Sipdex®

M-200 IPPBX

User Guide

For System Administrator

Contents

Chapter 1: hardware Setup

M-200 IPPBX Overview

Codecs & Protocols

- SIP 2.0 (RFC3261) / IAX2 compliant
- Audio Codec: G.722 / G.711-Ulaw / G.711-Alaw / G.726 / G.729 / GSM / SPEEX
- Video Codec: H.261 / H.263 / H.263+ / H.264
- DTMF: RFC2833, SIP INFO, In-band

Network Features

- DDNS Client
- DHCP Server / SNMP v1/v2
- IEEE 802.1Q of VLAN
- IPv4 / IPv6
- Manual Configuration of Static Route Table
- Troubleshooting (Ping, Traceroute)

Security Features

- VPN Client (Supports N2N / L2TP / PPTP / OpenVPN)
- VPN Server (Supports PPTP / L2TP / OpenVPN Server)
- Permit IP for Extensions
- SIP Allowed Address
- Block SIP Register Failed
- Iptables Firewall / SRTP

Environment

Working Temperature	0 ~ 40°
Storage Temperature	- 20 ~ 55°
Humidity	5 ~ 95% Non-condensing

Unpack Your IPPBX

Thank you for purchasing SIPDEX M-200 IPPBX. This Quick Installation Guide will introduce how to finish the basic setting of connecting the web management interface and the Internet. Open the box of the IPPBX system and carefully unpack it. The box should contain the following items:

- Internet Telephony PBX System Unit x 1
- Quick Installation Guide x 1
- Power Adapter x 1 (12V)
- RJ-45 x 1
- Bracket x 2

If any of the above items are damaged or missing, please contact your dealer immediately.

Dimensions

- Dimensions: 343 (L)x 154 (W)x 35 (H)mm
- Net Weight: 1.4kg (without package)

Front Panel

LED definitions

Front Panel	State	Description
PWR	Steady Green Off	IPPBX Power ON IPPBX Power OFF
SYS	Blinking Green Off	System is working System is Off
ETH	Steady Red Off	IPPBX network connection established Waiting for network connection
FXO / GSM	Steady Red Flashing Off	Ready / Standby Ringing Module not available
FXS	Steady Green Flashing Off	Ready / Standby Ringing Module not available
T1 / E1	LED 1,2,4 Steady in Red	Configured to T1 / E1
USB	Steady Green	Connected External Disk

Rear Panel

Reset	The reset button, when pressed, resets the IP PBX without the need to unplug the power cord.
12V DC	12V DC Power input outlet
ETH	The ETH port supports auto negotiating Fast Ethernet 10/100 Base-TX networks. This port allows your IP PBX to be connected to an Internet Access device, e.g., router, cable modem and ADSL modem through a CAT.5 twisted pair Ethernet cable.
Console	For Console Interface configuration
USB	For External USB Storage
SLOT1	support 4FXS 4FXO 2FXOS 2GSM 4SGM 1PRI
SLOT2	support 4FXS 4FXO 2FXOS 2GSM 4SGM

Accessories

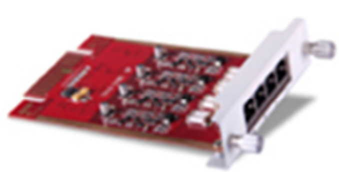
Adopting an innovative modular design, it is very easy to add telephony ports to expand the Phone System.

IPPBX-M200 supports industry standard SIP trunks as well as analog PSTN trunks (FXO module), mobile GSM trunks (GSM module), analog stations (FXS module), as well as digital trunks (E1/T1 module).

