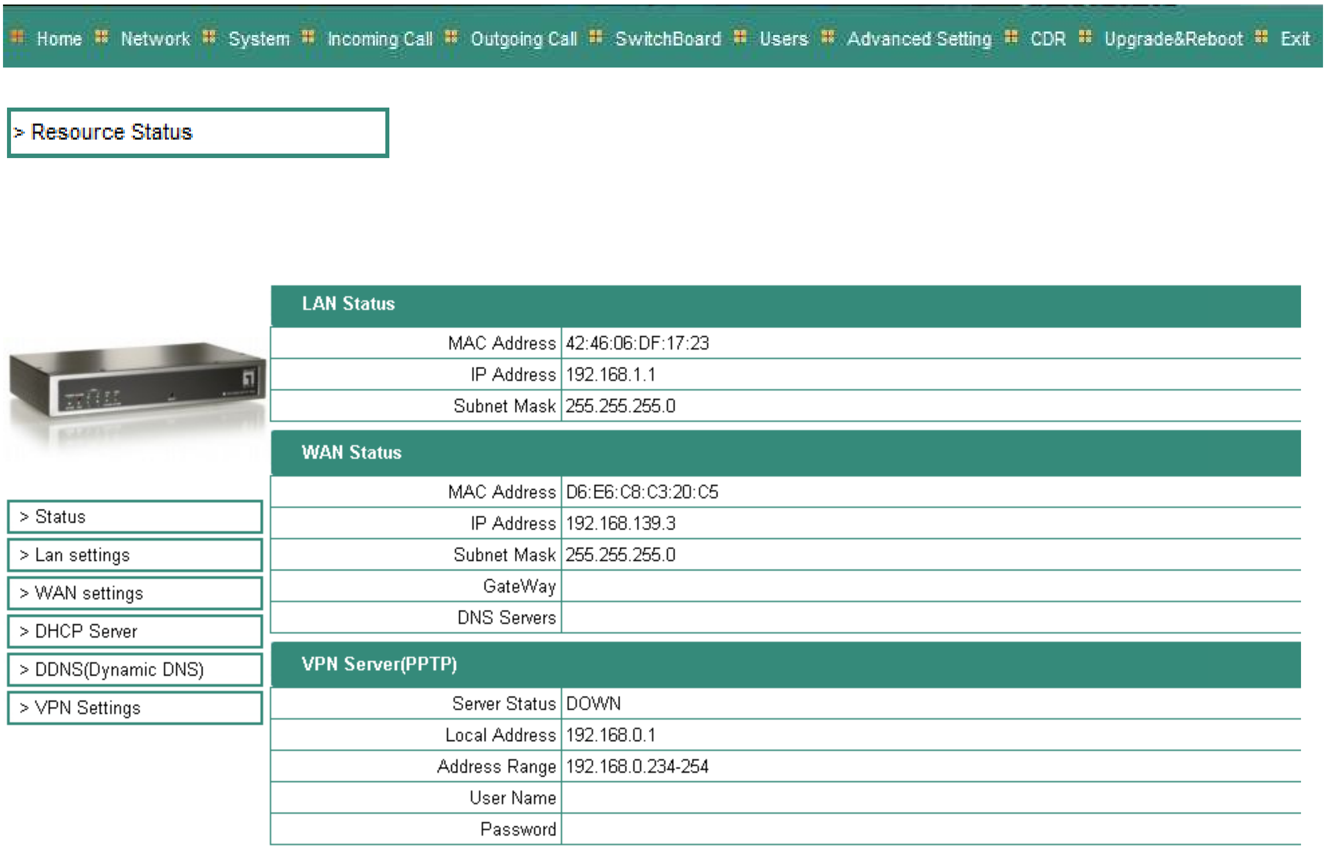
2). Please enter the username and password into the blank field. The default settings are:
Username: **admin**
Password: **123456**

3). Click the "**Login**" button to enter the VOI-9300 for web configurations. Whenever you change the setting in each Web page, remember to click the "Submit" button to save into the non-volatile memory and click the "Reboot" button to activate the new settings.

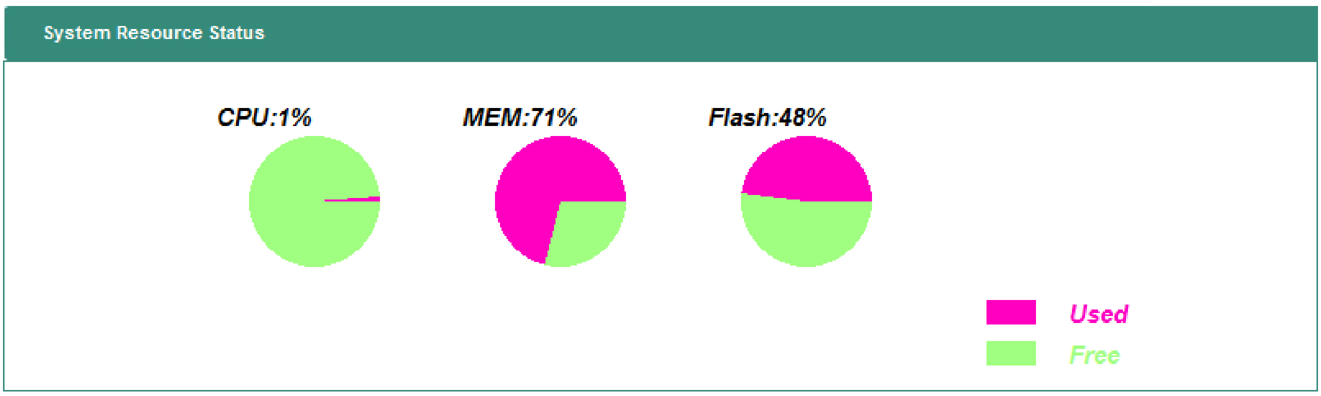
WAN & LAN Network IP Address and Mask

NIC	IP Address	Mask
WAN	**192.168.139.3**	**255.255.255.0**
LAN	**192.168.1.1**	**255.255.255.0**

4). VOI-9300 provides 100 SIP user ID number accounts which can be configured as well by Web browser. The preset user ID numbers are from **2001~2010** with same password **123456.** The SIP service port is default at **5060**.

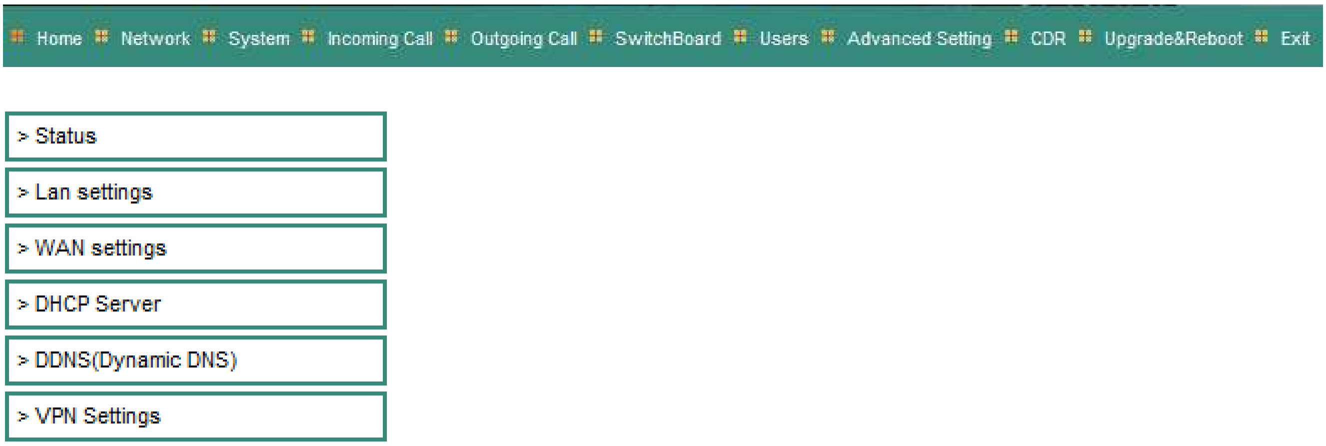


Resource Status



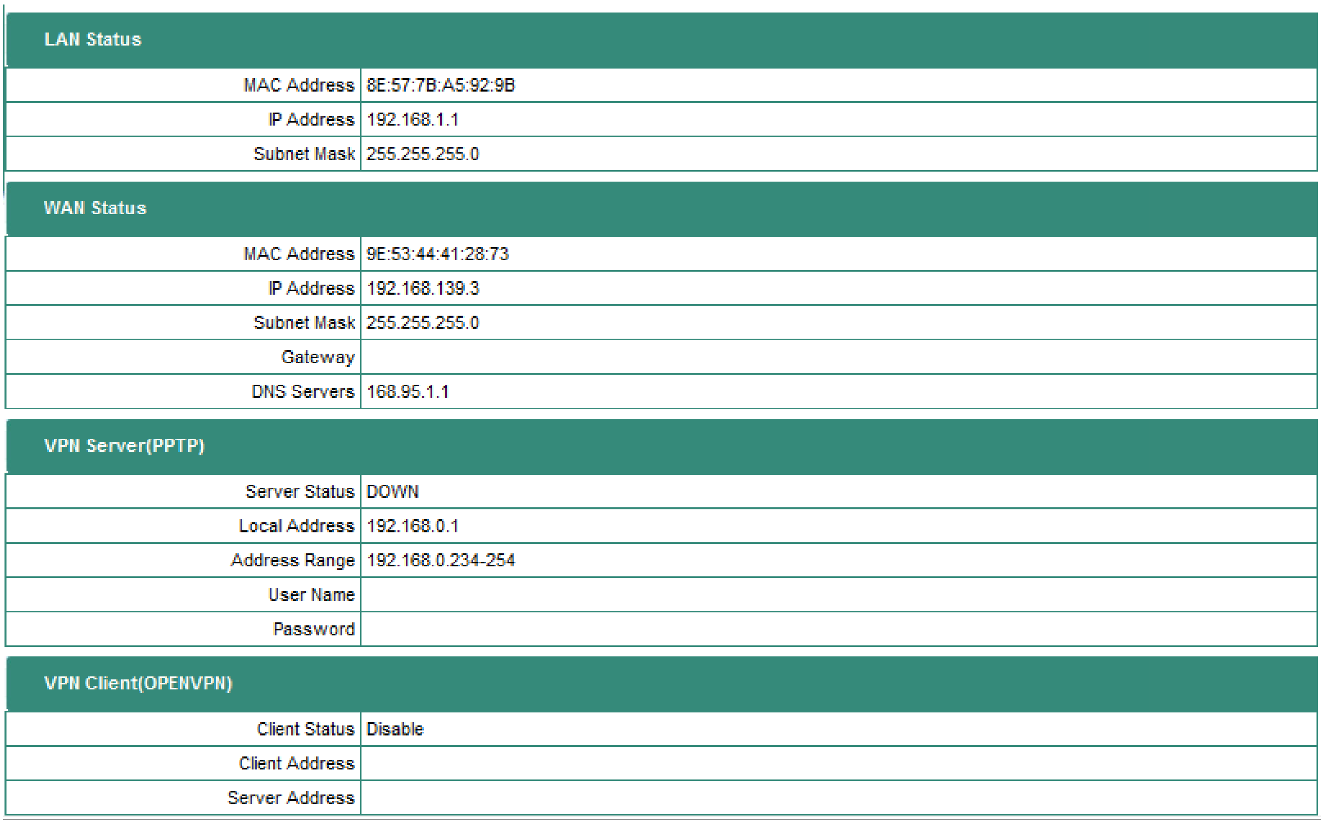
8.1 Network

VOI-9300 IP-PBX provides two RJ45 connectors for LAN and WAN ports at 10/100M Ethernet interfaces. The Network will display the current status for LAN, WAN, DHCP, DDNS, and VPN settings.



8.1.1 Network Status

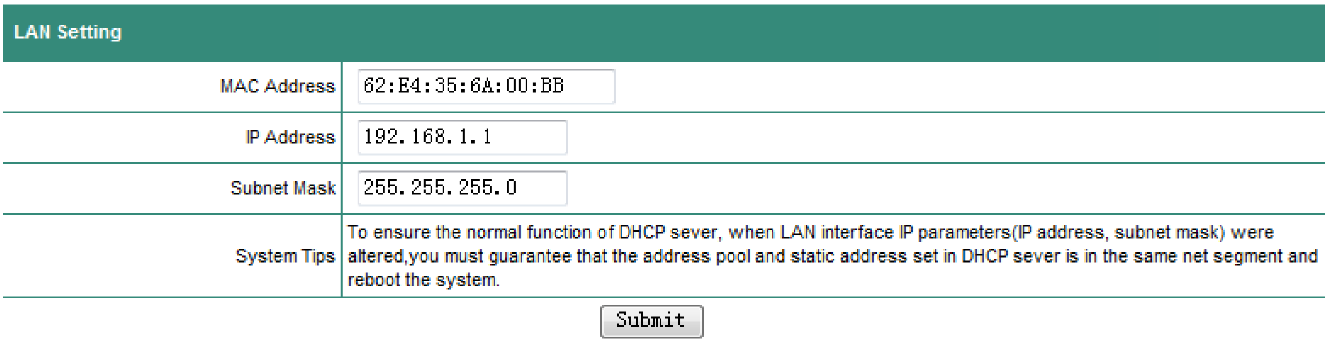
Network Status shows all the IP addresses for LAN, WAN, VPN server and VPN clients.



LAN Status	
MAC Address	8E:57:7B:A5:92:9B
IP Address	192.168.1.1
Subnet Mask	255.255.255.0
WAN Status	
MAC Address	9E:53:44:41:28:73
IP Address	192.168.139.3
Subnet Mask	255.255.255.0
Gateway	
DNS Servers	168.95.1.1
VPN Server(PPTP)	
Server Status	DOWN
Local Address	192.168.0.1
Address Range	192.168.0.234-254
User Name	
Password	
VPN Client(OPENVPN)	
Client Status	Disable
Client Address	
Server Address	

8.1.2 LAN Setting

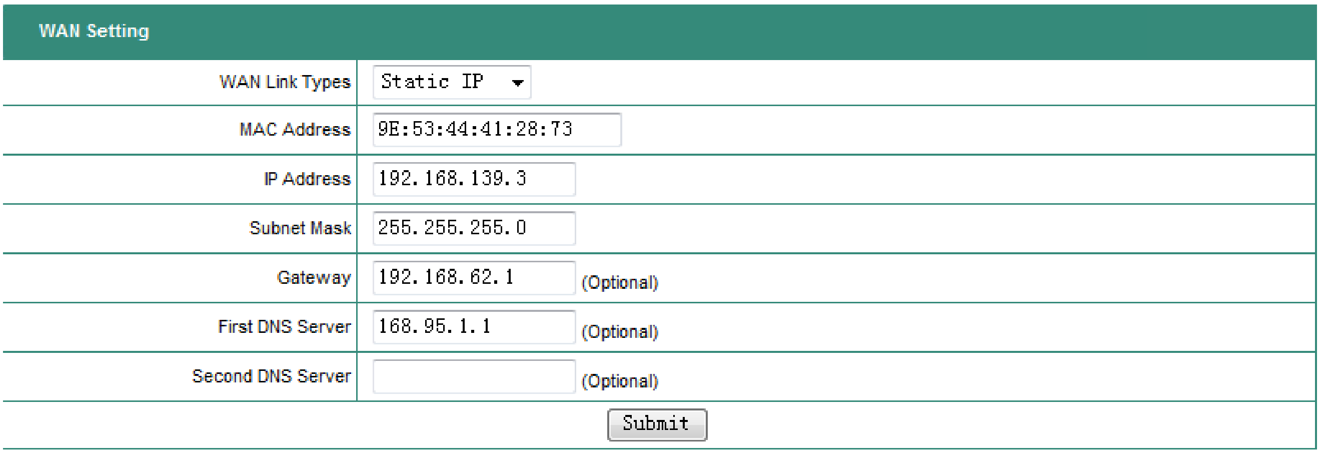
LAN Port can be used for IP-PBX to connect to a Notebook PC for configurations. The embedded DHCP Server will automatically assign IP address through the LAN port.



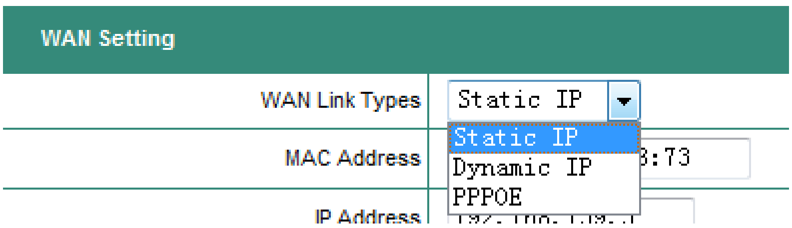
MAC Address should be unique in the same network, and IP Address must be in the format of xxx.xxx.xxx.xxx and xxx is from 0 to 255, e.g. 192.168.1.1. Subnet Mask is used for network segmentation. Please make sure the mask is correct and all the VoIP devices are within the same network as VOI-9300.

8.1.3 WAN Settings

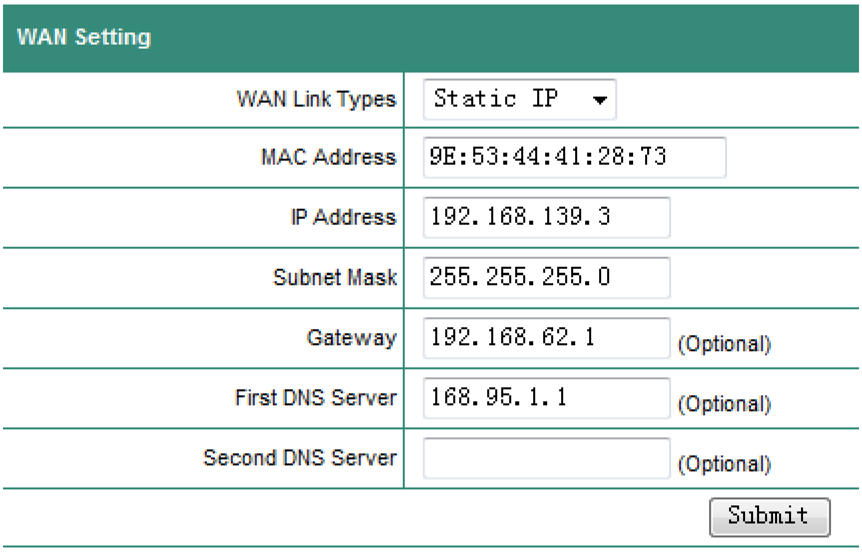
WAN port is to connect to ADSL modem for Internet access. There are 3 options for WAN settings; DHCP, Static IP, and PPPoE. The following example shows a Static IP type for WAN Setting.



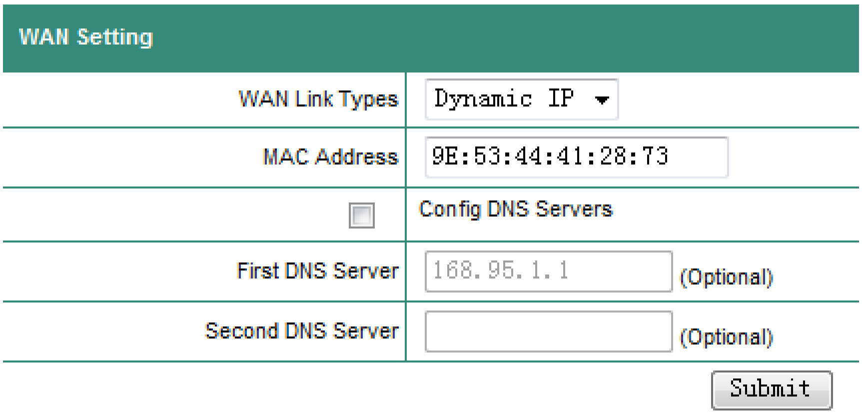
WAN Link Types



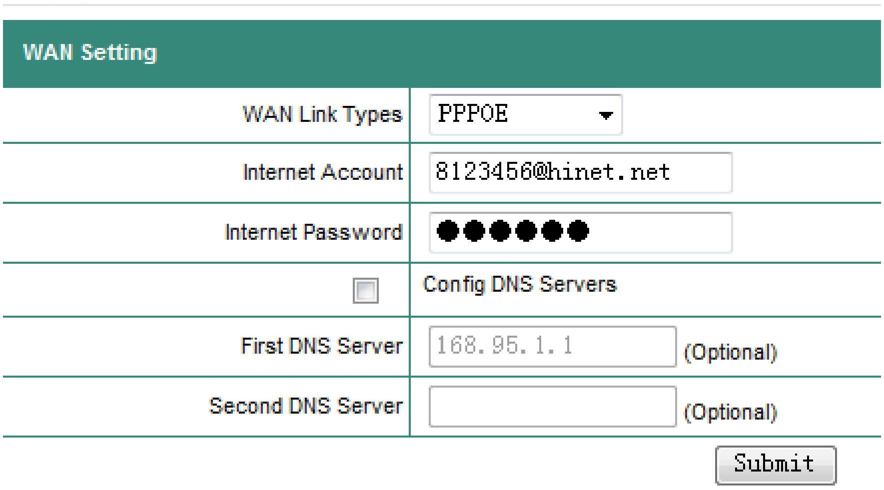
Static IP mode



Dynamic IP Mode



PPPoE Mode



When VOI-9300 IP-PBX connects to ADSL Modem with PPPOE link, you may need to enter the account name and password for PPPOE. In addition, you may select the DNS server and enter the IP address for the First and Second DNS servers.

