SPEEDYTEL TECHNOLOGY LIMITED

Deploying the

PBX220 IP Telephone System

Technical Manual

PBX220

Welcome to

Wi-Fi IP PBX

PBX220

1. Getting Started

Thank you for purchasing Speedytel PBX220 (WiFi Router+ IP PBX).It is one cost-efficient yet easy-to-configure IP PBX in the market today.

Administrating a VoIP system can be a daunting task for administrators unfamiliar with VoIP. This guide is designed to help you plan and configure Speedytel PBX220 Voice over IP (VoIP) deployments.

1.1 Introduction

The Speedytel PBX220 (300M WiFi Router+ IP PBX) is the ideal system for small businesses and home offices requiring a pint-sized yet powerful on-premise wifi router IP PBX.

The compact solid-state device supports 16 extensions and offers a wide range of IP PBX telephony features.

Keeping up with the demands of sustainability, the speedytel PBX220 is based on a low-power, high performance MIPS processor, providing the complicated communication features including the hardest HD communication protocol, complete router features and QoS (Ensure the voice quality in a case the bandwidth is not enough).Meanwhile, the feature of one touch to deploy the phones makes the configuration of phones easy and enjoyable thing.

1.2 Packing list

1 unit PBX220

1 Piece Power Supply (12V,1A)

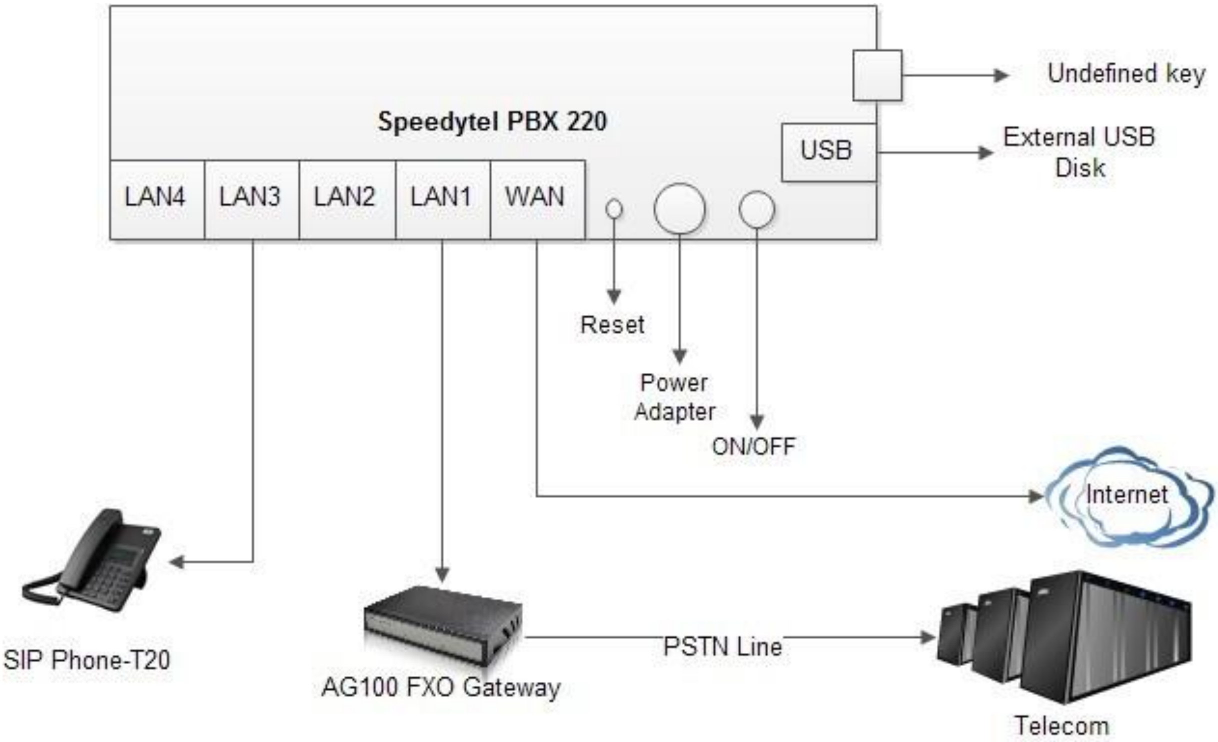
1 piece of 2-meter Network cables

1.3 Specification

Hardware	Processor: MIPS RAM: 64MB FLASH: 16MB WAN: 1xRJ45 10/100MB Ethernet port LAN: 4xRJ45 10/100MB Ethernet port Button: Reset Button, Feature Button Power adapter: AC 100~240V input and DC 12V/1A output Power Consumption: 1.2-2.0W Operating humidity: 10~95% Operating temperature: 0~45°C Net weight 230g, Gross weight 600g Dimension: 168X106X71mm Box dimension: 19.5X13.8X8.5cm
WiFi Router	Channels: Support 1-13 channels and auto mode TxPower: 802.11n 300M 14dBm, 802.11g 54M 16dBm, 802.11b 11M 17-19dBm Antenna: Internal 2T2R 3dBi Coverage Area: Indoor 100 meters, Outdoor 300 meters Encryption:WEP-Auto,WPA-PSK,WPA-PSK2 HT Bandwidth: 20Mhz, 40Mhz DHCP: Server / Client Port Forward: Supported Firewall: Support flood attack protect DMZ: Supported uPNP & NAT-PMP: Supported Qos: WMM, DSCP External Disk: USB 2.0 in Fat32 , Ext3, Ext4
Networks	WAN: DHCP, STATIC, PPPoE LAN: Static IP Time Zone: Multinational Dynamic DNS: Supported VPN: PPTP in Tunnel, PPTP in PBX only Wan Mac Clone
	Extensions: 16 SIP extensions, up to 32 SIP extensions in Turbo mode 4 concurrent calls, up to 8 concurrent in Turbo mode Extension Features: Support Ring Group, Follow Me, Extension Outbound Routes,

PBX Features	Extension dialing intelligent analysis, Extension for BLF support, MWI(Message Waiting Indicator) Support SIP trunk： Registration mode、IP docking mode(SIP Direct) SIP Trunk Failover Trunk features：DID set, routes way, Inbound call system support routing, Support for multiple devices in the cluster, Support automatic looking for the callee’s number. Call Pickup, Voicemail One Touch to Config Codec: .722,G.711U/A,G.729,GSM, H.264 SIP Protocol: Over TCP/UDP, RFC3261 DTMF：In-Band，RFC2833,SIP Info Conference: Up to 4 parties conference system, support multiple independent meeting room Queues: Call queue strategy support random，ring all，rotate members call IVR: Support voice menu, multi-level and key recognition Background Music: Supported IVR Voice file: play, record or upload Call record: Call record will automatically store in the external disk Time Frames: Support the IVR auto-switch based on the time SIP Behind NAT: Supported AMI port allows to connect PBX220 to the third parties’ software

1.4 Hardware Setup



You may check the picture above to config your system

Step 1: Connect the LAN port of PBX220 with your corporate IP network. Before you connect the PBX220 to the network, please check if your network can work normally.

Step 2: Plug in and open your browser to visit the web address: Http://192.168.1.1

Make sure your PC IP address is 192.168.1.XXX

(If you use IE6 and above, the prefix address http:// can't be left out)

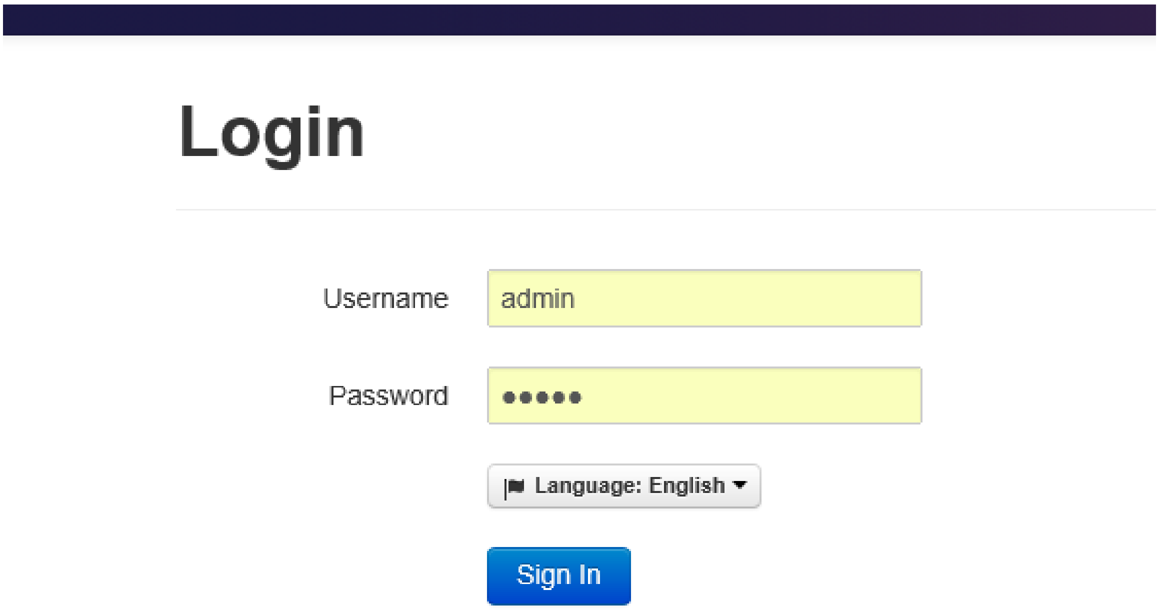
Now we access to the Wizard page.

Username: admin (By default)

Password: admin (By default)

Or you can find the WiFi SSID : WifiRouterPBX_XXXXXX and log in. And visit the web:

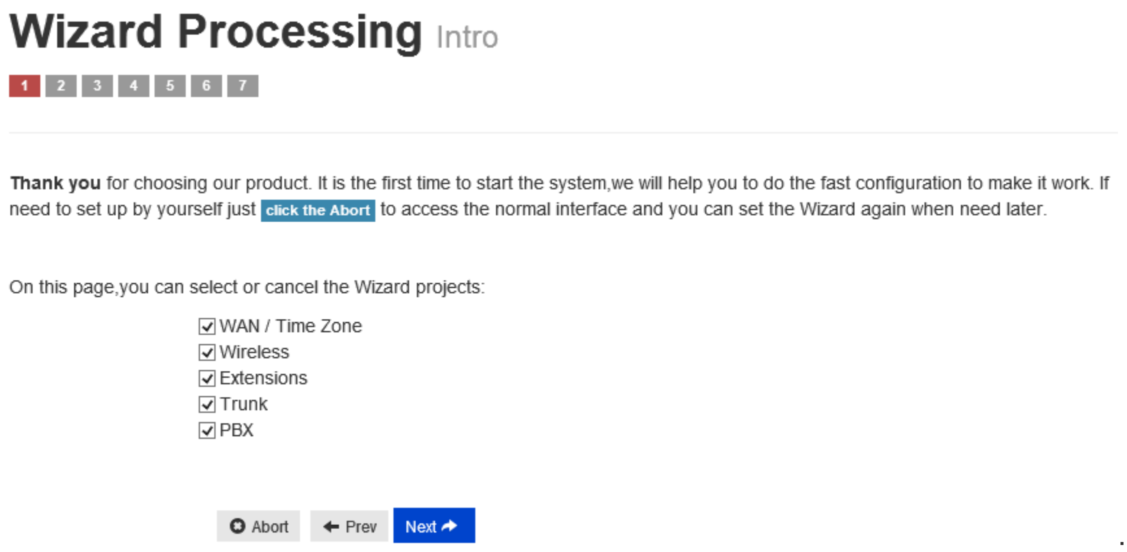
Http://192.168.1.1 You can also choose the web language.



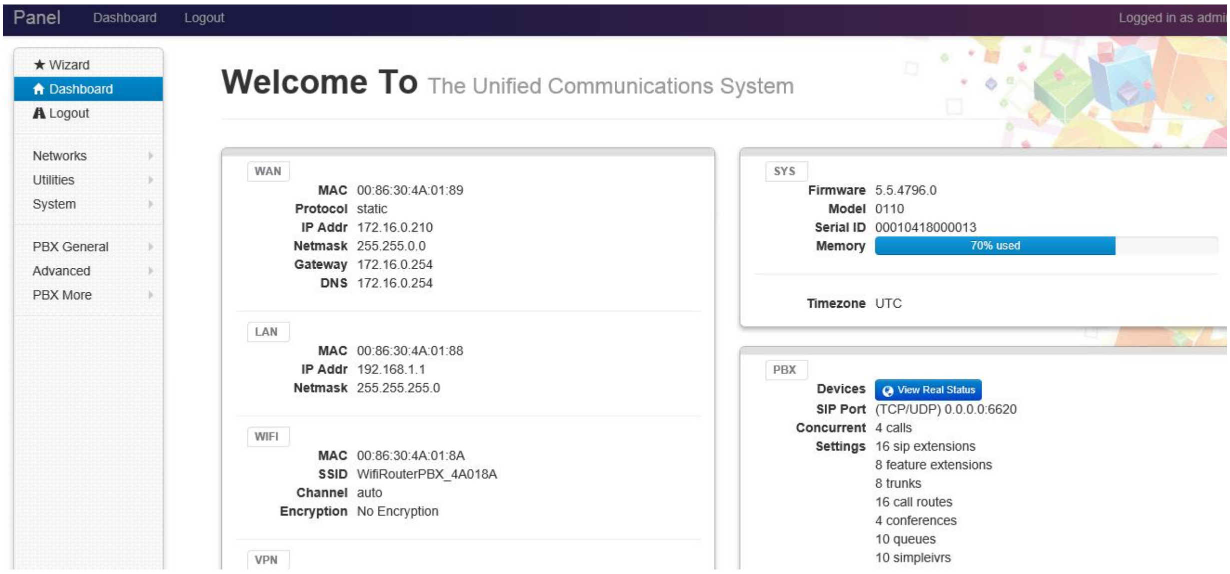
Note: If you connect PBX220 to your exist network, please make sure that Lan IP of PBX220 (default is 192.168.1.1) is different from your exist network. Otherwise, it will be a network conflict.

1.5 First Login to Wizard

This is your first time to log in, it will show Wizard Processing. It is simple and brief to deploy. In most cases, the default settings can be used for the rest of the configuration.



If you want to quit the Wizard, just Abort it. And if you want to access Wizard, just click the Wizad.



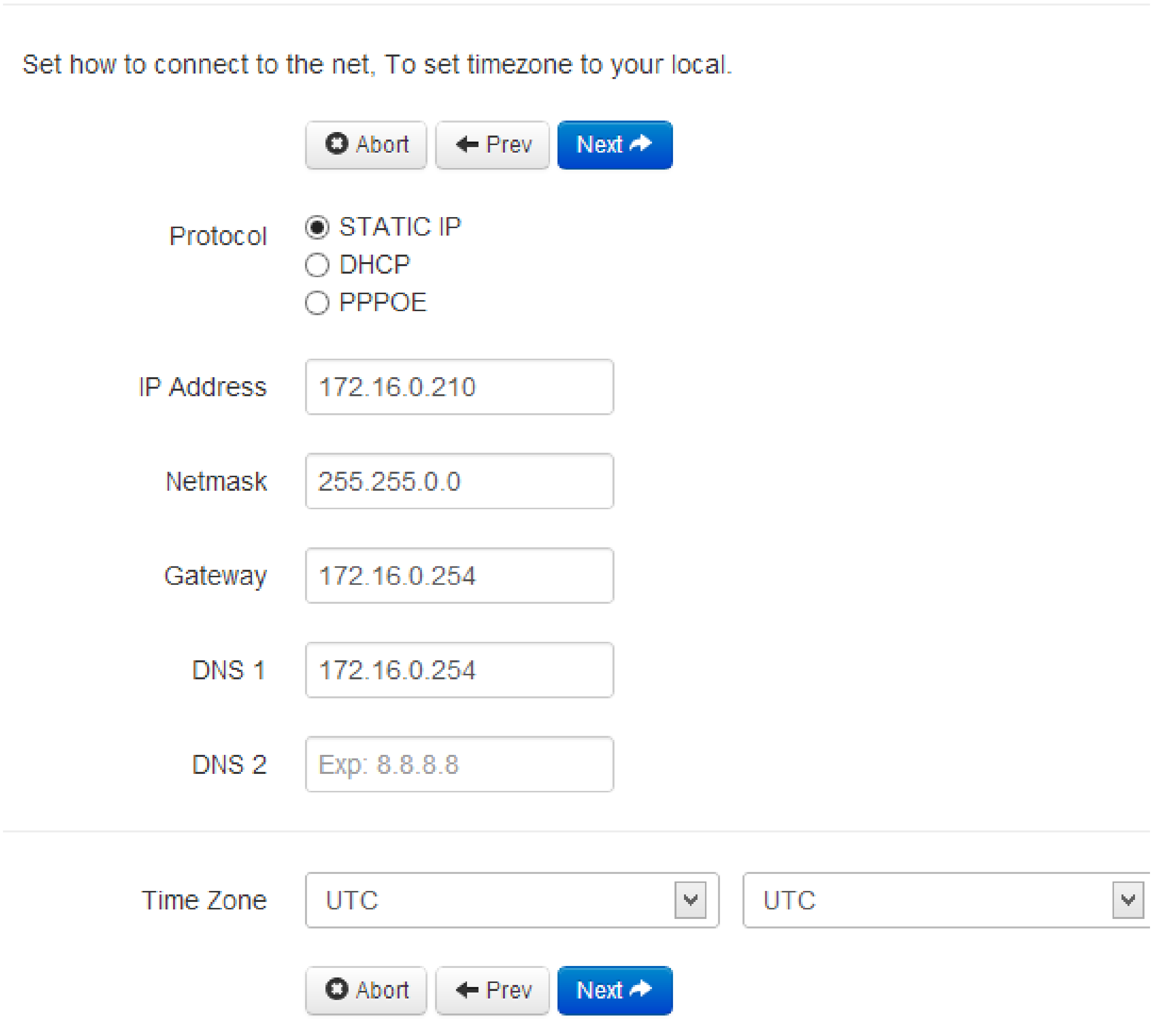
You can click Prev to return and Next to do the following steps.

On the Wizard page, you can set the WAN IP and Time Zone, WiFi, Extension, line provider and PBX.

You can select which part need to set then do the Next.

(You can also follow all the default settings and confirm)

Next is the WAN and Time Zone as below:



You can choose the Protocol Static IP, DHCP, PPPoE.

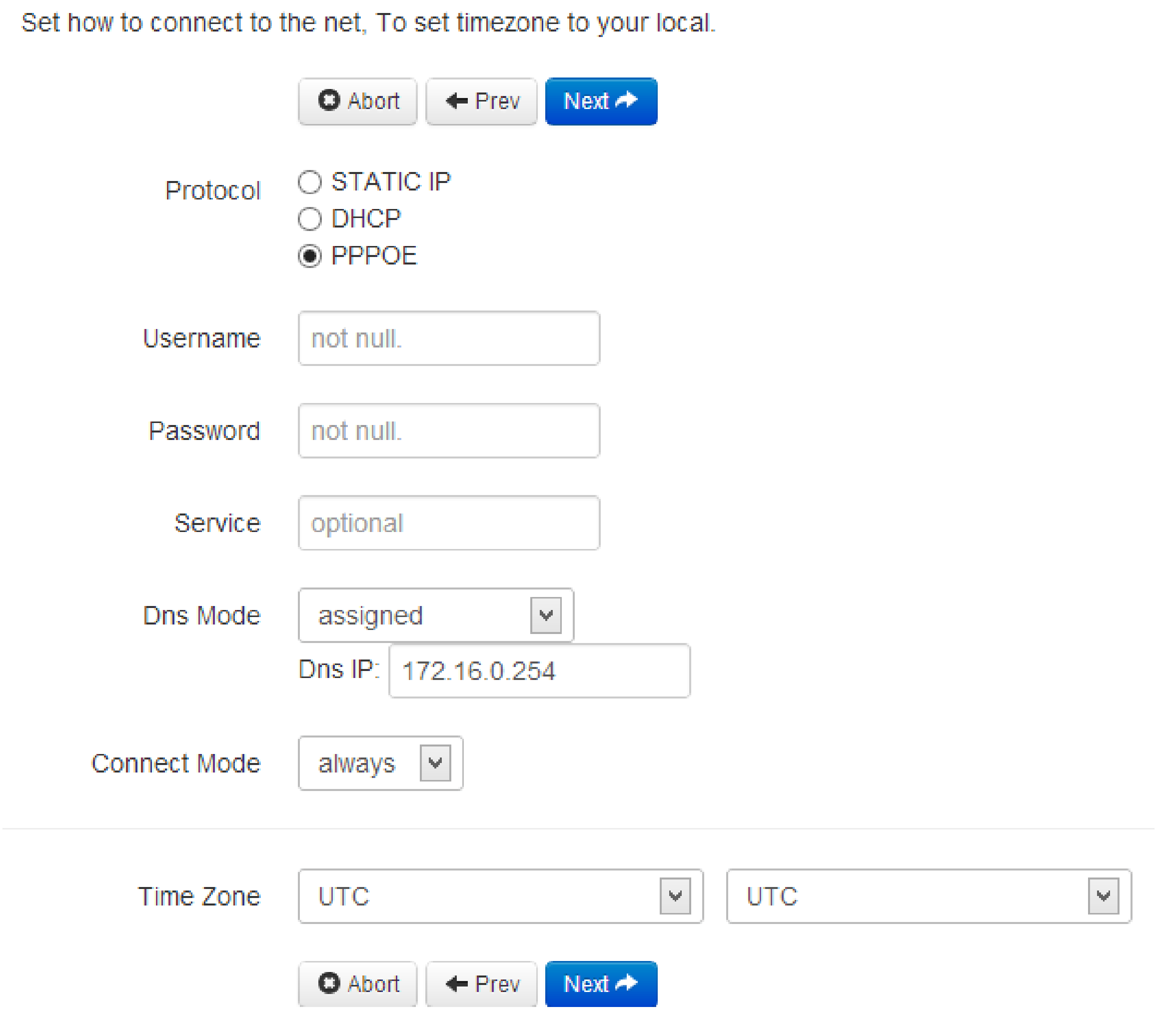
Static IP

Using a static IP address is the most reliable way to ensure your server IP address does not change. To find an IP address that is not in use on your network and will not be used for another client by the DHCP server or used by some other devices.

DHCP（Dynamic Host Configuration Protocol, DHCP）

PPPoE (Point to Point Protocol over Ethernet)

Fill out the account information from your telecom operator and **Next.**

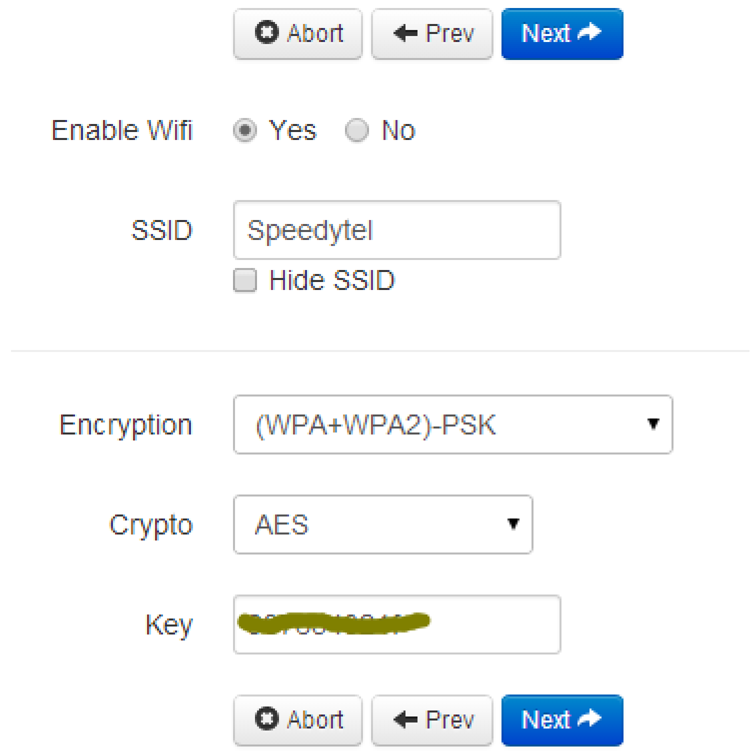


Time Zone

You can set the local time here and it is important for generating accurate call reports for the system. And Time Frame will also analysis the system time to switch to the proper IVR.

If you select the incorrect time zone, or you move to a different time zone later, you can change it in the Wizard or in the Network-WAN/LAN/Time Zone.

Now we came to the WiFi setting page:



Here you can set the WiFi option.

Enable WiFi and set the SSID(Service Set Identifier)

According to the tip in the black, choose your encryption, crypto and key.

Caution: PBX220's WiFi is enabled by default. Setup your WiFi access password here to protect your network

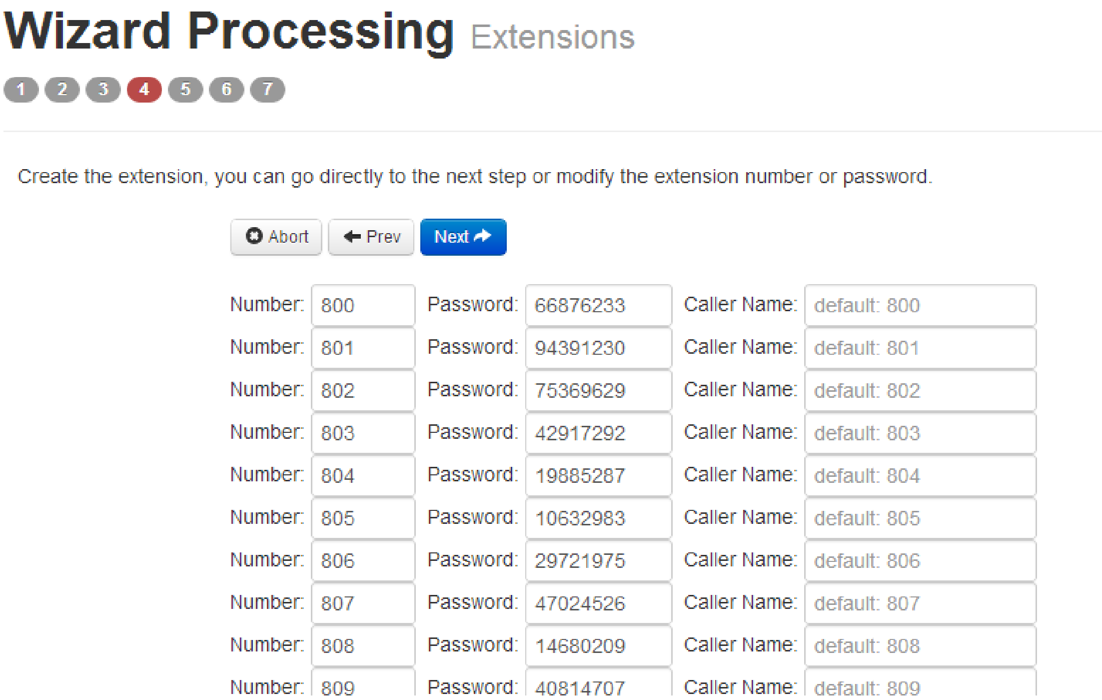
Now we come to the Extension page.

The system has already auto generated 16 SIP extension by default.