1 PRI Module

2 GSM Module

4 GSM Module



4 FXO Module

4 FXS Module

2 FXOS Module

Chapter 2: Connect to the Network & Access IPPBX

Use an Internet Brower to Access the IPPBX

1. Connect a computer to an ETH port on the IPPBX.
2. Your PC must set up to the same subnet 192.168.1.X
3. Open a web browser (Internet Explorer (version 6 and higher, Firefox, Chrome or Safari)
4. Enter the default URL of the IPPBX System: ***http://192.168.1.100:9999***

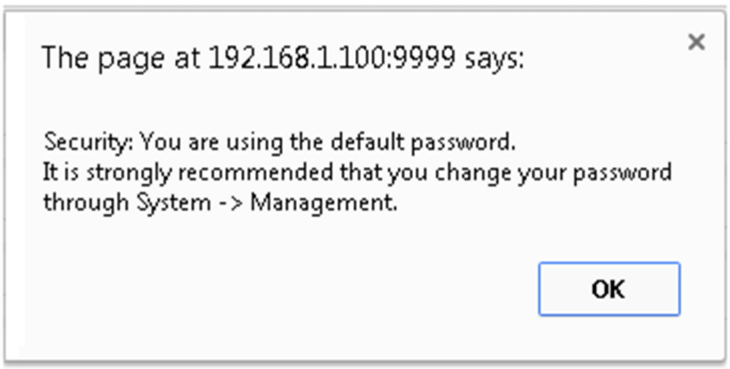
Enter the default username ***admin*** and password ***admin***, then click Login to enter Web-based user interface.

Default URL	http://192.168.1.100:9999
Username	admin
Password	admin

Login page of the SIPDIX M-200 IPPBX

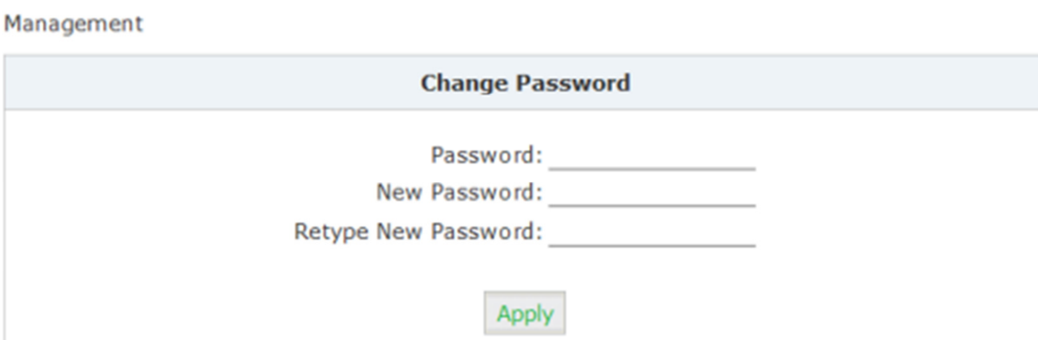
Change Password

Security message will pop-up when you login to the system, until you change the password.



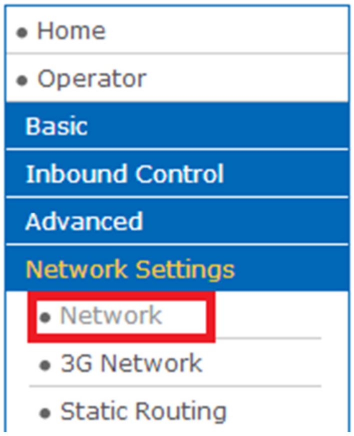
For security reason, please change and memorize the new password after this first setup.

Click **System** -> **Management**:

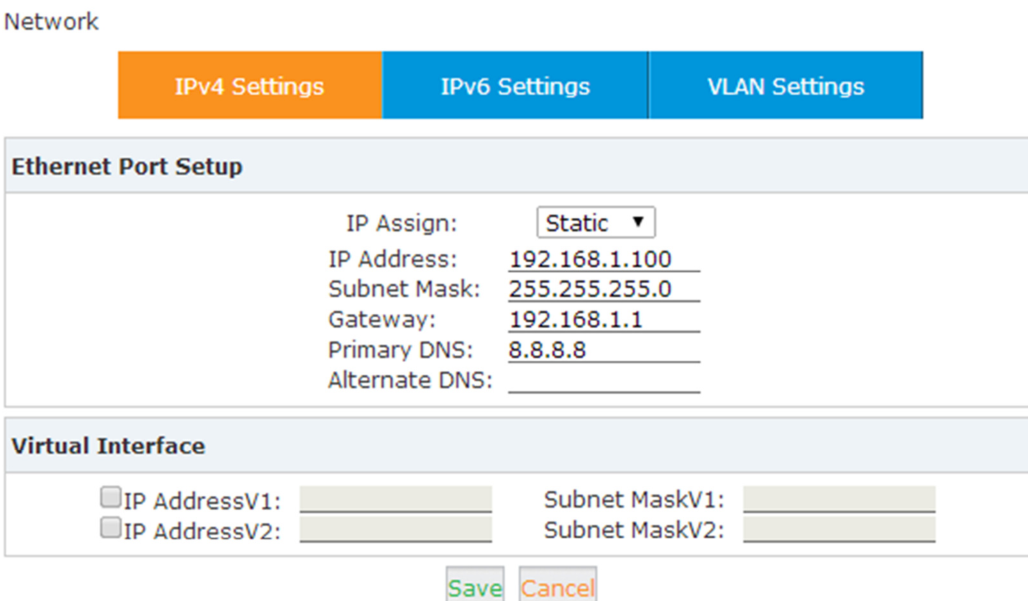


Network Settings

1. Go to **Network Settings** -> **Network**

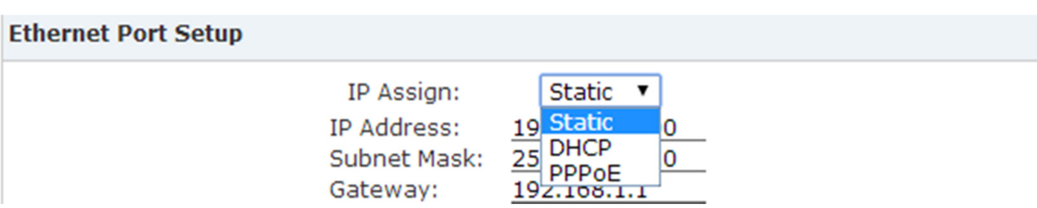


In **IPv4 Settings**



2. Edit your ETH port IP information.

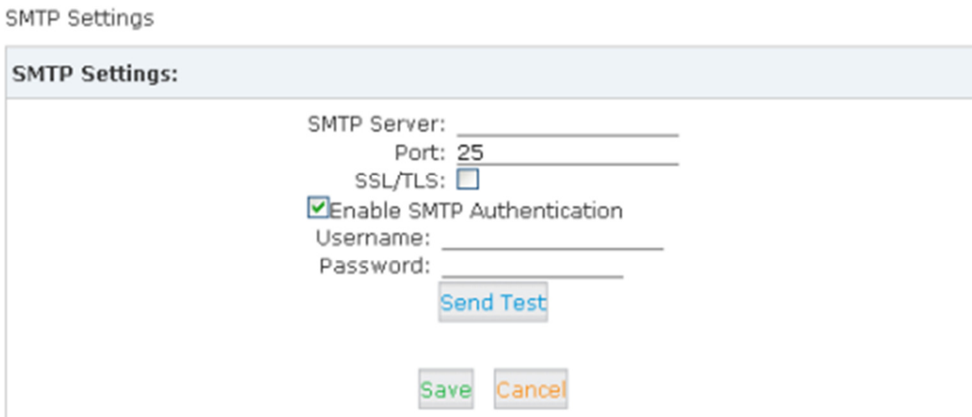
There are three types of Ethernet port connection. They are Static IP, PPPoE (Point-to-Point Protocol over Ethernet), DHCP. You can find detailed setting process in the user manual.



SMTP Settings

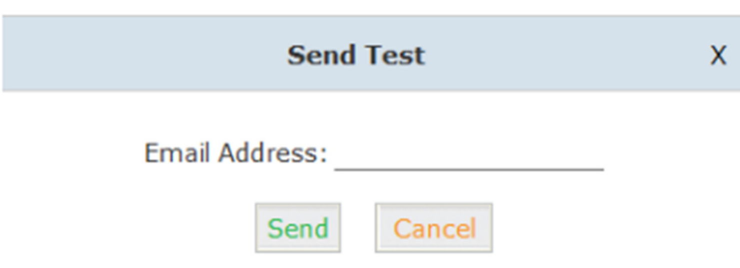
If the IPPBX require the SMTP Server for send out Email. (Voicemail to email, Fax to email)

Click **Advanced** -> **SMTP Settings** to configure



Item	Explanation
SMTP server	In order to send e-mail notifications of your voicemail, set the IP address or domain name of a SMTP server that you're IP PBX may connect to. e.g. mail.yourcompany.com
Port	The port number the SMTP server runs is generally port 25. If SSL is encrypted, please use port 465 instead.
SSL/TSL	Enable SSL/TLS to send secure messages to server.
Enable SMTP Authentication	If your SMTP server needs Authentication, please enable SMTP Authentication, and configure the following information.
User Name	Input username of your email box.
Password	Input password of your email box.