8.1.4 DHCP Server

The embedded DHCP server in NAT will automatically assign IP address to the network devices.

DHCP Server Status: To show the current DHCP server status

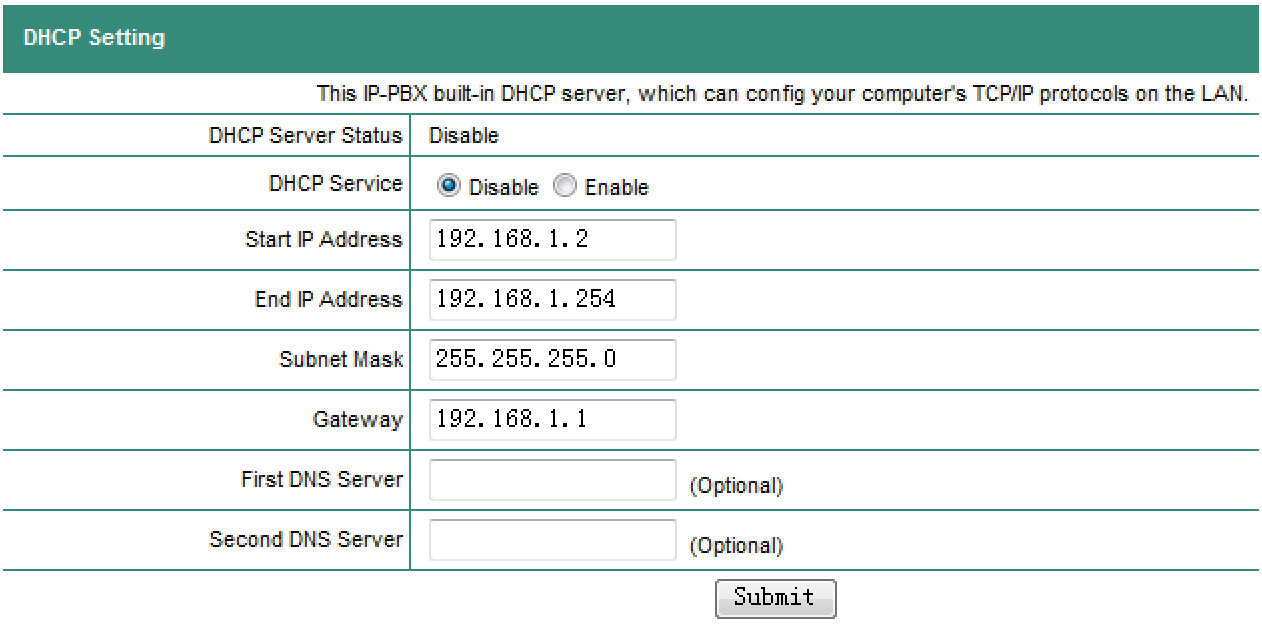
DHCP Server Start/Stop: To enable/disable DHCP Server

Start/End IP Address: DHCP Server will assign an IP within the start/end IP address range, e.g. 192.168.1.100 — 192.168.1.200. Note that the start IP and end IP must be in the same 192.168.1.xxx network.

Mask: Usually 255.255.255.0 for subnet mask

Default Gateway: The IP address for NAT gateway.

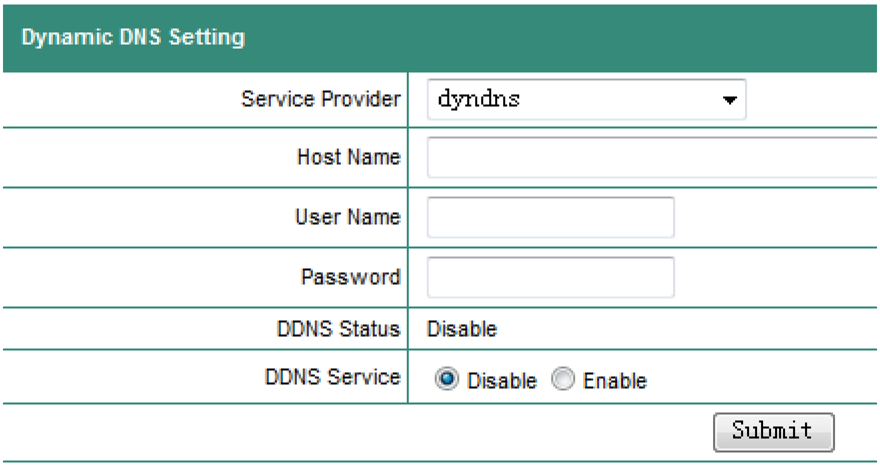
DNS Server: The Domain Name Server IP address.



8.1.5 DDNS Settings

DDNS is an abbreviation for Dynamic Domain Name Service. In general, ISP will allocate IP PBX a dynamic IP address by ADSL PPPoE mode connection when you access Internet, and the IP address may change day by day. In other words, the IP-PBX IP address may change and can NOT be reached by a fixed IP address. In such a case, DDNS can be a resolution.

DDNS resolution is used to associate domain name with dynamic IP address acquired from ISP. Users would only need to know the domain name, and do not care about the current PBX IP address. For DDNS, you should firstly register an account onto, for example, www.3322.org or www.dyndns.com for free. The IP-PBX supports the DDNS settings for these two most popular DDNS; 3322.org and dyndns.org.



Service Provider: Choose DDNS server that updates your IP address.

Host Name: Set your host name that need to do DDNS update.

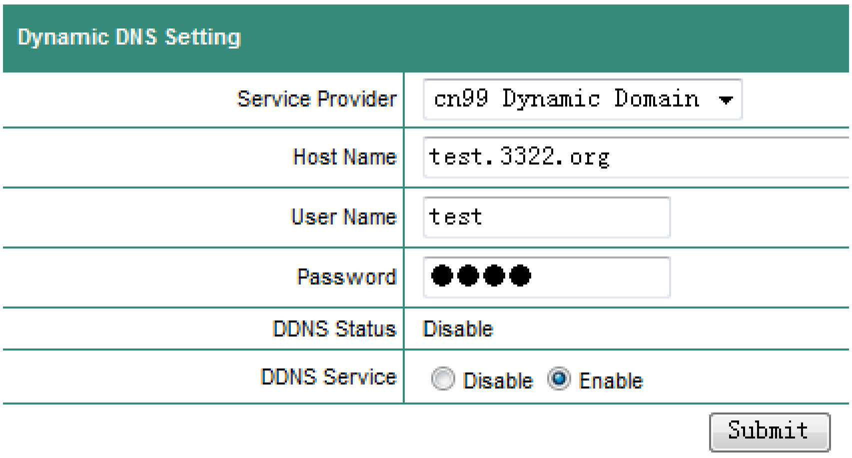
User Name: The DDNS server requires you to supply a name and password to update your IP address. You have to register the user name and password offline to the specific DDNS server.

Password: The DDNS server requires you to supply a name and password to update your IP address. You have to register the user name and password offline to the specific DDNS server.

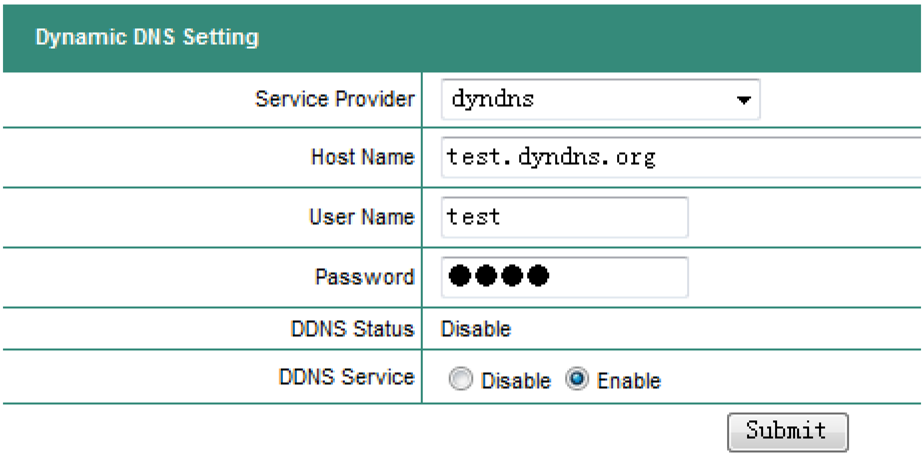
DDNS Status: Disable or Enable have to show.

DDNS Service: Disable/Enabled DDNS service.

Example for www.3322.org



Example for www.dyndns.com

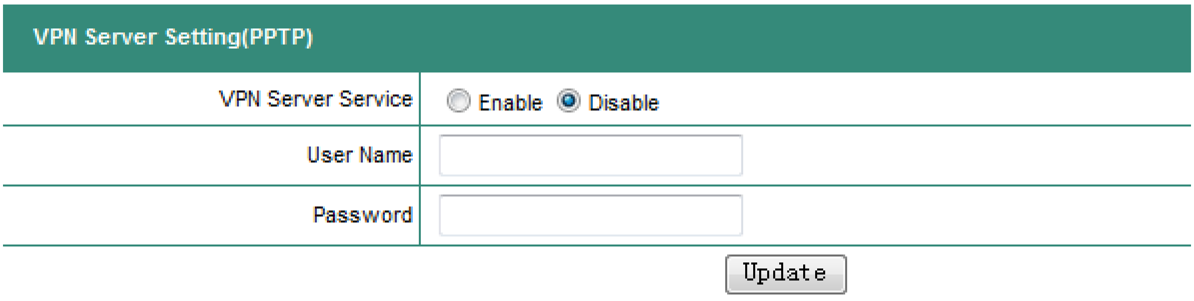


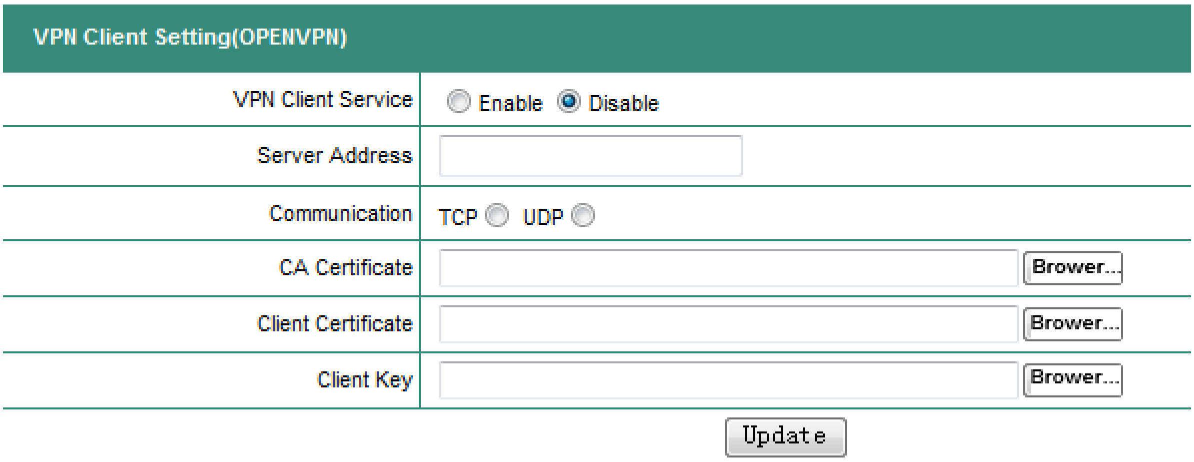
DDNS Settings

This shows the current registration of **www.3322.org** or **www.dyndns.org** for dynamic DNS service.

8.1.6 VPN Settings

VPN is namely virtual private network. VPN establishes a temporary and safely connection over public network (generally Internet). Usually VPN is an extension of an intranet. It helps remote users, company branches, commercial partners and suppliers to establish a reliable and safely connection with intranet and guarantees safe data transmission. The IP-PBX supports two kinds of VPN Link: PPTP VPN server and OpenVPN client.





VPN Server Configuration for PPTP (Point-to-Point Tunnel Protocol)

VPN Server Setting(PPTP)	
VPN Server Service	(•) Enable ( ) Disable
User Name	test
Password	test
	Update

VPN Server: You may enable or disable VPN Server Service.
UserName: For remote user when connected to VPN server.
Password: For remote user when connected to VPN server.
Press Update to save the configurations, and press Reboot to activate the new configurations.

VPN Client Configuration for OPENVPN

OpenVPN is used when the IP-PBX register as a VPN client to a remote VPN server.

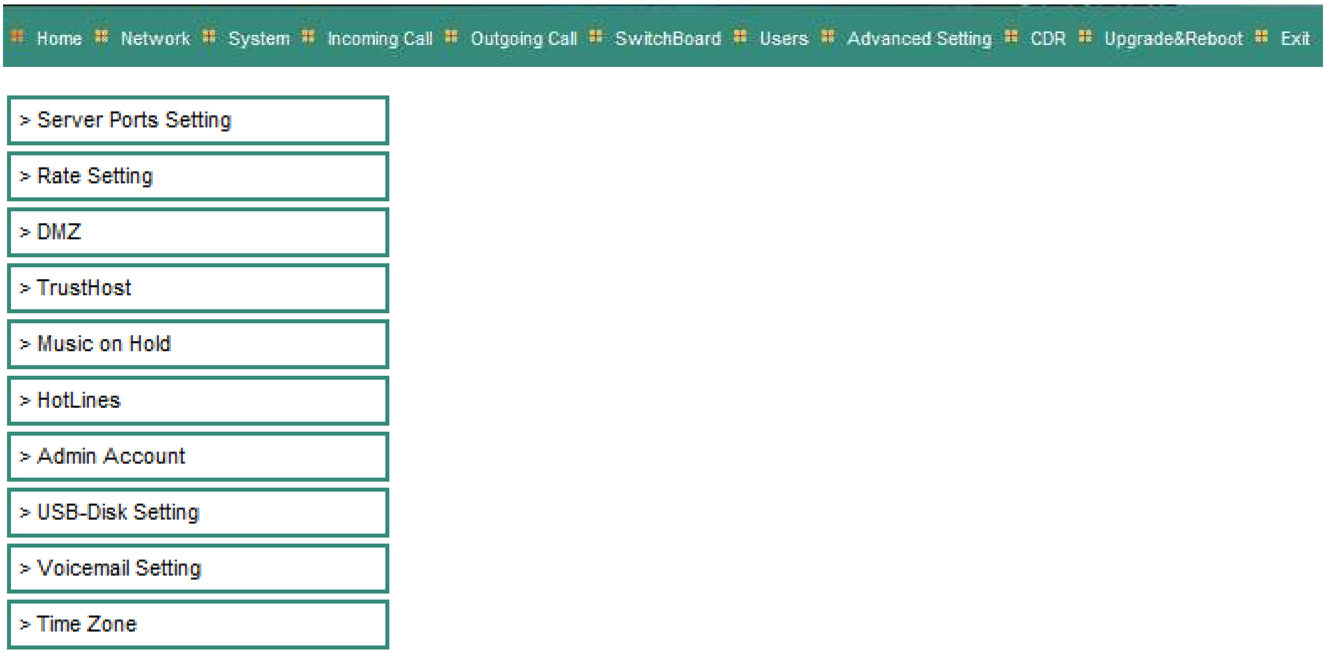
VPN Client Setting(OPENVPN)	
VPN Client Service	(•) Enable ( ) Disable
Server Address	220.229.128.75:1194
Communication	TCP (•) UDP ( )
CA Certificate	Brower...
Client Certificate	Brower...
Client Key	Brower...
	Update

VPN Client Service: You can enable or disable VPN Client Service.
Server Address: The IP address of remote VPN server with format of “address:port”.
Communication: The communication mode between clients and servers.
CA certificate: The CA certificate given by the VPN server.
Client certificate: The Client certificate given by the VPN server.
Client Key: The Client key given by the VPN server.

Press Update to save the configurations, and press Reboot to activate the new configurations.

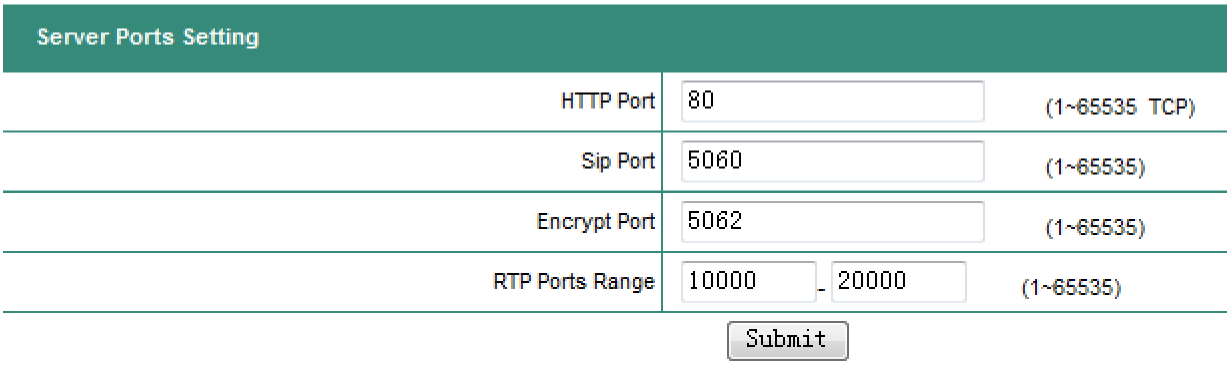
8.2 System

The VOI-9300 IP PBX System configurations can be set in this section. The settings include Server Ports Setting, Rate Setting, DMZ, TrustHost, Music on Hold, Hotlines, Admin Accounts, USB_Disk Setting, Voicemail Setting, and Time Zone.



8.2.1 SIP Port

The SIP default port number is 5060 for VoIP Applications. The value for SIP port can be from 1 to 65535. Note that the SIP port number must be the same for all the IP-PBX and VoIP phones and TAs.



HTTP Port : HTTP port number, default at 80.

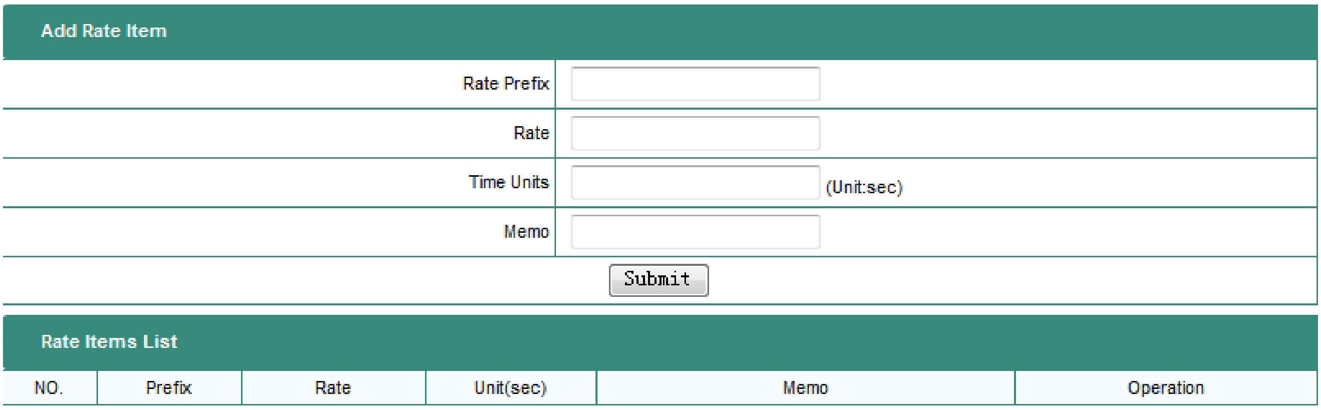
SIP Port : Default at 5060 .

Encrypt Port : Support SIP packet encryption with VP301/VP306 VoIP Phone and VS110/210/211 TA.

RTP Ports Range : RTP port for VoIP packets. Default at 10000~20000.

8.2.2 Rate Settings

Rate setting is used to calculate the charges for each call. The IP PBX will generate a call record for the charges. All the charges are based on this rate table.