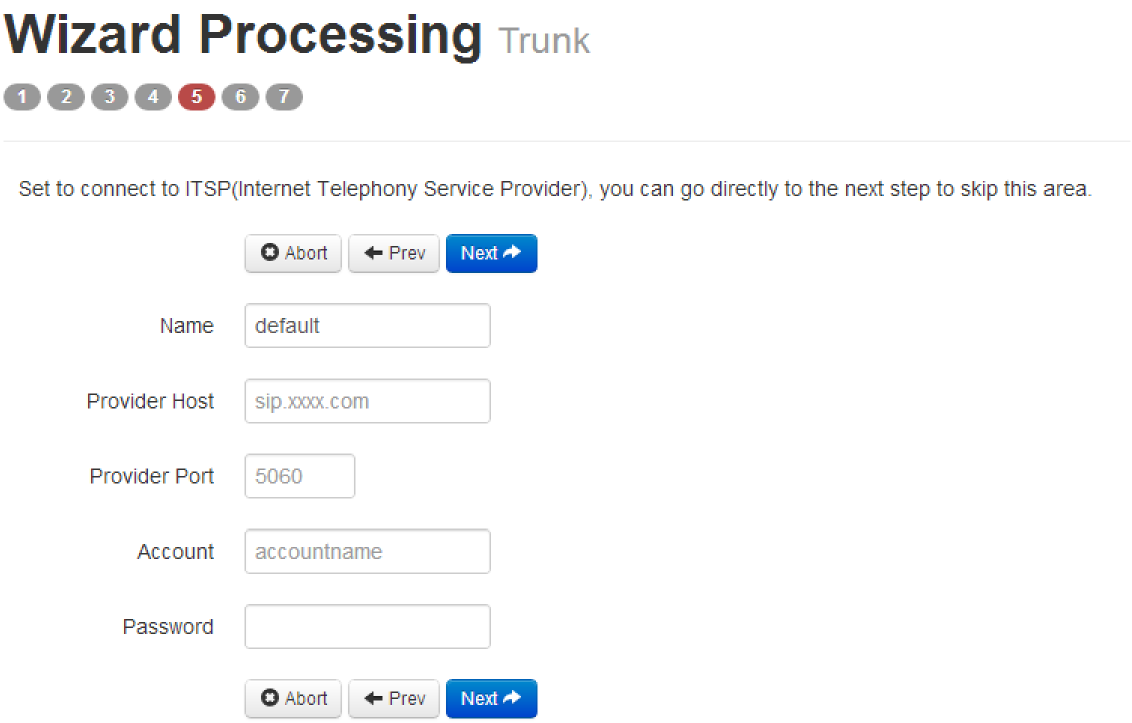
(PBX220 supports max 16 SIP extensions.)

You can set the extension number, password and the caller name manually

If want to set the extensions quantity as you like, just amend this in **Extension** table.



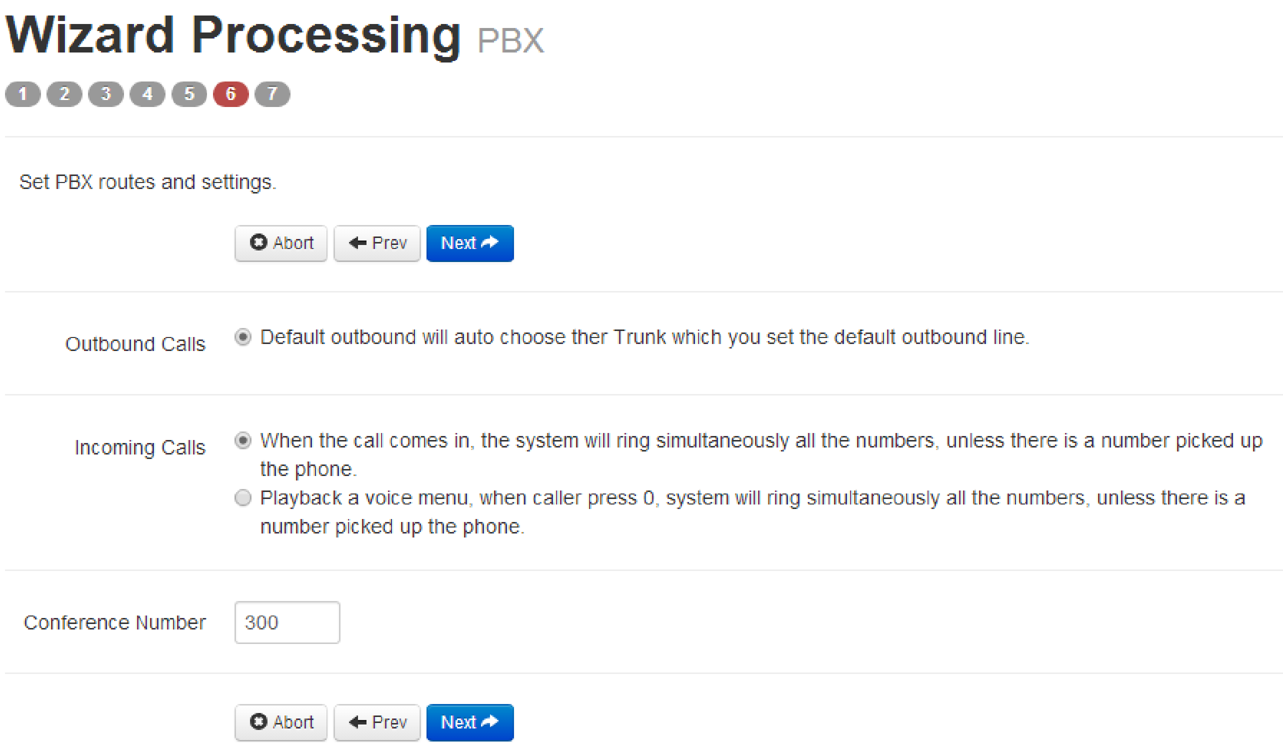
Next we come to **Trunk** Setting.



Set a SIP Trunk here. Fill out the Provider Host, account and password from your Internet Telephony Service Provider then one SIP Trunk will be built.

PBX220 supports maximum 8 SIP trunks. If you want to build more, just do it in the **Line Provider**.

Next is **PBX** setting :



Here you can set how the system will deal with when make outbound calls or receive the inbound calls.

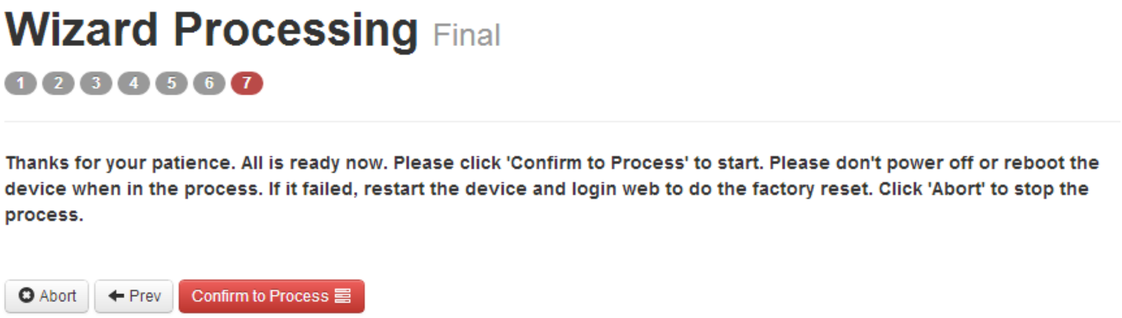
Outbound calls: Auto-select the Trunk which you set it as the default outbound line.

Incoming calls: Ring all the numbers when the call comes in or just into a IVR.

If want to set other routes, you can do it in **Outbound Routes** and **Inbound Routes** after the Wizard.

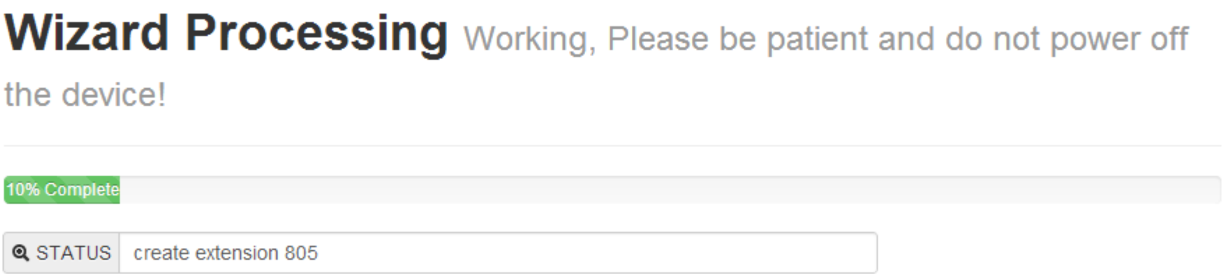
You can also set the conference number here. By default is 300.

Next, well done! Just confirm to process.



Then the PBX220 begin to configure. Now enjoy the HTML5 interface.

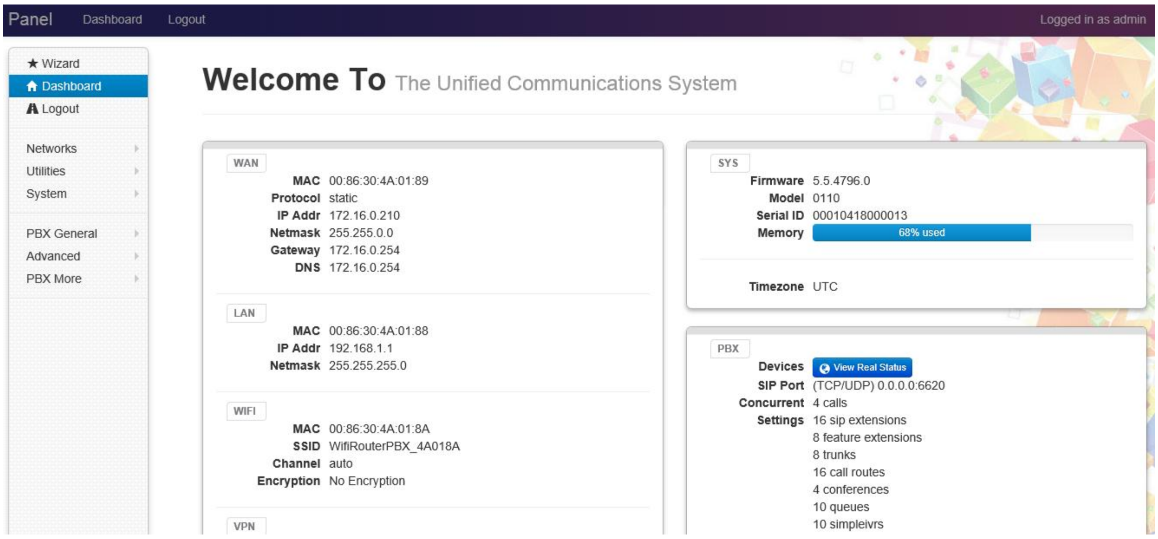
It will auto-restart the PBX and if you can't see the page you can refresh it. If you amended the static IP address, re-visit the changed IP. Re-log in, you will see its IP at the dashboard of PBX220.



OK. This is the first time to access the PBX220.

1.6 Dashboard

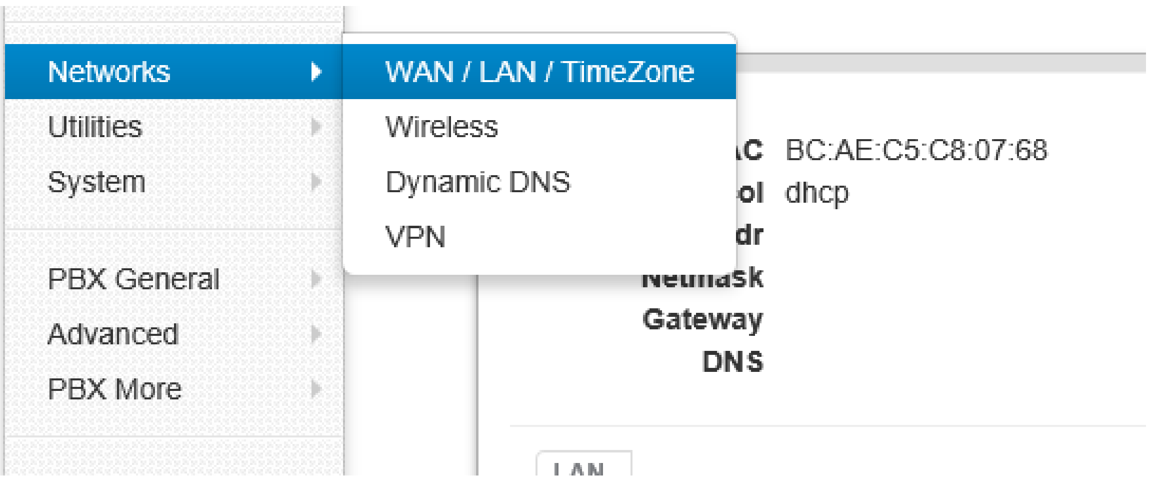
You can see the status of WAN, LAN, VPN, WiFi, System and PBX in the dashboard.



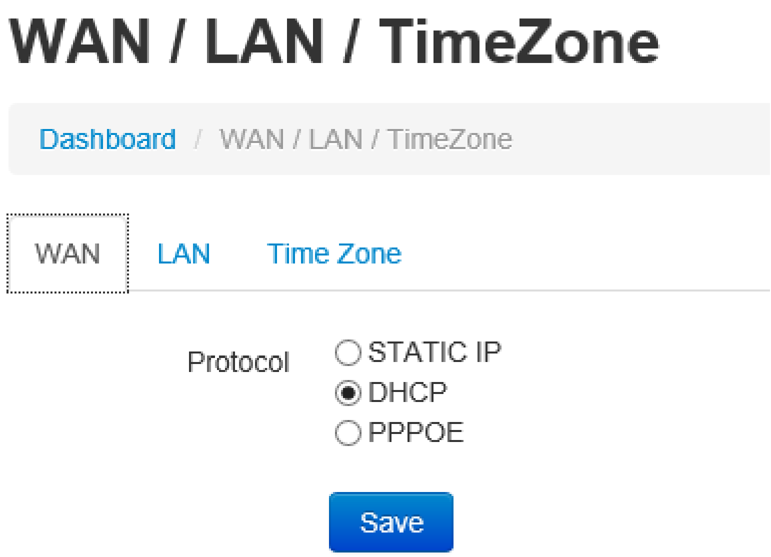
2. Networks

1.7 WAN/LAN/Time Zone

After successfully connecting the PBX220 to the network for the first time, users could login the Web GUI and go to Networks to configure the network parameters for the device.

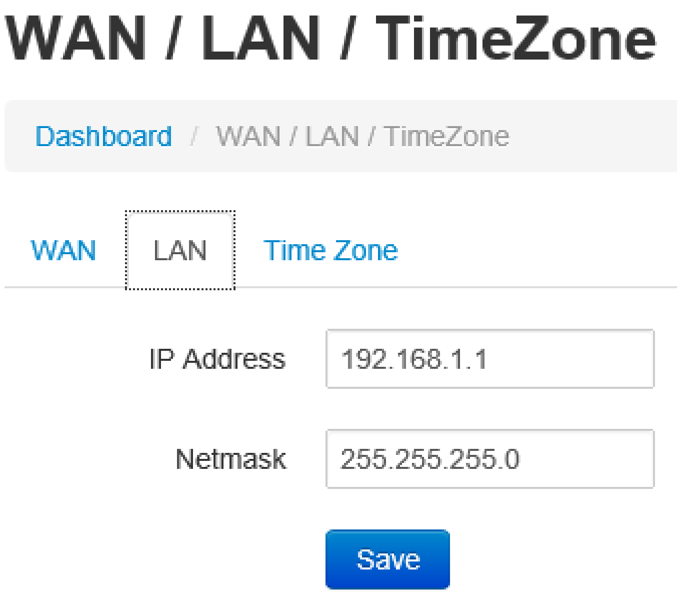


WAN/LAN/Time Zone



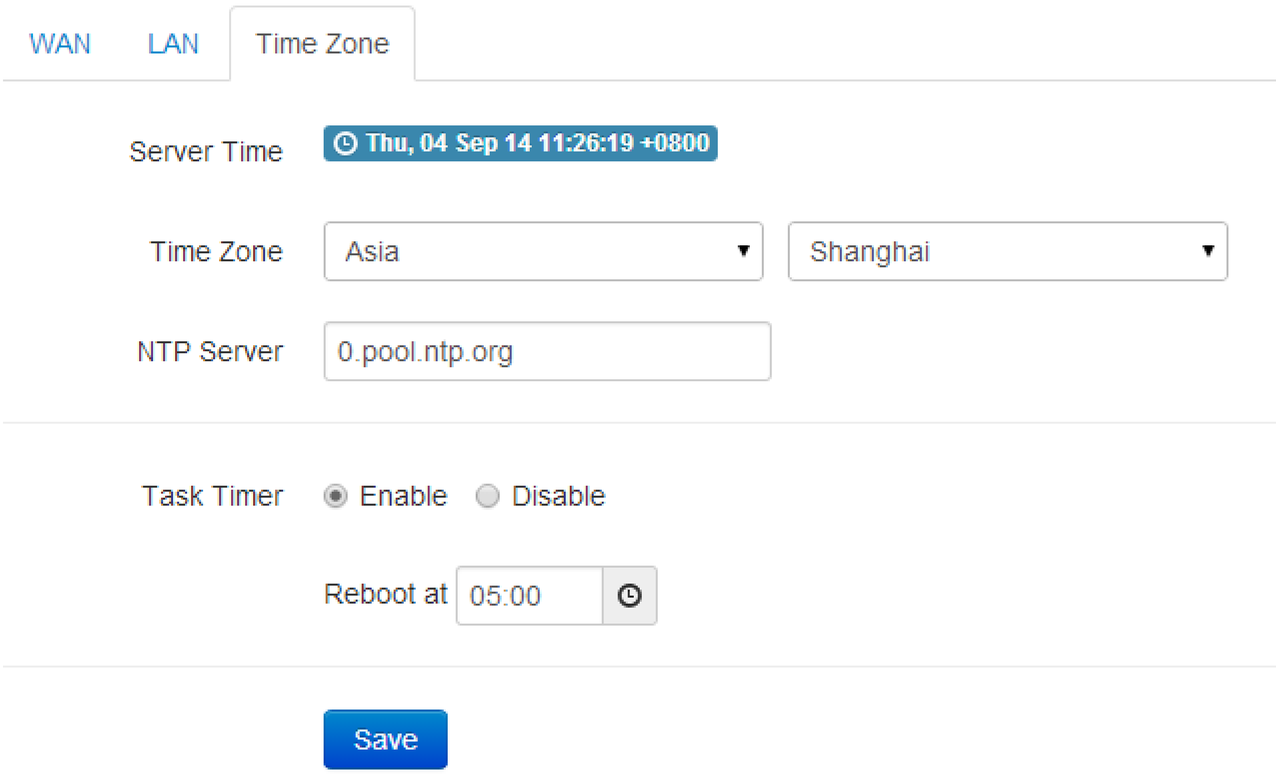
Please refer to the following tables for basic network configuration parameters on WAN setting.

WAN Setting	
Protocol	Select DHCP, Static IP, or PPPoE. The default setting is DHCP
IP Address	Enter the IP address for static IP settings.
Netmask	Enter the subnet mask address for static IP settings.
Gateway	Enter the gateway IP address for static IP settings.
DNS 1	Enter the DNS server 1 address for static IP settings.
DNS 2	Enter the DNS server 2 address for static IP settings.
Username	Enter the user name to connect via PPPoE
Password	Enter the pass word to connect via PPPoE
Service	Enter the ISP service name(optional)
DNS Mode	Select the DNS mode for PPPoE: **Assigned:** DNS is assigned by your ISP. **Set:** Manually set your DNS
Connect Mode	Select the either **always** or **demand**



Please refer to the following tables for basic network configuration parameters on LAN setting.

LAN Setting	
IP Address	Enter the IP address assigned to LAN port. The default setting is 192.168.1.1
Netmask	Enter the subnet mask. The default setting is 255.255.255.0

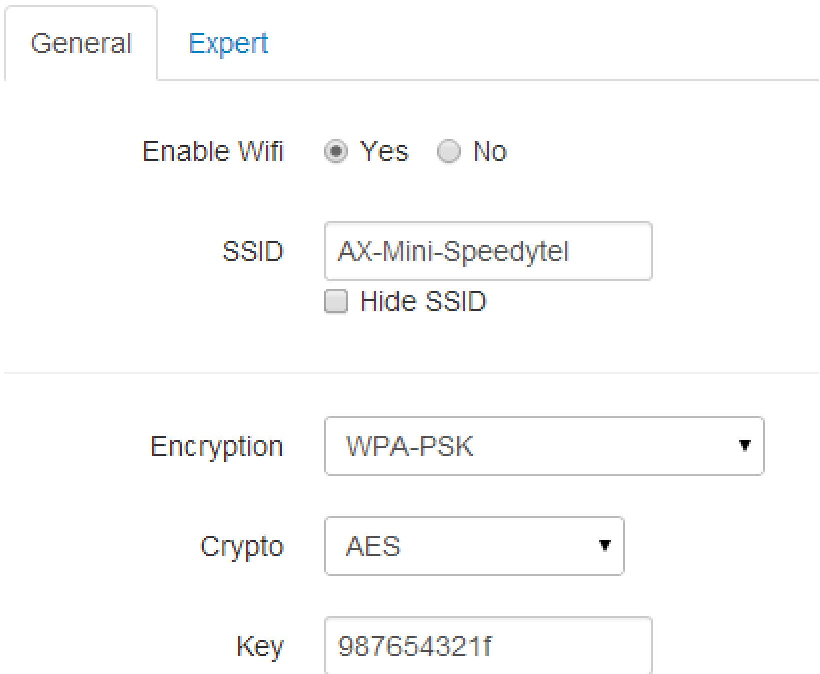


Please refer to the following tables for basic network configuration parameters on Time Zone setting.

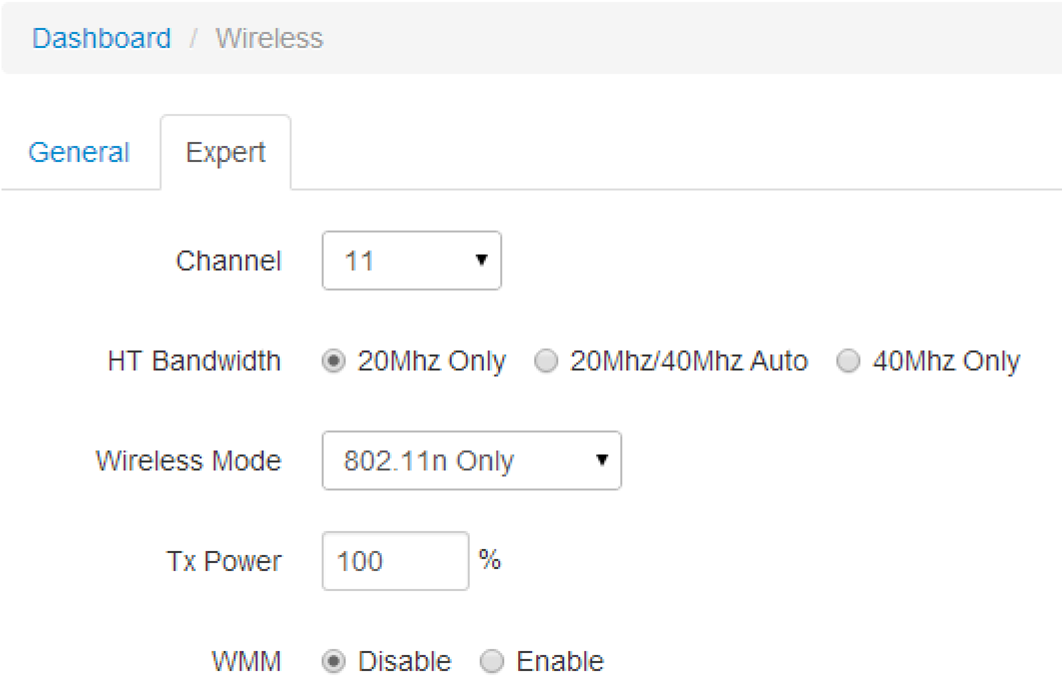
Time Zone Setting	
Server Time	The current time from the NTP server
Time Zone	Select the proper time zone option for PBX220
NTP Server	Specify the URL or IP address of the NTP server for the PBX220 to synchronize the date and time.
Task Timer	Select enable or disable task timer
Reboot at	Select the auto reboot time

Wireless

Under WEB GUI--**Network**—**Wireless**, click **Wireless** to config the WiFi setting of PBX220



General Setting	
Enable WiFi	Select Yes to enable your WiFi network, No to disable it
SSID	SSID is the name of your wireless network. Create your name here. Check Hide SSID box to hide your wireless network name.
Encryption	Select the encryption option for your wireless network
Crypto	Select the crypto option
Key	Enter the password for your wireless network

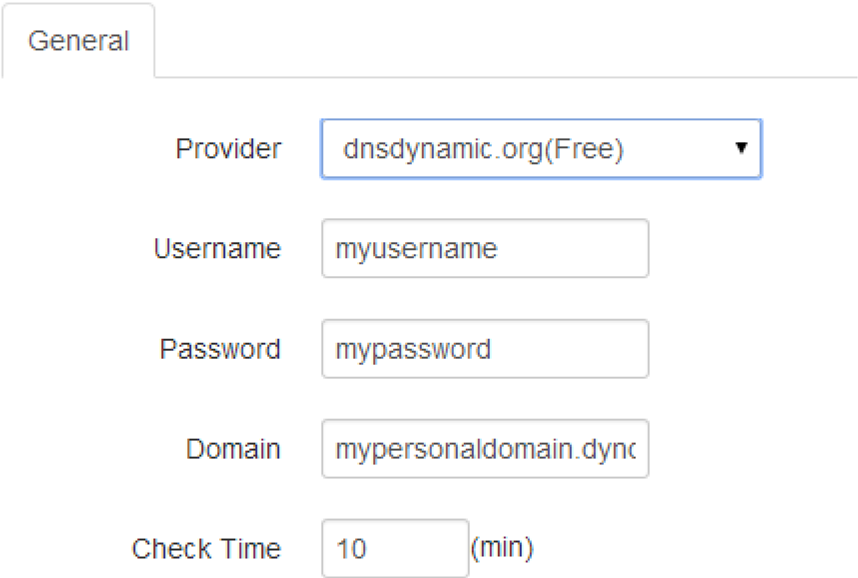


Expert Setting	
Channel	Select the channel setting for PBX220.
HT Bandwidth	Configure HT bandwidth option.
Wireless Mode	Select one of the following: **802.11n only-**Select if all your wireless clients are 802.11n **802.11g/n-**Select if you are using both 802.11g and 802.11n wireless clients **802.11b/g/n-**Select if you are using a mix of 802.11b, 802.11g and 802.11n wireless clients
Tx Power	Manually set the Tx rate
WMM	Select to enable or disable WiFi Multimedia

Dynamic DNS

Under WEB GUI--**Network—Dynamic DNS**, click **it** to config the DDNS setting of PBX220.