Click **Send Test** after configuration, the following diagram will be displayed to ask you to input the Email for receiving.

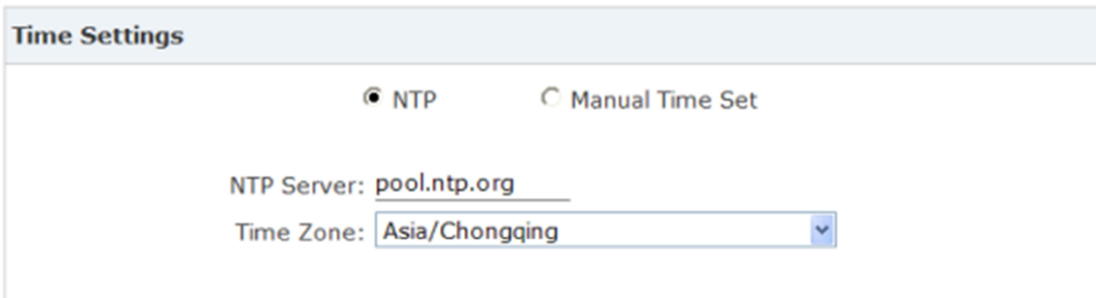


Input the Email Address and click **send** to send the test email. Login to your Email to check configuration is successful if you receive the test email otherwise, it fails. Please check your email settings again.

System Time Settings

The system supports either Sync with NTP Server or Manual Time Set.

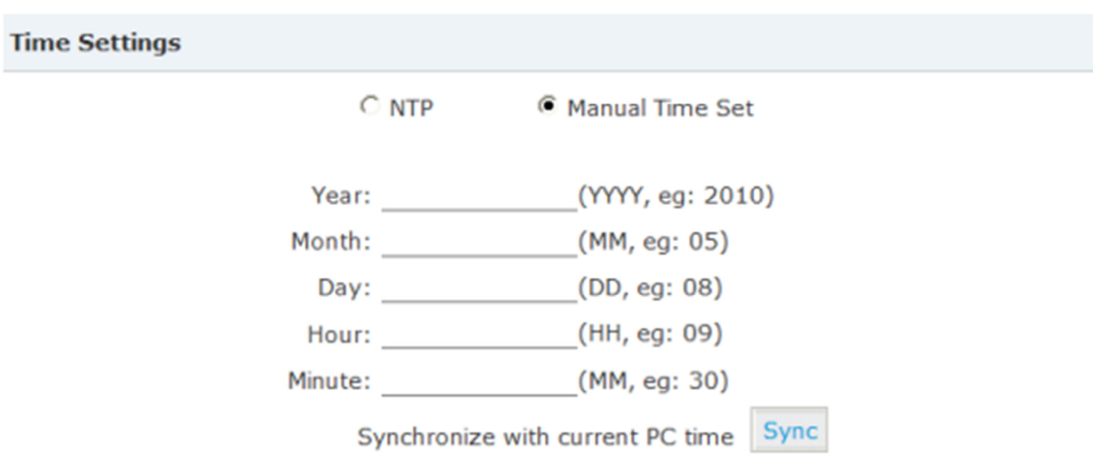
Click **System** -> **Time Settings** to configure



Reference:

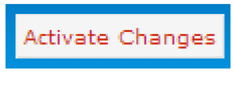
Item	Explanation
NTP Server	Define the NTP Server. You can input the IP address or domain of this server. Default server is pool.ntp.org.
Time Zone	Select your time zone so that the system will set time based on the time zone.