The rate is based on the prefix to calculate the charges:

Prefix: 0

Rate: 15 (cent)

Time Unit： 60 (second)

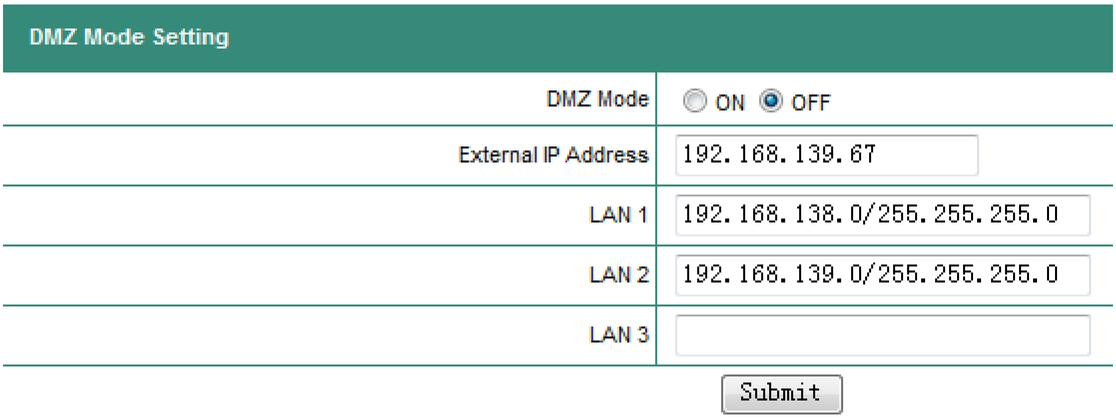
This means when calling a number with 0 prefix (e.g., 01010086)，The rate is 15 cents for every 60 seconds , and the duration less than 60 seconds will be charged the same as 60 seconds. In addition, the prefix is for the longer one. For example, if there are rates for prefixes 0 and 00, the call number with 00 prefix will be charged per the rate of 00 instead 0.

If there is no rate for the calling prefix, this implies the rate is 0.

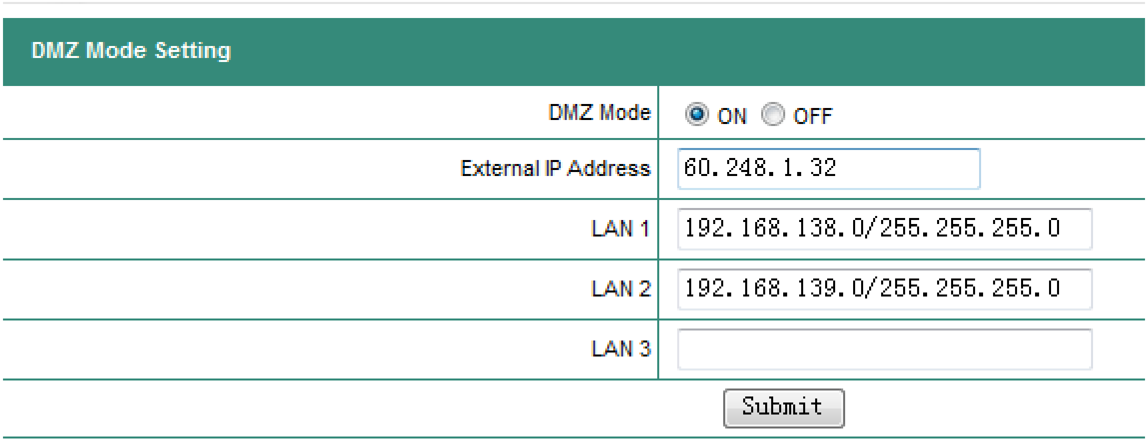
8.2.3 DMZ Settings

DMZ is an abbreviation for demilitarized zone. It is a buffer between safe system and unsafe system to solve the problem that facilities in the outer net access servers behind firewall in the private network. The buffer is in a small network zone between enterprise internet and outer net where it can put some servers that must be public, for example Web server, FTP server and so on.

The IP-PBX supports DMZ by cooperating with routers which provides DMZ function.



The DMZ must be set ON if the DMZ of firewall is activated.



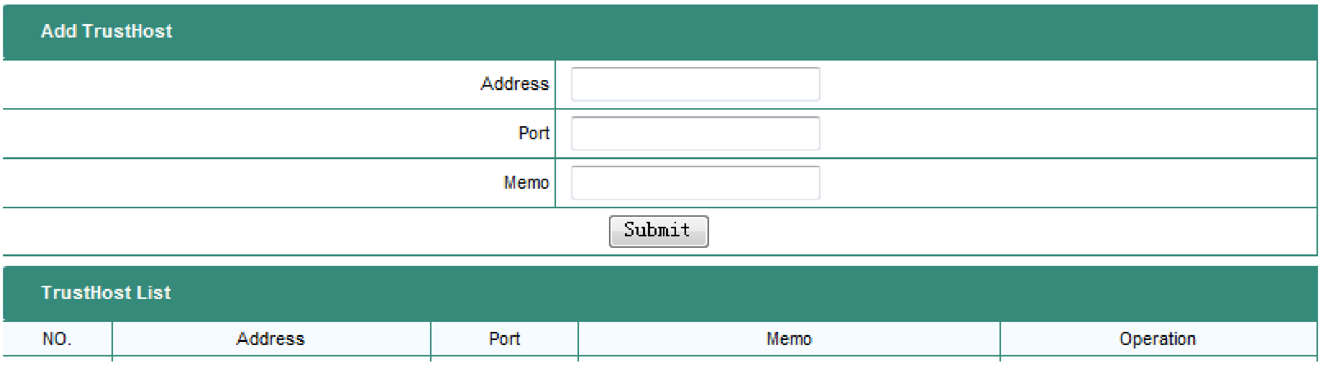
DMZ mode: You can enable DMZ mode or disable.

External IP Address: External address can be accessed by facilities from Internet.

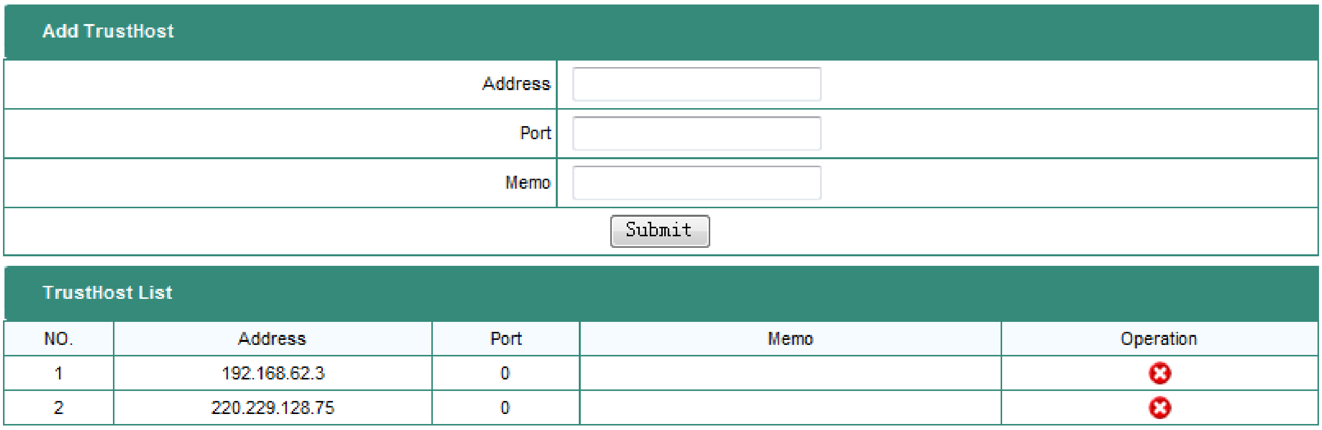
LAN n: This is the private network segments where the IP-PBX resides in. Update your settings, Submit -> Reboot.

8.2.4 Trust Host

It is not need to have authentication from the call of Trusted IP Address. If the port is set at 0, all the ports from this trusted address will not need authentications. The trusted address could be either IP address or domain name.



The TrustHost address can be added or deleted, and need to be configured in case of the following conditions;



Add TrustHost	
Address	
Port	
Memo	
Submit	

TrustHost List				
NO.	Address	Port	Memo	Operation
1	192.168.62.3	0		
2	220.229.128.75	0		

(1) Need to work with FXO ports,

(2) Need to connect with VOI-9300,

(3) Need to connect with another SIP Server.

In summary, the TrustHost address can be set whenever the authentication is not needed.

8.2.5 Music ON Hold

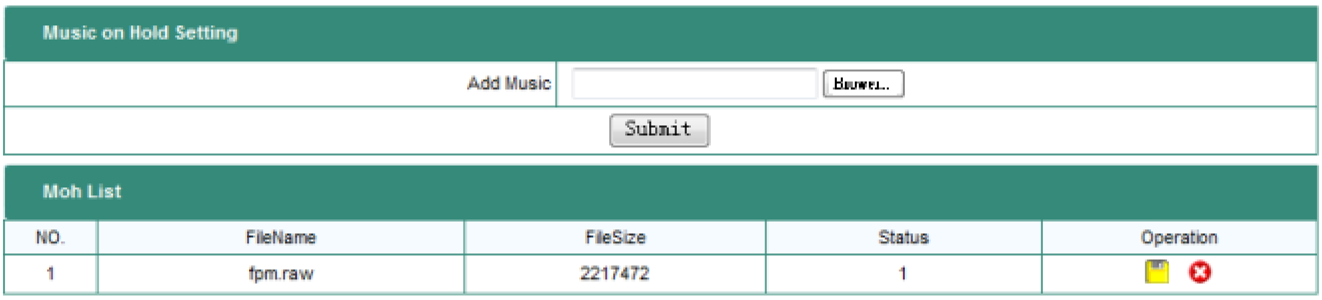
The IP PBX will play music when a call is on hold due to the following situations;

(1) When the call is transferred to attendant and waiting for answer.

(2) When the call is hold and waiting for answer.

(3) When the call is transferred and waiting for answer.

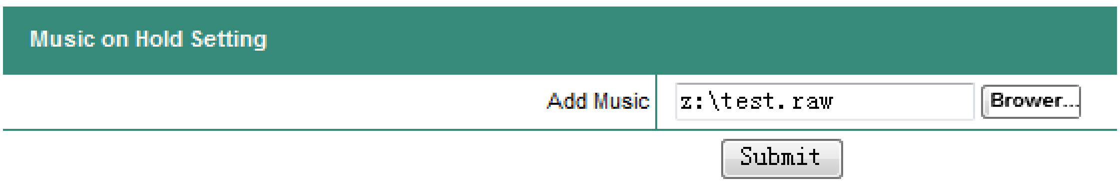
You may choose one of the music files for MUSIC ON Hold. You can upload more music , But only One can Play , The size is unlimited. Propose < 200KB.



Music on Hold Setting	
Add Music	Brower..
Submit	

Moh List				
NO.	FileName	FileSize	Status	Operation
1	fpm.raw	2217472	1	

Music ON Hold requires a unique file format, and any MP3 files must be converted to **.raw** format before upload to VOI-9300 for use of Music ON Hold.



Music on Hold Setting	
Add Music	z:\test.raw Brower...
Submit	

8.2.6 Hot Lines

Hot line function numbers can be entered into the list. Make sure each of the Hot line function number is unique, and not the same as any extension number or PSTN numbers.

The Hotline list displays the function number of the PBX, and it will activate the function service when pressing the corresponding function numbers.

HotLine List

NO.	Description	HotLine	Operation
1	AA	112	
2	CID Reader	117	
3	Queue Settings Hotline	1600	
4	Operators Hotline	1601	
5	Exten Settings Hotline	1602	
6	My Own Voicemail	1603	
7	Voicemail	1604	
8	Conference Room1	1650	
9	Conference Room2	1651	
10	Queue1	1701	
11	Queue2	1702	
12	Queue3	1703	
13	Queue4	1704	
14	Operators	9	

AA: Auto Attendant Voice Prompt Flow Chart

CID Reader: Voice Prompt for the Current IP Phone extension number

Queue Settings Hotline: For Q1-Q4 function settings

Operators Hotline: To assign or unassign extension number as the operator

Exten Settings Hotline:For setting Hotline extension number

My Own Voicemail: To enter Individual voicemail

Voicemail: To enter the wholevoicemail system

Record Prompt: For recording voice prompt

Conference Room1: To enter Conference Room 1

Conference Room2: To enter Conference Room 2

Queue1: Enter Queue1 operators .

Queue2: Enter Queue2 operators .

Queue3: Enter Queue3 operators .

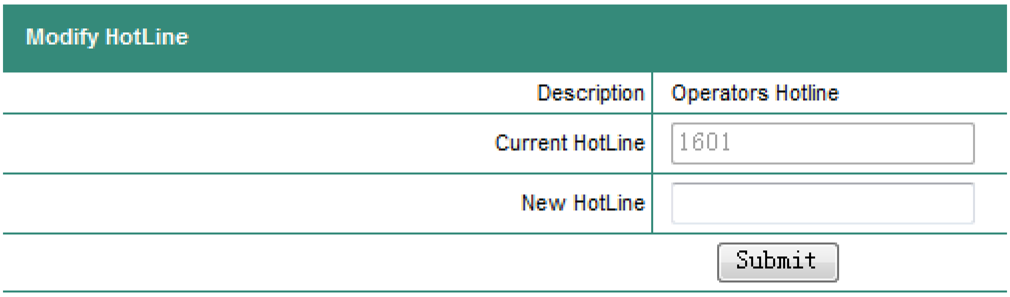
Queue4: Enter Queue4 operators .

IP BroadCast : To Broadcast on the IP Phone exten of Group 0.

Operators : The extension number of operator

Features :

- Standard: Standard SIP Protocol
- CBCOM: Encryption protocol for SIP and Voice RTP



Detailed Descriptions for Hotline Function Numbers

Hotline Numbers	Functions	Descriptions
*8	Call pickup	Press ***8#** to pickup calls within the same group
9	Operator	Press **9#** to call Operator.
112	Auto Attendant (AA)	Press **112#** to enter Auto Attendant IVR followed with the voice prompts.
1600	Queues Setting	Any extension phone can join or leave for one of the Queues. **Example:** To join as a member of Queue 1 for extension 201 **Procedures:** From 201 IP Phone, Dial **1600** -> **1701** -> **123456** (password of Queue1) -> **1** -> hang up. **Example:** To leave Queue 1 for extension 201 **Procedures:** From 201 IP Phone, Dial **1600** -> **1701** -> **123456** (password of Queue1) -> **2** -> hang up.
1601	Operators Setting	Any extension phone can join or leave for the Operators. **Example:** To join as one of operators for extension 200 **Procedures:** From 200 IP Phone, Dial **1601** -> **123456** (password of Operator) -> **1** -> hang up. **Example:** To quit as one of operators for extension 200 **Procedures:** From 200 IP Phone, Dial **1601** -> **123456** (password of Operator) -> **2** -> hang up.
1602	Extension functions	Forward , Find me , Binding, and Password can be set up for each extension number. **Example:** To set extension 202 busy forward to 203

		Procedures: From 202 IP Phone, Dial **1602** -> **1** (set forward) -> **2** (busy forward)-> **203#** -> **1** (submit) -> hang up.
1603	Play Voicemail for Current Extension	To play & to delete the voicemails for the current IP phone extension number. **Example:** To hear the first voice message and delete message afterward, and to hear the following voice message and delete afterward for extension 203. **Procedures:** From 203 IP Phone, Dial **1603** -> **1** -> **7** -> **6** -> **7** -> **#** to hang up.
1604	Play Voicemail for Other Extension	To play & to delete the voicemails for the other IP phone extension number. **Example:** To hear the voicemail of extension 203 from extension 200 **Procedures:** From 200 IP Phone, Dial **1604** -> **203** -> **123456**(password of extension 203) ->**1** -> **7** -> **6** -> **7** -> **#** to hang up.
1650	Conference Room1	Hotline number of Conference Room1
1651	Conference Room2	Hotline number of Conference Room2
1701	Queue1	Hotline number of Queue1
1702	Queue2	Hotline number of Queue2
1703	Queue3	Hotline number of Queue3
1704	Queue4	Hotline number of Queue4

Hotline Applications

Example:

A customer calls into Operator and wants to talk to the IP extensions 203, but the 203 user are not near the phone.

Procedures for Operator:

(1) Call Parking for the incoming call

(2) IP Broadcasting for the 203 user

(3) The 203 user may pickup any IP phone and dial 701 to take the waiting call

Example:

If the IP PBX prompts an incoming call with the prompt message “Press 1 for customer service, Press

2 for technical support, Press 0 for directory", then you may set as the following;

(1) Call hold 1 for customer service and connect extension 1;

(2) Call hold 2 for technical support and connect extension 2;

(3) Modified the prompt voice messages.

1) Change 1701 hotline number to 1, and 1702 hotline number to 2.

System > Hotlines > Queue1 > Modify of Operation.

Modify HotLine	
Description	Queue1
Current HotLine	1701
New HotLine	1
	Submit

Click Submit to finish.

System > Hotlines > Queue2 > Modify of Operation.

Modify HotLine	
Description	Queue2
Current HotLine	1702
New HotLine	2
	Submit

Click Submit to finish.

2) Add to the Queues for an existing extension number (208).

Advanced Setting > Queue Setting > Agent Setting

Current Queue List

NO.	Queue Extension	Queue Passwd	Strategy	Queue Length	Queue Desc.	Operation
1	1	123456	Roundrobin	10		
2	1702	123456	Roundrobin	10		Agent Setting

Add User as Agent	
Queue Extension	1
User-ID	208
Priority	5
Memo	
	Submit

Submit to add the number 208 for Queue1.

Current Queue Agents List					
NO.	Extension	Agent User-ID	Priority	Memo	Operation
1	1	208	5		

The number will be displayed in the Current Queue Agent List for Queue 1.

Add User as Agent	
Queue Extension	2
User-ID	203
Priority	5
Memo	
	Submit

Submit to add the number 203 for Queue2.

Current Queue Agents List					
NO.	Extension	Agent User-ID	Priority	Memo	Operation
1	2	203	5		

The number will show in the Current Queue Agent List for Queue2 .

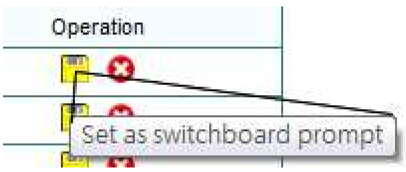
3) Add Voice prompts to Switchboard.

Add Prompt	
New Prompt	C:\work\welcome.gsm Brower..
	Submit

Submit.

Prompts List				
NO.	FileName	FileSize(Bytes)	Status	Operation
1	welcome.gsm	23892	0	
2	sippbx.gsm	10791	0	
3	greeting.gsm	23892	1	

It will display voice prompts in the List.



Press the ICON to Activate the Voice.

Prompts List				
NO.	FileName	FileSize(Bytes)	Status	Operation
1	welcome.gsm	23892	1	
2	sippbx.gsm	10791	0	
3	greeting.gsm	23892	0	

The No. 1 of Status will show "1" to indicate activate status.

8.2.7 Admin Account

Modify Password	
Admin Name	admin
New Password	
Confirm Password	
	Submit

8.2.8 USB Disk Setting

USB disk Service	
Tips	Ensure that USB Disk have been inserted When you record voice-mail in USB disk.Ensure that you have cancelled the use of USB Disk recording voice-mail when you remove the USB Disk.
Operation&Status	Insert USB Disk
Record voice-mail in USB Disk	Enable ○ Disable ◉
Submit	

When you need to store the voice mails, please do the steps as follows;

USB disk Service	
Tips	Ensure that USB Disk have been inserted When you record voice-mail in USB disk.Ensure that you have cancelled the use of USB Disk recording voice-mail when you remove the USB Disk.
Operation&Status	Insert USB Disk
Record voice-mail in USB Disk	Enable ◉ Disable ○
Submit	

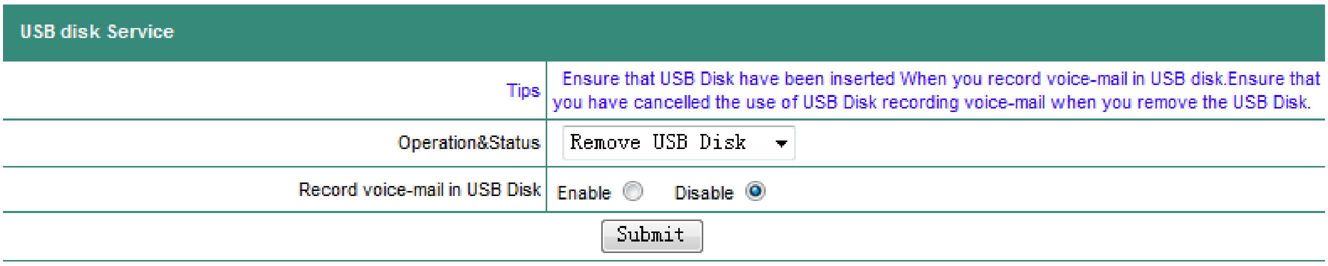
(1) Insert USB disk

(2) Select "Insert USB Disk" in the current status

(3) Select Yes to enable USB voice mail recording.

(4) Click Submit button.