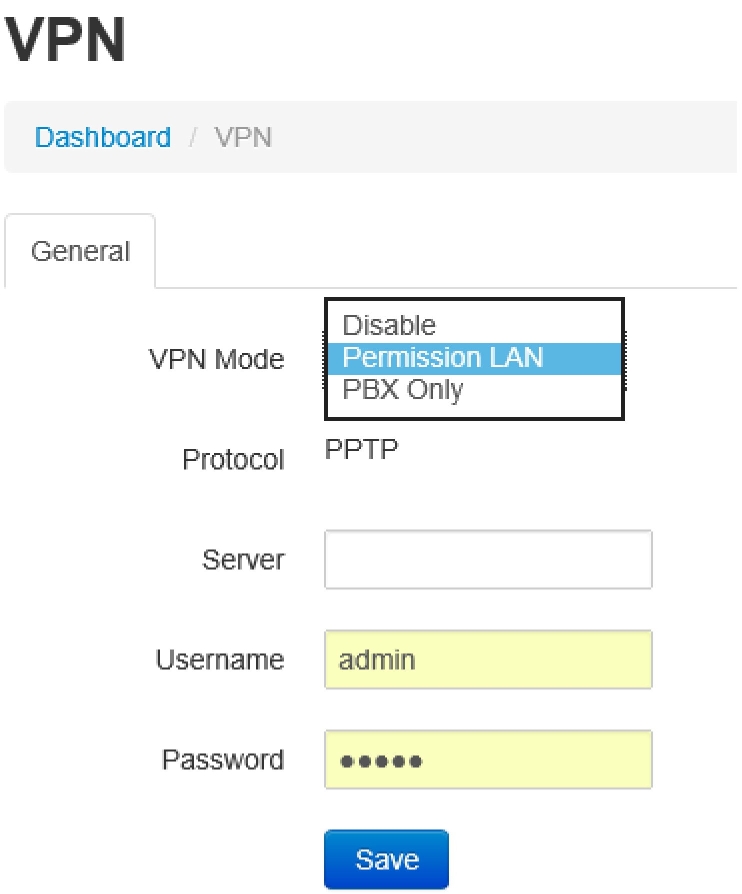
Dynamic Domain Name System is a method of keeping a domain name linked to a changing IP address.



General Setting	
Provider	Select one DDNS provider from the list.
Username	Enter the user name of your DDNS account.
Password	Enter the password of your DDNS account.
Domain	Enter the domain name that you registered with your DDNS service provider
Check Time	Enter the check time value

VPN

Under **WEB GUI—Networks—VPN**, click **VPN** to config the VPN setting of PBX220. PBX220 supports PPTP VPN as a server endpoint.



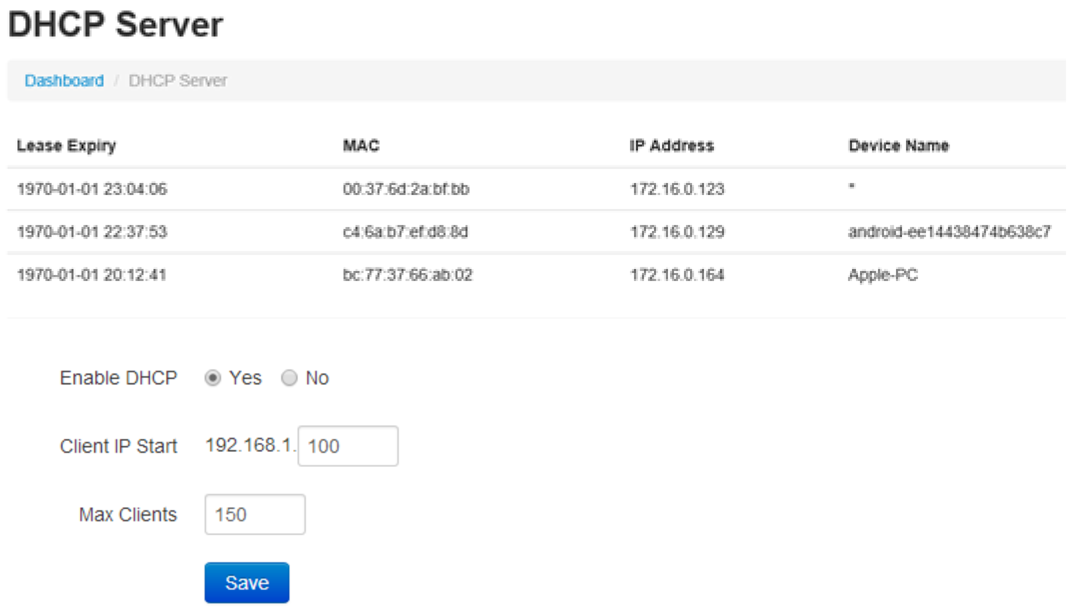
VPN Setting	
VPN Mode	Select one of the following; **Disable**—disable VPN **Permission Lan**—all the data from the Lan will go via VPN **PBX only**—only the PBX data (VoIP)will go via VPN
Protocol	PBX220 supports PPTP VPN protocol.
Server	Enter the VPN server IP address.
Username	Enter the user name of your VPN account.
Password	Enter the password of your VON account.

1.8 Utilities

DHCP Server

DCHP stands for Dynamic Host Control Protocol. PBX220 has a built in DHCP server. The

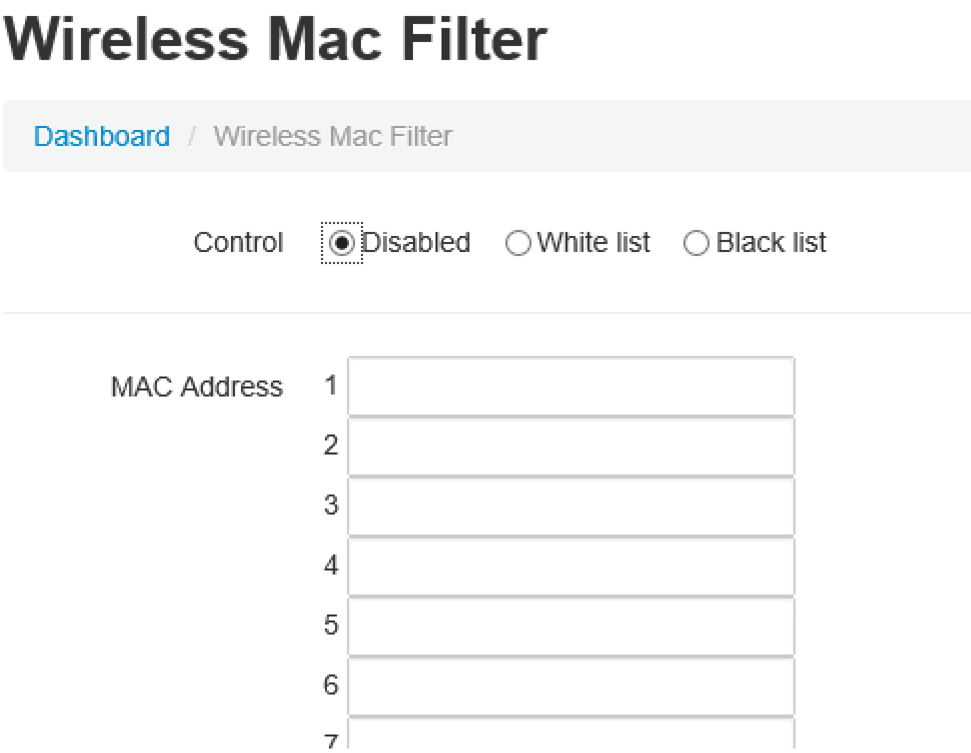
DHCP server will automatically assign an IP address to the computers on the LAN. Be sure to set your computers to be DHCP clients by setting their TCP/IP setting to “Obtain an IP address automatically”. The DHCP server will automatically allocate an unused IP address from the IP address pool. You must specify the starting IP address and the clients’ quantity.



DHCP Server Setting	
Lease Expiry	DHCP client expiry time
MAC	DHCP client MAC address
IP Address	DHCP client assigned IP address
Device Name	DHCP client name
Enable DHCP	Select Yes to enable, no to disable
Client IP Start	Specify the starting IP for the DHCP server assignment
Max Clients	Enter the DHCP clients max quantity

Wireless Mac Filter

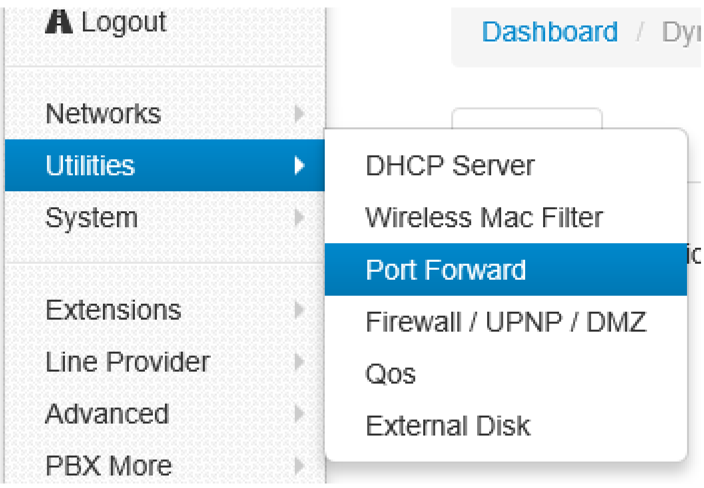
Use Wireless Mac Filter to allow or deny wireless devices by their MAC addresses from accessing the network. You can manually set the white list or black list to manage the access.

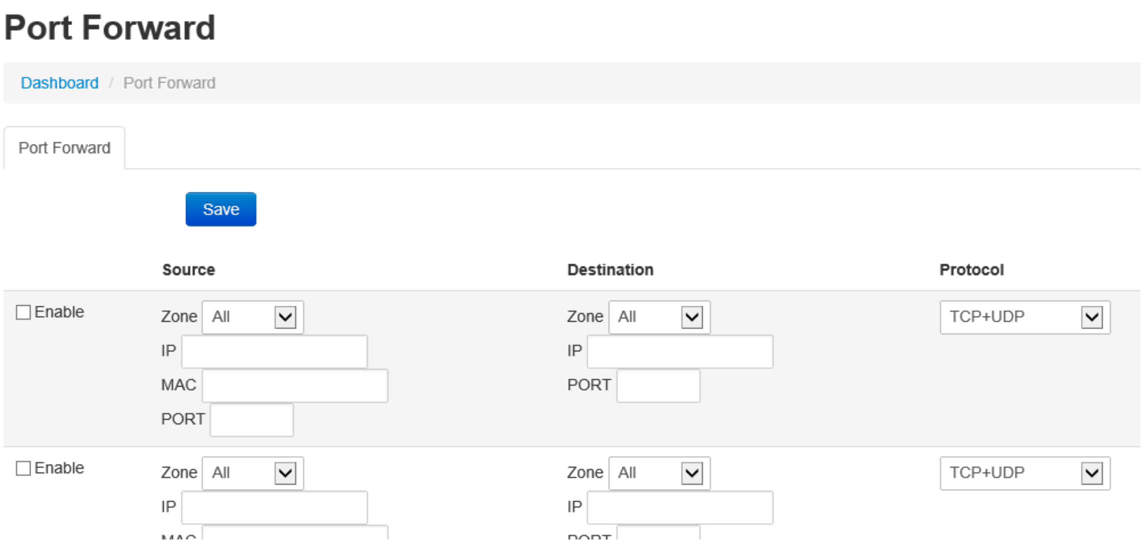


Wireless MAC Filter Setting	
Control	Select filter list type. MAC addresses in white list are permitted to access the WiFi network MAC address in black list are forbidden to access the WiFi network
MAC Address	Manually add the filter MAC address into the list

Port Forward

This section allows to open a single port or range of ports. Set the IP address, Mac, port from the source and destination, choose the protocol and enable.

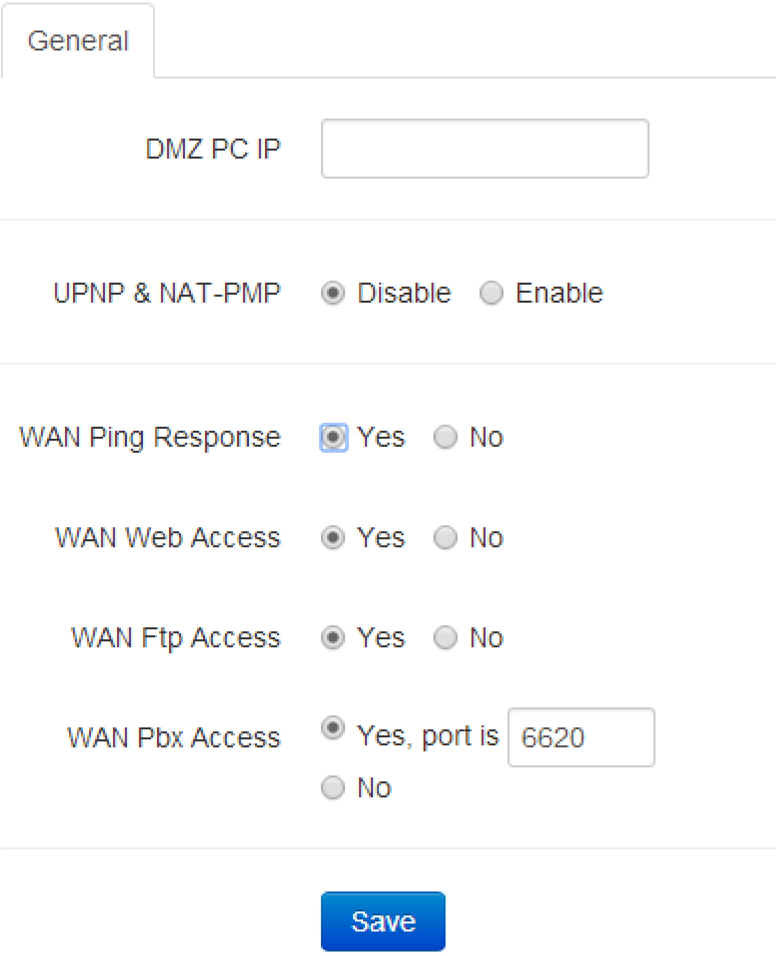




Port Forwarding Setting	
Enable	Check the box to enable the port forwarding rule
Source	**Zone:** Configure Zone setting for source. **IP:** Enter the IP address of the source device. **Port:** Enter a port or port range.
Destination	**Zone:** Configure Zone setting for destination. **IP:** Enter the IP address of the device on your local network that you want to allow the incoming service to. **Port:** Enter a port or port range for the public and private port
Protocol	Select the type of protocol you'd like to assign to the rule

Firewall/UPNP/DMZ

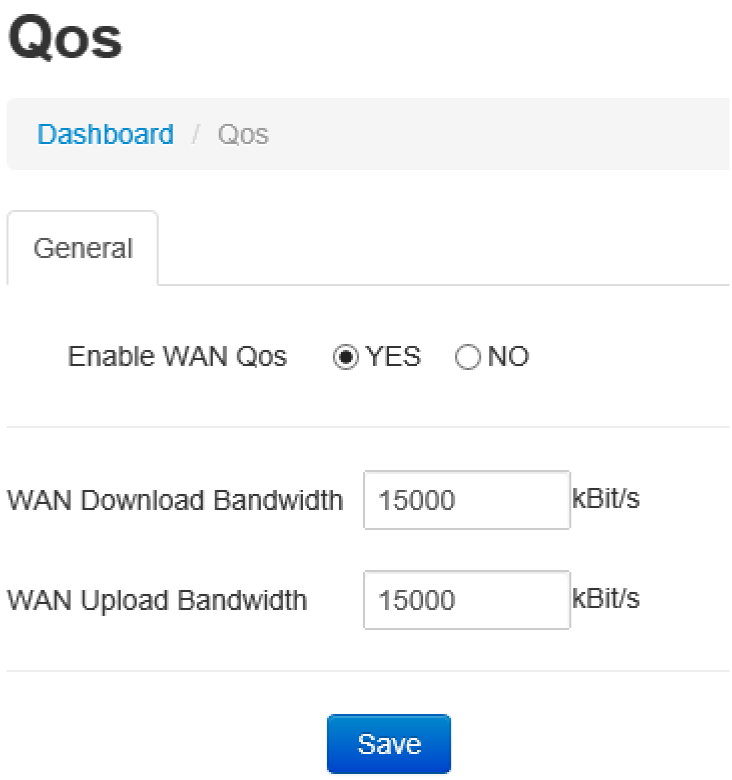
A firewall protects your network from the outside work. PBX220 provides a simple protection of your network.



Firewall, DMZ&UPNP Setting	
DMZ PC IP	Specify the IP address of the device on the LAN that you want to have unrestricted Internet communication.
UPNP&NAT-PMP	Enable UPNP&NAT-PMP or Disable UPNP&NAT-PMP
WAN Ping Response	Select Yes to enable WAN Ping Response, Select No to disable it.
WAN Ftp Access	Select Yes to enable WAN Ftp Access, Select No to disable it.
WAN Web Access	Select Yes to enable WAN Web Access, Slect No to disable it. **Note:**WAN Web Access is disable by default. If you would like to visit the WEB GUI through WAN of PBX220, please enable it at the first time when you log into the system.
WAN PBX Access	Select Yes to enable Wan PBX Access and set the access port no. , Select no to disabl e it. **Note:** Default Wan PBX Access port no. is 6620. Please change your SIP phone's SIP port no. to 6620 before you register it to PBX220.

QoS

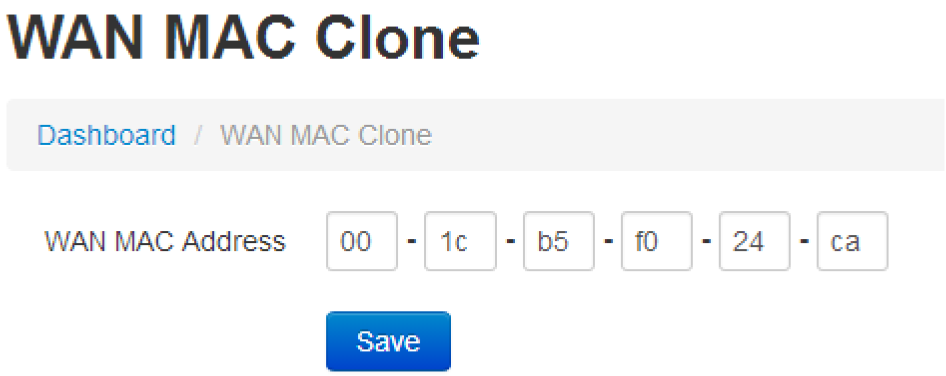
Quality of service. It ensures the voice quality in case the bandwidth is not enough. You can also set the WAN download/upload bandwidth.



QoS Setting	
Enable WAN QoS	Select Yes to enable WAN QoS, Select No to disable it. The default setting is disable.
WAN Download Bandwidth	Enter the WAN download bandwidth value.
WAN Upload Bandwidth	Enter the Wan upload bandwidth value.

WAN MAC Clone

Some ISPs require that you clone (copy) the MAC address of your computer's network card into the Router. If you are not sure then simply clone the MAC address of the computer that was originally connected to the modem before installing the Router. Cloning your MAC address will not cause any problems with your network.



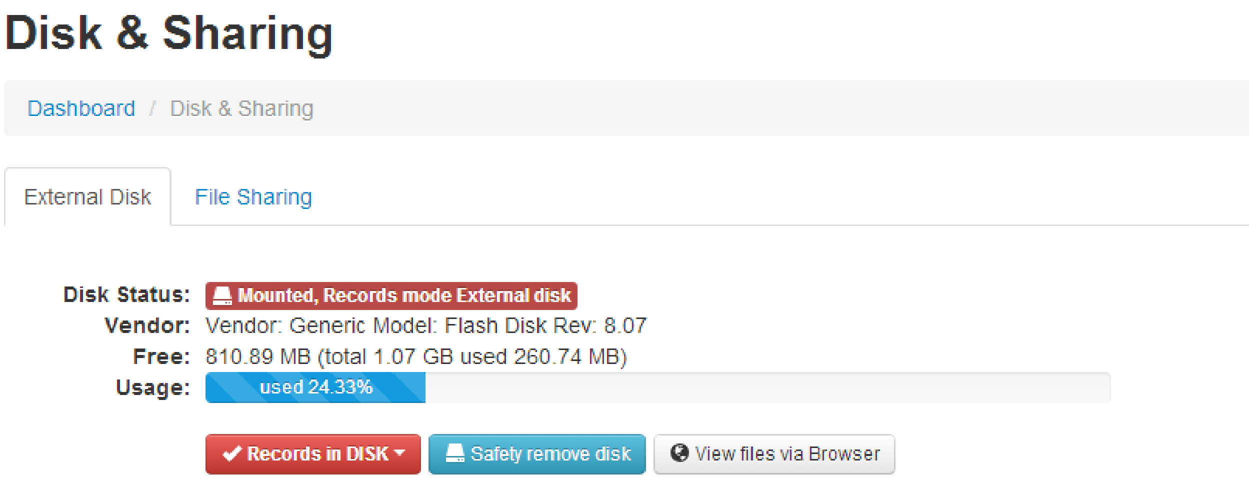
WAN MAC Clone Setting	
WAN MAC Address	Enter the MAC address you would like to clone

Disk and Sharing

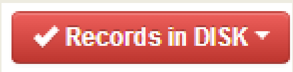
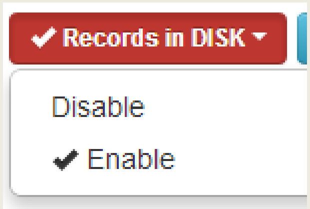
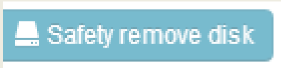
First, you need to insert a USB drive to the USB port at the left side of PBX220.

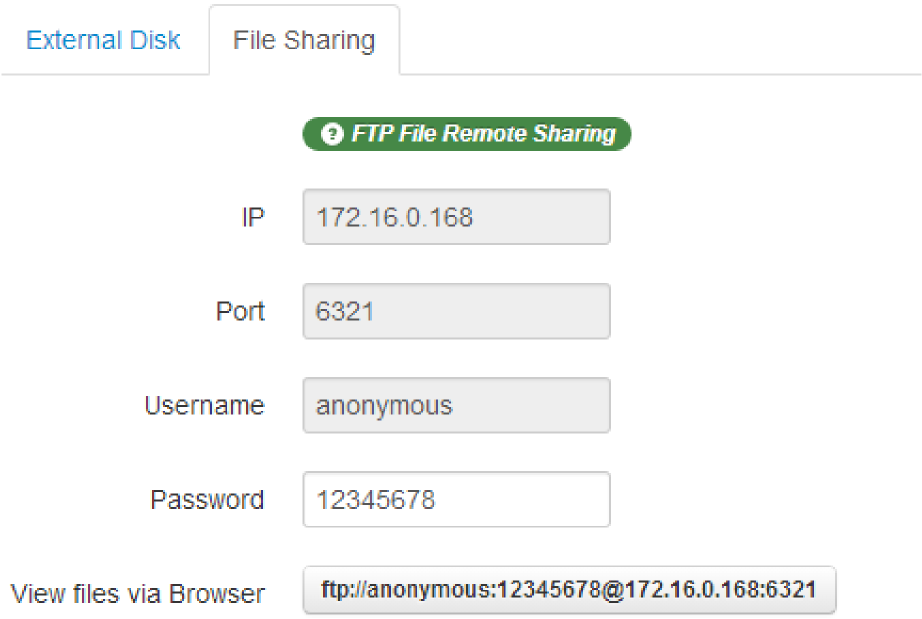
Click **Utilities-Disk and Sharing** to configure the setting of external USB drive of PBX220.

It supports FAT32, EXT4 or based the MLC USB disk.



External Disk Setting	
Disk Status	External USB drive detected  No external USB drive detected 
Vendor	Info of the external USB drive

Free	It displays the unused space of the external USB drive.
Usage	It shows the used space percentage of the external USB drive.
	Click to enable or disable records in disk.  **Note:** Records in Disk has to be enabled if you need to record calls or config voice mail of each extension.
	Click it before you remove the disk
	Click to view the files in web browser.



File Sharing Setting	
IP	FTP server IP address
Port	FTP port no.
Username	FTP access user name **Note:** Administration FTP access user name is admin
Password	Enter FTP access password. **Note:** Administration FTP access password is admin
View files via browser	Click to view the files via web browser

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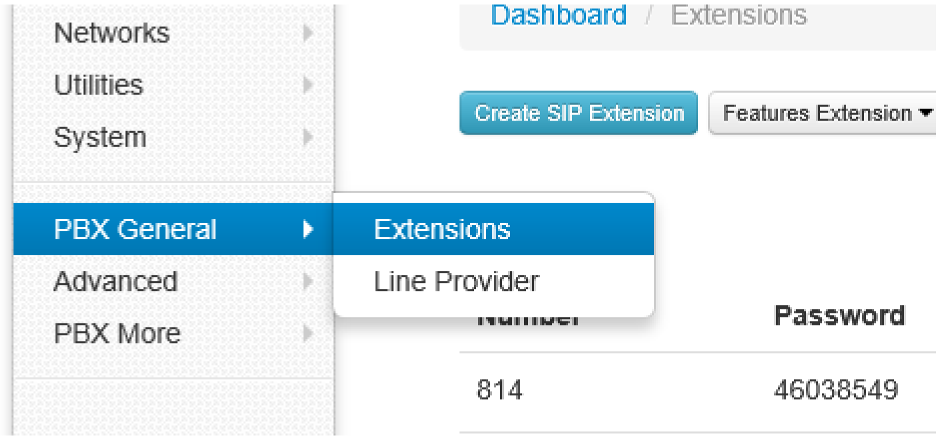
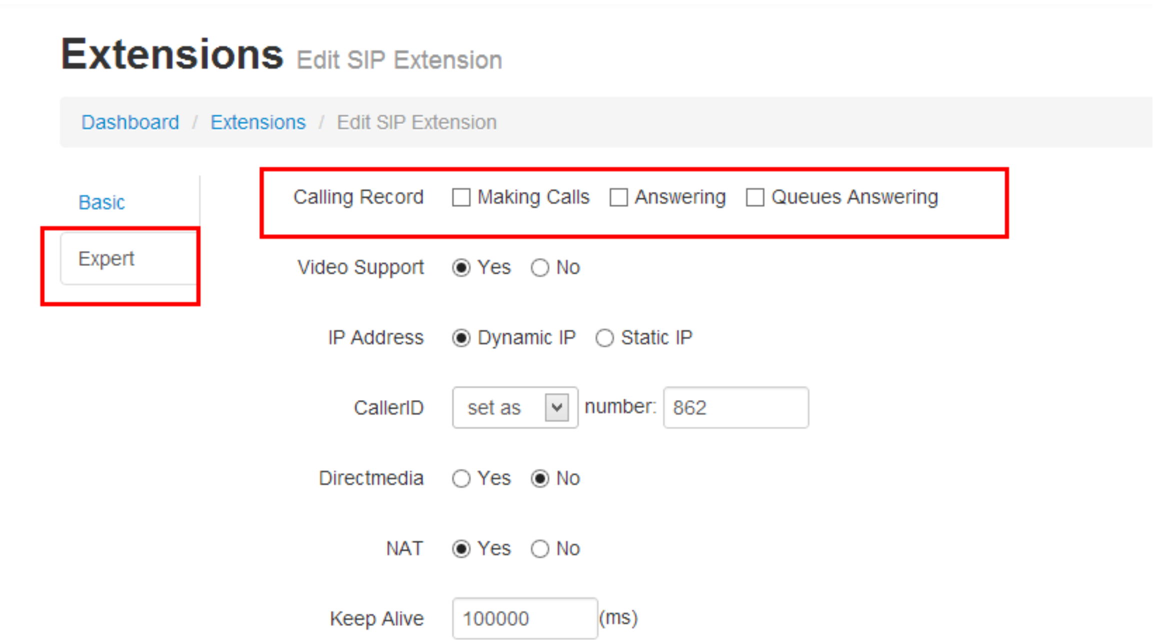
Note: You can use any standard FTP client to access the FTP server and share files. FTP IP, Port and

User name can't be changed.