Manual Time Set:

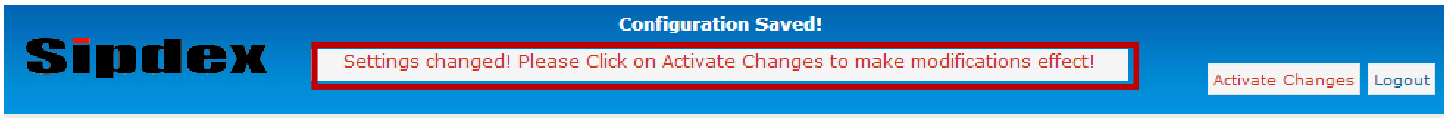


After entering Year/ Month/ Day/ Hour/ Minute, then save and activate or, you can click **Sync** to synchronize with current PC time.

Make the settings effect now

If you change the settings in Web GUI M-200 IPPBX will not take effect immediately. It will show the banner on the top of the website. Please click 

to take effect the changes.



Connect Module to the IPPBX

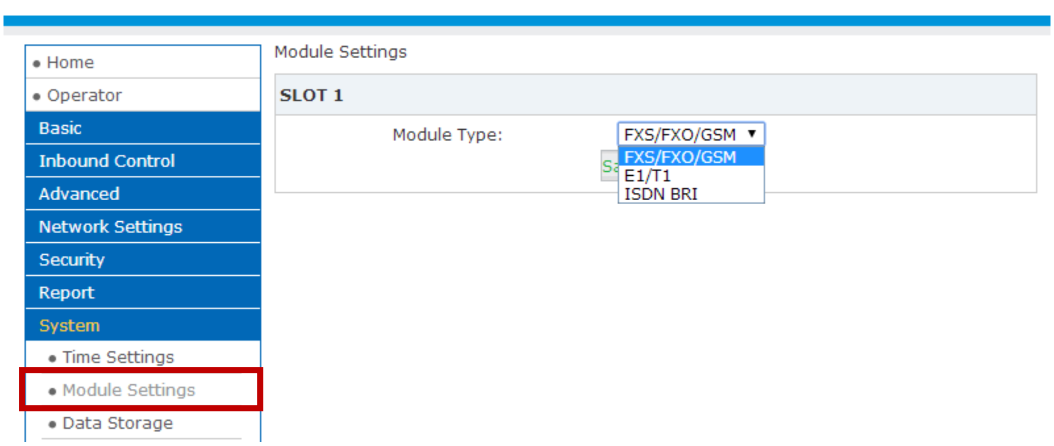
NOTE:

1. Please unplug the IPPBX Power, Before connect the Module.
2. All Module NOT support Hot-plug / Hot-swap

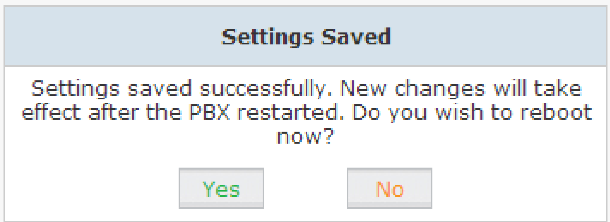
SLOT	**Module**
SLOT1	supported 4FXS 4FXO 2FXOS 2GSM 4SGM 1PRI
SLOT2	supported 4FXS 4FXO 2FXOS 2GSM 4SGM

Step for connect the Module.

1. Unplug the power supply.
2. Plug the card to the Slot 1
3. Go to **System** -> **Module Settings**
4. On **Module Type** to select SLOT 1 which Module was inserted, then click **save**.