When the voice mails is not needed, please do the steps as follows;



(1) Select “Insert USB Disk” in the current status

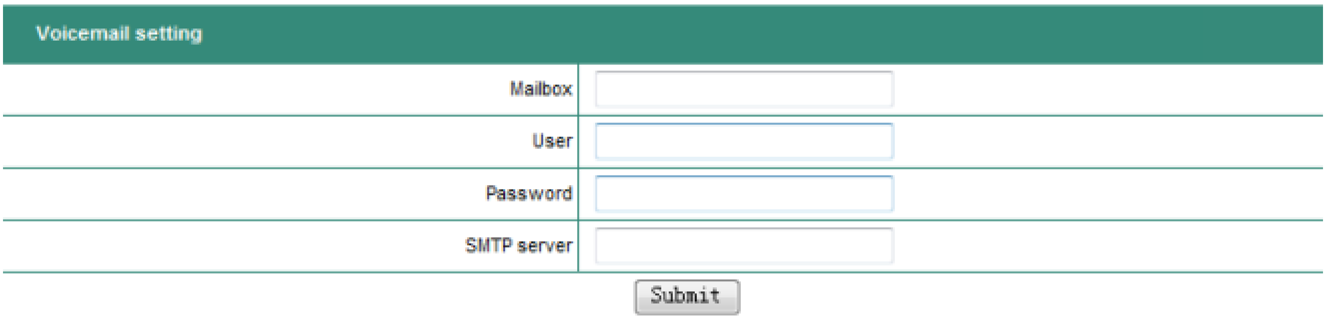
(2) Select No to disable USB voice mail recording.

(3) Click Submit button.

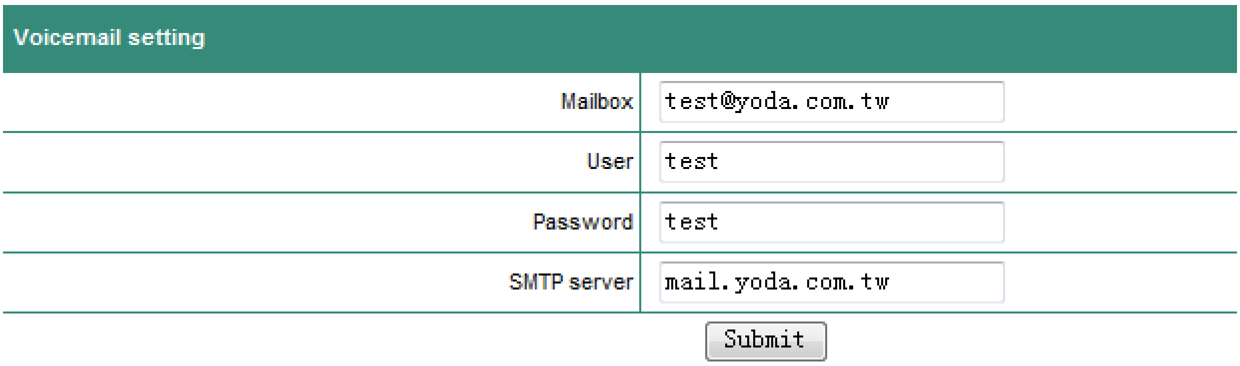
(4) Pull out the USB disk.

8.2.9 Voicemail Setting

Voicemail is the voice messages of incoming calls. When the call is not answered, the VOI-9300 will play the voice prompt and the caller may leave voice message. You can play the caller’s voice message in time. This setting must be compatible with the Voice-Mail Service of Users Function Setting .



General settings :



MailBox : The e-mail address of mailbox.

User : The user name used by the e-mail address .

Password : The password used by the e-mail address .

SMTP Server : This is the e-mail address used by SMTP of e-mail server domain name .

Use SSL Mode : Need to be checked If SSL mode is used. Do not check if General settings is used.

SSL for Gmail :

The IP-PBX supports voicemail via Gmail server by using SSL mode.

MailBox: This is used for the sending account in Gmail. This must be registered in Gmail in advance.

User: The user name reistered to Gmail.

Password: The password registered to Gmail.

SMTP Server: Gmail uses port 465 for the SSL SMTP server.

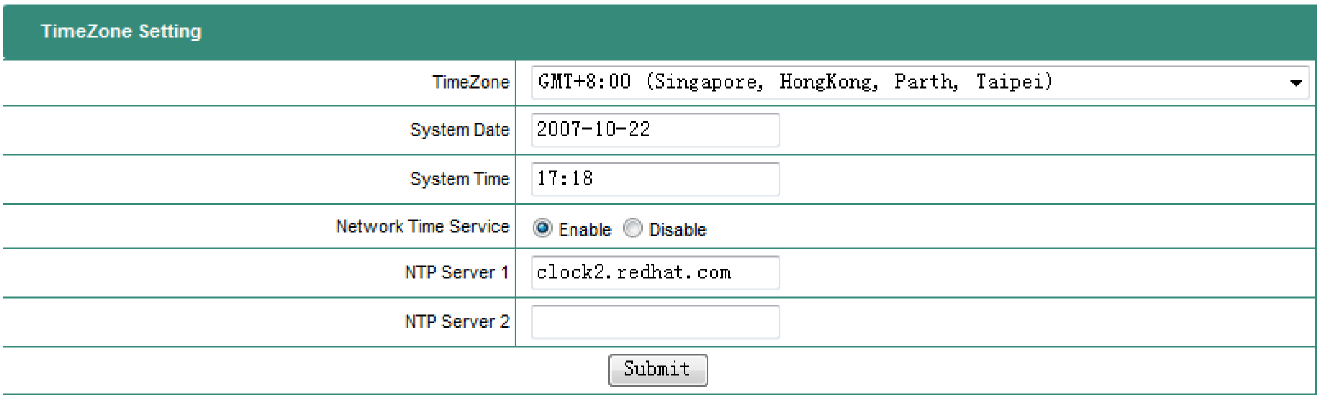
Use SSL Mode: Check to activate the function.

Press Submit to save the configurations, and press Reboot to activate the new configurations

8.2.10 Time-Zone Setting

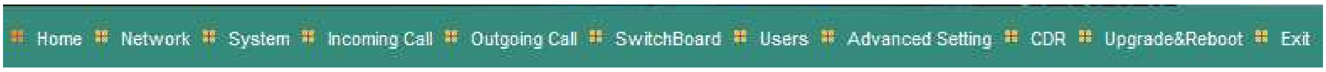
This provides system date and time information. The time will be recorded for use of CDR, or the CDR may record wrong timing if the system date and time were not set. The Network Time Service can be enabled to synchronize with NTP server. You still need to set the TimeZone, Date, and Time. It will display the synchronized NTP time when you re-enter this setting.

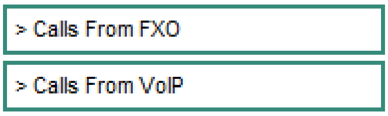
Press Submit to save the configurations, and press Reboot to activate the new configurations.



8.3 Incoming Calls Settings

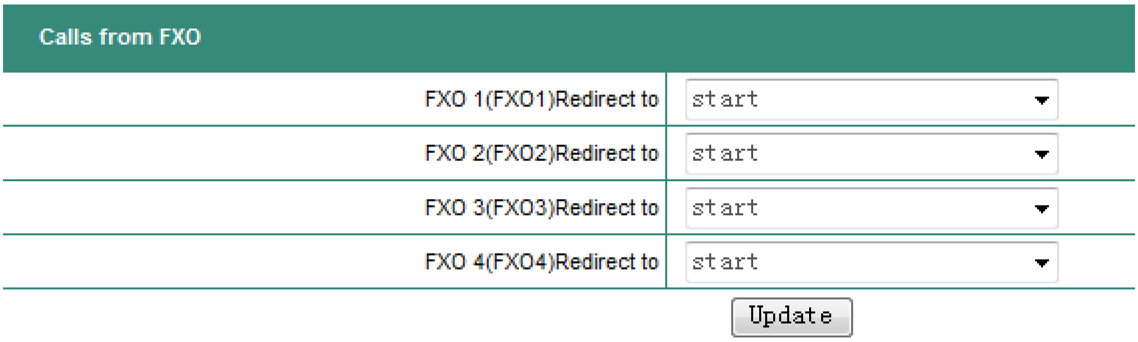
The IP PBX provides two ways of incoming calls; one is from local PSTN through the FXO ports, and the other is from the calls from ITSP. In some case, the ITSP may provide the real PSTN numbers for the IP PBX from the VoIP.

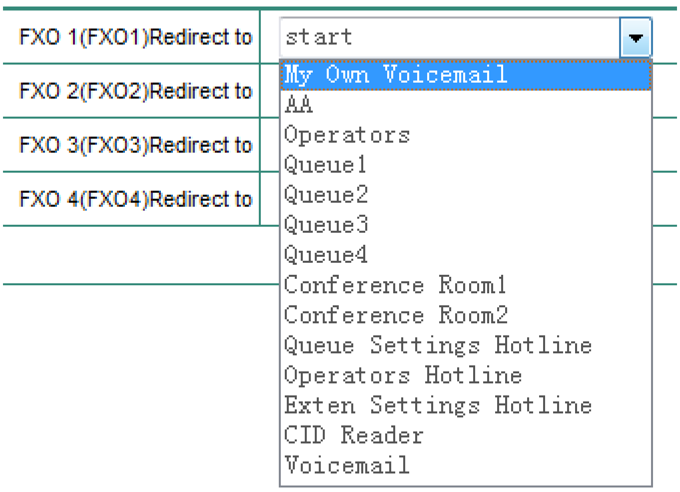




8.3.1 Calls from FXO ports

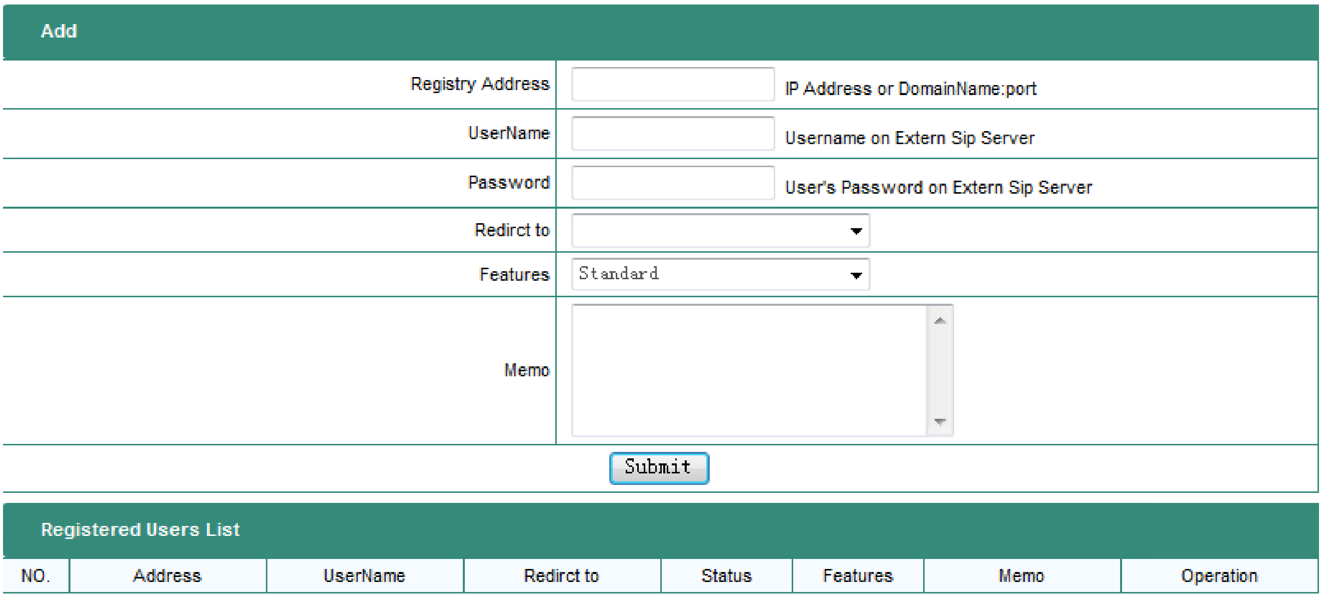
The IP PBX provides 4 FXO ports to allow PSTN incoming calls. For any PSTN incoming calls, you may configure to redirect to either switchboard, extension number, or conference room, etc.

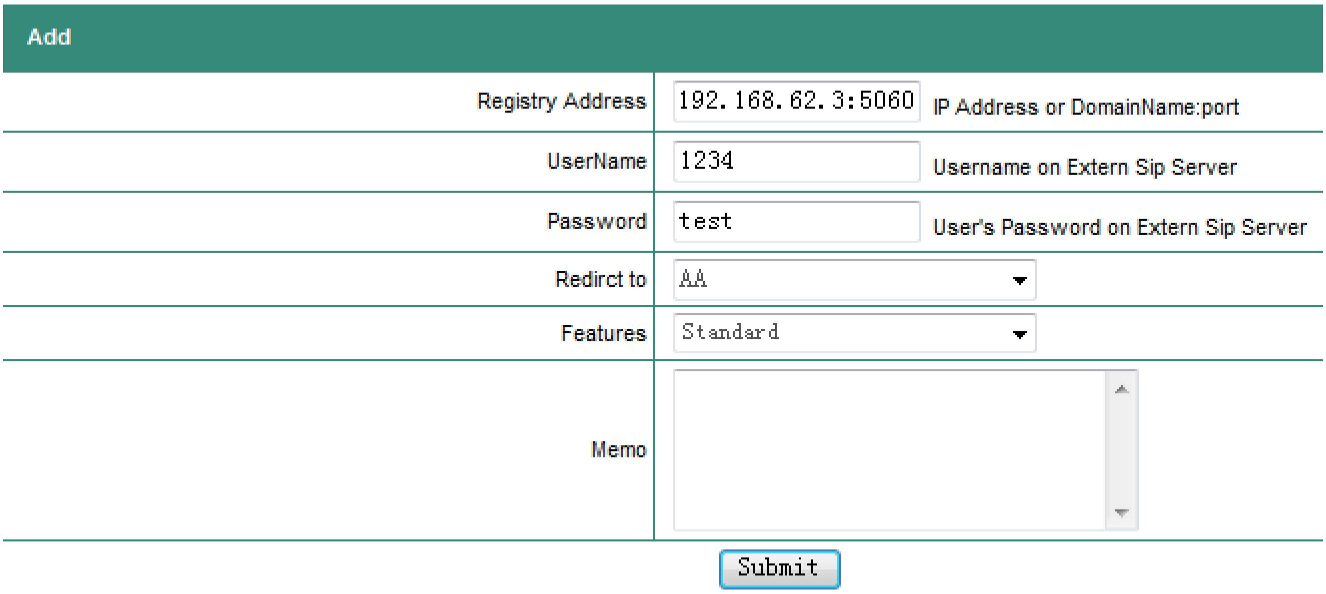




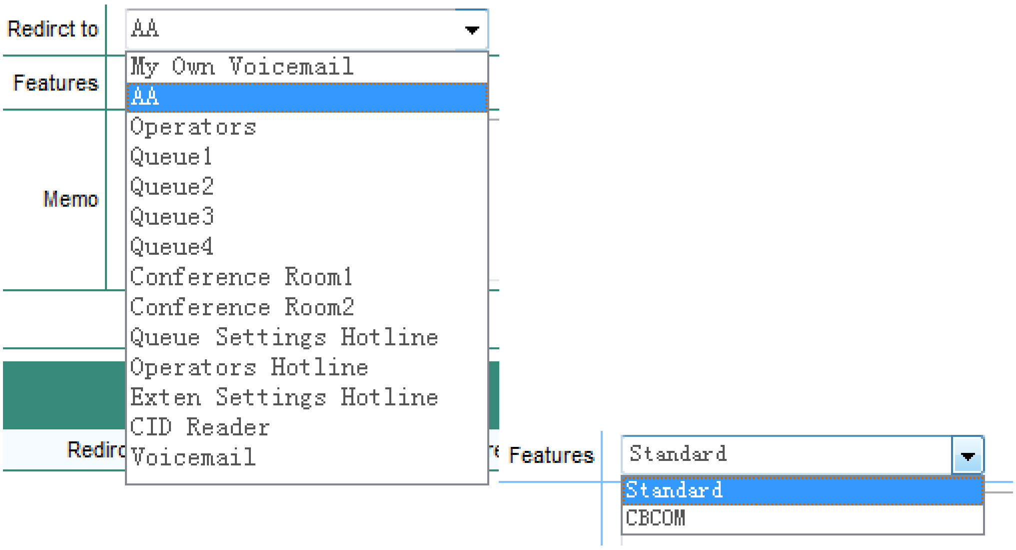
8.3.2 Calls from VoIP

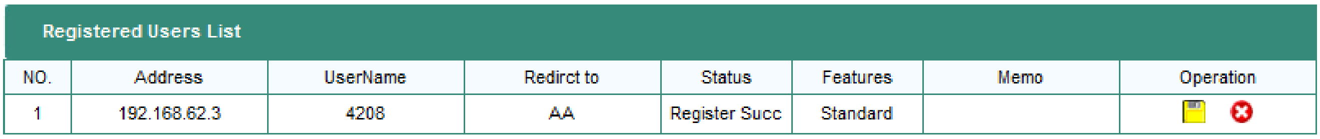
The IP PBX can function as a terminal CPE to register into a SIP server or another IP PBX. For any external VoIP calls, for example, 1234 into this IP PBX, you may redirect to either Auto Attendant, Operator, Conference rooms, etc.





- Registry Address : The SIP Registration Server IP/URL:Port number provided from ITSP, other IP PBX, or SIP Server.
- User Name : The user name provided from ITSP, other IP PBX, or SIP Server.
- Password : The password provided from ITSP, other IP PBX, or SIP Server.
- Redirect To: AA, Operators, Conference rooms, etc.

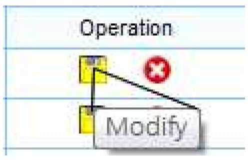




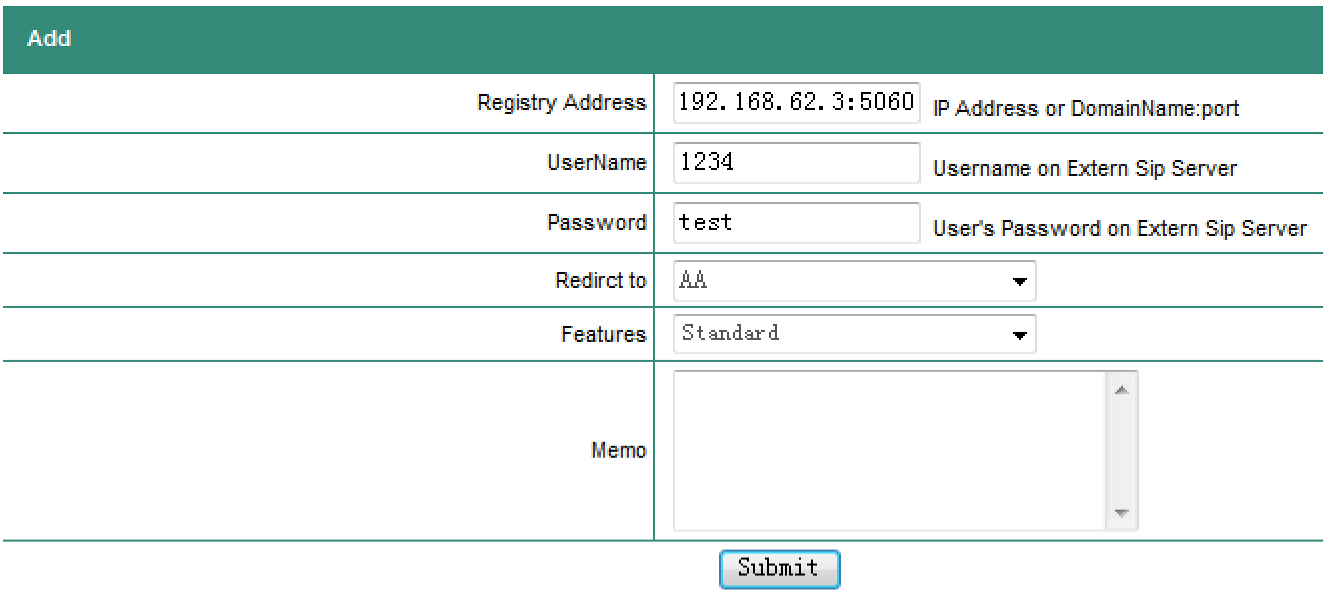
Registered Users List

NO.	Address	UserName	Redirct to	Status	Features	Memo	Operation
1	192.168.62.3	4208	AA	Register Succ	Standard		

The list will display registered users with status after submit.



You may also press the icon to modify the setting.



Add

Registry Address	192.168.62.3:5060	IP Address or DomainName:port
UserName	1234	Username on Extern Sip Server
Password	test	User's Password on Extern Sip Server
Redirct to	AA	
Features	Standard	
Memo		

Submit

You may press the icon to delete this setting.

8.3.3 Access Number

Access Number is used to supplement the function of incoming calls from VoIP SIP Server in the previous section. For some ITSP SIP Servers, one way voice connection may be missing due to protocol incompatibility. In this case, the Access Number can be set up to register into ITSP SIP Server and to redirect VoIP calls from ITSP into the IP PBX.