Now you can find the **call recording option** in the Extensions-Extension-Expert. Recording file will be saving on USB external disk memory only.

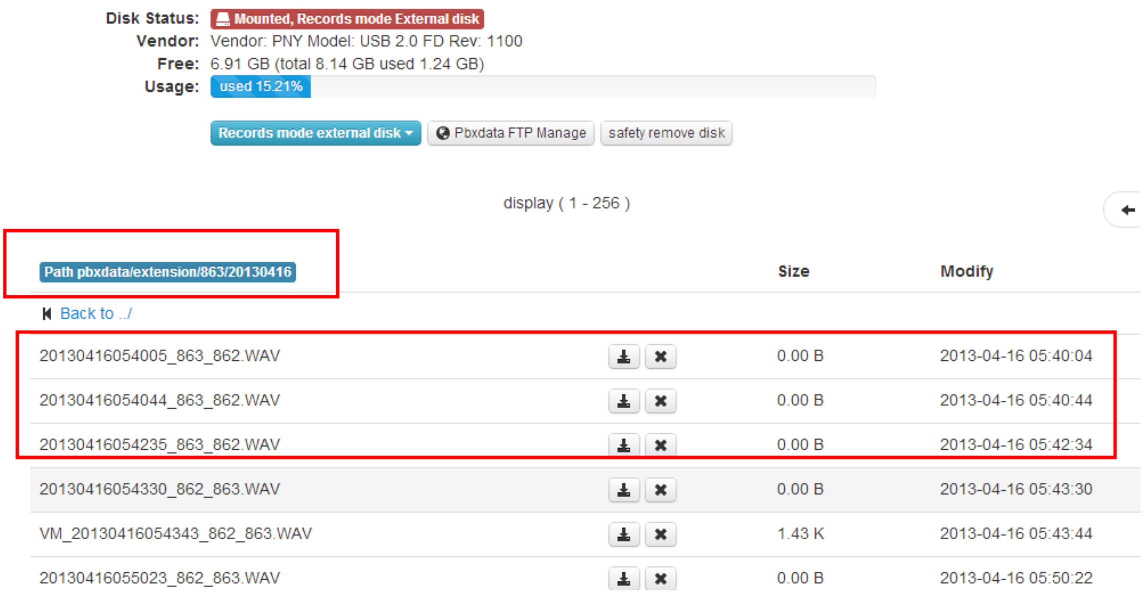


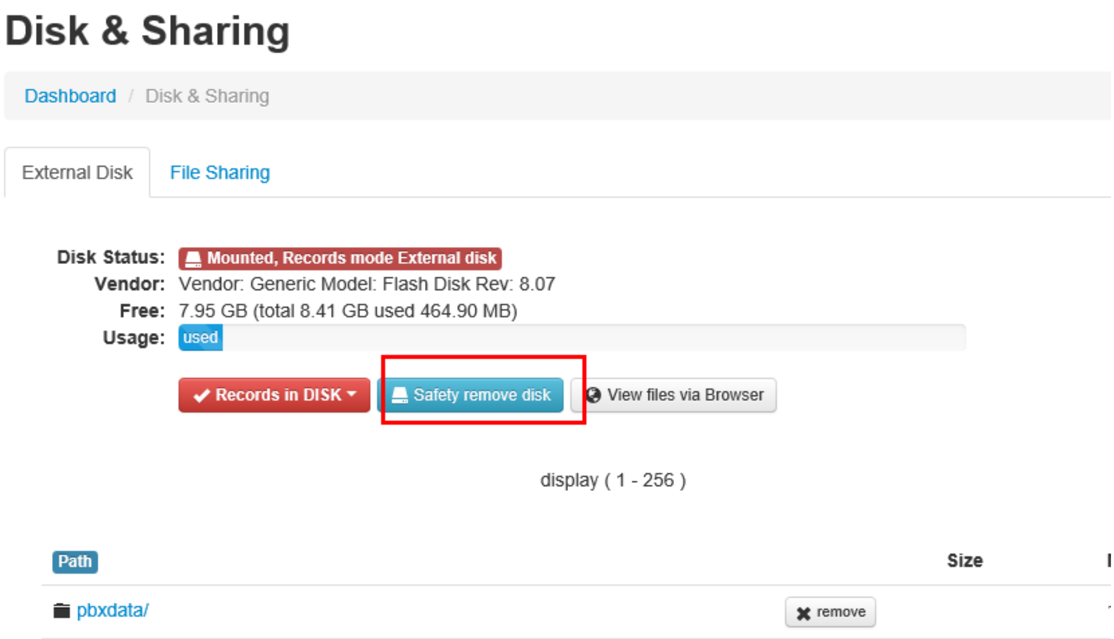


You can also check the PBX data-Extensions to find the record file.

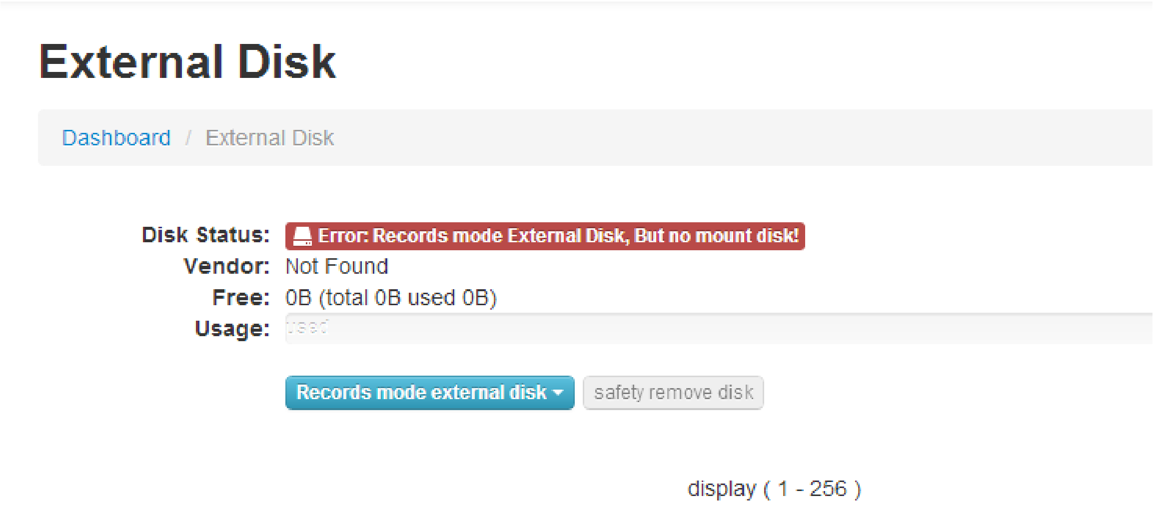
You can download and delete the files.

Or you can check them in FTP as below.





When you want to extract the USB, close all the files related to USB before click the safety remove disk and wait around 1-3 seconds. Then it will be ok as below.



File Sharing

You can treat the PBX220 as a file sharing server now.

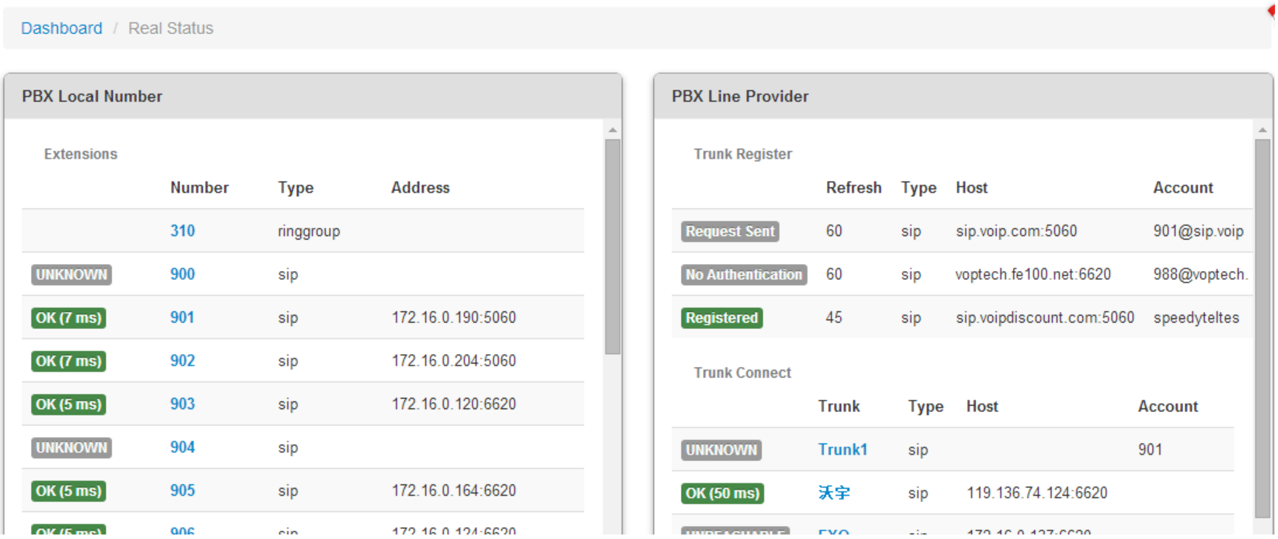
1.9 System

1.10 Real Status

The PBX220 system status can be accessed via **WEB GUI-System-Real Status,** which displays the following system information.

PBX Location number status including extension status and other source.

PBX Line provider status including trunk register and trunk connect status.



Extension Status	
Status	OK (6 ms) Registered UNKNOWN Registered failed
Number	It shows the extension number
Type	Extension number type
Address	Extension IP address
Trunk Register	
Status	Registered Registered Request Sent Register request sent, but no response from server. No Authentication Filed to authorize the user and password. Check your account and password.
Refresh	Refresh time
Type	Trunk type
Host	Host server address
Account	Trunk account
Trunk Connect	
Status	OK (6 ms) Registered UNKNOWN Failed
Trunk	Trunk name
Type	Trunk type
Host	Host server address
Account	Trunk account

Call Details Report

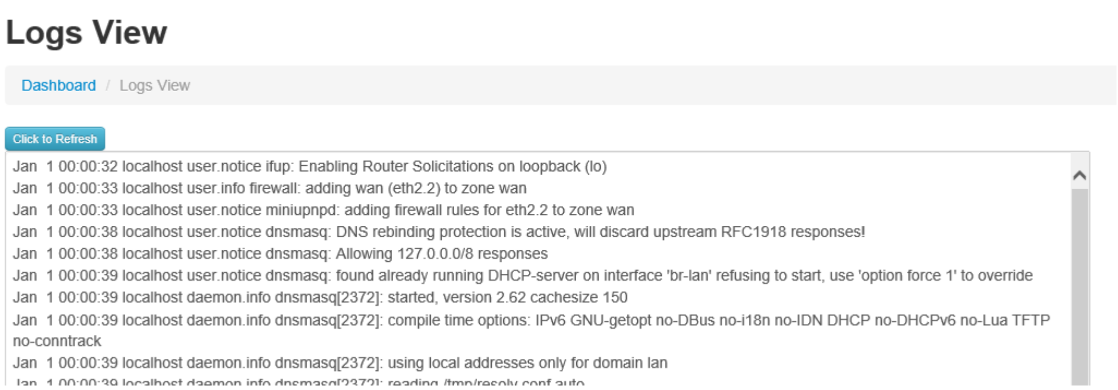
A call details record is a data record produced by telephone exchange activities or other telecommunication equipment documenting the details of a phone call that pass through the PBX. The CDR is composed of the following data files on the PBX220

Account	Source	Destination	Calldate	Duration/Answer	Status
812	812	00902122179595	2014-09-12 15:59:29	0.61/0	FAILED
812	812	00390287187252	2014-09-12 15:49:46	0.61/0	FAILED

Call Detail Record	
Account	Extension
Source	Caller ID
Destination	Callee ID
Calldate	Call Date
Duration/Answer	Duration: Call time Answer: Talk time
Status	Format NO ANSWER, BUSY, ANSWERED OR FAILED

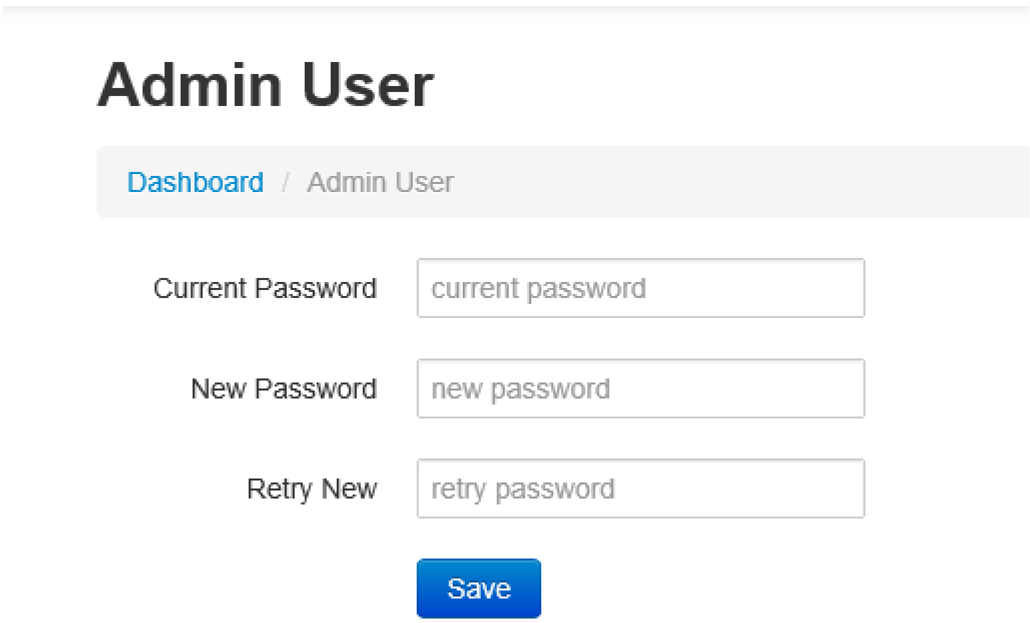
Logs View

Under **WEG GUI-System-Logs View**, you can find the system logs here. It is very convenient to check any problem. Click to Refresh, it will show the latest logs.



Admin User

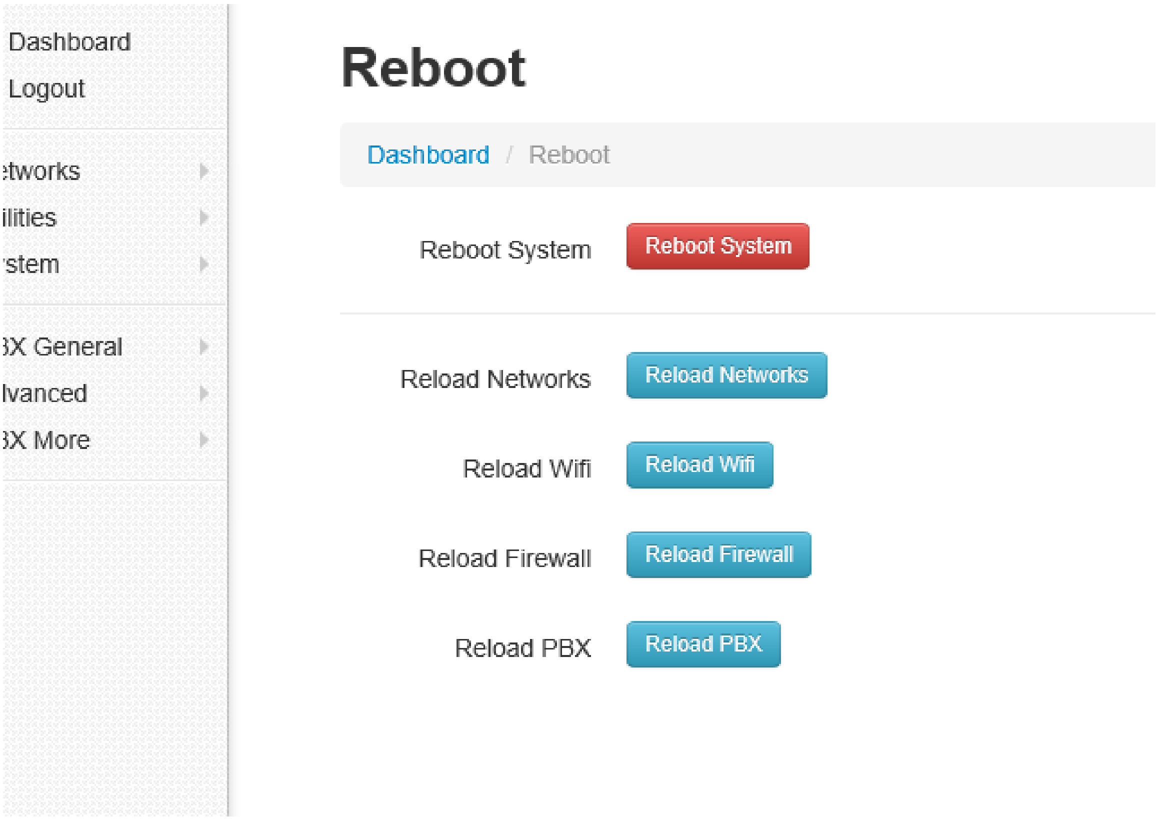
Under **WEB GUI-System-Admin User**, you can change the sign-in password here.



Reboot

User could perform reboot under **WEB GUI-System-Reboot**

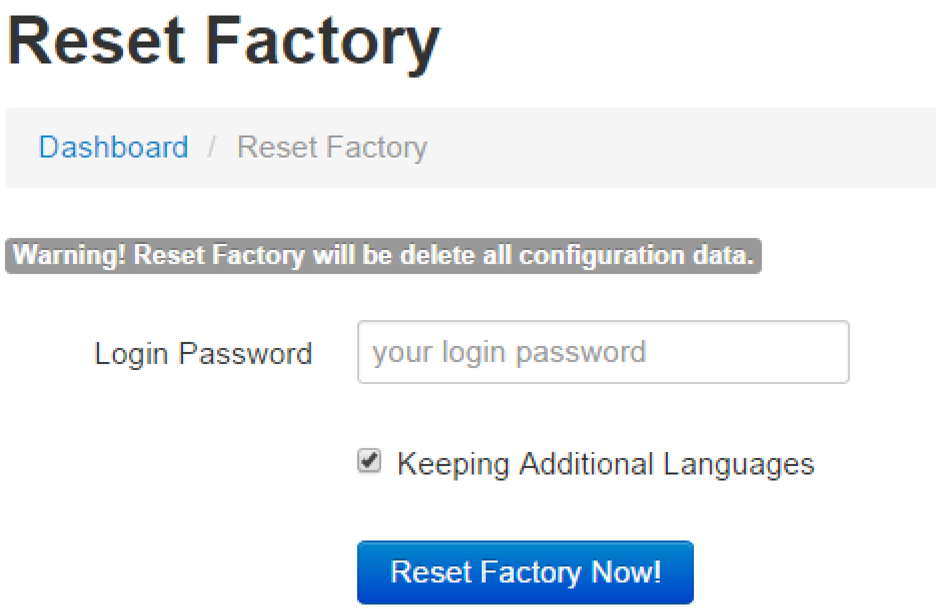
Also you can reload Networks, Wifi, Firewall, PBX.



Reset Factory

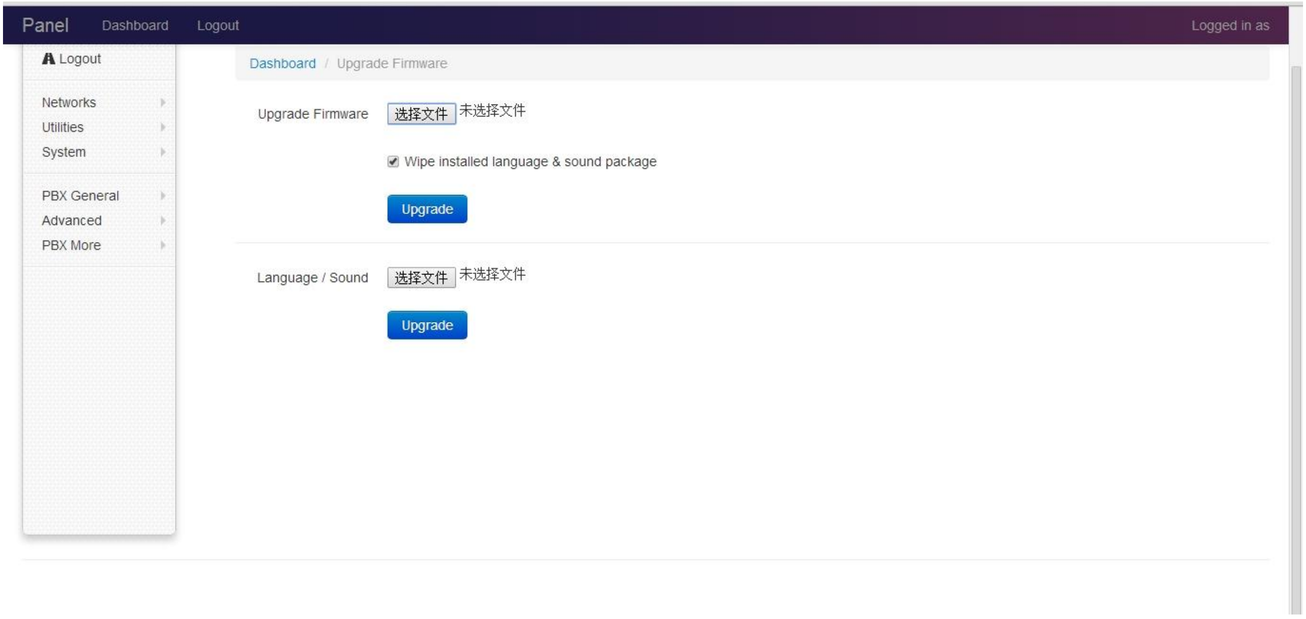
User could restore the factory default setting under **WEB GUI-Reset Factory.**

Factory Mode: Input your password to reset your device.(Admin by default)

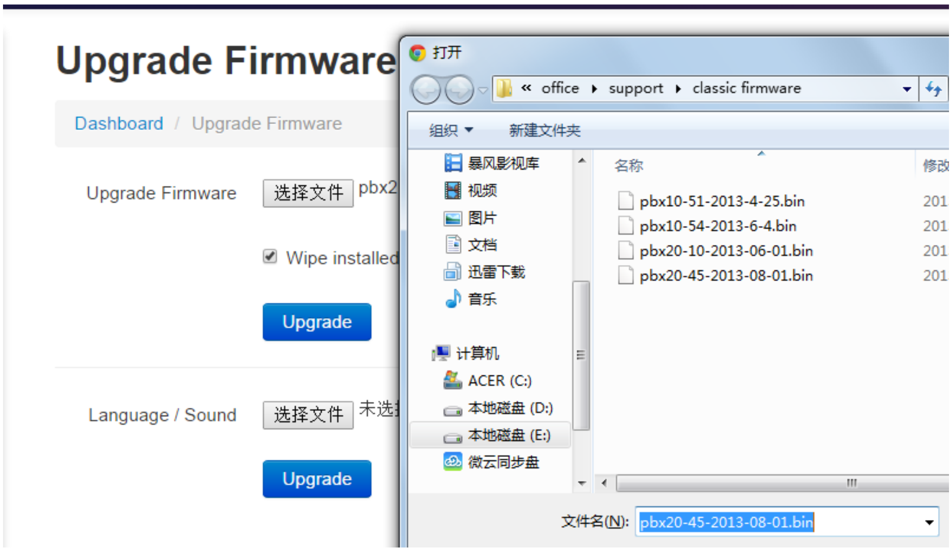


Reset Factory	
Login Password	Enter the login password to reset PBX220. Default is admin
Keeping Additional Languages	Check the box if you have additional languages to keep after reset

Upgrade Firmware



Go to WEB GUI-System-Upgrade Firmware, upgrade PBX220 firmware by clicking on 选择文件 and select the firmware file from your PC



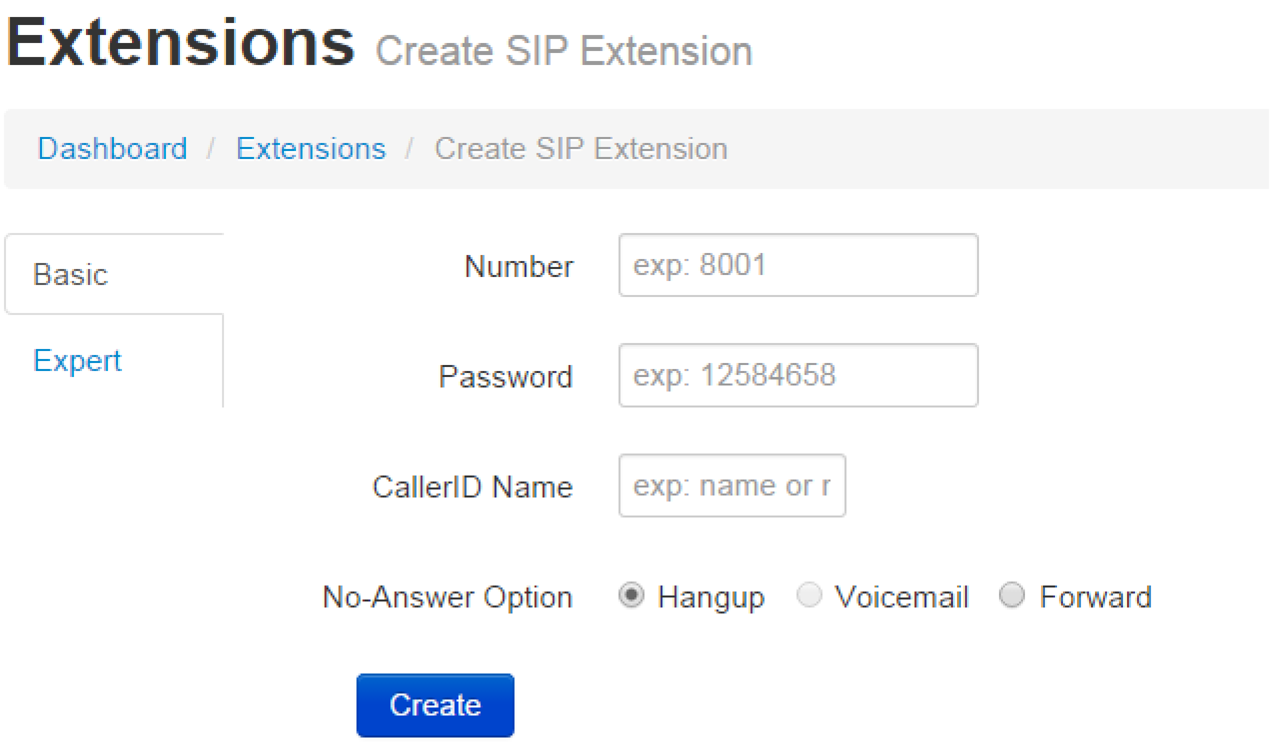
Click upgrade to start. Check ☑ Wipe installed language & sound package to wipe the current language and sound data inside PBX220.