When your registered, for example, 636346 in some SIP server with one-way voice connection, you may resolve the problem by the following configurations;

(1) In Section 8.3.2 Calls from VoIP

Username: Enter 636346

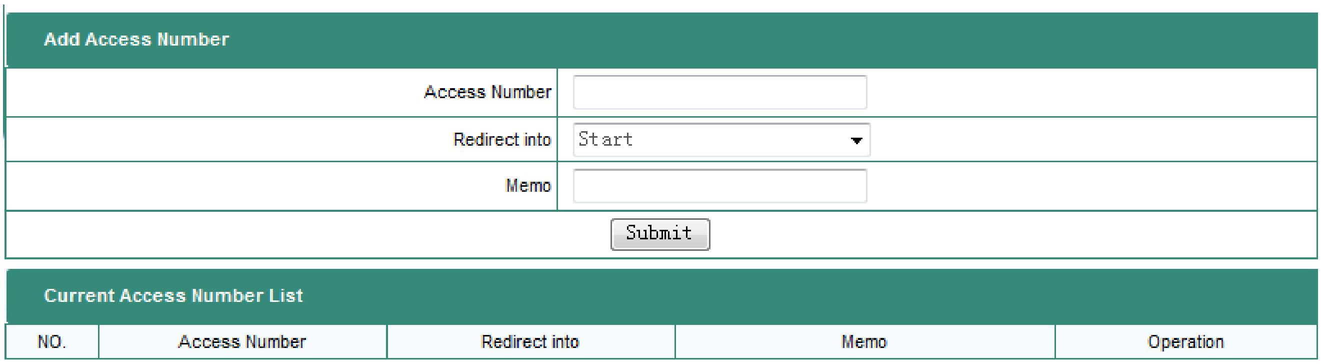
Redirect to: Enter 636346

(2) In Section 8.3.2 Access Number

Access Number: Enter 636346

Redirect to: Enter AA, Operator, or any Holine function call.

Any VoIP calling at 636346 from ITSP will be redirected into the function calls of VOI-9300, such as Auto Attendant, Operators, or Hotlines.



Add Access Number	
Access Number	636346
Redirect into	Start
Memo	FWD accound
	Submit

Access Number : The registered number to ITSP or SIP server.

Redirect into : One of the hotline functional calls.

Memo : For memo descriptions.

Current Access Number List

NO.	Access Number	Redirect into	Memo	Operation
1	636346	Start	FWD accound	

The Access Number will be displayed after pressing Submit button.

8.4 Outgoing Call

The IP PBX provides two kinds of outgoing calls; one is via FXO port to local PSTN lines, and the other is via VoIP calls. The settings are as follows:

Home | Network | System | Incoming Call | Outgoing Call | SwitchBoard | Users | Advanced Setting | CDR | Upgrade&Reboot | Exit

> Outgoing calls Via FXO

> Outgoing calls Via VoIP

8.4.1 Outgoing Calls via FXO ports

The IP PBX is equipped with 4 FXO ports for connection to local PSTN lines. All the extension number can then make outgoing call to local PSTN numbers. You need to assign the group number to each of the FXO port to activate the FXO outgoing PSTN line. If any FXO port is not assigned any group number, the outgoing call for this FXO port will be prohibited and only incoming calls can be accepted. For outgoing calls, you need to specify the group number for calling PSTN numbers, and the IP PBX will select one of the available FXO ports from the group to make an outgoing calls. This make the grouping become flexible in outgoing calls.

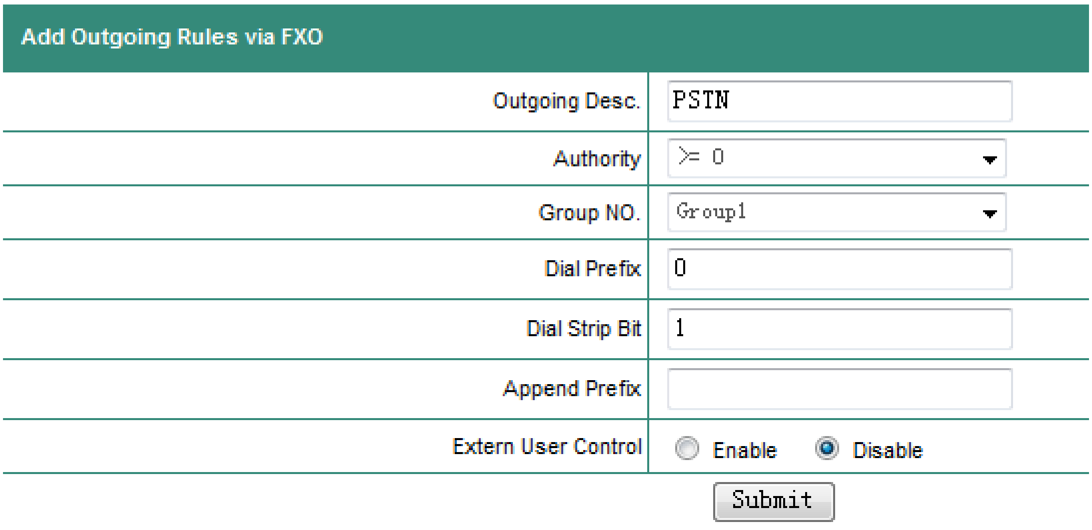
FXO Groups

FXO Group NO.	Group 1	Group 2	Group 3	Group 4	Group 5	Group 6	Group 7	Group 8
FXO1	☑	☐	☐	☐	☐	☐	☐	☐
FXO2	☐	☑	☐	☐	☐	☐	☐	☐
FXO3	☐	☐	☑	☐	☐	☐	☐	☐
FXO4	☐	☐	☐	☑	☐	☐	☐	☐
				Submit				

Having set the rules for local outgoing PSTN calls, the authority is used to restrict the call. For example, for authority >=5, only the user extension number with authority >=5 can make an outgoing call through the FXO ports in this group.

The user extension prefix is used to indicate the group number for making an outgoing call. The delete digits is used to delete the group number with additional prefix before the dialing string. The external use is to allow the external user to use the same rule for redial out to local PSTN numbers, when calling into the IP PBX.

Current Outgoing Rules List Via FXO [Add Outgoing Rules via FXO]



Here, you can configure outgoing call rules for FXO ports.

Outgoing Desc.: Brief description of this rule.

Authority: Used to restrict user's call, only user whose Authority is equal or greater than this value can use this rule to make an outgoing call.

Group NO.: Used to specify the FXO group, for each outgoing call, the IP-PBX selects an available FXO port from this group.

Dial Prefix: The prefix number when user uses this rule to make an outgoing call.

Dial Strip Bit: Indicates how many bits will be stripped from the dialed number.

ppend Prefix: Indicates what added on the dialed number.

Extern User Control: If select "Enable", users from the third party system can use this rule to make an outgoing call.

Example:

When one user press 056789 to apply the outgoing rule, the IP PBX will use one of the FXO port in the group 1, delete 0 (one digit), The final dialing string will be 56789 to the PSTN line.

8.4.2 Outgoing Calls via VoIP

SIP Trunk is a logical pipeline connecting IP-PBX and SIP ITSP. By SIP Trunk, extension users in this system can communicate with users in the third-party VoIP system.

The VOI-9300 IP-PBX supports outgoing call through SIP Trunk, connecting with the third-party VoIP system, which means, we can either call users in other system, or call PSTN phone through other system.

There are two ways in connecting with the third party system:

1) The third party system provides an account and a password; we use this account and a password to authenticate outgoing call. In this case, we need to fill “UserName” and “Password” fields when we add an outgoing rule via VoIP.

2) The third party system set us as a trust host, do not authenticate our outgoing call but accept it directly. In this case, no need to fill “UserName” and “Password”.

When using the first way, the caller number of our outgoing call is the account the other system provides.

When using the second way, the caller number of our outgoing call is the transformed number of our extension number. The transformation depends on the values of “User Prifix” and “User Strip Bit” fields.

Current Outgoing Rules List via VOIP [Add Outgoing Rules via VOIP]

NO.	Authority	IP Address	UserName	Password	User Prefix	User Strip Bit	Dial Prefix	Dial Strip Bit	Append Prefix	Features	Operation

Current Outgoing Rules List via VOIP [Add Outgoing Rules via VOIP]

Add Outgoing rules via VOIP	
Description	VOIP out
Authority	>= 0
IP/Domain Address	192.168.62.3:5060
UserName	1234
Password	test
Dial Prefix	9 TIPS:Dial Prefix can not be in FXO,or system will not work normally.
Dial Strip Bit	1
Append Prefix	
User Prefix	
User Strip Bit	0
Feature	Standard
Outside User Limit	Enable / Disable

Submit

Description: Brief description of this rule.

Authority: Used to restrict user's call, only user whose Authority is equal or greater than this value can use this rule to make an outgoing call.

IP/Domain Address: The IP address of the third party system.

UserName: The account provided by the third party system.

Password: Password corresponding to the account.

Dial Prefix: The prefix number when user uses this rule to make an outgoing call.

Dial Strip Bit: Indicates how many bits will be stripped from the dialed number.

Append Prefix: Indicates what added on the dialed number.

User Prefix: Indicates what added on the caller number.

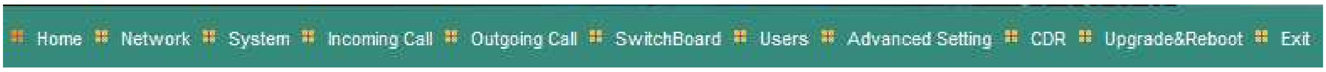
User Strip Bit: Indicates how many bits will be stripped from the caller number.

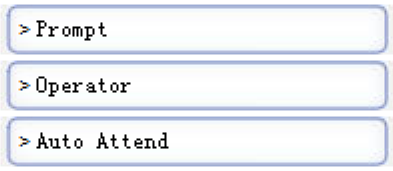
Feature: If encrypt SIP signaling or not.

Outside User Limit: If select "Enable", users from the third party system can use this rule to make an outgoing call.

8.5 SwitchBoard (Auto Attendant Settings)

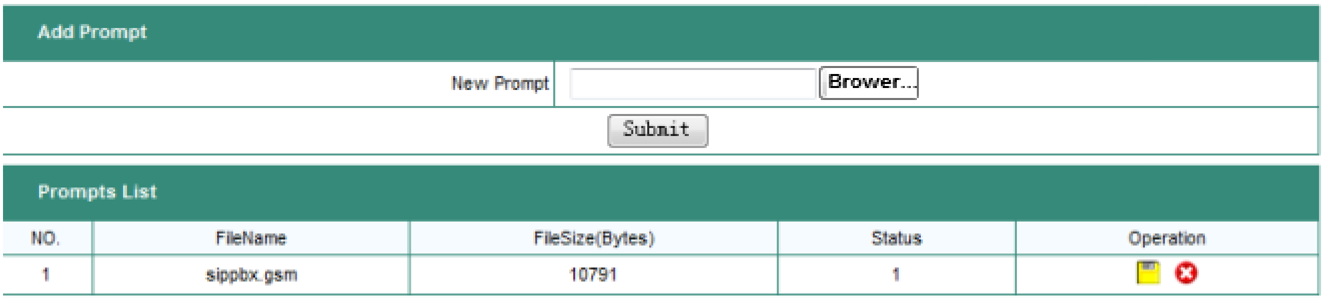
The IP PBX provides an Auto Attendant for user to call. When receiving an incoming call, the auto attendant will play a welcome prompt message or call another extension number. The settings include welcome prompt message, operator, and auto attend.





8.5.1 Prompt Message

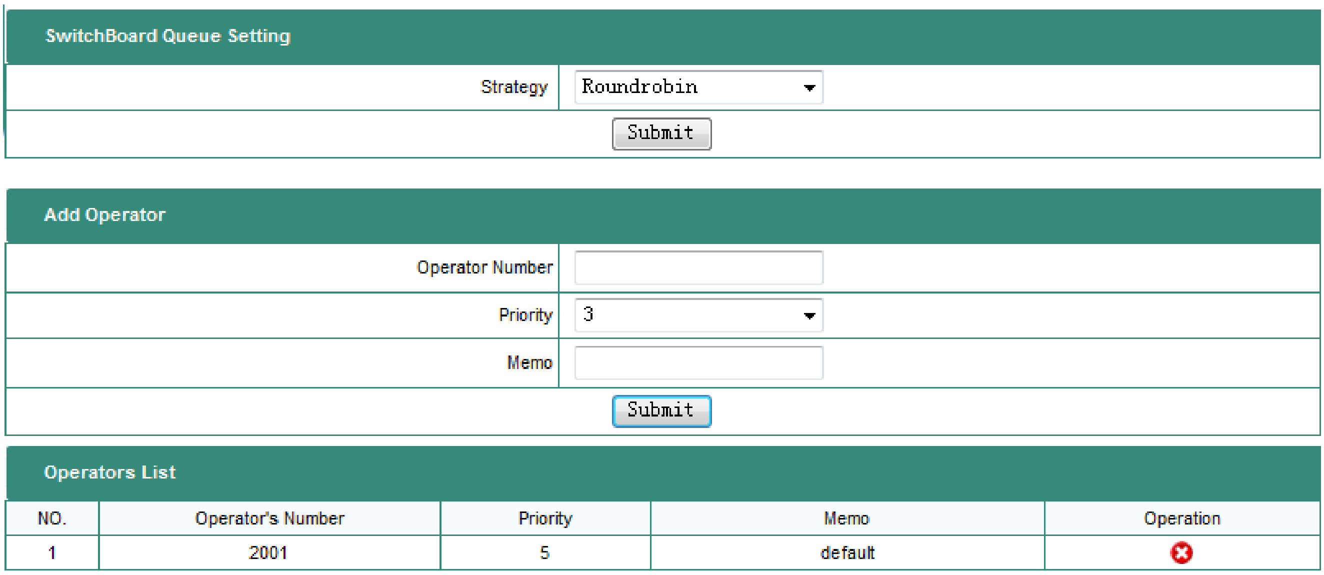
You may choose the prompt message for the auto attendant to play while receiving incoming calls. You need to upload the messages to the prompt list for setting choices.



Note: The prompt file must be GSM format.

8.5.2 Operator Settings

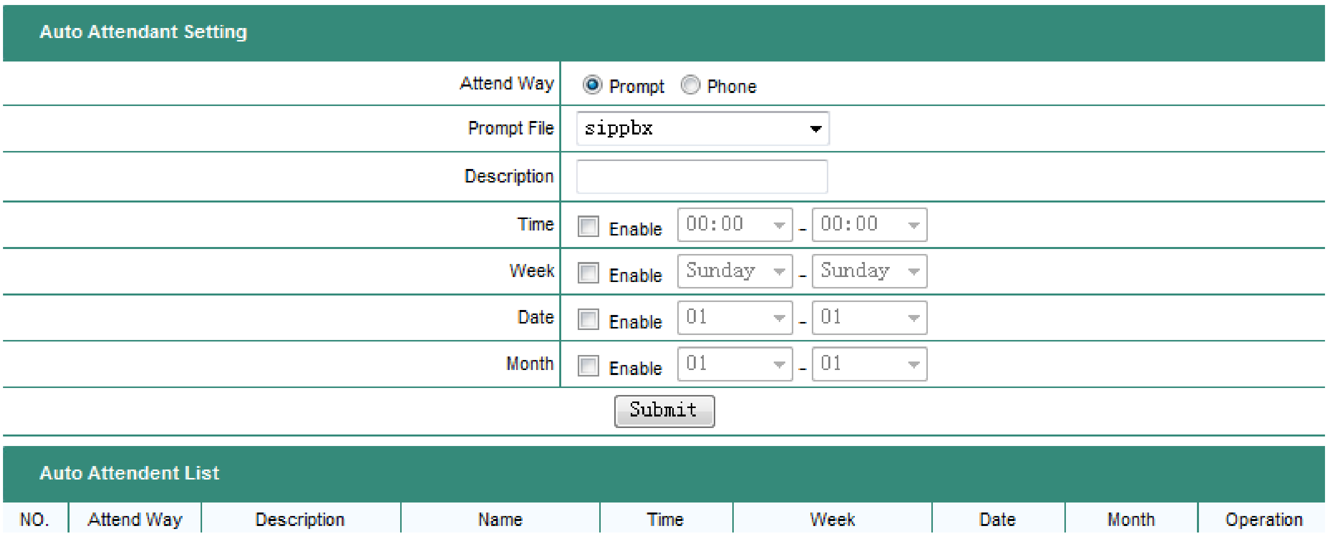
When a transfer call can not be transferred to the desired extension number, the call will be transferred to operator. The operator can be any extension or PSTN number with assigned priority, and the call will be forwarded by priority.



8.5.3 Auto Attendant

Auto Attendant will handle all the incoming calls when no one can answer in the company. There are two ways for attendant. The first one is Prompt with playing answering prompt messages, and the second one is Phone with direct transfer to the phone number.

- Prompt : An answering prompt message will be played to answer the incoming call. Make sure the answering message is uploaded and chosen. The IP PBX provides many options for time durations. If the time durations are not set, the handling incoming call will be always effective.
- Phone : All the incoming calls will be transferred to the preset phone number. The phone number can be either extension number or PSTN number.



- Attend Way : You can select the attend way is prompt or phone.
- Promot File : If you selected the prompt way . You can choose one prompt file to be use .

- Number : If you select the Number way . The extension number will answer the call.
- Time/Week/Date/Month: The time of these time field set (Time & Week & Date & Month), the extension user will answer the call.

8.6 Users Management

This section describes the account opening, closing, and management. The extension number (also known as user name) must not exceed 32 digits. For easy management, it is recommended to assign different prefix number for different departments.

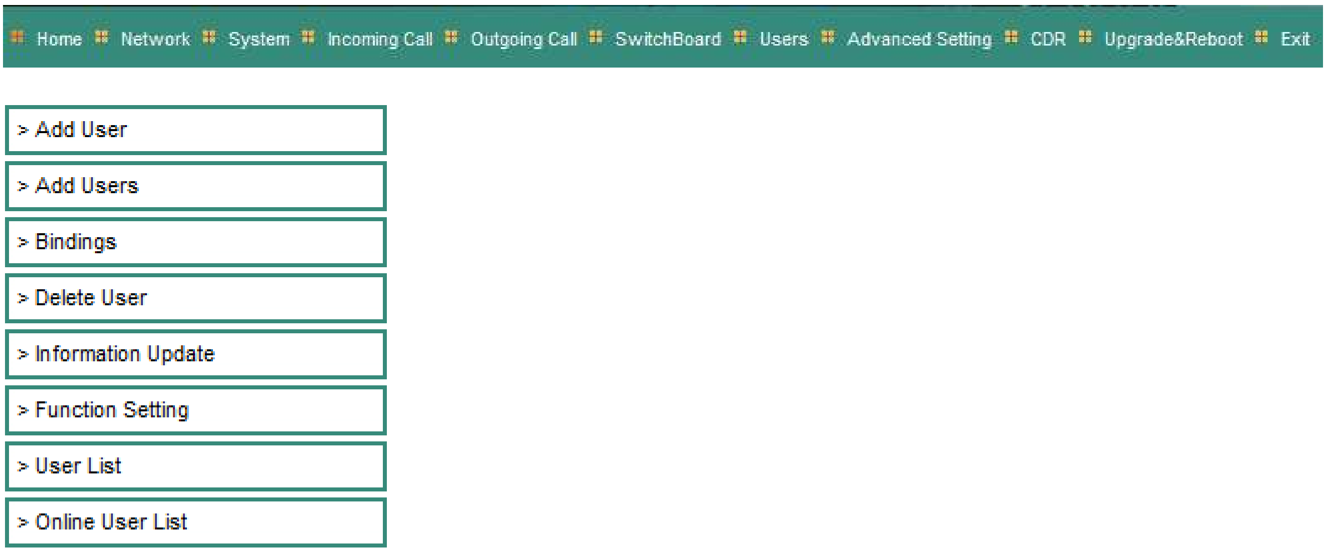
A user number should have the following;

(1) **User name:** The user name is a string of number digits with length less or equal to 32. For easy management, it is recommended to assign different prefix for different departments.

(2) **Password:** Every user name will have its own password with length less than or equal to 32 alpha-numeric digits. The CPE VoIP phone must use the same password for authentications.