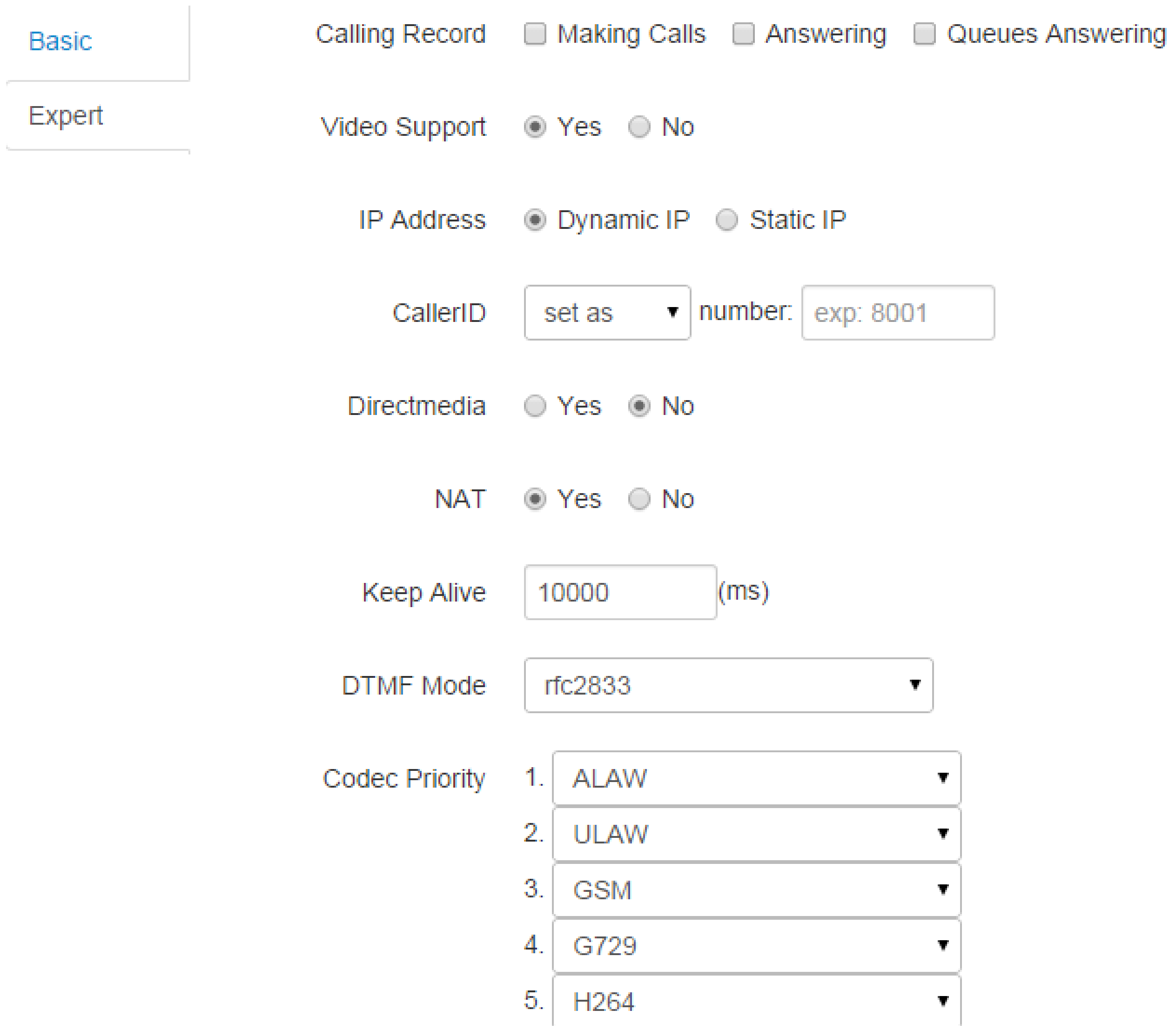
3. VoIP

1.11 PBX General

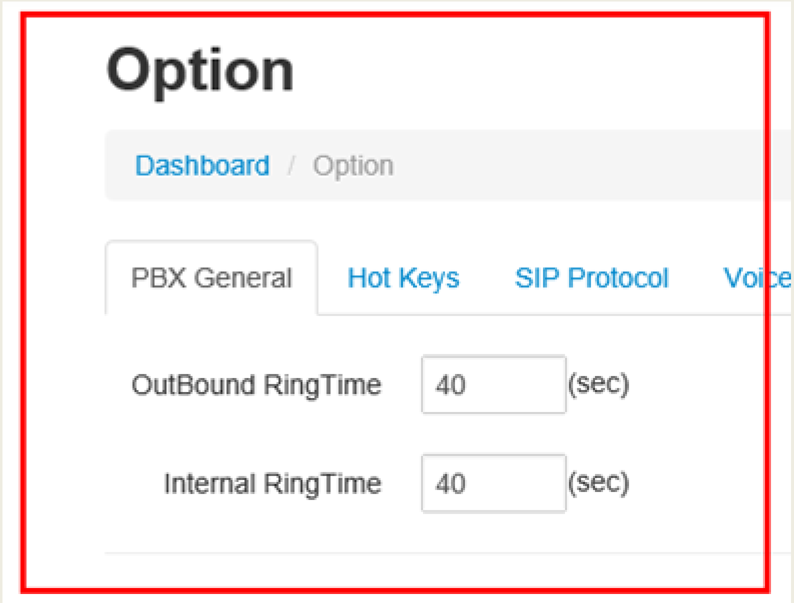
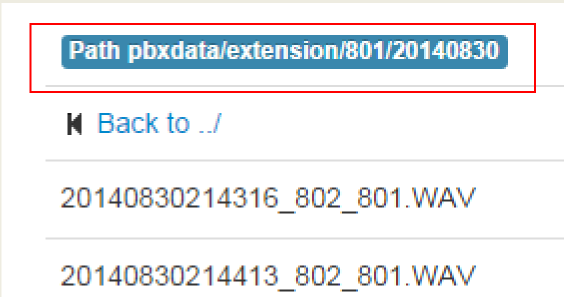
Extensions

To manually create new SIP user, go to WEB GUI-PBX More-Extension, click on "Create new SIP Extension" and a new dialog window will show for users to fill in the extension information. The configuration parameters are as follows.

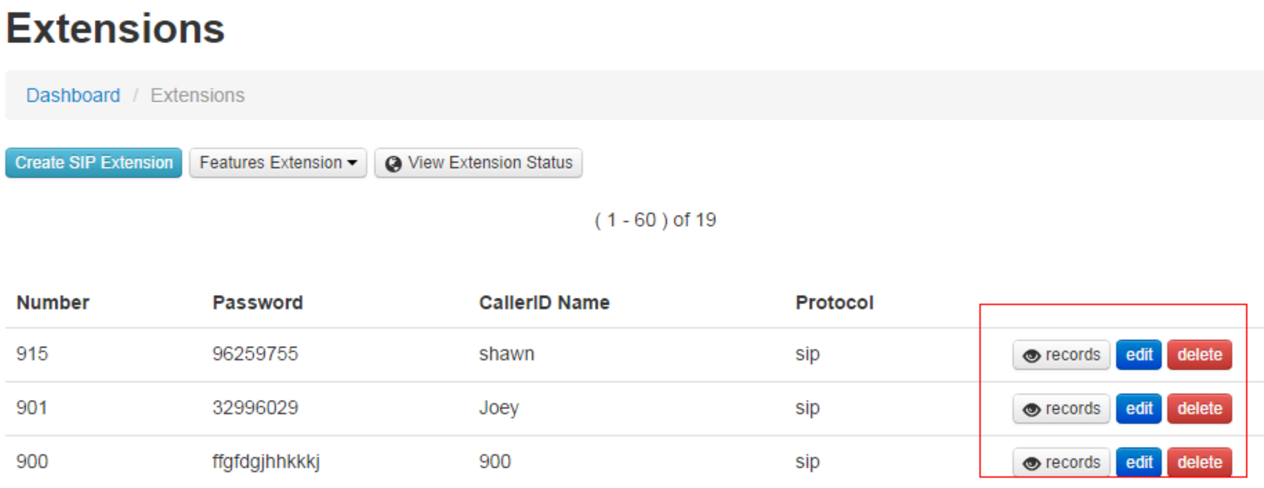




Extension Basic	
Number	The extension number associated with the user, not less than 3 digits
Password	Configure the password for the user, not less than 8 digits
CallerID Name	Configure the caller ID name that would be applied for outbound calls from this user
No-Answer Option	Hangup: if nobody answer after ring, the call will be hangup Voicemail: if nobody answer after ring, allow to leave a voice message. Forward: if nobody answer after ring, forward the call to a specify number The ring time could be configured at WEB GUI-PBX More-Option

	 **Note:Voice mail only valid when an external USB disk is mounted and record mode is external disk.**
Extension Expert	
Calling Record	Check the recording option for the user. Default is disabled. The recording files can be accessed under **WEB GUI-Utilities-Disk&Sharing-External Disk** 
Video Support	Check Yes to enable video, check no to disable video of the user
IP Address	Configure IP address option.
Caller ID	Default: use the default caller ID number Set as: configure a number as the caller ID number
Directmedia	If enabled, the voice data will not be transferred. Default setting is disable
NAT	Use NAT when the PBX220 is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports. The default setting is enabled
Keep Alive	Configure the Keep-alive interval (in milliseconds) to check if the host is up. The default setting is 10000 seconds
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Codec Priority	Select audio and video codec for the extension. The available codecs

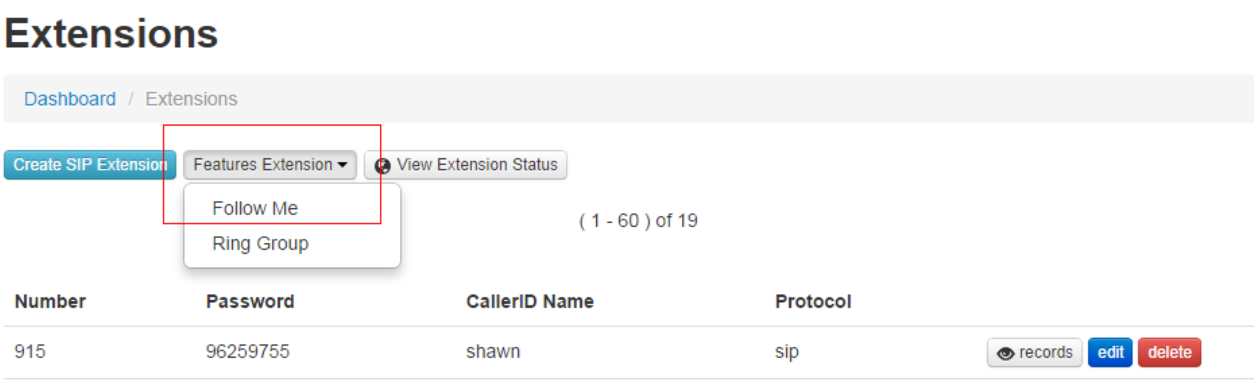
	are: GSM, G.729, G.722, ALAW, ULAW,H.264, H.263



If need to modify the setting or have a view, just click edit. To remove an extension from your system permanently, click delete. Click records, you can go to check the call record directly.

Follow Me

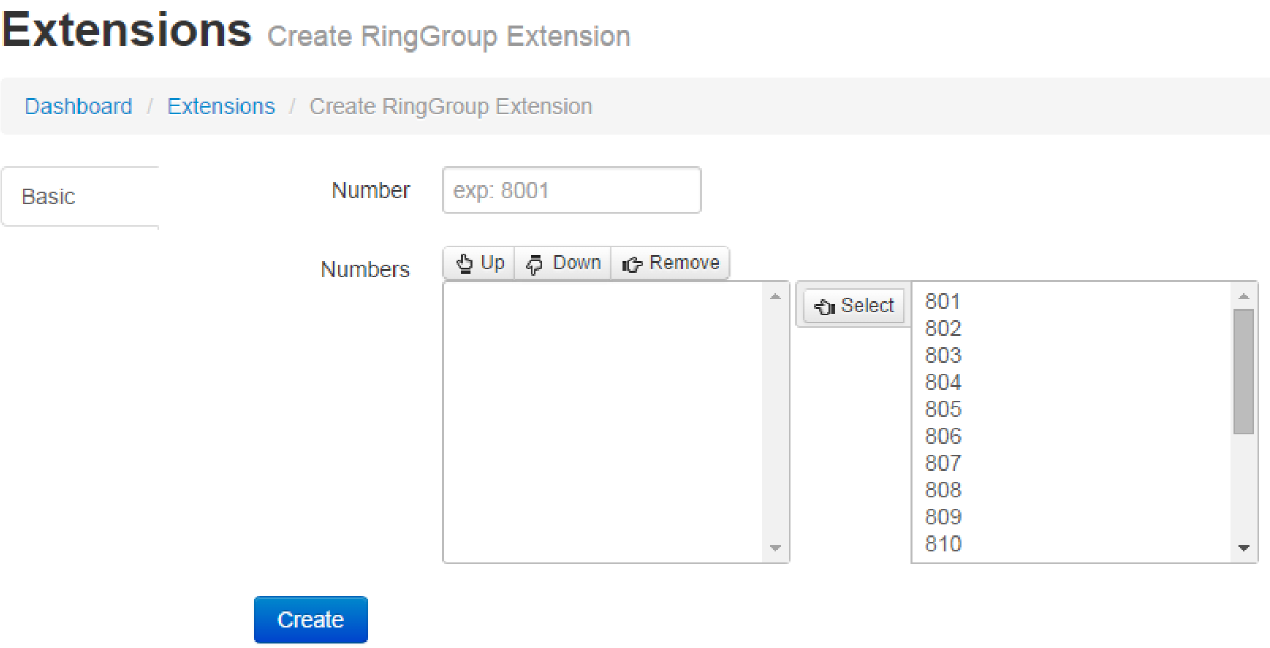
If you have more than one number, when a call comes in, the system will ring your numbers one by one until you answer it. This feature called follow me. The configuration parameters are as follows.



Follow Me	
Number	Enter the extension number for follow me
Internal Extensions	Select the extensions from the list on the right side to the left side for follow me. The order can be selected by clicking on 
External Number	Enter the external number as a forward option if the internal extensions failed to answer

Ring Group

Users could assign extensions to different ring groups to better manage the configurations on the PBX220. For example, when there is a sales hotline, users could select a group of sales extensions instead of each person's extension to assign. When a call comes in, the system will ring all extensions in the group simultaneously. This feature simplifies the configuration process and helps manage and categorize the extensions for business environment.



Ring Group	
Number	Enter the extension number for the ring group
Numbers	Select the extensions from the list on the right side to the left side for ring group

Line Provider

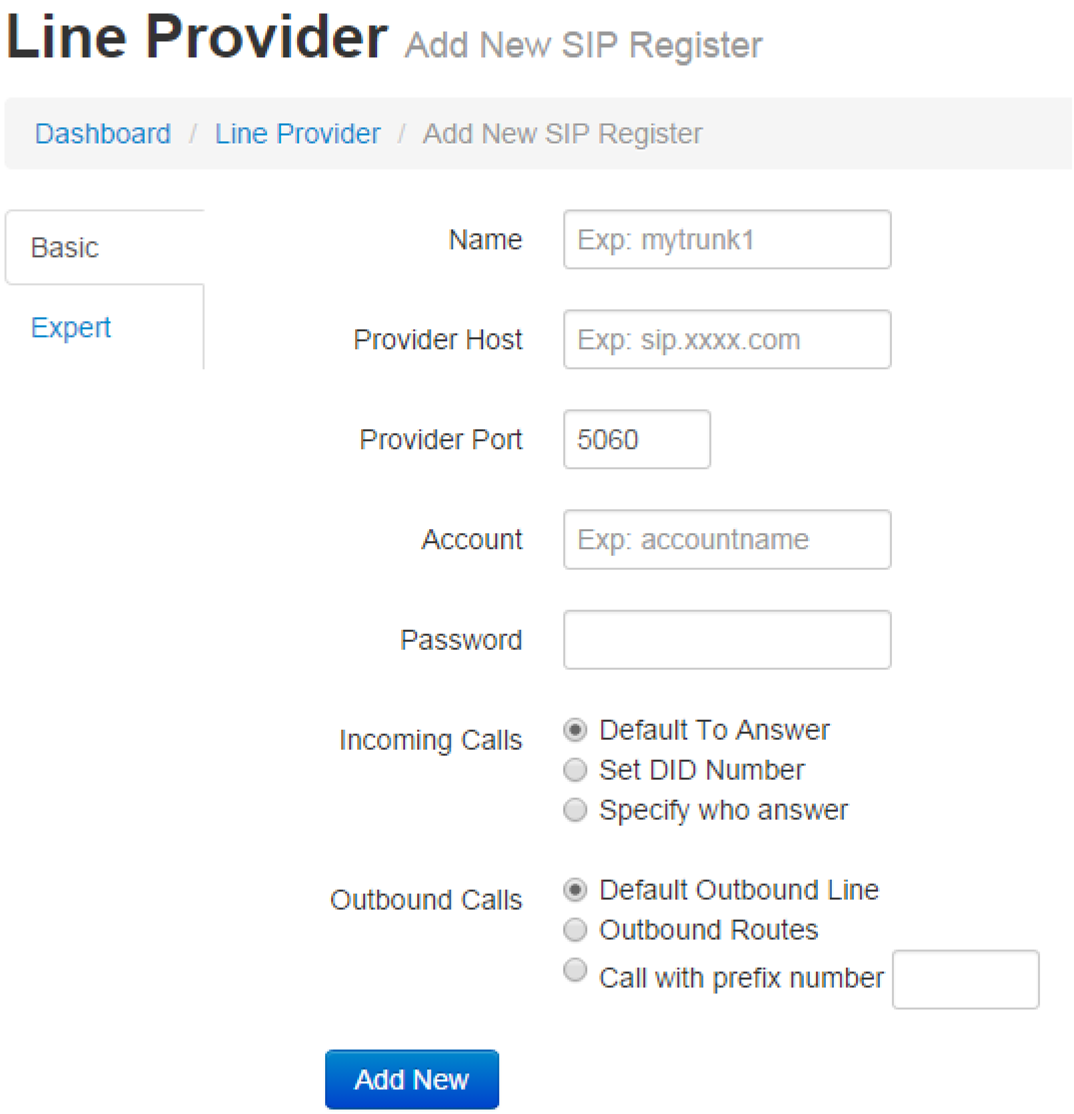
Line provider can be configured in PBX220 under **WEB GUI-PBX General-Line Provider**

Click “SIP Register” to use SIP account and password to register to Internet Telephony Service Provider.

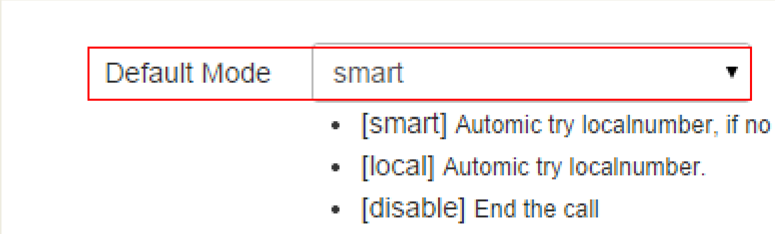
Click “SIP Direct” for directly point to point connect other SIP server and authenticate by IP and port.

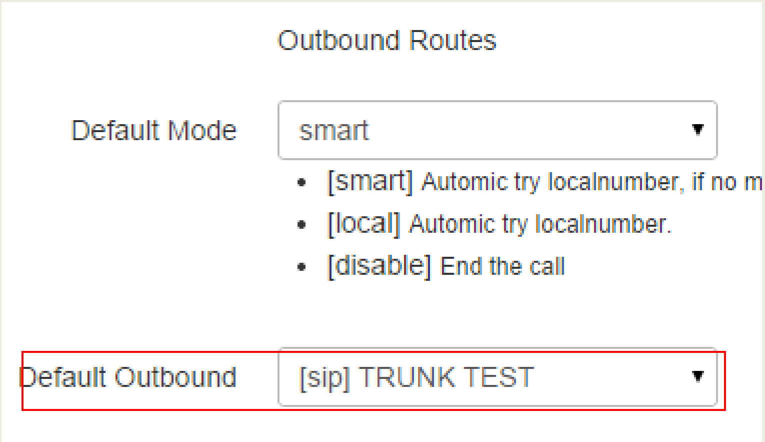
SIP Register

The SIP Register options are listed in the table below.



Line Provider Basic	
Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Provider Host	Configure the IP address or URL for the VoIP provider's server of the trunk
Provider Port	Enter the SIP port number of the trunk
Account	Enter the username to register to the trunk from the provider
Password	Enter the password to register to the trunk from the provider
Incoming Calls	Check how to process the inbound calls from the provider

<table>
<tr>
<td></td>
<td>Default to Answer: The inbound calls will be handled by the default rule. The default rule of handing the inbound calls can be set under WEB GUI-PBX More-Option


Set DID Number: Configure a DID number for the inbound calls
Specify who answer: Specify a number to answer the inbound calls</td>
</tr>
<tr>
<td>Outbound Calls</td>
<td>Check how to process the outbound calls
Default Outbound Line: Use the default outbound line. It could be configured under WEB GUI-PBX More-Option


Outbound Routes: Execute the matched outbound routes when dialing
Call with Prefix Number: Enter a prefix number to use this line when calling out</td>
</tr>
</table>

Line Provider Add New SIP Register

Dashboard / Line Provider / Add New SIP Register

Basic

Expert

From Domain

Auth. Name

Auth. Contact is ◉ UniqueID ○ Account ○ Auth. Name

Default reg expiry 60 (sec)

Failover Trunk 1 ALL-FAIL To Disable

Failover Trunk 2 ALL-FAIL To Disable

Outbound force callerid

Callee Number ◉ Default ○ Search To Field

Allow callin ◉ Yes ○ No

SIP Progress ◉ Inband ○ Outband

Keep alive 10000 (ms)

NAT ○ Yes ◉ No

Video support ◉ Yes ○ No

DTMF mode rfc2833

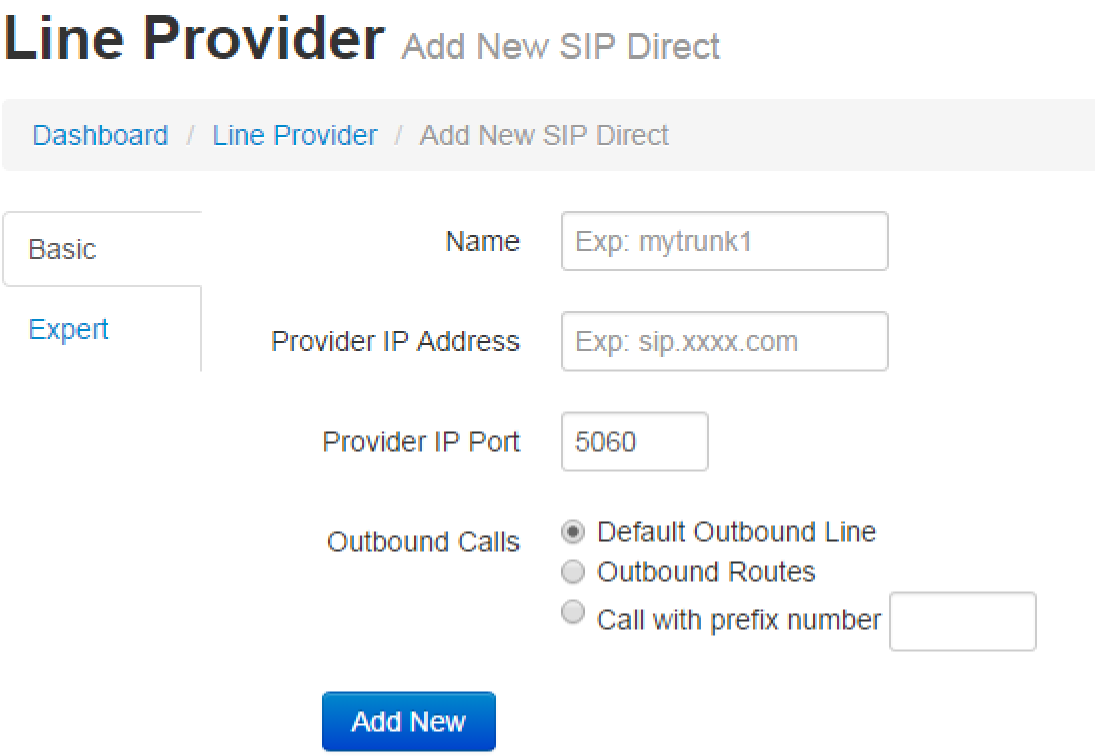
Codec priority 1. ALAW

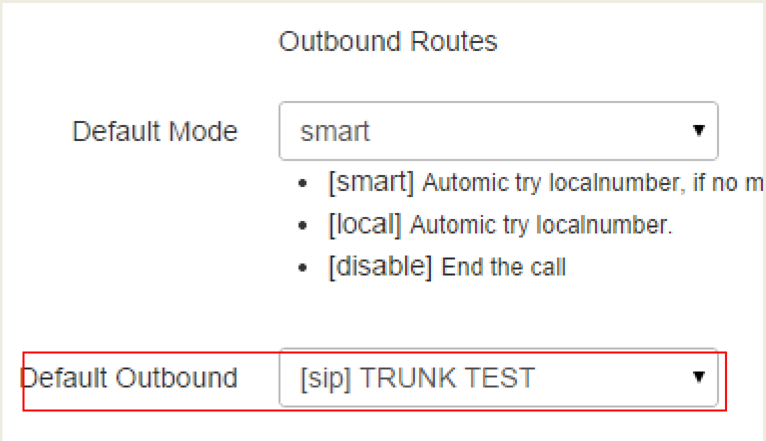
2. ULAW

Line Provider Expert	
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the From Header. For example, “trunk.pbx220.provider.com” is the from domain in from header: sip: 1234567@trunk.pbx220.provider.com
Auth. Name	This is the authentication name for the PBX220 to register to the trunk if required by the provider. If not specified, the CallerID name will be used for authentication
Auth. Contac is	Check to choose the auth. Contact
Default Reg. Expiry	Configure the refresh interval when register to the trunk
Failover Trunk 1	Configure failover trunk 1 for the current SIP trunk Busy: the current SIP trunk server returns “busy” No Answer: the current SIP trunk server returns “no answer” Cancel: the current SIP trunk server returns “cancel” Congestion: the current SIP trunk server returns “congestion” Chanunavail: the current SIP trunk server returns “Chanunavailable”
Failover Trunk 2	Configure failover trunk 2 for the current SIP trunk
Outbound force callerid	Configure the outbound force callerid.
Callee Number	Check the callee number option.
Allow Callin	Check yes to allow callin, check no to disable it
SIP Progress	Check the SIP progress option.
Keep Alive	Configure the Keep-alive interval (in milliseconds) to check if the host is up. The default setting is 10000 seconds
NAT	Check Yes to enable NAT, check No to disable it
Video Support	Check Yes to enable Video, check No to disable it
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Codec Priority	Select audio and video codec for the extension. The available codecs are: GSM, G.729, G.722, ALAW, ULAW,H.264, H.263

SIP Direct

The SIP Direct options are listed below.



SIP Direct Basic	
Name	Configure the provider name for the VoIP trunk. This is a unique label to identify the trunk when listed in outbound rules, inbound rules and etc.
Provider IP Address	Configure the IP address or URL for the VoIP provider server of the trunk.
Provider IP Port	Enter the SIP port number of the trunk
Outbound Calls	Check how to process the outbound calls Default Outbound Line: Use the default outbound line. It could be configured under **WEB GUI-PBX More-Option**  Outbound Routes: Execute the matched outbound routes when dialing Call with Prefix Number: Enter a prefix number to use this line when calling out
Line Provider Expert	
From Domain	Configure the actual domain name where the extension comes from. This can be used to override the From Header. For example, “trunk.pbx220.provider.com” is the from domain in from header: sip: 1234567@trunk.pbx220.provider.com

Default Reg. Expiry	Configure the refresh interval when register to the trunk
Failover Trunk 1	Configure failover trunk 1 for the current SIP trunk Busy: the current SIP trunk server returns “busy” No Answer: the current SIP trunk server returns “no answer” Cancel: the current SIP trunk server returns “cancel” Congestion: the current SIP trunk server returns “congestion” Chanunavail: the current SIP trunk server returns “Chanunavailable”
Failover Trunk 2	Configure failover trunk 2 for the current SIP trunk
Outbound force callerid	Configure the outbound force callerid.
Allow Callin	Check yes to allow callin, check no to disable it
SIP Progress	Check the SIP progress option.
Keep Alive	Configure the Keep-alive interval (in milliseconds) to check if the host is up. The default setting is 10000 seconds
NAT	Check Yes to enable NAT, check No to disable it
Video Support	Check Yes to enable Video, check No to disable it
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC2833". If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit PCMU and PCMA are required. When "Auto" is selected, RFC2833 will be used if offered, otherwise "Inband" will be used.
Codec Priority	Select audio and video codec for the extension. The available codecs are: GSM, G.729, G.722, ALAW, ULAW,H.264, H.263

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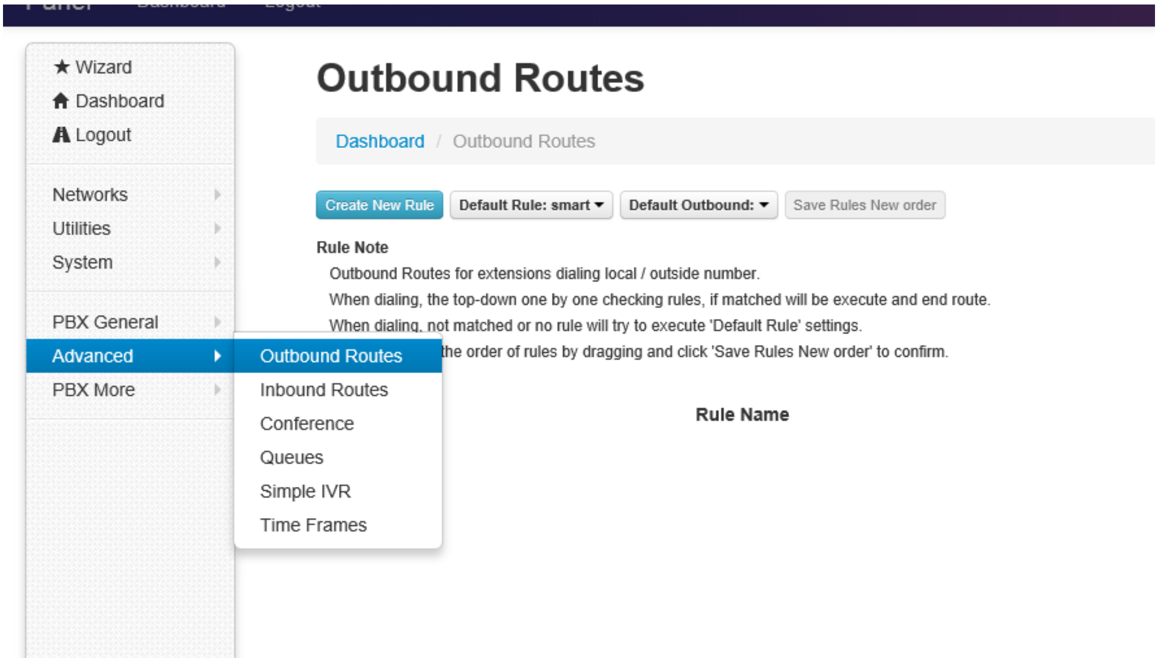
1.12 Advanced

Outbound Routes



Outbound Routes for extensions dialing local / outside number. Go to **WEB GUI-Advanced-Outbound Routes** to add and edit outbound routes.

Click **"Create New Rule"** to add a new outbound route.

Click **"Default Rule"** to select the options for default rule.

[Smart]Automatically try local number, if not matched then try to select [Default Outbound] and call out.

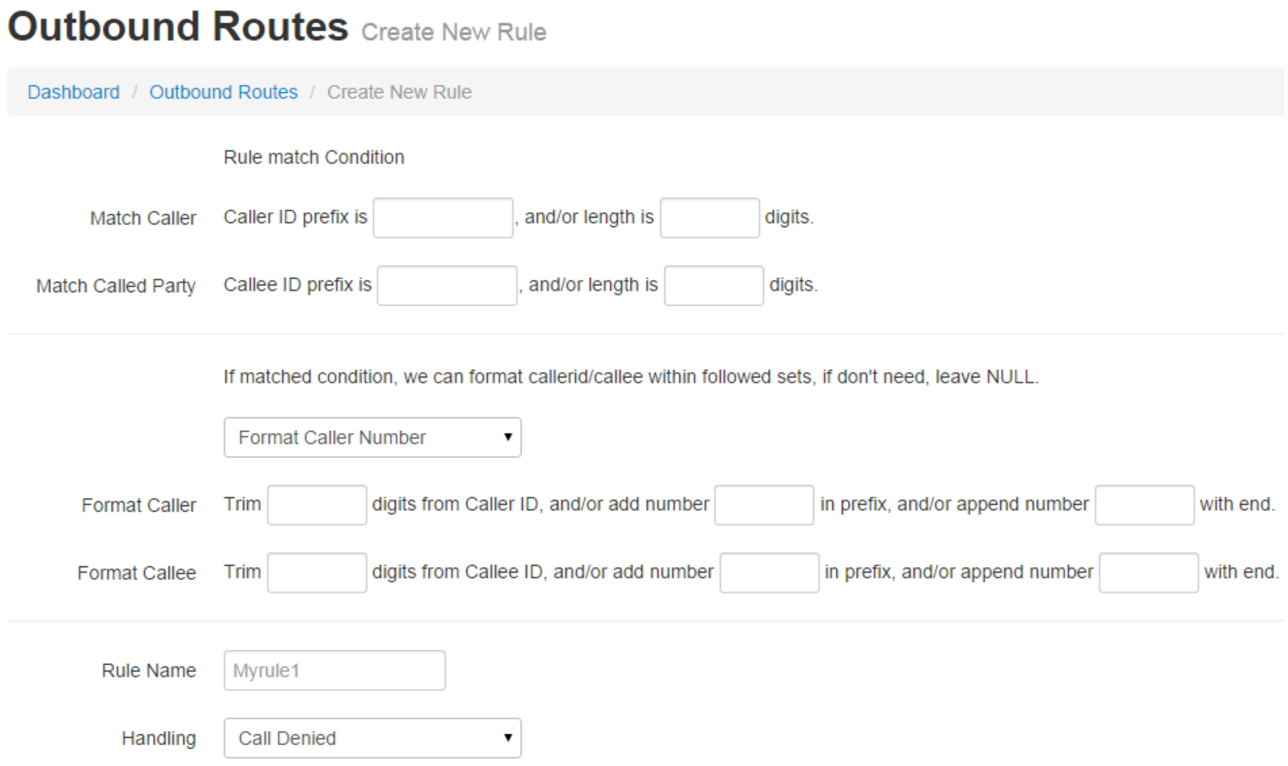
[Local] Automatically try local number

[Disable]End the call

Click **"Default Outbound"** to select the default outbound trunk.

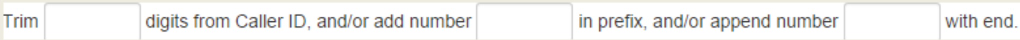
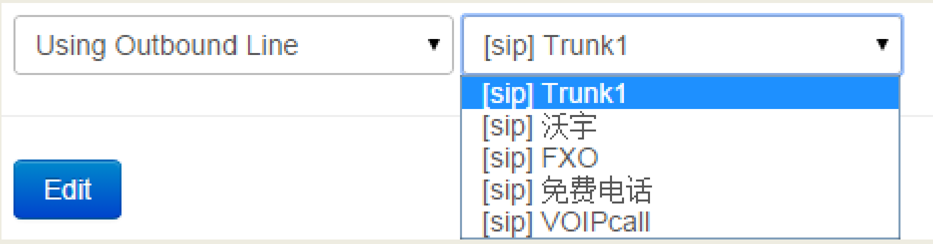
Click **"Save rules new order"** to confirm the order changes of the rules by dragging them on the page. The outbound rule listed on the top has higher priority.

Create New Rule first:

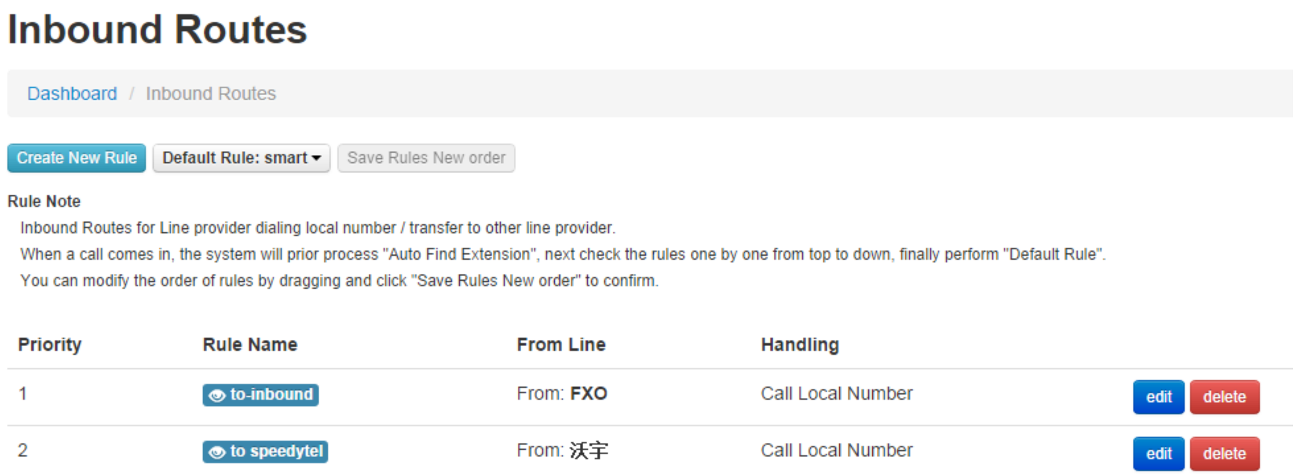


Outbound Routes	
Rule Match Condition	The rule can be matched by caller or by called party. Caller ID prefix is [ ], and/or length is [ ] digits. Examples by caller: 1. Configure caller ID prefix is 9, leave length as Null. When the caller ID with prefix 9, the condition is matched no matter how many digits it has 2. Configure length is 4, leave caller ID prefix as Null. When the caller ID is 4 digits, the condition is matched no matter what prefix it has 3. Configure caller ID prefix is 9 and length is 4. When the caller ID with prefix 9 and has 4 digits, the condition is matched Callee ID prefix is [ ], and/or length is [ ] digits. Examples by callee: 1. Configure callee ID prefix is 9, leave length as Null. When the callee ID with prefix 9, the condition is matched no matter how many digits it has 2. Configure length is 4, leave callee ID prefix as Null. When the callee ID is 4 digits, the condition is matched no matter what prefix it has 3. Configure callee ID prefix is 9 and length is 4. When the callee ID with prefix 9 and has 4 digits, the condition is matched
CallerID Format Option	Once the condition is matched, select the format option at the drop down list if necessary. Default setting is format caller number.

	Format caller number: Once the condition is matched, only format the caller number Format caller name: Once the condition is matched, only format the caller name Format caller both: Once the condition is matched, format both the number and name of the caller
Caller&Callee Format Configuration	Format caller at **“Format caller number”** option Trim [ ] digits from Caller ID, and/or add number [ ] in prefix, and/or append number [ ] with end. Example: 1. Configure Trim 1 digits from caller ID, leave other settings as Null. When you dial number from extension 910 and extension name is May, caller ID number 10 and name May will be sent to the called party. 2. Configure add number 8 in prefix, leave other settings as Null. When you dial number from extension 910 and name May, caller ID number 8910 and name May will be sent to the called party. 3. Configure append number 8 with end, leave other settings as Null. When you dial number from extension 910 and May, caller ID number 9108 and name May will be sent to the called party 4. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When you dial number from extension 910 and name May, caller ID number 8108 and name May will be sent to the called party Format caller at **“Format caller name”** option Example: 1. Configure Trim 1 digits from caller ID, leave other settings as Null. When you dial number from extension 910 and extension name is May, caller ID number 910 and name ay will be sent to the called party. 2. Configure add number 8 in prefix, leave other settings as Null. When you dial number from extension 910 and name May, caller ID number 910 and name 8May will be sent to the called party. 3. Configure append number 8 with end, leave other settings as Null. When you dial number from extension 910 and May, caller ID number 910 and name May8 will be sent to the called party 4. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When you dial number from extension 910 and name May, caller ID number 910 and name 8ay8 will be sent to the called party. Format caller at **“Format caller both”** option Example: 1. Configure Trim 1 digits from caller ID, leave other settings as Null. When you dial number from extension 910 and extension name is May, caller ID number 10 and name ay will be sent to the called party. 2. Configure add number 8 in prefix, leave other settings as Null. When you dial number from extension 910 and name May, caller ID number 8910 and name 8May will be sent to the called party. 3. Configure append number 8 with end, leave other settings as Null. When you dial

	number from extension 910 and May, caller ID number 9108 and name May8 will be sent to the called party 4. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When you dial number from extension 910 and name May, caller ID number 8108 and name 8ay8 will be sent to the called party. **Format callee**  Example: 1. Configure Trim 1 digits from callee ID, leave other settings as Null. When you dial 810, number 10 will be dialed. 2. Configure add number 8 in prefix, leave other settings as Null. When you dial 810, number 8810 will be dialed. 3. Configure append number 8 with end, leave other settings as Null. When you dial 810, number 8108 will be dialed. 4. Configure Trim 1 digits from callee ID, and add number 8 in prefix and append number 8 with end. When you dial number 810, number 8108 will be dialed.
Rule Name	Configure the name of outbound route
Handling	Call Deny: Deny the call Call Local Number: Call local number in PBX220 Using Outbound Line: Select the outbound trunk for the call 

Inbound Routes



Inbound Routes for Line provider dialing local number / transfer to other line provider. Go to **WEB GUI-Advanced-Inbound Routes** to add and edit inbound routes.

Click **“Create New Rule”** to add a new inbound route.

Click **“Default Rule”** to select the options for default rule.

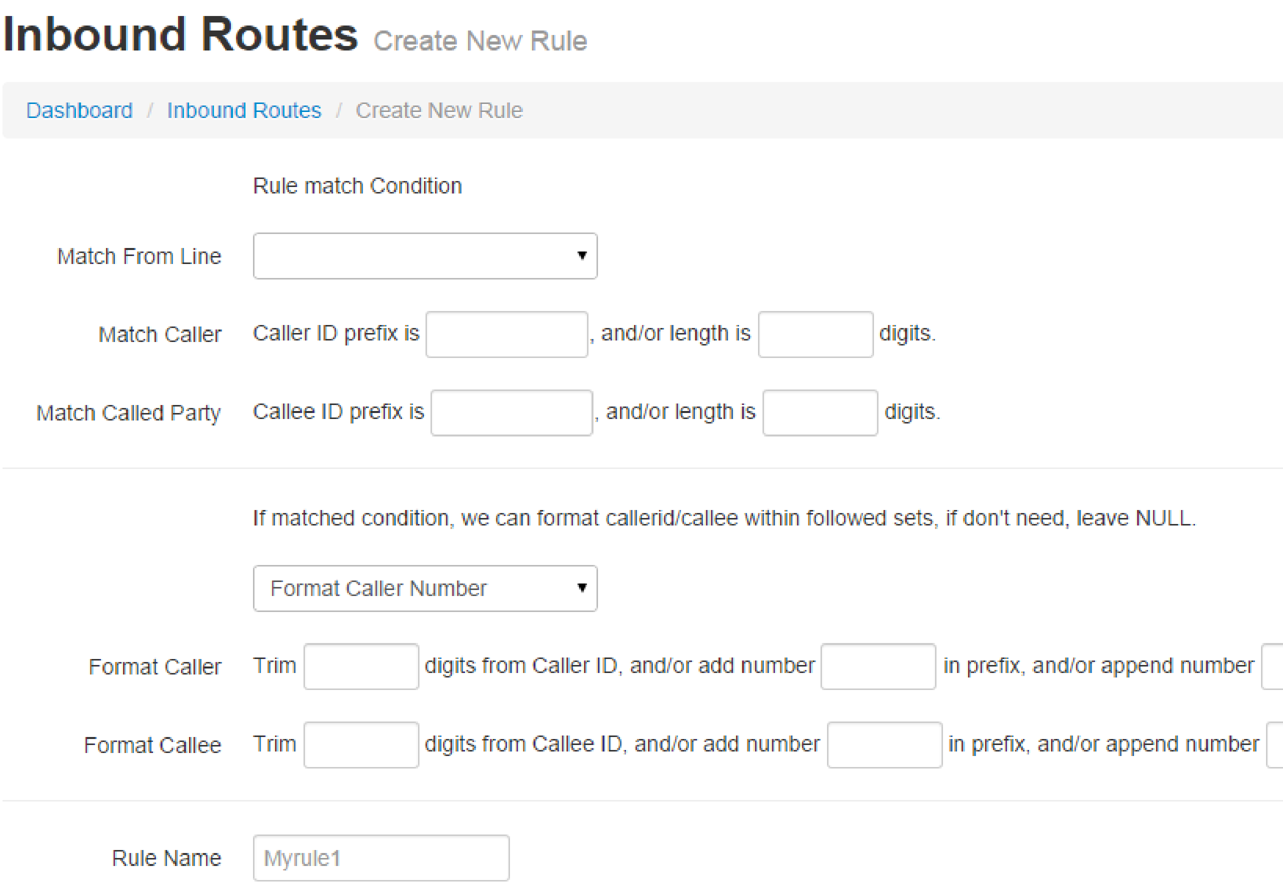
[Smart]Automatically try local number, if not matched then try to call 310 as the extension.

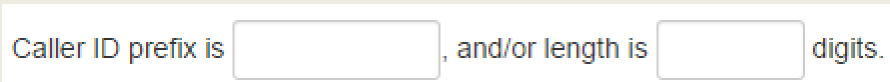
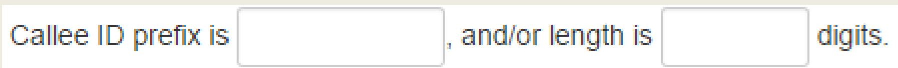
[Local] Automatically try local number

[Disable]End the call

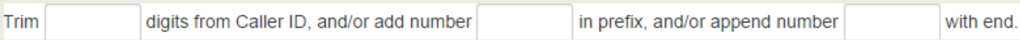
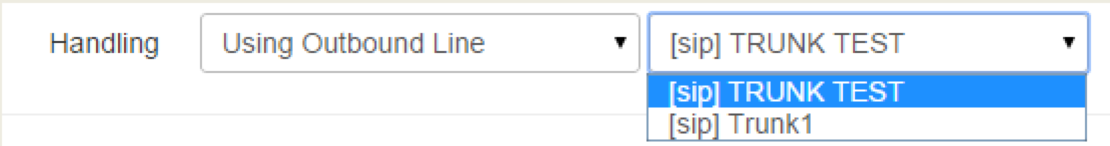
Click **“Save rules new order”** to confirm the order changes of the rules by dragging them on the page. The inbound rule listed on the top has higher priority.

Create a new rule as below.



Inbound Routes	
Match From Line	Select the SIP trunk line to configure the inbound route.
Rule Match Condition	The rule can be matched by caller or by called party.  Examples by caller: 1. Configure caller ID prefix is 9, leave length as Null. When the caller ID with prefix 9, the condition is matched no matter how many digits it has 2. Configure length is 4, leave caller ID prefix as Null. When the caller ID is 4 digits, the condition is matched no matter what prefix it has 3. Configure caller ID prefix is 9 and length is 4. When the caller ID with prefix 9 and has 4 digits, the condition is matched  Examples by callee: 3. Configure callee ID prefix is 9, leave length as Null. When the callee ID with prefix 9, the condition is matched no matter how many digits it has 4. Configure length is 4, leave callee ID prefix as Null. When the callee ID is 4 digits, the condition is matched no matter what prefix it has

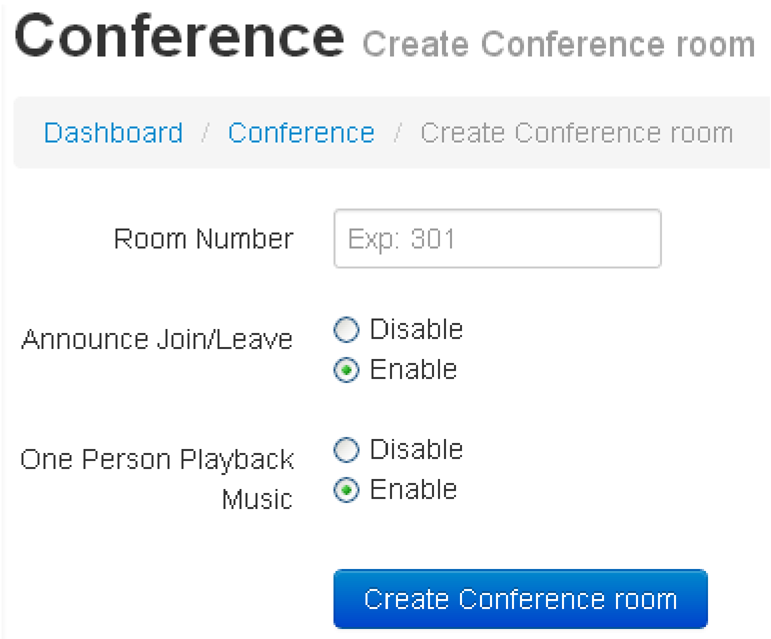
	3. Configure callee ID prefix is 9 and length is 4. When the callee ID with prefix 9 and has 4 digits, the condition is matched
CallerID Format Option	Once the condition is matched, select the format option at the drop down list if necessary. Default setting is format caller number. Format caller number: Once the condition is matched, only format the caller number Format caller name: Once the condition is matched, only format the caller name Format caller both: Once the condition is matched, format both the number and name of the caller
Caller&Callee Format Configuration	Format caller at **“Format caller number”** option Trim [ ] digits from Caller ID, and/or add number [ ] in prefix, and/or append number [ ] with end. Example: 5. Configure Trim 1 digits from caller ID, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 10 and name May will be sent to the called party. 6. Configure add number 8 in prefix, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 8910 and name May will be sent to the called party. 7. Configure append number 8 with end, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 9108 and name May will be sent to the called party 8. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When the inbound caller ID is 910 and name May, caller ID number 8108 and name May will be sent to the called party Format caller at **“Format caller name”** option Example: 5. Configure Trim 1 digits from caller ID, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 910 and name ay will be sent to the called party. 6. Configure add number 8 in prefix, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 910 and name 8May will be sent to the called party. 7. Configure append number 8 with end, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 910 and name May8 will be sent to the called party 8. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When the inbound caller ID is 910 and name is May, caller ID number 910 and name 8ay8 will be sent to the called party. Format caller at **“Format caller both”** option Example: 5. Configure Trim 1 digits from caller ID, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 10 and name ay will be sent to the called party. 6. Configure add number 8 in prefix, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 8910 and name 8May will be sent to the called party. 7. Configure append number 8 with end, leave other settings as Null. When the inbound caller ID is 910 and name is May, caller ID number 9108 and name May8 will be sent to the called party

	8. Configure Trim 1 digits from caller ID, and add number 8 in prefix and append number 8 with end. When the inbound caller ID is 910 and name is May, caller ID number 8108 and name 8ay8 will be sent to the called party. **Format callee**  Example: 5. Configure Trim 1 digits from callee ID, leave other settings as Null. When callee ID is 810, number 10 will be dialed. 6. Configure add number 8 in prefix, leave other settings as Null. When callee ID is 810, number 8810 will be dialed. 7. Configure append number 8 with end, leave other settings as Null. When callee ID 810, number 8108 will be dialed. 8. Configure Trim 1 digits from callee ID, and add number 8 in prefix and append number 8 with end. When callee ID is 810, number 8108 will be dialed.
Rule Name	Configure the name of inbound route
Handling	Call Deny: Deny the call Call Local Number: Call local number in PBX220 Using Outbound Line: Select the outbound trunk for the call 

Conference

PBX220 supports conference room allowing four parties to hold a phone conference. The conference configurations can be accessed under **WEG GUI-Advanced-Conference**.

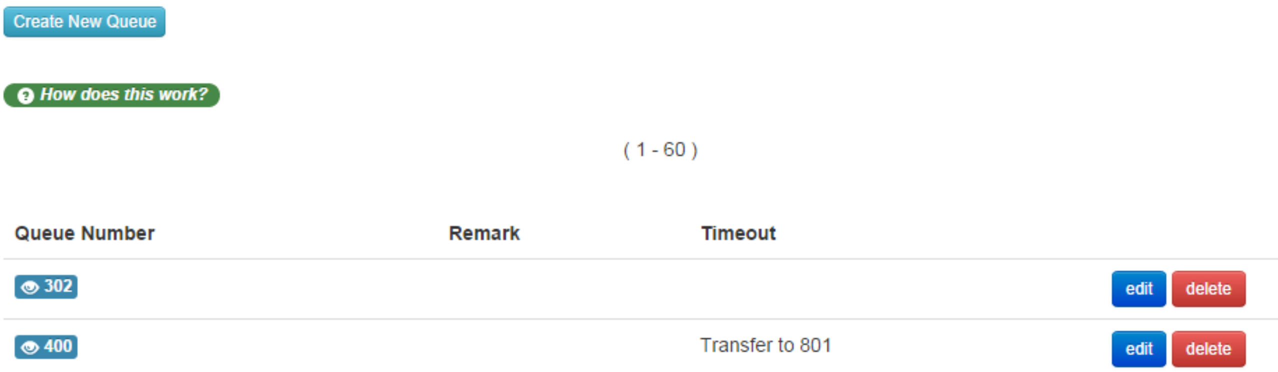
Click “Create Conference Room” to add a new conference room.