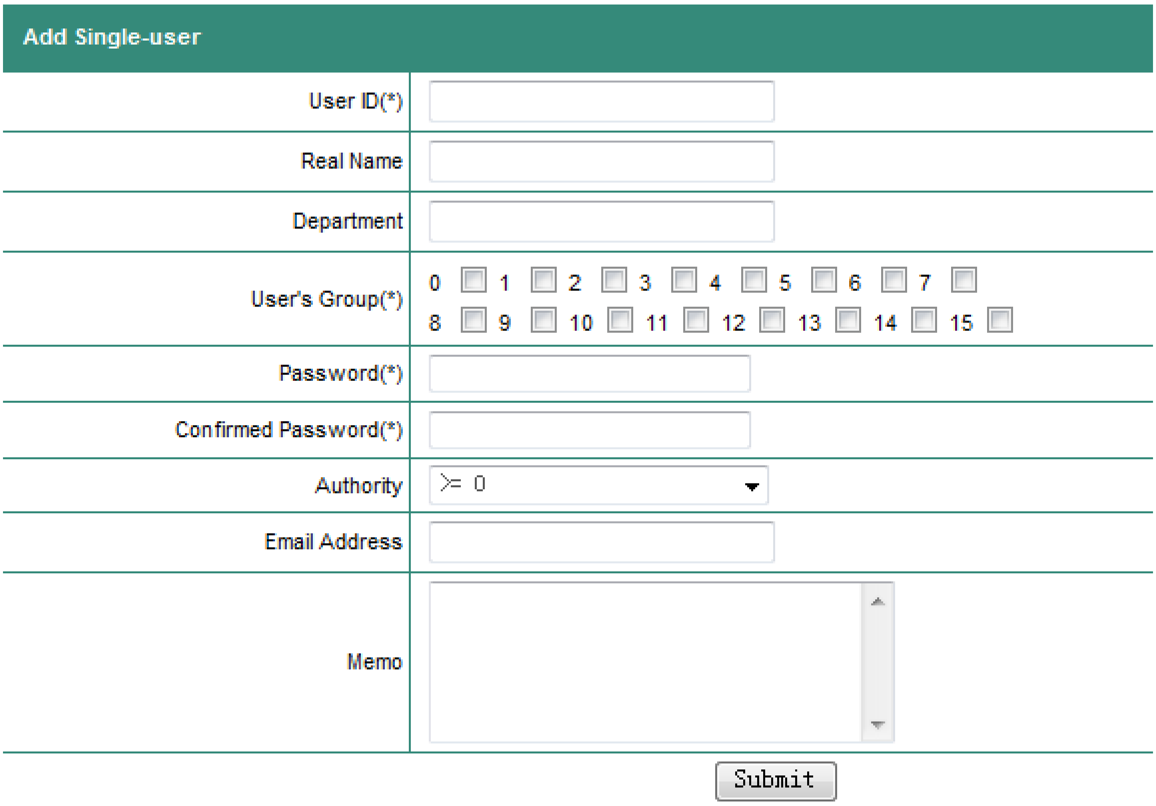
(3) **Call Authority:** from 0-15. This is used to restrict the call authority. The user authority must be equal or greater than the predefined authority to make the defined outgoing calls.

(4) **User Group:** Each user name can belong to one or multi-groups. User can answer incoming calls for another user within the same group by pressing "***8**". There are 16 groups can be assigned.



8.6.1 Single User Account Opening → Add User

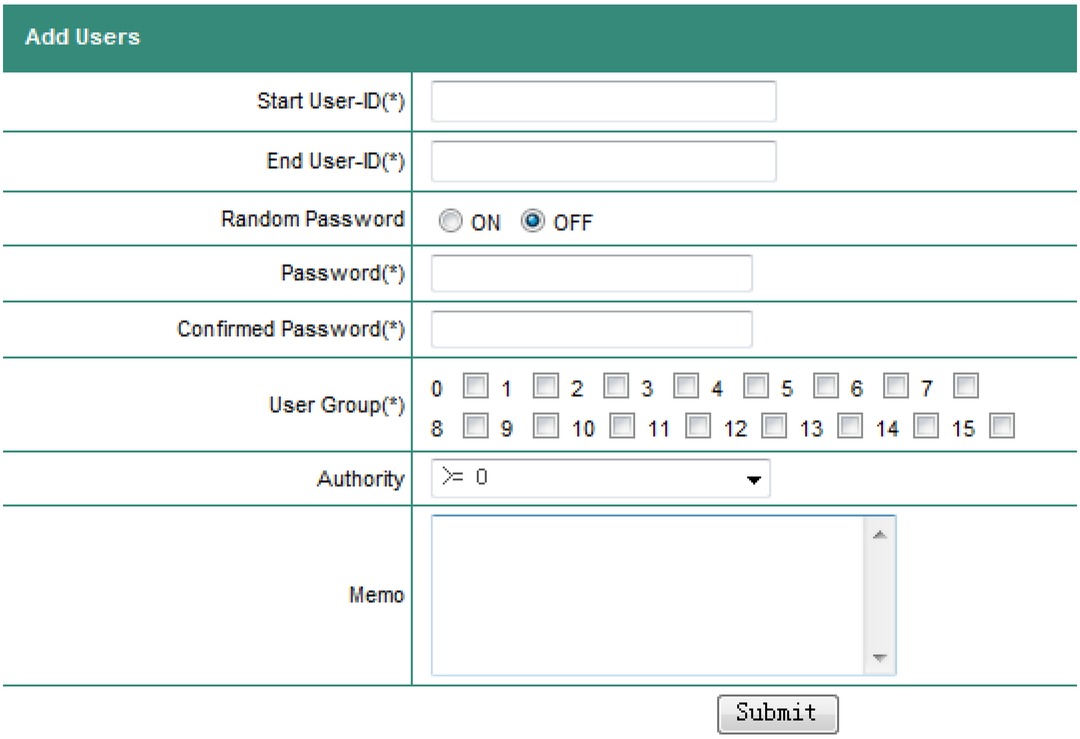
You may add single user account in this section by entering user name, password, groups, and call authority. Remember the length should not exceed 32.



Note: when the number of registered users reach the limit, the function of add user will not work.

8.6.2 Group Users Account Opening→ Add Users

You may create group users accounts and assign group numbers with priority to all the accounts. You may also assign password to each account or generated by IP PBX. After settings, you may display all the created accounts.



8.6.3 Bindings

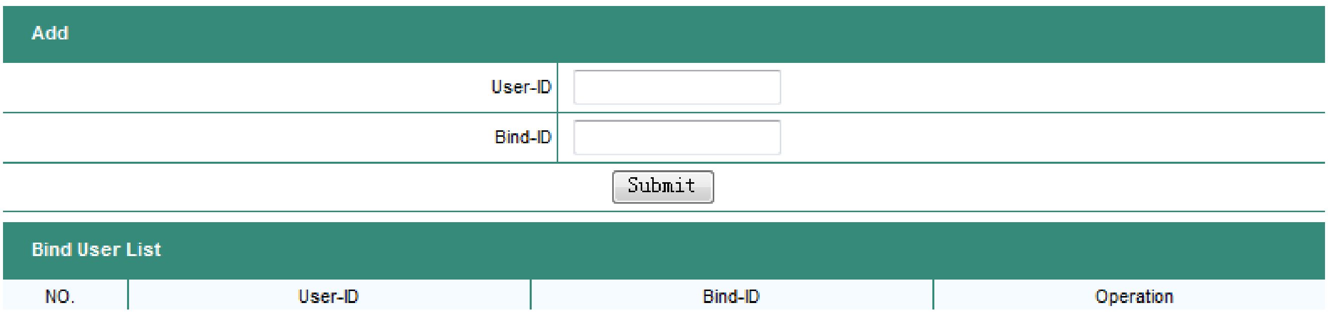
Extension binding is used to bind many extension numbers together as a group. When calling to one of the extension, the other binding extension will ring as well. This is also referred to as Group Ringing.

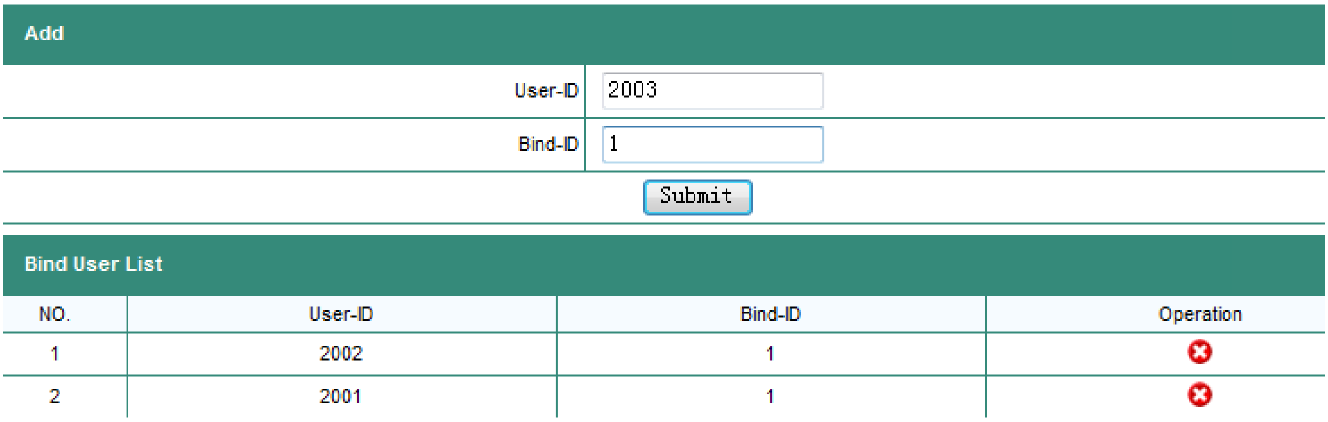
(1) User-ID: To enter extension number.

(2) Bind-ID: To enter the binding ID. The extension number with same binding ID will belong to the same binding group.

(3) The binding list will display the extension numbers.

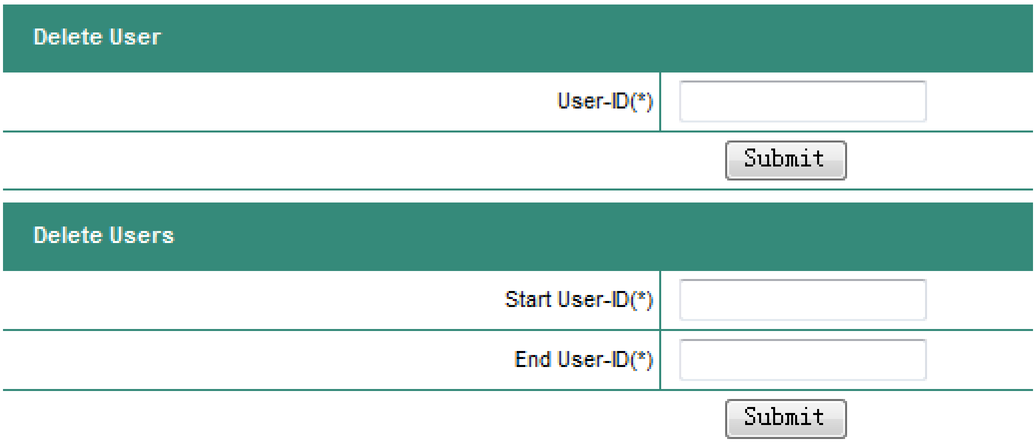
Note that one extension number can belong to multi-binding groups.





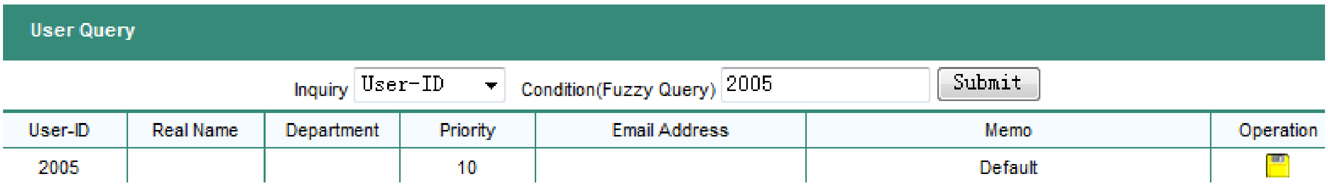
8.6.4 Delete User Accounts

There are two ways for user account deletion; one is for single account deletion, and the other is for group account deletions. For single account deletion, you need only to enter the extension number. For group account deletions, you need to enter a range of extension numbers. The IP PBX will delete all the user information for the extension within the range. Note that the information will not be recovered once deleted.

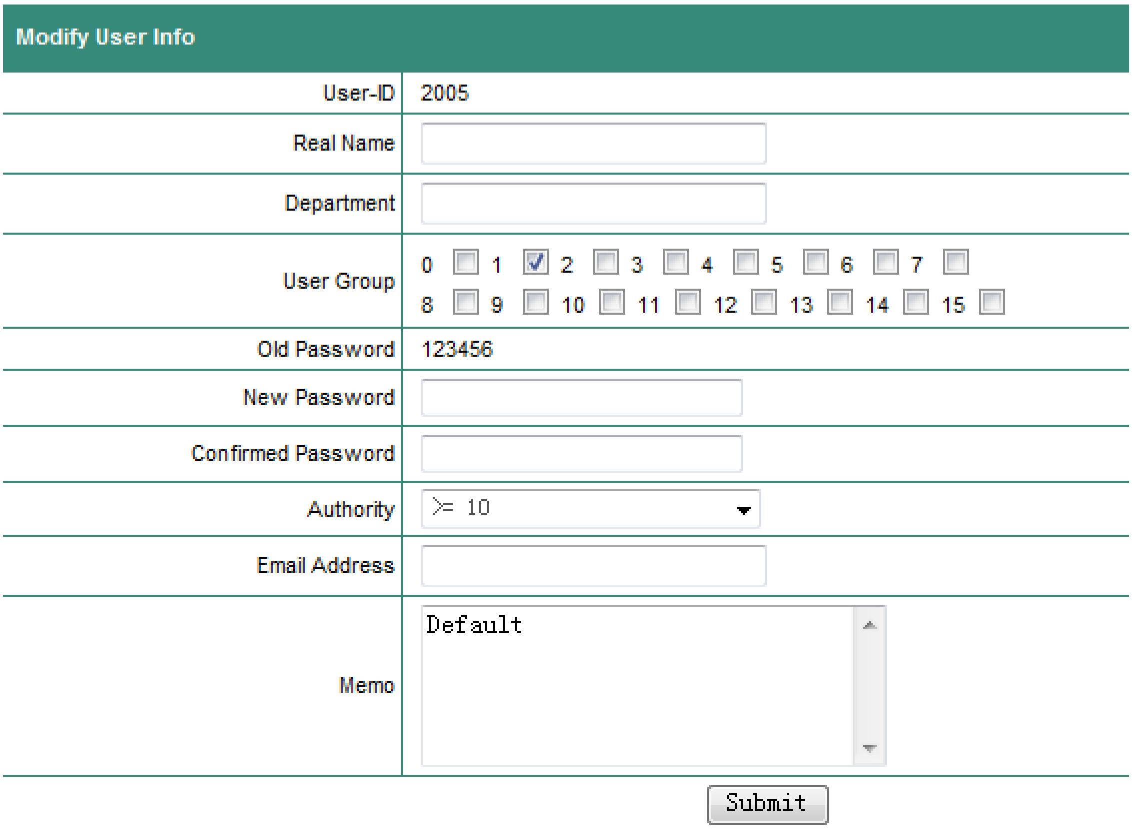


8.6.5 User Information Update

In this section, you may modify password, name, department, authority and group.



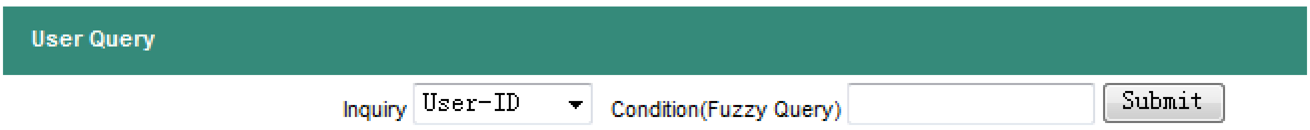
- Inquiry : Choose one User-ID or User Name for search .
- Condition (Fuzzy Query) : Enter the number or user name for search.

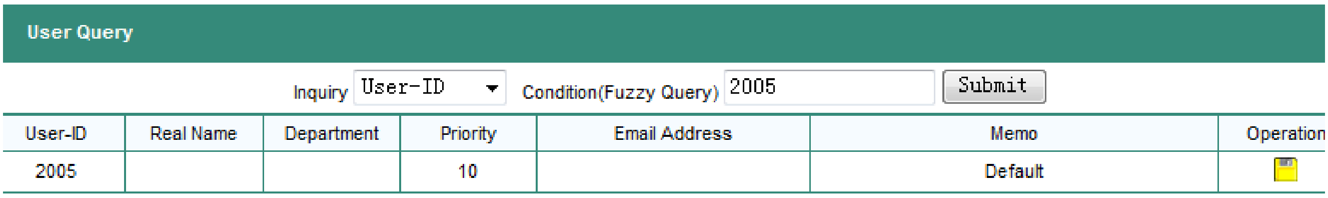


This page will show the Modify User settings.

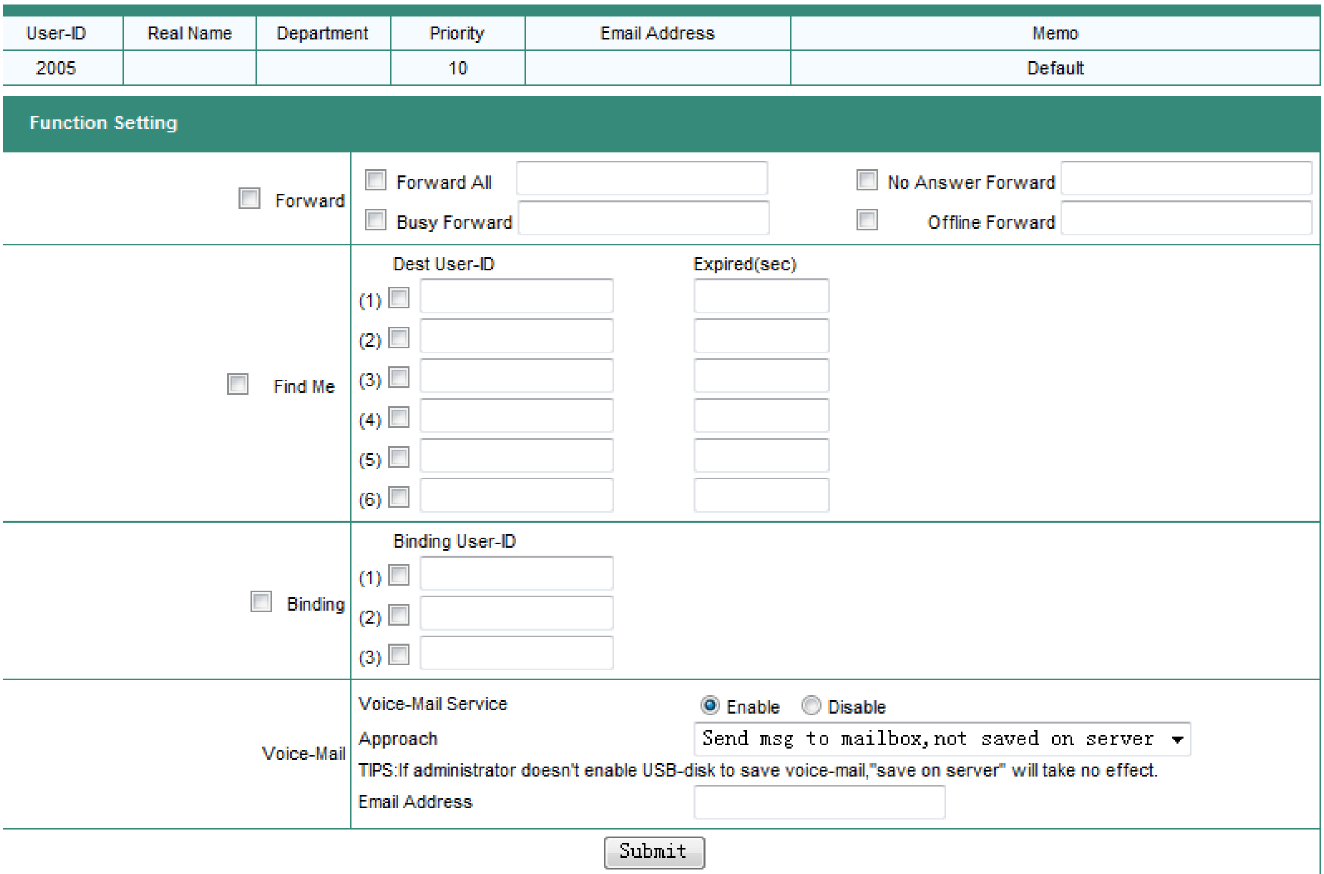
8.6.6 Function Settings

There are three functions provided as follows;





- Inquiry : Choose one User-ID or User Name for search .
- Condition(Fuzzy Query) : Enter the number or user name for search.



This page shows the Modify User's Function setting.

(1) Call Forward

- Unconditional Forward: Any incoming call to this extension will be forwarded directly to the set extension number.
- No-Response Forward: When no answer in one minute, the call will be transferred to the set extension number.
- Busy Forward: When busy, the new incoming calls will be transferred to the set extension number.
- Off-Line Forward: When the extension number is off-line (unregistered), all the incoming call will be transferred directly to the set extension number.

(2) Find Me

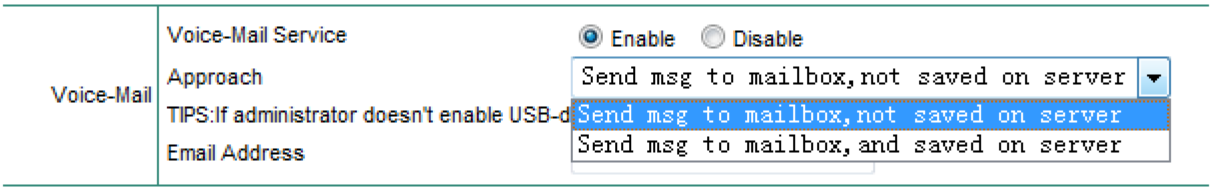
When one user is set for “Find Me”, the call will be transfer to the first extension number when no answer for the incoming call. If no answer again, the call will be transferred to the next extension number until the call is answered. The maximum is 6 extensions numbers. After that, the call will be disconnected.

(3) Telephone Binding

The telephone binding is to bind the extension number with a PSTN number. For incoming call to the extension number, the PSTN phone number will ring simultaneously.

(4) Voice Mail

You may enable the voice mail for extension numbers.



You may to choose one for Approach to store the Voice-mail method .

1) Send msg to mailbox, not saved on server : It will only send the message to the Email Address without saving a copy on the IP PBX.
2) Send msg to mailbox, and saved on server : It will send the message to Email address with a copy saved on the IP PBX.

Note :

(1) Every user may select only one at a time out of the three functions.

(2) You need to check on the icon to select the desired function.

(3) The dialing number must follow the rules for outgoing calls.

8.6.7 User List

There are two kinds of list; one for all the users, and the other for the on-line registered users.

All Users List

NO.	User-ID	Real Name	Password	Group NO.	Authority	Call Features	Operation
1	2001		123456	1	10	Binding	
2	2002		123456	1	10	Binding	
3	2003		123456	1	10	Normal	
4	2004		123456	1	10	Normal	
5	2005		123456	1	10	Normal	
6	2006		123456	1	10	Normal	
7	2007		123456	1	10	Normal	
8	2008		123456	1	10	Normal	
9	2009		123456	1	10	Normal	
10	2010		123456	1	10	Normal	

The list will show the current setting status for each user as follows;

(1) Normal

(2) Unconditional Transfer

(3) Call Transfer; including busy, off-line, and no answer transfer.

(4) Find Me

(5) Telephone Binding

You may modify and update all the user settings directly from the entry of the list.

8.6.8 On-Line User List

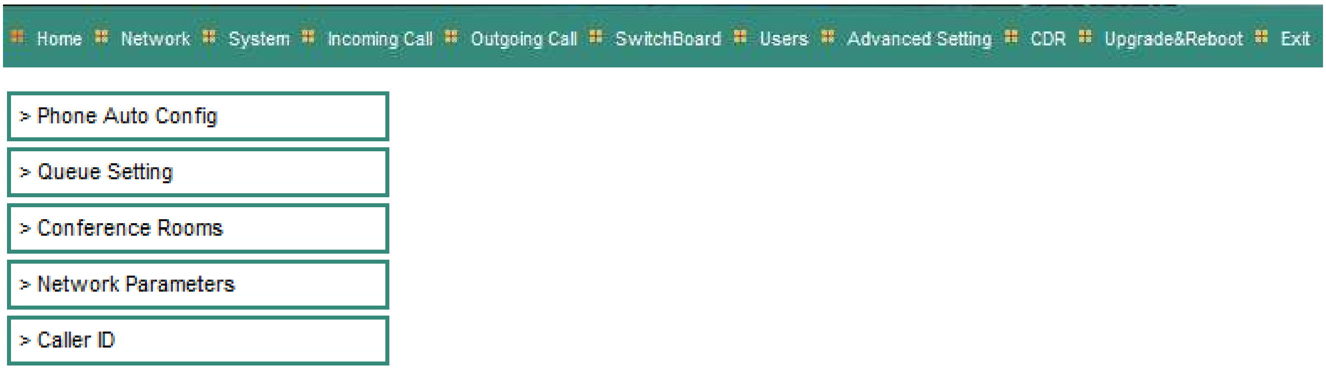
The on-line users are for the current registered users.

Current OnLine List

NO.	User-ID	Real Name	Department	UA Info	IP Address
1	201	201	201	VP306o (708240)	192.168.62.102
2	202	202	202	PHONE 7000C	192.168.62.182
3	203	203	203	VS211	67.170.238.55
4	208	208	208	VP306 (709050)	192.168.62.146

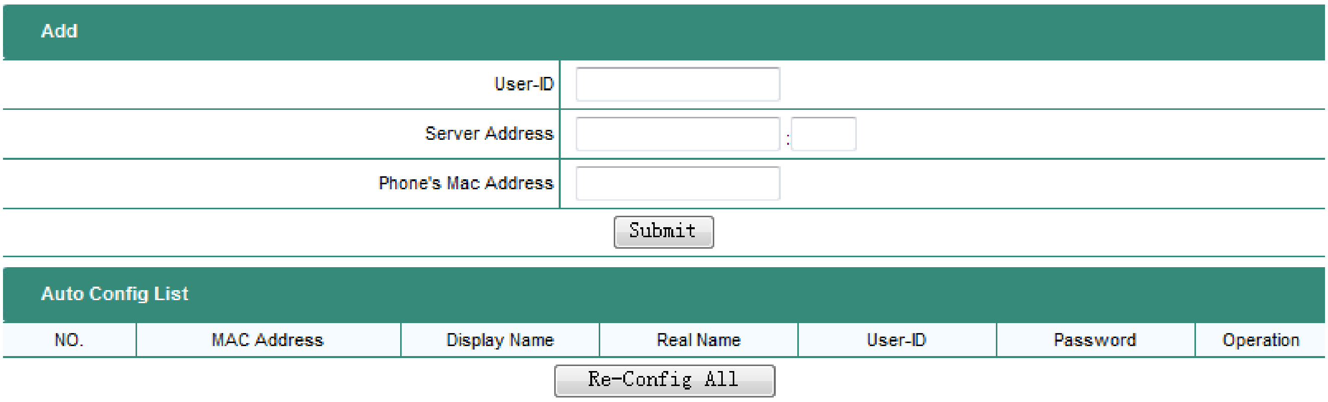
8.7 Advanced Settings

The advanced Settings cover some call waiting, voice conference rooms, and IVR upload process.

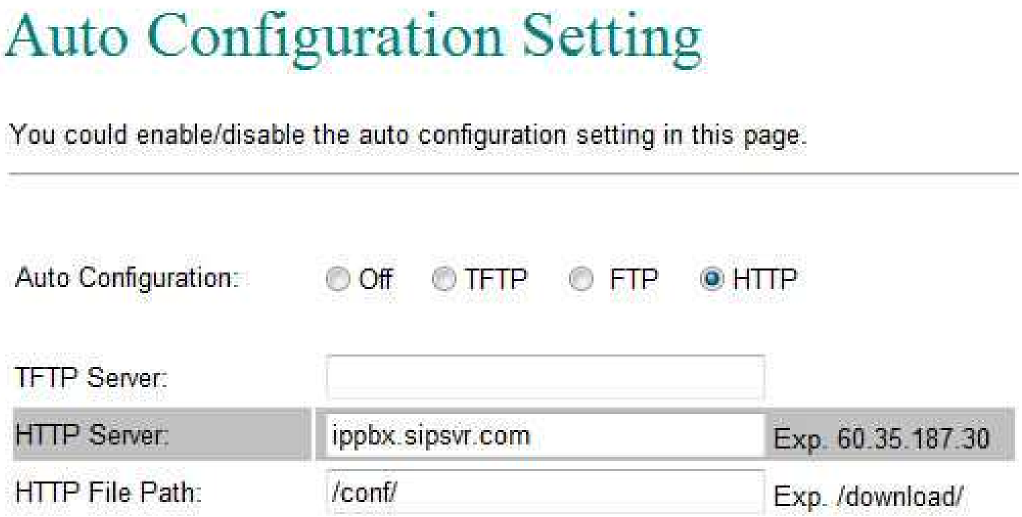


8.7.1 Phone Auto Config

The IP-PBX system can configure automatically the IP Phone with auto-configuration capability by HTTP protocol.

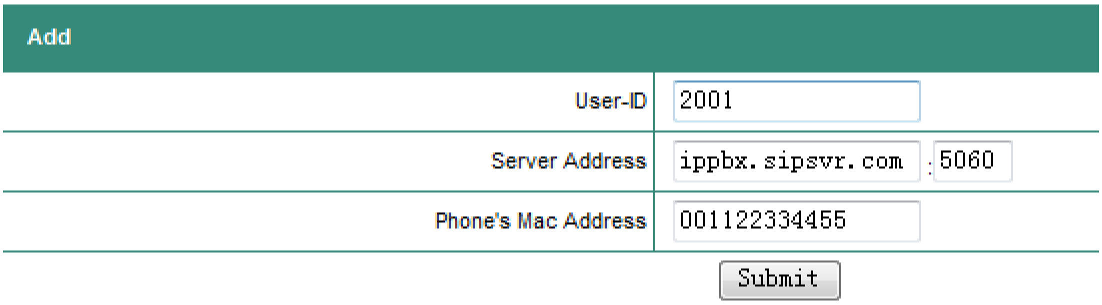


Firstly, you need to set up auto configuration for the IP Phone. The following example is for reference only.



- Auto Configuration: Select the HTTP protocol.
- HTTP Server: IP address or URL of the IP-PBX.
- HTTP File Path: Configuration file's saved path, it must be "/conf/".

Secondly, you need to configure for the IP PBX.



- User-ID : The extension number of the telephone.
- Server Address : The IP address or URL & port of the IP PBX.
- Phone's Mac Address : The Mac address of the IP Phone.

Auto Config List

NO.	MAC Address	Display Name	Real Name	User-ID	Password	Operation
1	001122334455	2001	2001	2001	123456	

Re-Config All

8.7.2 Queue Settings

Call waiting is to queue all the incoming calls and to assign based on rules to the desired extension numbers. If all the available extensions are busy, the system will play the waiting music for the queuing incoming calls. Once available, the system will connect to the desired extensions.

Current Queue List

NO.	Queue Extension	Queue Passwd	Strategy	Queue Length	Queue Desc.	Operation
1	1701	123456	Roundrobin	10		
2	1702	123456	Roundrobin	10		
3	1703	123456	Roundrobin	10		
4	1704	123456	Roundrobin	10		

Queue Info Setting

Queue Extension	1701
Queue Password	123456
New Password	
Confirmed Password	
Strategy	Roundrobin
Queue Length	10
Queue Desc.	

Submit

Roundrobin

- ring all
- Roundrobin
- leastrecent
- fewestcalls
- random

Add User as Agent

Queue Extension	1701
User-ID	
Priority	5
Memo	

Submit

Current Queue Agents List

NO.	Extension	Agent User-ID	Priority	Memo	Operation

The IP PBX provides 4 call waiting queues. The waiting queues can be set to a simple calling center, and the user may define individual waiting queues for its own purpose.

The waiting queues can be configured by the IE Web browser to define the functions as follows;

(1) Update new incoming calls;

(2) Setting the password

(3) Set the desired extension

(4) Set the waiting time length

(5) Add/Delete waiting number

Functions:

1） Update the call waiting list

2） Modify the call waiting information

The call waiting password will be required when the call is entering the waiting queue.

For each waiting call, the IP PBX may provide one of the following call assignment:

(1) Round-Robin: for any incoming call, the call will be forwarded to the next extension number.

(2) Random: The incoming call will be randomly assigned to the extension numbers.

(3) The least-answer: the new incoming call will be transferred to the extension with the least answering.

(4) The longest idle: the new incoming call will be transferred to the extension with the longest idle time.

(5) All Ringing: the new incoming call will ring all the extension numbers. Anyone pick up the phone will answer the call. Waiting length means the maximum numbers for call waiting.

3） To Add/Delete the waiting list

The user may add/delete the call waiting queues.

8.7.3 Voice Conference Rooms

The IP PBX provides two standard voice conference rooms. The details are as follow;

Current Conference Room List

NO.	Conf. NO.	Conf. Passwd	Admin Passwd	Extension	Max. Number	Status	Memo.	Operation
1	001	123456	654321	1650	10	Idle		
2	002	123456	654321	1651	10	Idle		

(1) Conference Room Password

When a user calls to the conference room, he will be required to enter the password, or the call will be denied. It is recommended to set the password for the conference room.

Please refer to the conference room setting for password.

(2) Maximum attendants of conference room

If set at 0, it has no limitation. Please refer to the conference room settings.

(3) Web Control of Conference Room

Current Attendance List					
NO.	Conf. NO.	Attendance User-ID	Attendance Name	Status	Operation
1	C001	208	208	Listen&Talk	
2	C001	4208	4208	Listen&Talk	

Current Attendance List					
NO.	Conf. NO.	Attendance User-ID	Attendance Name	Status	Operation
1	C001	208	208	Listen&Talk	
2	C001	4208	4208	Listen&Talk	

Listen&Talk

Current Attendance List					
NO.	Conf. NO.	Attendance User-ID	Attendance Name	Status	Operation
1	C001	208	208	Listen&Talk	
2	C001	4208	4208	Listen&Talk	

Listen

Current Attendance List					
NO.	Conf. NO.	Attendance User-ID	Attendance Name	Status	Operation
1	C001	208	208	Listen&Talk	
2	C001	4208	4208	Listen&Talk	

Kick out

There are three icons for the three operation functions;

(1) The extension numbers are in conversation dialog status

(2) The extension is in monitoring status.

(3) Kick out the extension number. The extension will be forced out in 5 minutes.)

(4) IVR Control of Conference Room

The user after entering the conference room may press * key for an IVR playback. The user may interact with the IVR messages.

General User Extension:

1) Make oneself mute. The user voice will not be heard in the conference room. To let one's voice heard in the conference, just repeat the same procedure as mute. Press * key for an IVR playback, then press 1 key to resume conference attending.

2) Make oneself manager. The user needs to enter the manager password. Please refer to conference room settings for the manager password.

Manager has the following functions:

1, Make oneself mute. The user voice will not be heard in the conference room.

2. Make the current conference room in lock, and any user trying to enter the conference room will hear the prompt message “**Conference Room is in Lock**”, and get disconnected. To unlock the conference room, simply repeat the same procedure in lock. That means to press * first, and then press 2 to unlock the current conference room.

3. To kick out all the general users. (Note that all managers will not be kicked out.).

4. To mute all the general users. All the general users in the conference room will not be able to speak but to listen only.

5. To disable mute all the general users. All the general users in the conference room will be able to speak and listen. (Note that if the user mutes oneself he needs to disable mute by himself to speak in the conference room. The manager can not disable mute for the user.)

6. Conference Invitation.

If you want to invite an outside user to join the conference room, you may follow the step by the IVR message operations. This outside user can be registered in the IP PBX or the PSTN number.

Note: Conference invitation is making an invitation only and do not acknowledge if invitation is successful or not. If not successful, the invitation will repeat after 60 seconds. If not successful after three times of conference invitation, the invitation will stop without any notice.

(5) Support IVR Conference Invitation

You may make conference invitation by IVR. Please refer to the IVR conference room control.

(6) Modify Voice conference room service number

You may change the conference room service number if the default number is not desired.

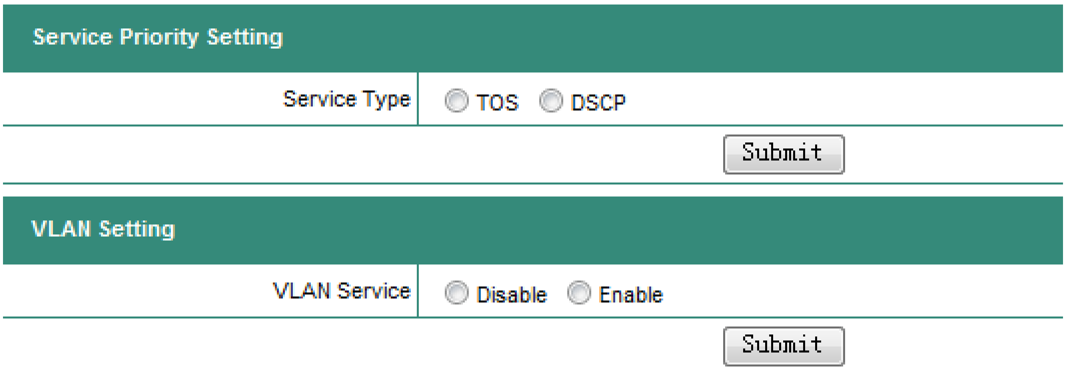
The conference room settings are as in the following page.

Conf. Room Setting		
Conf. NO.	C001	
Conf. Passwd	123456	If not set,you can enter the conf. without any Passwd.
Admin Passwd	654321	If not set,you can not get the manage authority.
Max. Number	10	TIPS:NULL or 0, there is NO restriction on MAX. Number
Memo		
	Submit	

The conference room password is used to enter the conference room. The maximum capacity is for the maximum number of users in the room.

8.7.4 Network Parameter

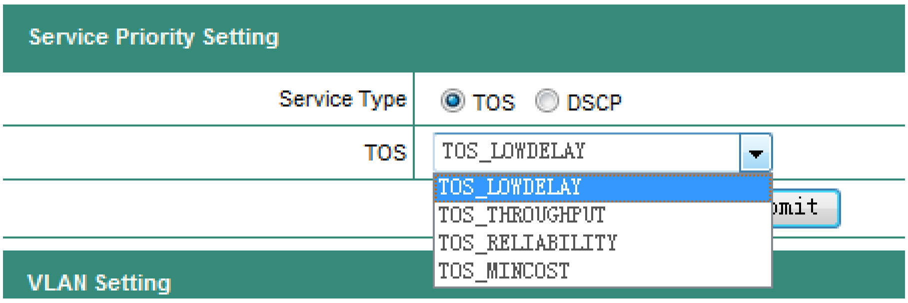
The network parameters allow to enable/disable IP network functions.



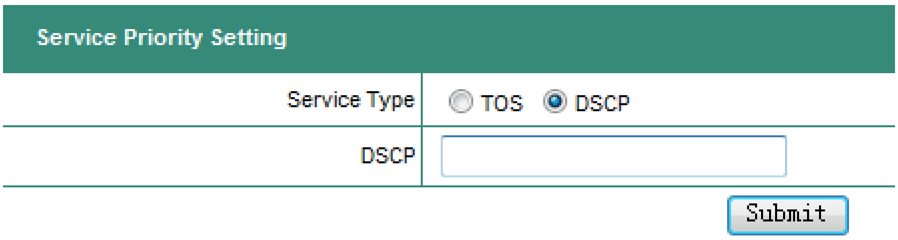
Support Voice Priority Tag (TOS/DSCP)

TOS (Type of Service) and DSCP (Differentiated Services Code Point) are used to set different priority to data packet and data flow, thus enabling QoS in IP communication.

TOS has four values; they are minimum delay (TOS_LOWDELAY), max throughput (TOS_THROUGHPUT), max reliability (TOS_RELIABILITY) and minimum cost (TOS_MINCOST). TOS can be one of them in practice.

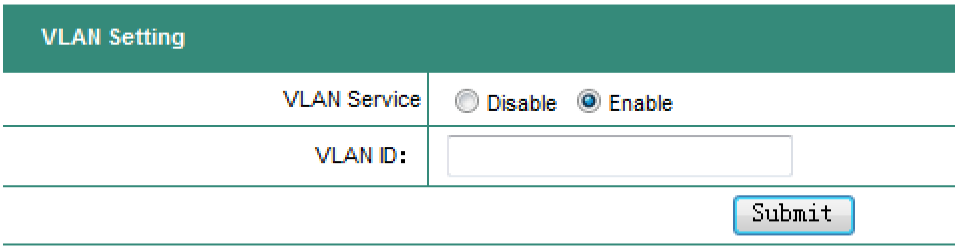


DSCP field is a superset of TOS, its definition is backward-compatible with TOS, and its value can be from 0 to 63, with 0 for minimum priority, 63 for max priority.



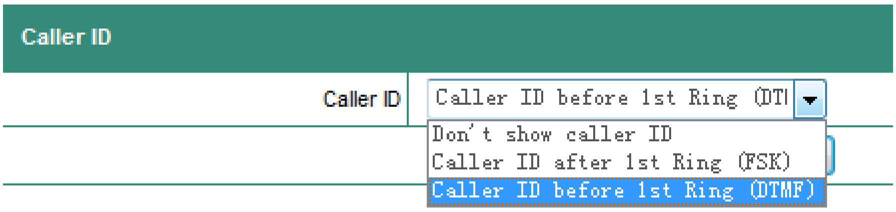
Support Tag-based VLAN 802.1P/802.1Q.