Conference	
Room Number	Configure the conference number for the users to dial into the conference.
Announce Join/Leave	Check to disable or enable the announcement when join and leave of the conference
One Person Playback Music	Check to disable or enable the one person play back music

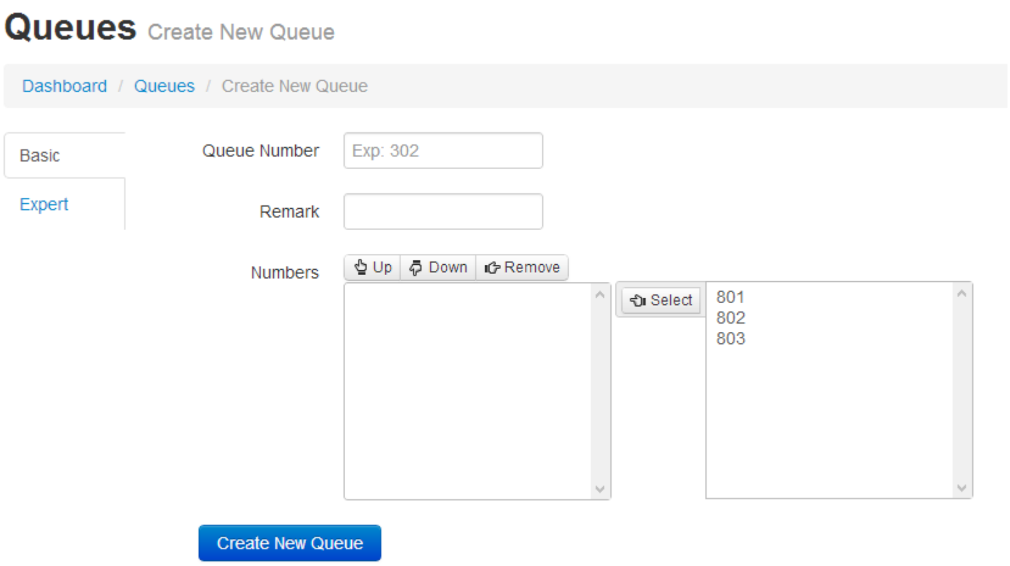
Queues

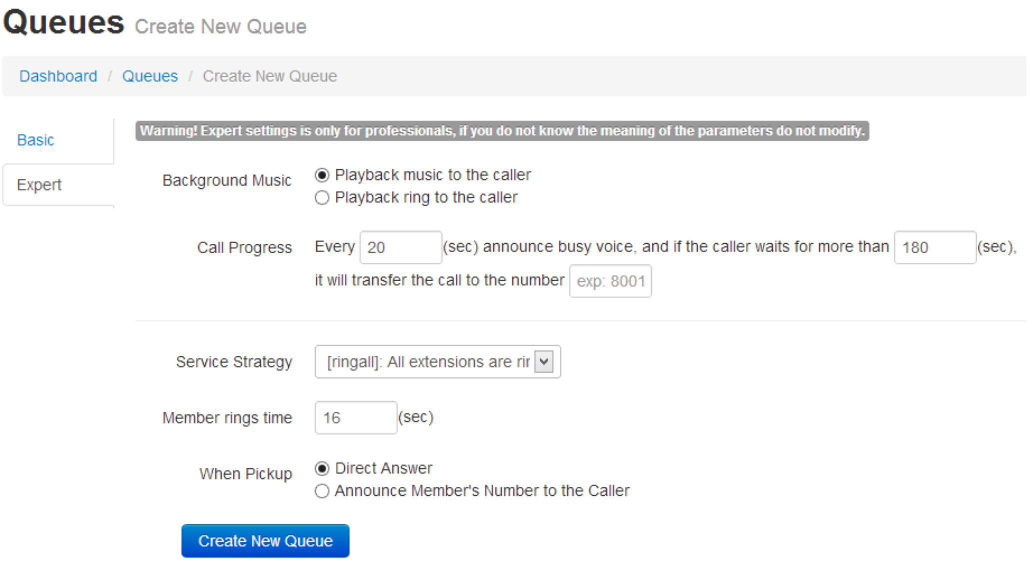
The PBX220 supports call queues. Call queues are often used to dial into a particular department or group; for example, the extension for the accounting department might be a call queue. This section describes the configuration of call queues under **WEG GUI—Advanced-Queues**



Click “Create New Queue” to add a new call queues.

Click “Edit” to edit the call queues. The queues configuration parameters are listed in the table below.

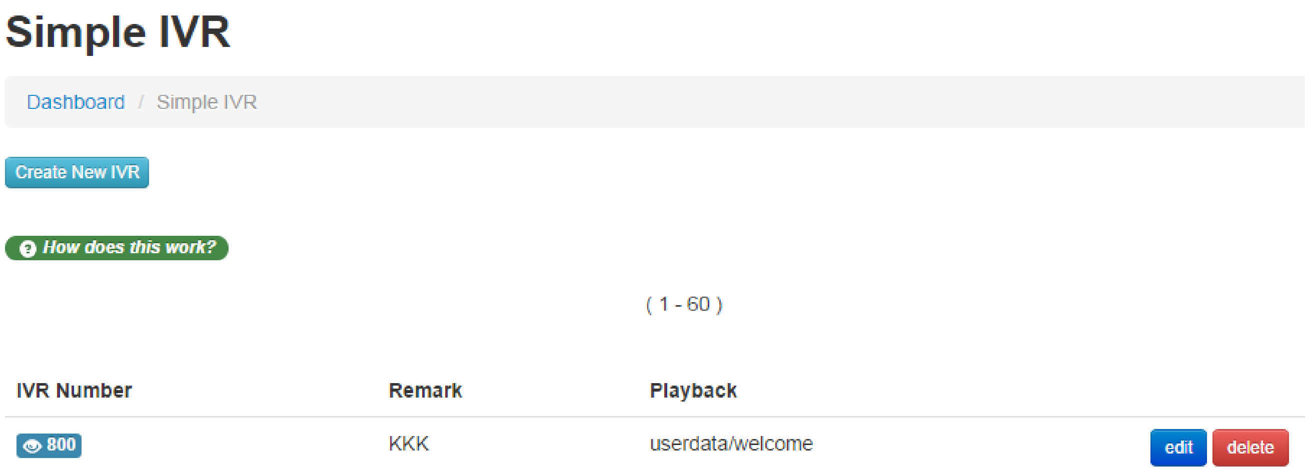




Queues Basic	
Queues Number	Configure the call queues extension number.
Remark	Configure the call queues name to identify the queues.
Numbers	Select the available extensions to be the agents in the queues. Choose the extensions on the right to the agents list on the left. Click Up Down Remove to arrange the order or remove extensions from the list.
Queues Expert	
Background Music	Check Playback music to the caller to playback music to the caller in queues. Check Playback ring to the caller to playback ring tone to the caller in queues.
Call Progress	Configure call progress details.
Service Strategy	Select the service strategy option **Ring All:** All extensions ring in the queues until one extension picks the call up **Random:** Randomly select one extension to serve the caller **Round Robin:** Round robin with memory, remember where we left off last ring pass.
Member Rings Time	Enter the member rings time.
When Pickup	Direct Answer: directly answer the caller. Announce member’s number to caller: announce the extension number to the caller.

Simple IVR

IVR configuration can be accessed under WEB GUI—Advanced—Simple IVR. Users can create, edit, view and delete an IVR.

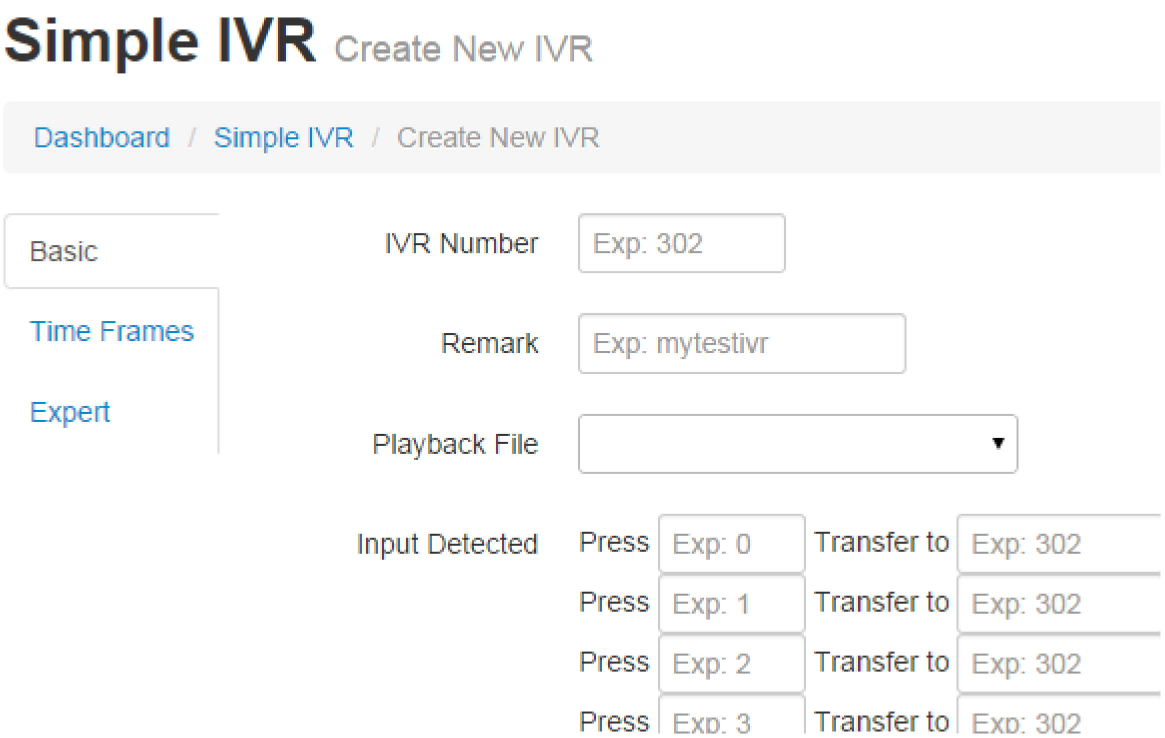


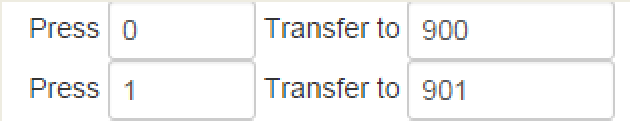
Click “Create New IVR” to add a new IVR.

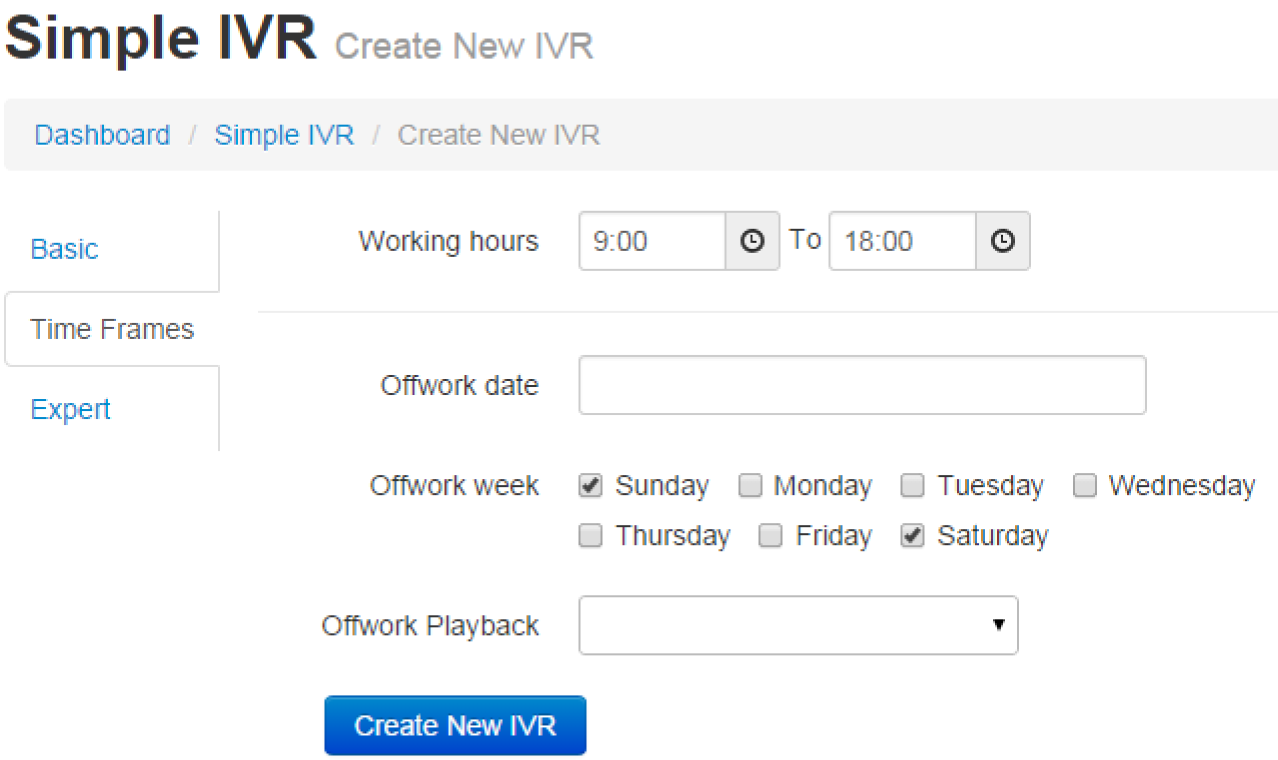
Click “Edit” to edit the IVR configuration.

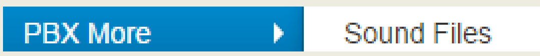
Click “Delete” to delete the IVR.

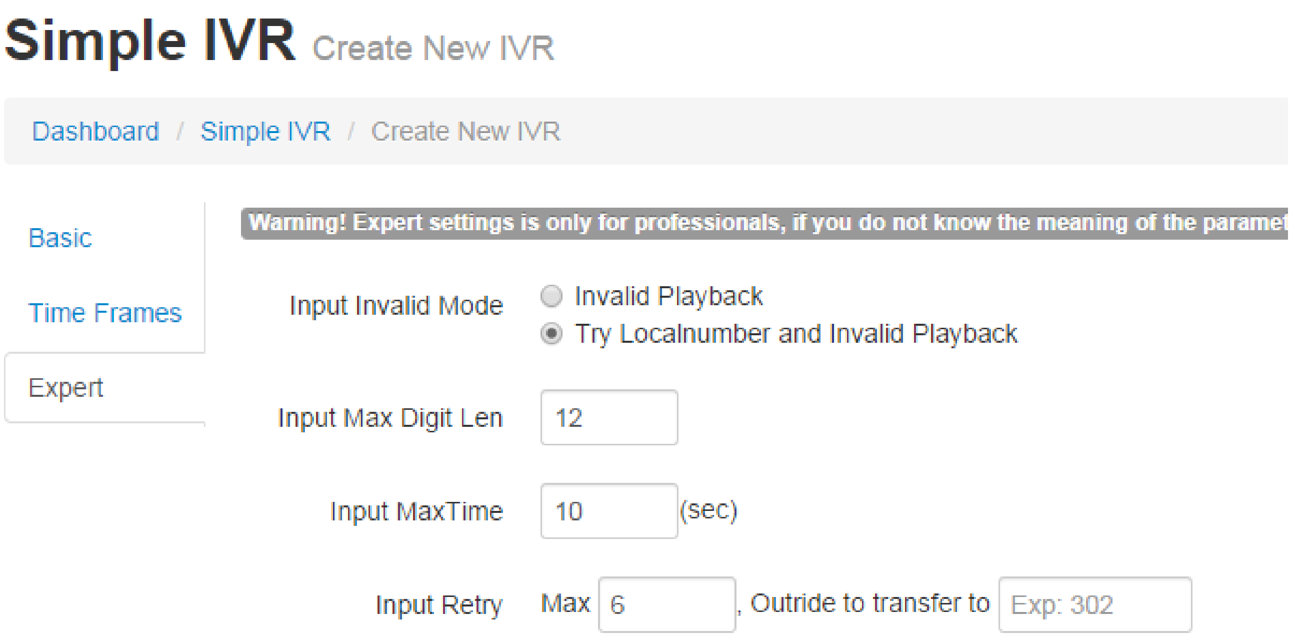
The simple IVR configuration parameters are listed in the tables below.



IVR Basic	
IVR Number	Configure the extension number for the IVR.
Remark	Configure the name of the IVR
Playback File	Select the audio file to play for the IVR. Add additional audio file under **WEB GUI—PBX More—Sound Files.** 
Input Detected	Configure the input digit and the transfer destination number. Example:  When caller dial 0 in IVR, transfer the call to number 900. Dial 1, transfer the call to number 901. The transferred number can be another IVR number, so user can arrange multi-level IVRs.



Time Frames	
Working hours	Configure the time frame to play the IVR.
Offwork date	Configure the offwork date for the IVR. Fill in with MM-DD format. Multiple comma-separated. Example: 12-24, 5-1 means Dec. 24th and May. 1st are the offework date.
Offwork week	Configure offwork date within one week.
Offwork Playback	Select the audio file to play for offwork time. Add additional audio file under **WEB GUI—PBX More—Sound Files.** 

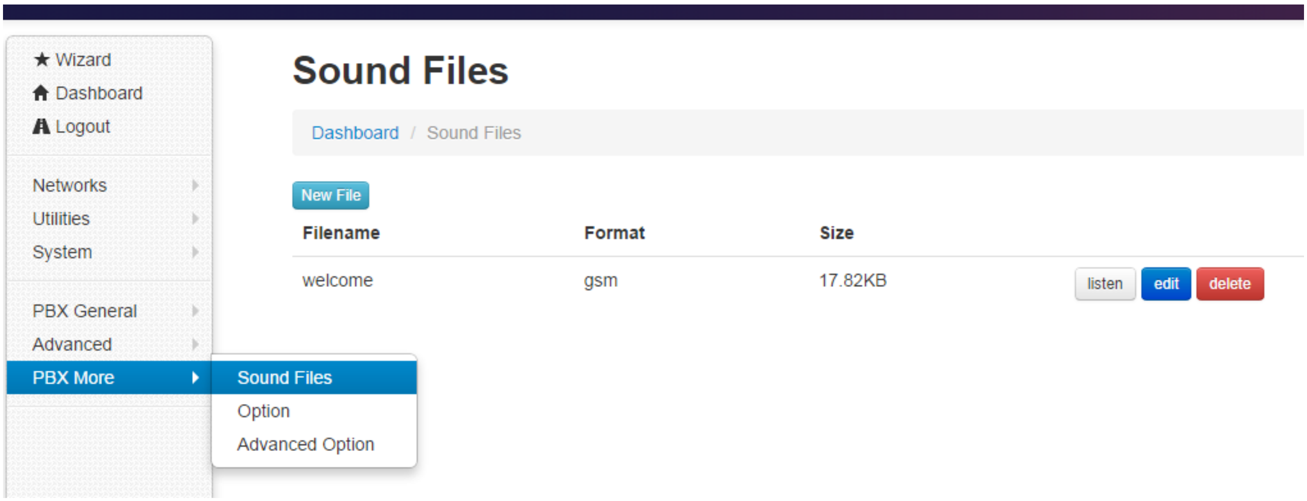


IVR Expert	
Input Invalid Mode	**Invalid Playback:** play an invalid voice prompt when the caller enters an invalid number. **Try Local number and Invalid Playback:** try call the local number first if failed then play an invalid voice prompt when the caller enters an invalid number.
Input Max Digit Len.	Configure the max digit length for the input.
Input Max Time	Configure the max timeout seconds for the input.
Input Retry	Configure the retry times for the input.
Outride to transfer to	Enter the extension number when the input exceeds the above configuration. Example: Input Max Digit Len is 12, Input Max Time is 10 seconds, Input retry max is 6, outride to transfer to 300. It means when the number the caller inputs exceeds 12, or input time exceeds 10 seconds, or retry times exceed 6, the call will be transferred to extension number 300.

1.13 PBX More

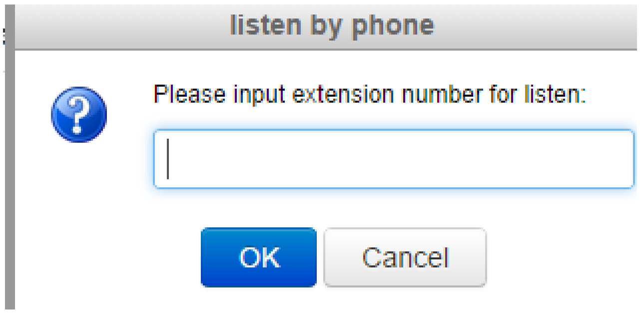
1.14 Sound Files

User can manage the sound files for IVR setting at this section. It can be accessed under WEG GUI—PBX More—Sound Files.



Click “New File” to add a new audio file.

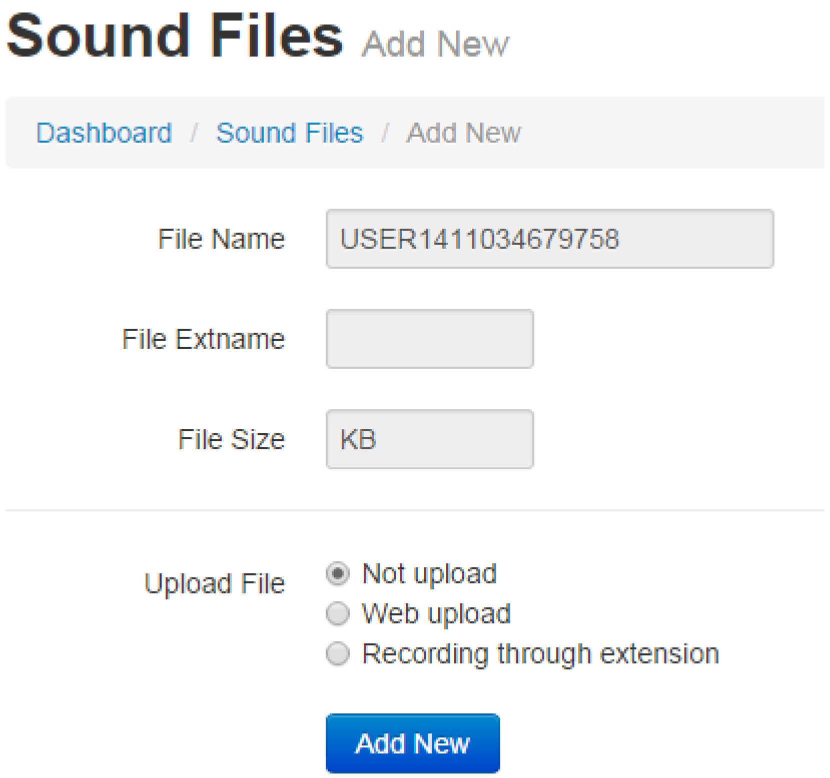
Click “Listen” and enter the extension number to listen the audio from the configured extension.

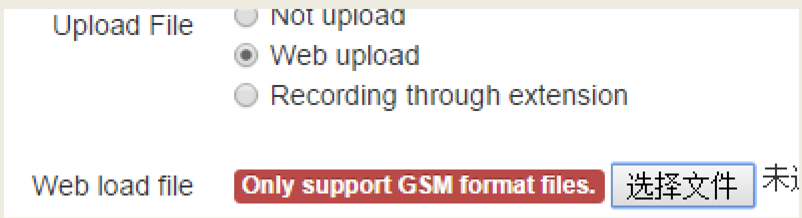
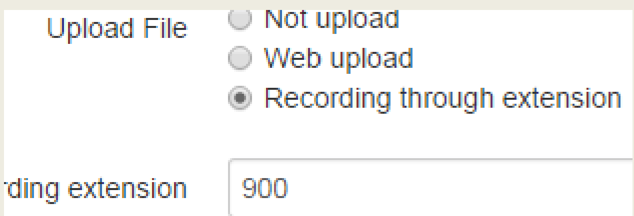


Click “Edit” to edit the audio setting.

Click “Delete” to delete the audio file.

The sound files configuration parameters are listed on the table below.



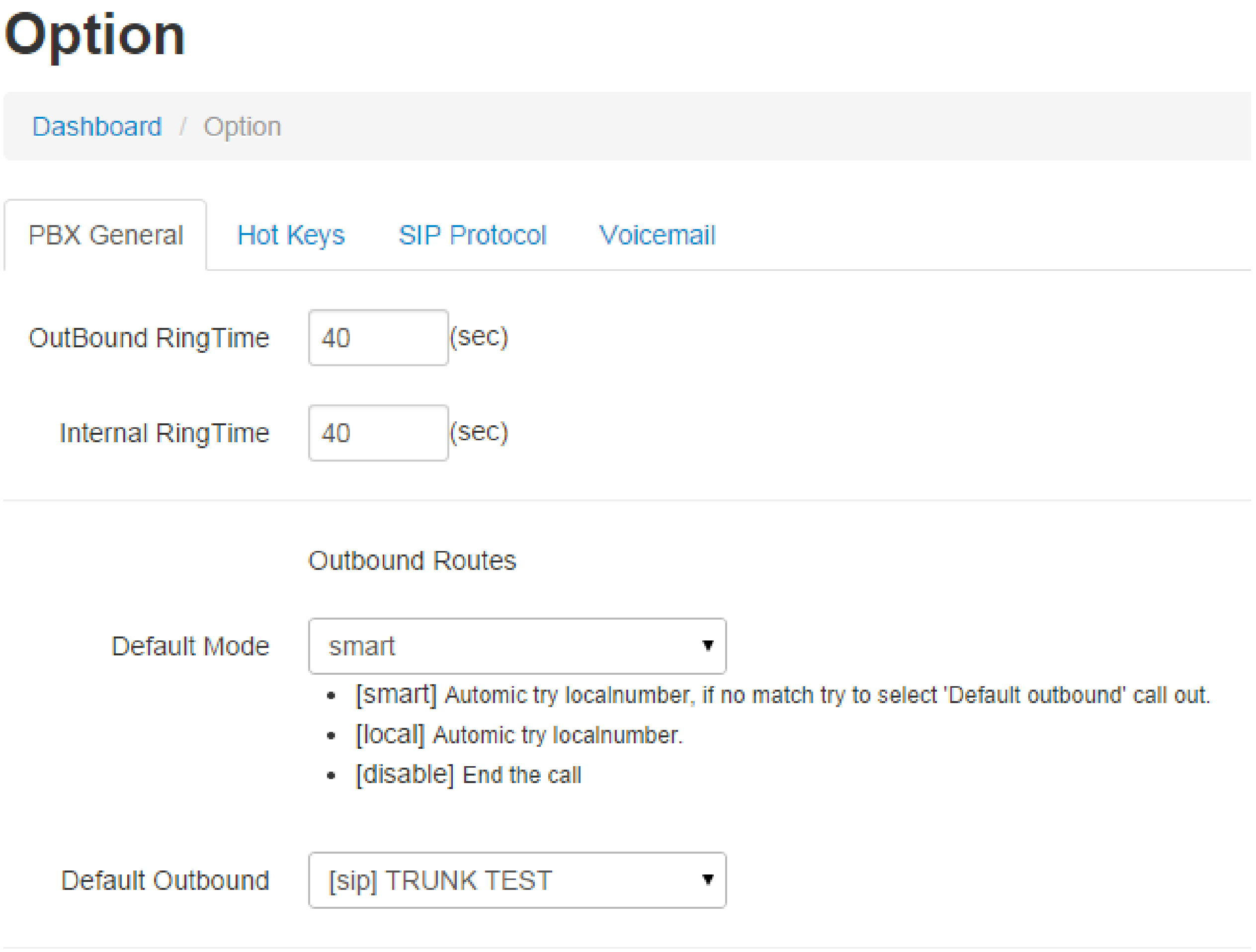
Sound Files	
File Name	The name can not be changed by default.
File Extname	It can not be changed by default.
File Size	The file size will be displayed.
Upload File	Configure the upload file option. **Web upload:**  Click 选择文件 to select the audio file on your PC. Only supports GSM format file. **Recording through extension:**  Enter the ext. and click Add New, the ext. will ring and you can pick up the

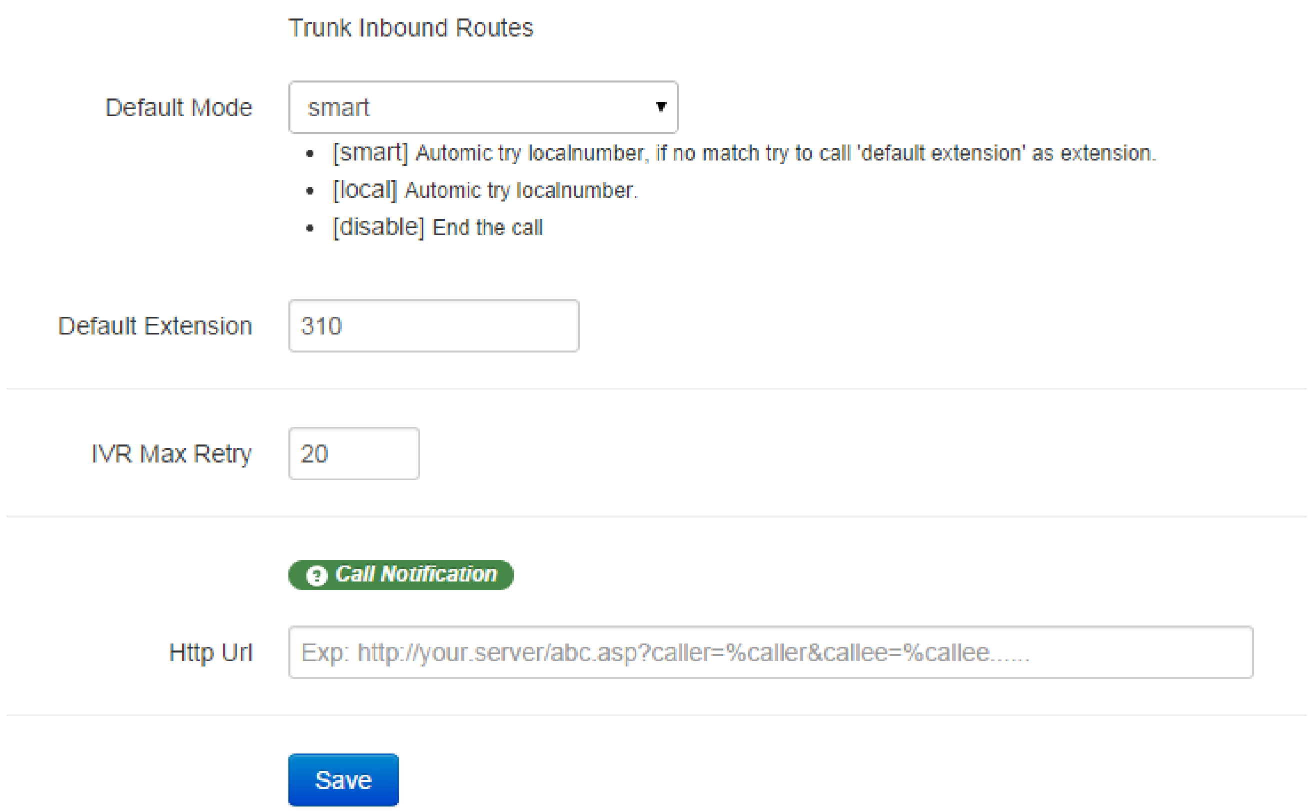
	phone and record the audio file through the extension.

Option

PBX General:

User could change the PBX general setting under **WEB GUI—PBX More—Option—PBX General.** The PBX general configuration parameters are listed in the tables below.

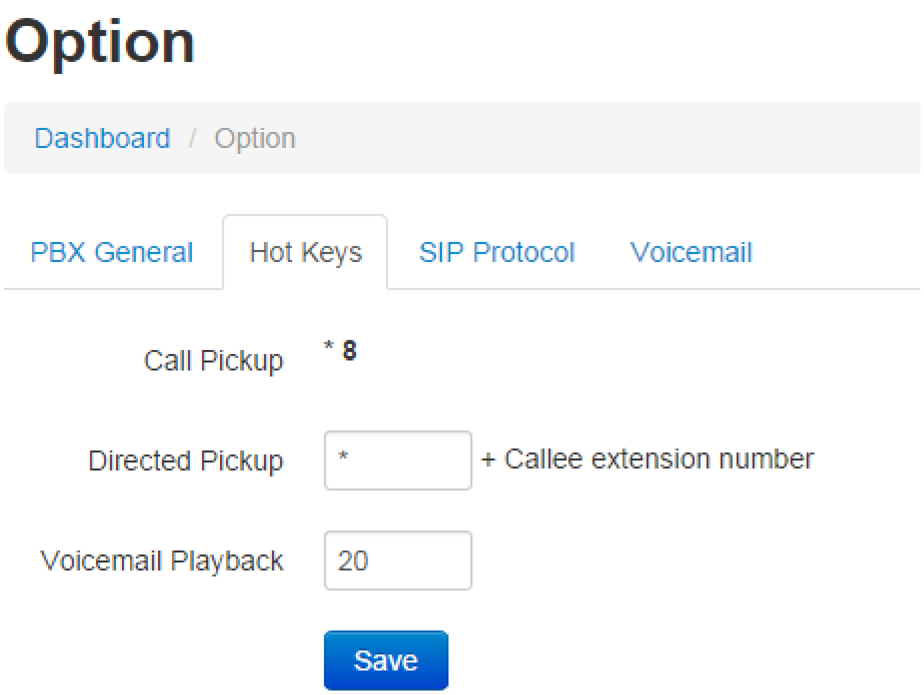




PBX General	
Outbound Ring Time	Trunk line has a timeout to determine if there was a hang up before the line is answered. This value can be used to configure how long it takes before the PBX220 considers a non-ringing line with hang up activity.
Internal Ring Time	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled).
Outbound Routes	Default mode **Smart:** Automatically try local number, if no match try to call out through default outbound. **Local:** Automatically try local number. **Disable:** End the call.
Default Outbound	Select the default outbound line.
Trunk Inbound Routes	Default mode **Smart:** Automatically try local number, if no match try to call “Default Extension”. **Local:** Automatically try local number. **Disable:** End the call.
Default Extension	Configure the default extension number.
IVR Max Retry	Configure IVR max retry times. Example, configure max retry as 2, the system will automatically try quite when the IVR plays twice.
Call Notification	Configure the call notification URL. When there is a call to extensions, group, queues, IVR. A notification request will be sent to the configured URL address.

Hot Keys

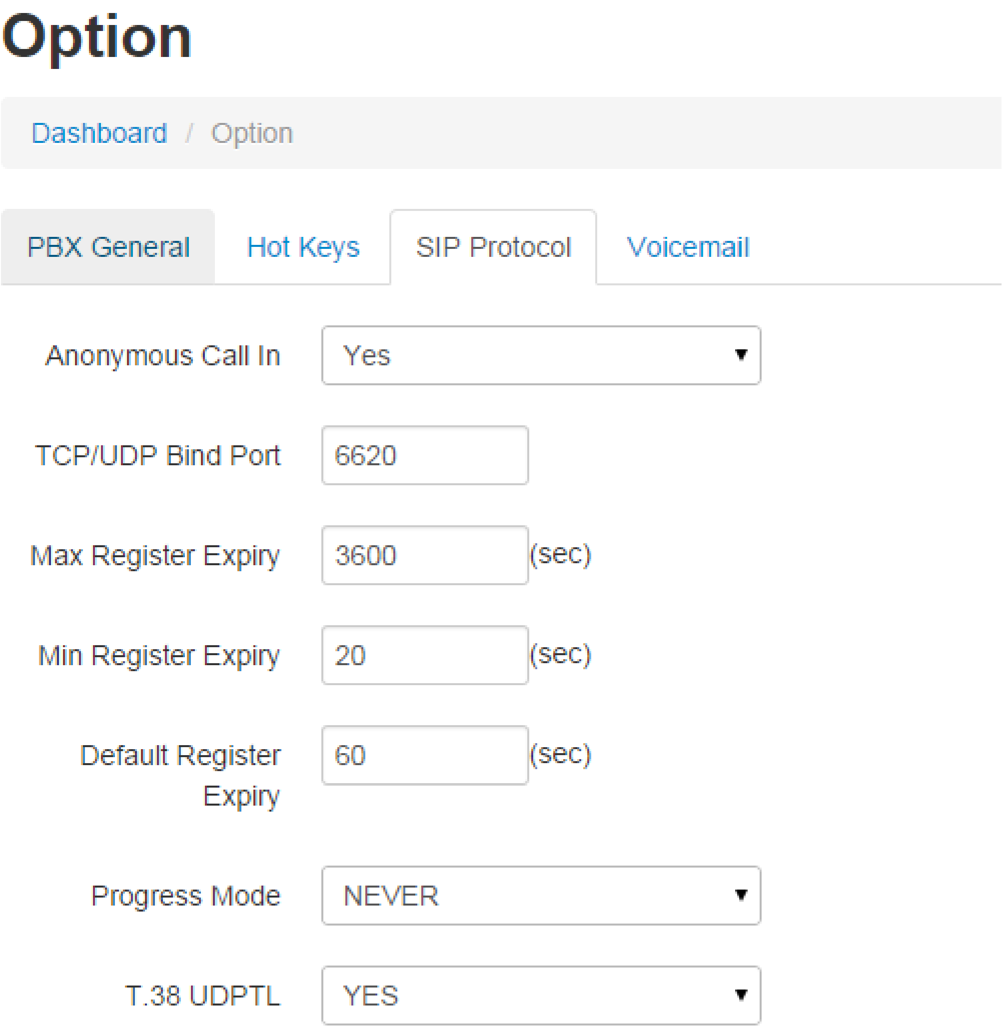
User can change the hot keys setting under **WEB GUI—PBX More—Option--Hot Keys.** The Hot Keys parameters are listed on the table below.



Hot Keys	
Call Pickup	Default call pick up setting is *8
Directed Pickup	Configure call pick up option directly from extension. Default is "*"+"Extension number"
Voicemail Playback	Configure voice mail playback number. Dial the number on the extension to listen to its voice mail.

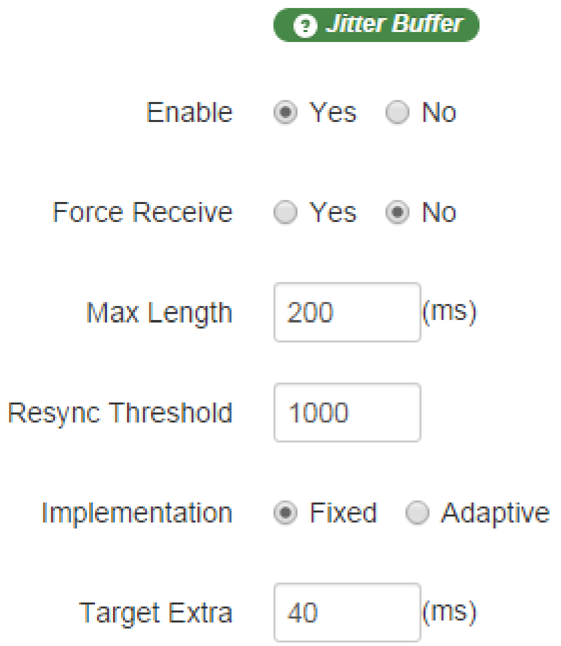
SIP Protocol

User could manage the SIP protocol setting under **WEB GUI—PBX More—Option—SIP Protocol.** The SIP Protocol parameters are listed on the table below.

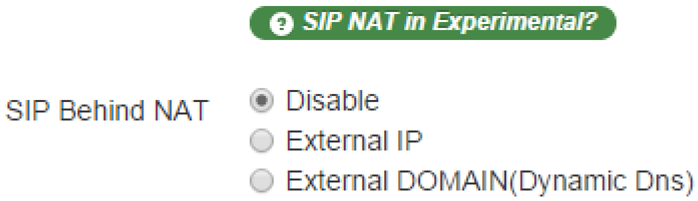


SIP Protocol	
Anonymous Call In	Check yes to allow anonymous call in, check no to disable it.
TCP/UDP Bind Port	Configure TCP/UDP port number used for SIP. Default setting is 6620. **Note:** When you register an IP Phone to PBX220, please modify your phone SIP port to 6620. Otherwise, you need to modify the port number 6620 in PBX220 to 5060.
Max Register Expiry	Configure the maximum period (in seconds) of registration. The default setting is 3600.
Min Register Expiry	Configure the minimum period (in seconds) of registration. The default setting is 20.
Default Register Expiry	Configure the default minimum period (in seconds) of registration. The default setting is 60.
Progress Mode	Configure whether PBX220 should generate inbound ringing or not. The default setting is "never".

	Yes: The PBX220 will send 180 Ringing followed by 183 Session Progress and in-band audio. **No:** The PBX220 will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing. **Never:** Whenever ringing occurs, the PBX220 will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly.
T.38UDPTL	Select the T.38UDPTL option from the pickup list.

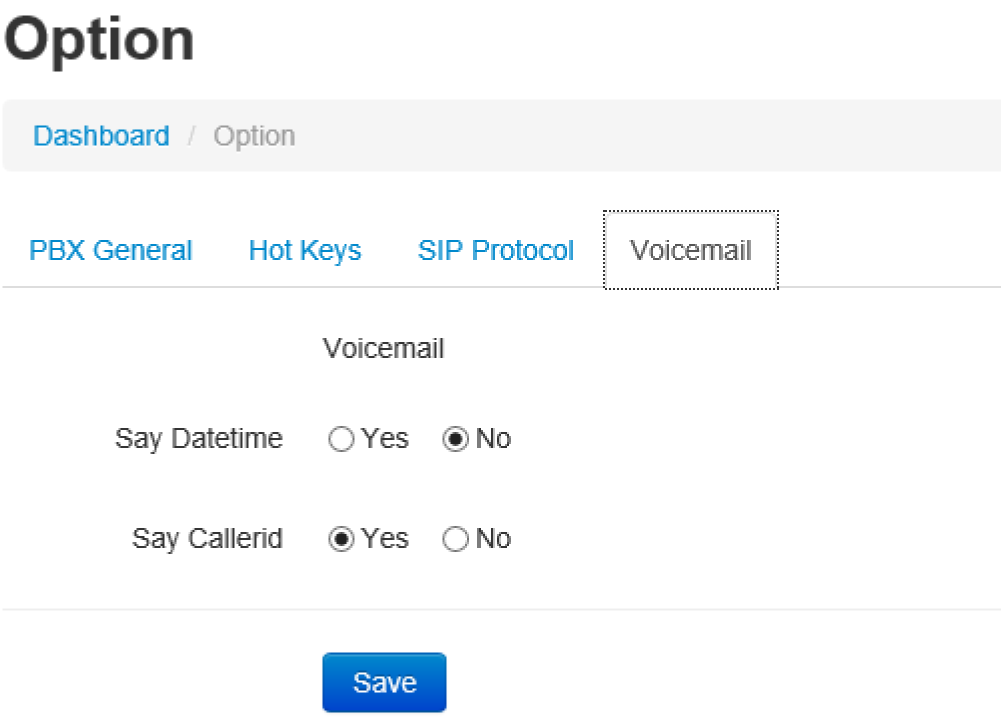


Jitter Buffer	
Enable	Check Yes to enable jitter buffer. Check No to disable it.
Force Receive	Check Yes to enable force receive. Check No to disable it.
Max Length	Configure the maximum time (in ms) to buffer for "Adaptive" jitter buffer implementation, or used as the jitter buffer size for "Fixed" jitter buffer implementation. The default setting is 200.
Resync Threshold	Eenter the resync threshold data.
Implementation	Configure the jitter buffer implementation on the sending side of a SIP channel. The default setting is "Fixed". **Fixed** The size is always equal to the value of "Max Jitter Buffer". **Adaptive** The size is adjusted automatically and the maximum value equals to the value of "Max Jitter Buffer".
Target Extra	Enter a target extra data.



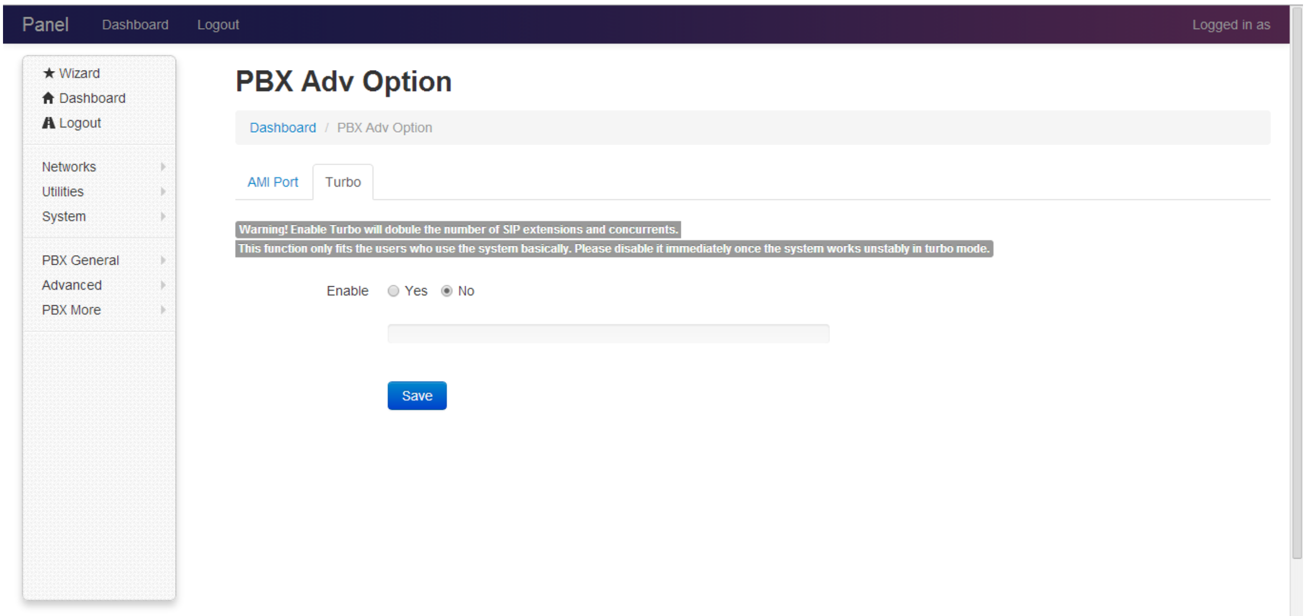
SIP NAT	
SIP Behind NAT	**Disable** Check to disable SIP behind Nat **External IP** Check to Configure a static address and port (optional) that will be used in outbound SIP messages if the PBX220 is behind NAT. If it's a hostname, it will only be looked up once. **External DOMAIN** Specify an external domain name, which is similar to External Address except the domain name will be looked up periodically.

Voicemail



Voice Mail	
Say Datetime	Check yes to say day time when play the voice mail.
Say Callerid	Check yes to say caller ID when play the voice mail.

Advanced Option-Turbo Mode



1: Via the Turbo in the Advanced Option, you can double the number of the PBX's Sip Extensions and the Concurrent Calls.

2: Choose"Yes" and Press"Save", and then this Operation come into effect.(After this Operation, PBX210 will support 64 Sip Extensions and 16 Concurrent Calls; PBX 220 will support 32 Sip Extensions and 8 Concurrent Calls).

3: Warning! This function only fits the users who use the system basically. Please disable it immediately once the system works unstably in the Turbo mode!

4. Glossary

ATA (Analog Telephony Adapter)

A device used to connect one or more standard analog telephones to a digital and/or non-standard telephone system such as a Voice Over IP based network.

DID (Direct Inward Dial)

A feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service, the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk. As calls are presented to the PBX, the dialed destination number (DNIS) is transmitted, usually partially (e.g., last four digits), so that the PBX can route the call directly to the desired telephone extension within the organization without the need for an operator or attendant.

DID numbers are assigned to a communications gateway connected by a trunk to the public switched telephone network (PSTN) and the VoIP network. The gateway routes and translates calls between the two networks for the VoIP user. Calls originating in the VoIP network will appear to users on the PSTN as originating from one of the assigned DID numbers.

DNS (domain name system)

The Internet's name/address resolution service that translates alphabetic domain names into numeric IP addresses. For example, the domain name www.pbx.com might translate to 198.105.232.4. If a computer cannot access DNS, the user's web browser will not be able to find web sites and the user will not be able to receive or send email. The DNS system consists of three components: DNS data,name servers, and Internet protocols for getting the data from the servers.

Domain name server

A computer that runs a program that converts a fully qualified domain name (FQDN) into its numeric IP address and vice versa.

DTMF (Dual-Tone Multi-Frequency)

The signal that is generated when a user presses the touch keys of an ordinary telephone. Also known as “Touchtone,” DTMF has essentially replaced pulse dialling. When a user presses touch keys, two tones of specific frequencies are generated (one from a high-frequency group and the other from a lowfrequency group), so it’s impossible for the voice to imitate the tones.

FTP (File Transfer Protocol)

A standard Internet protocol used to upload and download files between computers that are connected to the Internet. FTP uses the Internet’s TCP/IP protocols as does HTTP, which transfers displayable Web pages and related files, and SMTP, which transfers e-mail.

GSM (Global System for Mobile communication)

A wireless telephone standard in Europe and other parts of the world.GSM uses a variation of time division multiple access (TDMA), which is the most widely used of the three digital wireless telephony technologies (TDMA, GSM, and CDMA). GSM digitizes and compresses data, then sends it down a channel with two other streams of user data, each in its own time slot. It operates at either the 900 MHz or 1800 MHz frequency band.

IP-PBX (Internet Protocol Private Branch Exchange)

A telephone switch (see “PBX”) located on a customer’s premises that utilize VoIP to manage and deliver calls.

ITSP (Internet Telephone Service Provider)

A company that offers an Internet data service for making telephone calls using VoIP. Most ITSPs use SIP, H.323, or IAX for transmitting telephone calls as IP data packets. Customers may use VoIP phones or traditional telephones with an analog telephony adapter (ATA).

ITU (International Telecommunication Union)

A telecommunications standards body that is guided by the United Nations. It was founded as the International Telegraph Union in Paris on May 17, 1865. The ITU acts as the global focal point for governments and the private sector in developing networks and services and is comprised of more than 185 countries and produces over 200 standards recommendations annually in the areas of information technology, consumer electronics, broadcasting, and multimedia communications.

IVR (Interactive Voice Response)

A telephone technology that allows a caller to respond to configured voice menus through voice and

touch tone. The IVR system responds with pre-recorded audio to further direct callers on how to proceed.

LAN (Local Area Network)

A computer network covering a small physical area, like a home, office, or small group of buildings, such as a school, or an office park. LANs are connected primarily through Ethernet and can be connected to other LANs over any distance via telephone lines and radio waves. LANs have a high data transfer rate and are not very expensive to set up. See also “WAN.”

MAC (Media Access Control) address

A hardware address that uniquely identifies most network adapters or network interface cards (NICs) by the manufacturer for identification. The manufacturer’s registered identification number is usually part of the MAC address if it was assigned by the manufacturer. The MAC address is used by the Media Access Control protocol sub-layer of the Data-Link Layer (DLC) of telecommunication protocols.

MIPS (million instructions per second)

An old method for measuring a computer’s speed and power and, by implication, for determining the amount of work a computer can do. It measures the approximate number of machine instructions the computer can execute in 1 second (i.e., it measures CPU speed).

Because there are so many variables with computer performance (e.g., varying amounts of time for different instructions, importance of I/O speed, etc.), MIPS ratings are not used that often anymore. However, a MIPS rating can give you a general idea of a computer's speed.

NAT (Network Address Translation or Network Address Translator)

The method for translating an IP address used within one network to a different IP address known within another network (one network is designated the *inside* network and the other is the *outside* network). NAT allows as a router, for example, to act as an agent between the public network (e.g., the Internet) and a private network (i.e., a local network), which means that a single, unique IP address can represent an entire group of computers.

PBX (Private Branch exchange)

A telephone exchange that serves a particular business or office, as opposed to one that is owned by a common carrier or telephone company and is used by many businesses or the general public. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.PBXs have evolved over time, beginning as a manual switchboard or attendant console that was operated by a telephone operator (circuit switching) to the modern IP PBX. See also "IP PBX."

PSTN (Public Switched Telephone Network)

The network of the world's public circuit-switched telephone networks. Originally a network of fixed line analog telephone systems, the PSTN is now almost entirely digital in its core and includes mobile as well as fixed (plain old telephone service, POTS) telephones. The PSTN is largely governed by technical standards created by the ITU-T, and uses E.163/E.164 telephone numbers for addressing.

Proxy Server

A server (a computer system or an application program) that acts as an intermediary for requests from clients seeking resources from other servers. The VoIP proxy server is used in a DMZ of a company's secure internal communication network and receives VoIP control messages and VoIP media streams.

Using the MAC address and source IP address contained in the control message, the proxy server pushes a policy change to the internal network's external firewall to open call control protocol ports and Real Time Protocol (RTP) ports only for packets from the source IP address. The VoIP proxy server hides the company's internal network address and directs incoming VoIP packets to an IP-PBX connected to the company's internal network.

RAM (Random Access Memory)

A form of computer data storage that allows stored data to be accessed in any order (i.e., “random access”).

RAM is used by a computer's operating system, application programs, and currently used data, so that they can quickly be reached by the computer's processor. RAM is quickly readable and writeable compared to other kinds of computer storage (e.g., the hard disk, floppy disk, and CD-ROM);However, data in RAM remains only as long as the computer is running. Once the computer has been turned off, RAM loses its data. When the computer is turned on again, the operating system and other files are once again loaded into RAM.

Router

A device for connecting one or more computers to other computers, networked devices, or to other networks. Compared to hubs and switches (which are also connecting types of devices), a router is the smartest and most complicated of the three. Routers can be programmed to understand and route the data its being asked to handle. Configuration is done through a user interface. Larger routers are capable of being programmed to communicate with other routers to determine the best method of getting network traffic from point A to point B. Hubs work at the data link and network layers (layers
2 and 3) of the OSI model.

SIP (Session Initiation Protocol) [RFC 3261, 3262, 3263, 3264, and 3265]

A signalling protocol for initiating and terminating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality (it is used mainly for voice and video calls over the Internet or data networks).

SIP Trunk

A service offered by an ITSP that allows businesses that have a PBX for their internal calls to use VoIP to go outside the enterprise network by using the same connection as the Internet connection. Before SIP trunks can be deployed, a business must have a PBX with a

SIP-enabled trunk side, an enterprise edge device that understands SIP, and an ITSP. See "ITSP."

Soft-switch (software switch)

A term used to describe the software that is used to bridge a public switched telephone network (PSTN) and VoIP. This is done by separating the call control functions of a phone call from the media gateway (transport layer). The soft-switch is typically used to control connections at the junction point between circuit and packet networks.

UDP (User Datagram Protocol) [RFC 768]

A communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that is using the Internet Protocol (IP). UDP merely performs IP traffic demultiplexing based on UDP port numbers, after which it provides a checksum that can be used by end systems to determine whether the datagrams received were corrupted by the network.

WAN (Wide Area Network)

A computer network that covers a broad area (e.g., any network that links across metropolitan, regional, or national boundaries). WANs are similar to the Internet in that they are not owned by a single organization. They exist under collective or distributed ownership and management. For WAN connectivity over the longer distances, ATM, frame relay, and X.25 are used. Computers connected to a WAN can be connected via the telephone system, leased lines, or satellites. WANs have a lower data transfer rate when compared to LANs. See also “LAN.”