VLAN 802.1P/802.1Q are for local area network standard recommended by IEEE, which can partition network users from different physical locations into several logical subnets. The VOI-9300 IP-PBX supports this standard, and can add special VLAN Tag to data frames passing through.



8.7.5 Caller ID

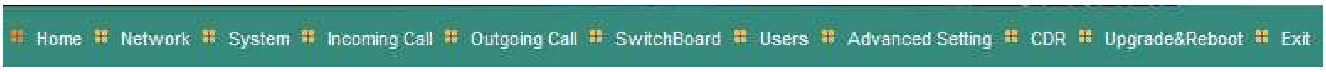
This allow to configure the caller ID fuctions and to show the incoming call numbers.

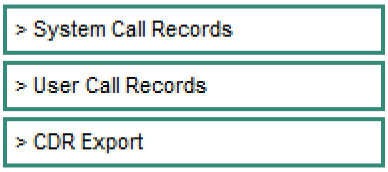


Default is "Caller ID after 1st Ring (FSK)".

8.8 Call Detail Records Query

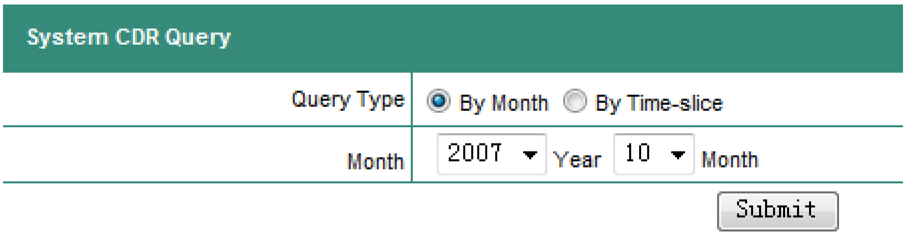
The IP PBX keeps call records for 2 months. You may have two ways for record queries. One is to query for the whole IP PBX, and the other is for the specific user. If you want to keep the call records, you may copy to USB disk before erased.



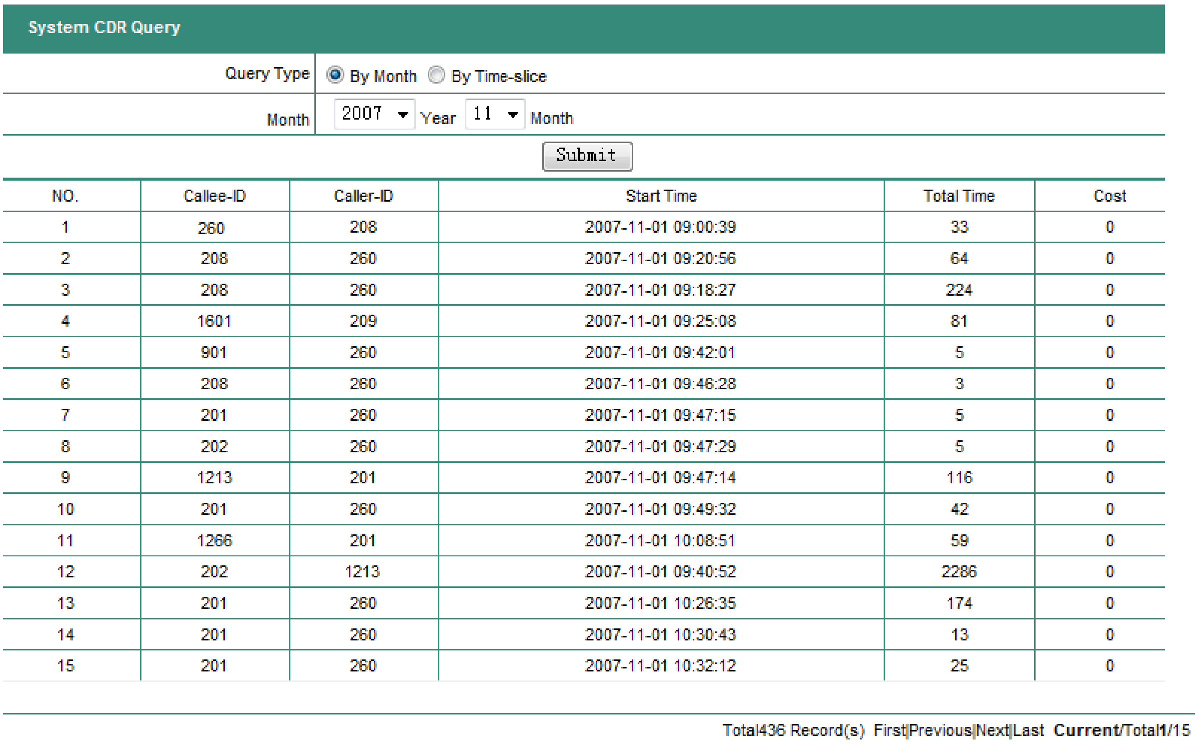


8.8.1 System Call Records query

You may query all the calls during the specified time frame and export to local storages.



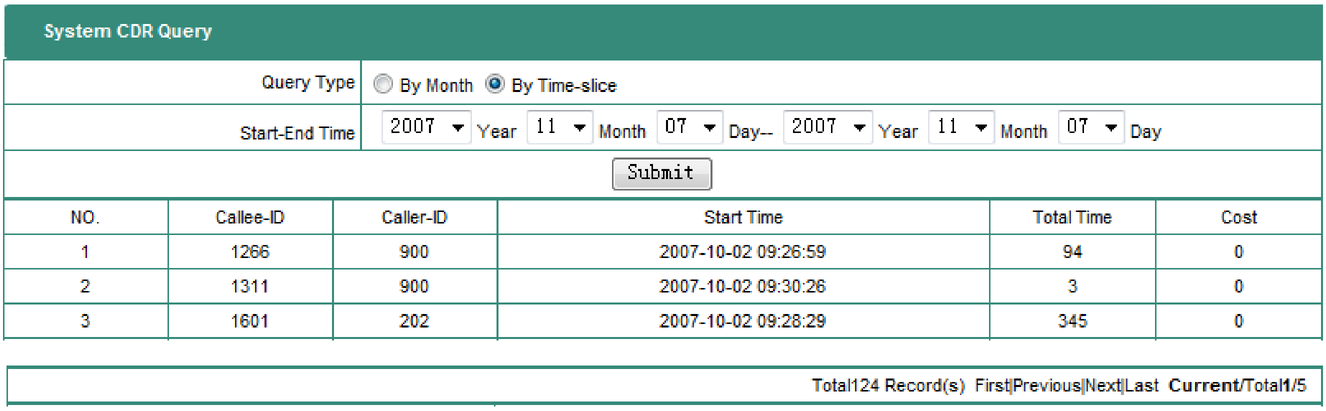
By Month



NO.	Callee-ID	Caller-ID	Start Time	Total Time	Cost
1	260	208	2007-11-01 09:00:39	33	0
2	208	260	2007-11-01 09:20:56	64	0
3	208	260	2007-11-01 09:18:27	224	0
4	1601	209	2007-11-01 09:25:08	81	0
5	901	260	2007-11-01 09:42:01	5	0
6	208	260	2007-11-01 09:46:28	3	0
7	201	260	2007-11-01 09:47:15	5	0
8	202	260	2007-11-01 09:47:29	5	0
9	1213	201	2007-11-01 09:47:14	116	0
10	201	260	2007-11-01 09:49:32	42	0
11	1266	201	2007-11-01 10:08:51	59	0
12	202	1213	2007-11-01 09:40:52	2286	0
13	201	260	2007-11-01 10:26:35	174	0
14	201	260	2007-11-01 10:30:43	13	0
15	201	260	2007-11-01 10:32:12	25	0

Total436 Record(s) First|Previous|Next|Last **Current**/Total**1**/15

By Time



System CDR Query

Query Type	By Month / By Time-slice
Start-End Time	2007 Year 11 Month 07 Day-- 2007 Year 11 Month 07 Day

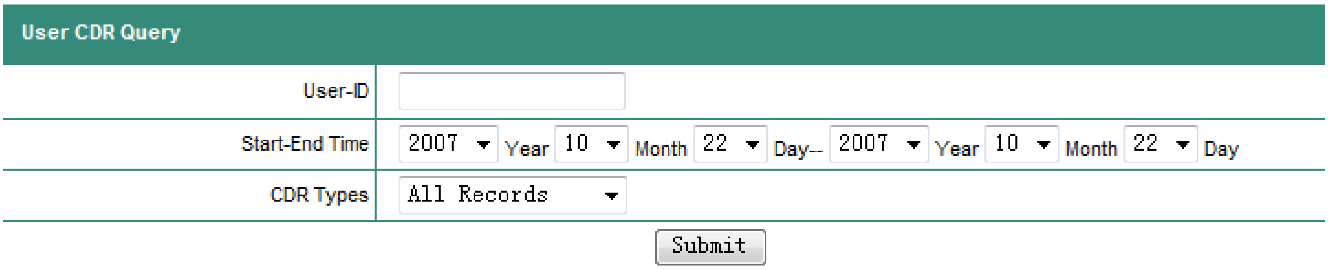
Submit

NO.	Callee-ID	Caller-ID	Start Time	Total Time	Cost
1	1266	900	2007-10-02 09:26:59	94	0
2	1311	900	2007-10-02 09:30:26	3	0
3	1601	202	2007-10-02 09:28:29	345	0

Total124 Record(s) First|Previous|Next|Last **Current/Total1/5**

8.8.2 User Call Records

You may query all the calls for certain user during the specified time frame and export to local storages.

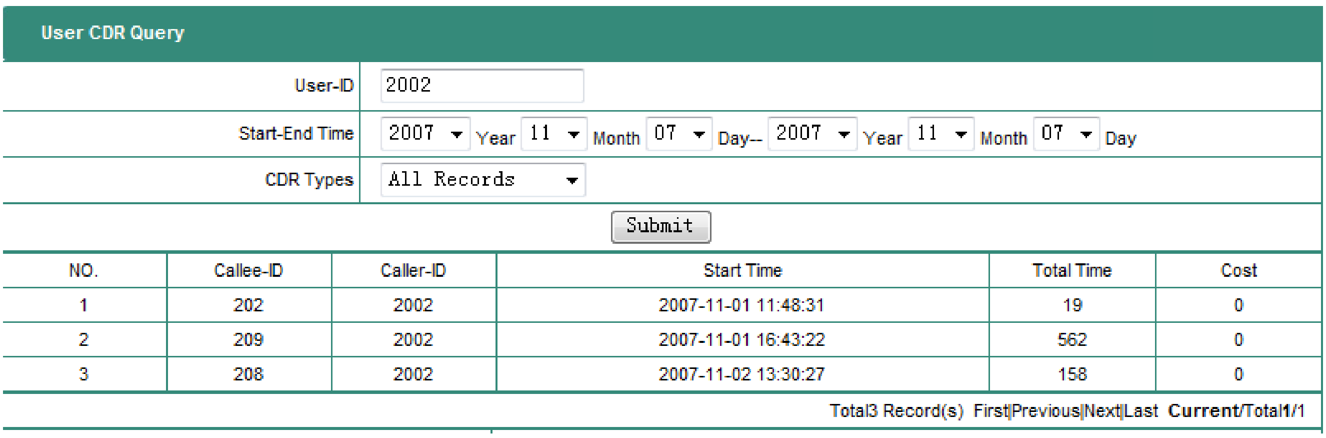


User CDR Query

User-ID	
Start-End Time	2007 Year 10 Month 22 Day-- 2007 Year 10 Month 22 Day
CDR Types	All Records

Submit

Example :



User CDR Query

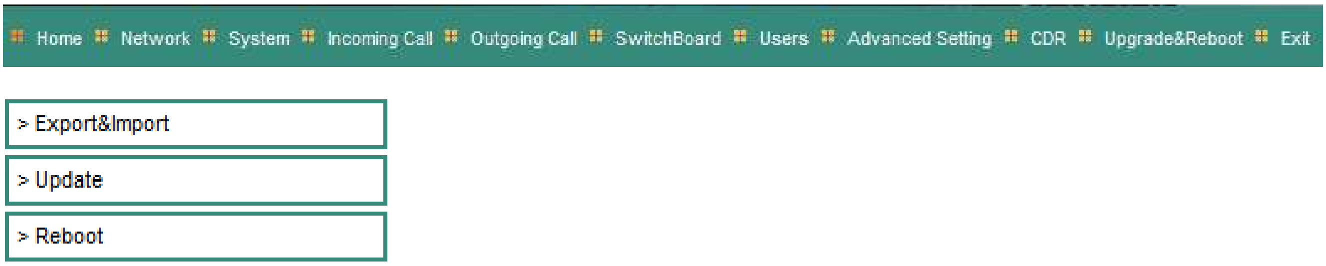
User-ID	2002
Start-End Time	2007 Year 11 Month 07 Day-- 2007 Year 11 Month 07 Day
CDR Types	All Records

Submit

NO.	Callee-ID	Caller-ID	Start Time	Total Time	Cost
1	202	2002	2007-11-01 11:48:31	19	0
2	209	2002	2007-11-01 16:43:22	562	0
3	208	2002	2007-11-02 13:30:27	158	0

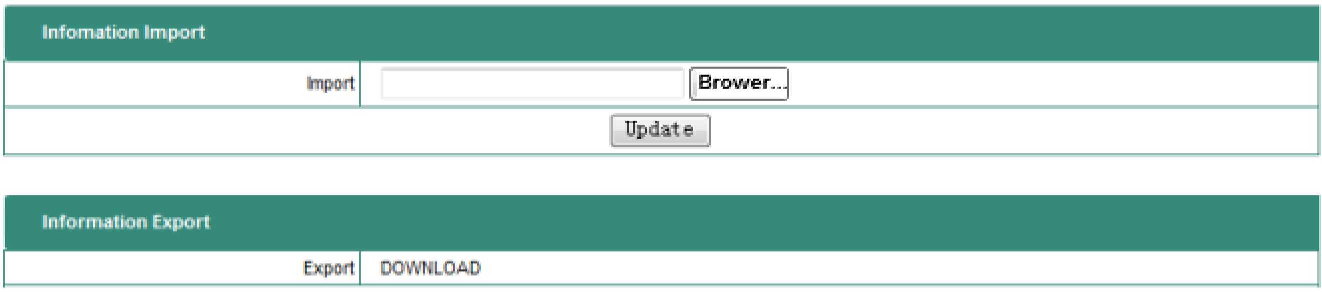
Total3 Record(s) First|Previous|Next|Last **Current/Total1/1**

8.9 Upgrade & Reboot



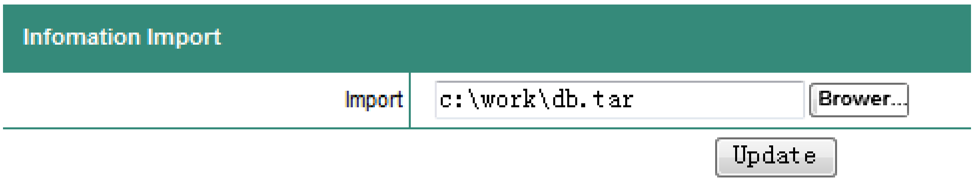
8.9.1 User information Export & Import

Before upgrading the VOI-9300 IP-PBX or reset to factory defaults, you may export all the user information to local PC storage, and import back after the upgrade or default settings are done.



Export: Click on “DOWNLOAD” button and select “Save as New File”.

Import: Select the desired file, and upload.



8.9.2 System Upgrade

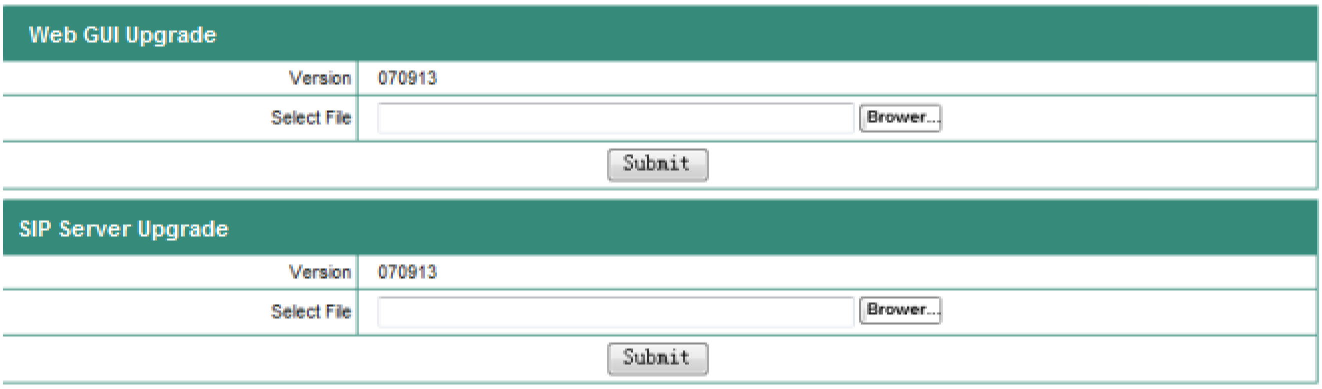
VOI-9300 IP-PBX provides WEB upgrading. Before upgrading, make sure the following;

(1) Get the desired upgrade version. The VOI-9300 support two different ways for upgrade; Web GUI Upgrade and SIP Server Upgrade.

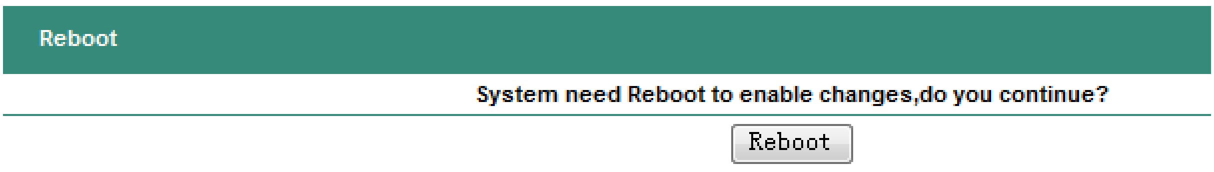
(2) Backup a copy of user information by import/export data files.

(3) Make sure the power is on while upgrading.

Then you may upgrade per the web instruction procedures.



8.9.3 Reboot



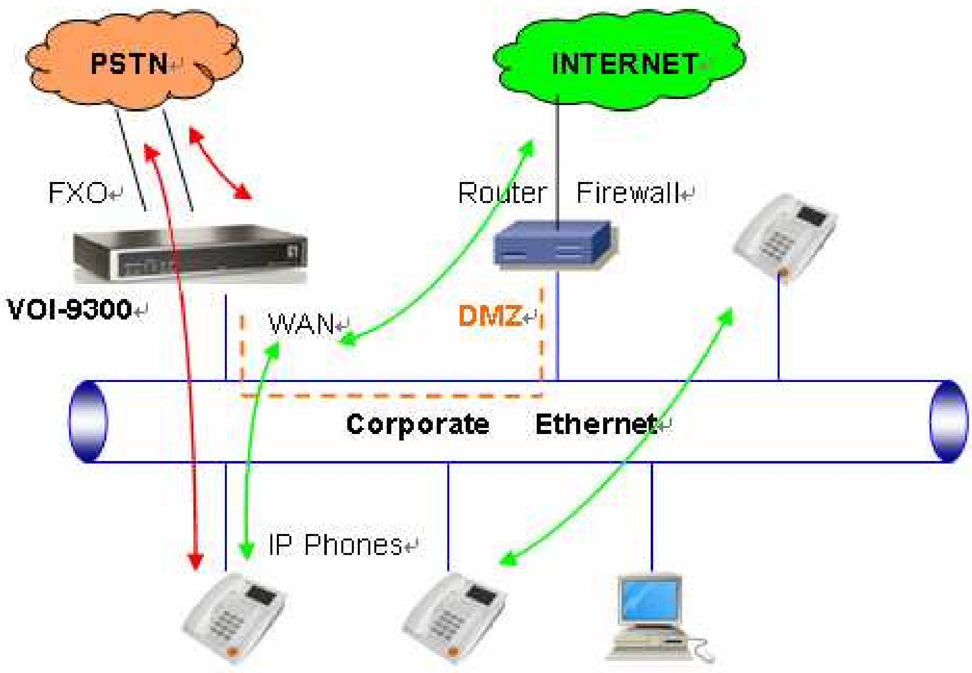
Reboot is used to enable the changes. Note that for Factory Default Setting, please refer to the chapter. 7 Reset to Factory Default.

9. Applications

Applications of IP PBX under Firewall with DMZ

This will protect corporate network security while allowing VOI-9300 to work as **IP-PBX** for VoIP applications. When VOI-9300 **IP-PBX** is operating under the corporate firewall, remember to enable and open the following service port numbers for VoIP applications.

- TCP Port：22, 80, 1723 for Telnet, Http, and PPTP.
- UDP Port：5060, 1194, 10000-20000 for SIP, OPENVPN, and Voice RTP Range.



Applications of IP PBX with ADSL

This VOI-9300 supports PPPOE to work with ADSL and to integrate **IP-PBX** into the corporate network for VoIP applications.