5. Click **Yes** to confirm and the system will reboot.

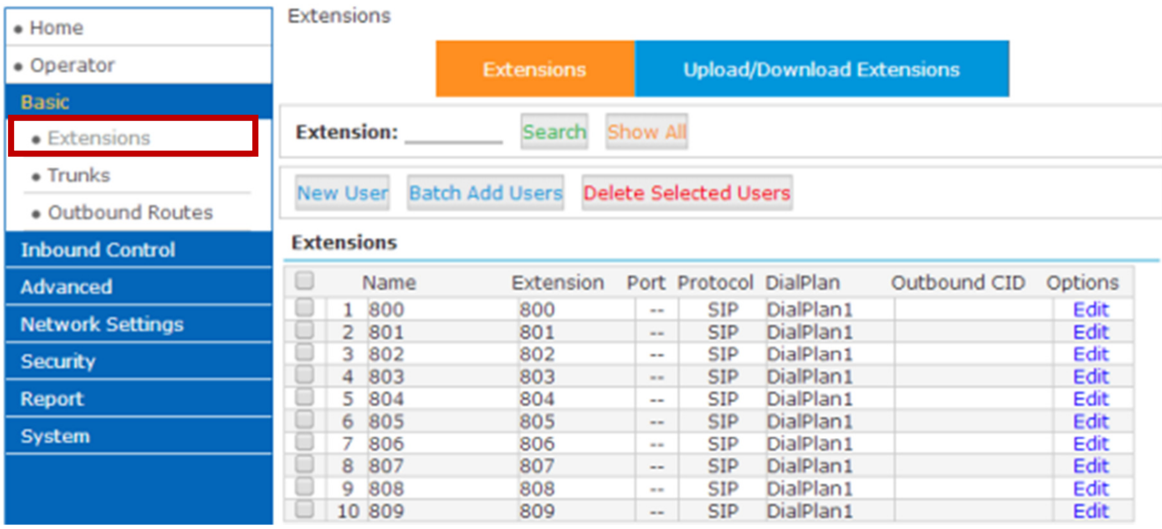


Chapter 3: Basic Configuration & Outgoing Call

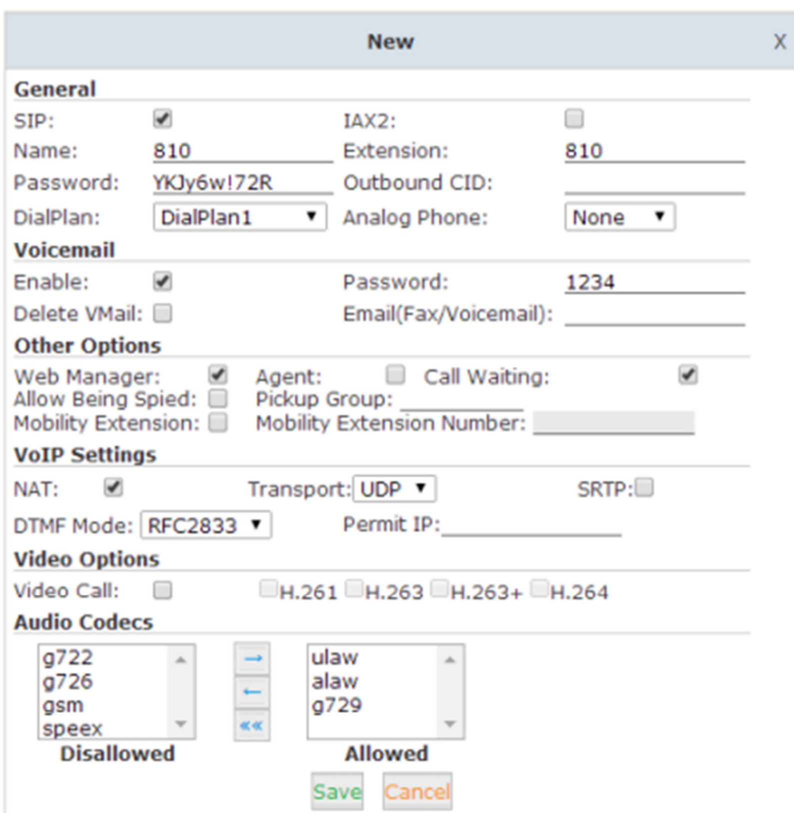
Extension Configuration

SIPDEX M-200 IPPBX Supports SIP/ IAX2 and analog extensions as well as the ability to "Batch Add Users" by uploading extensions file.

Click **Basic** -> **Extensions** to configure:



Click **New User** to see the extension configuration interface as shown below:



Extension Settings

Item	Explanation
SIP / IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g. Tom.
Extension	Extension Number connected to the phone, e.g. 888.
Password	Password must be 6-16 digits, (e.g.12u3b6)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu "Outbound Routes".
Analog Phone	Please select the related FXS port for your analog phone.
Voicemail	Select this option to open the voicemail account
VM Password	Set password for Voicemail, (e.g. "1234")
Delete VMail	Check this option to delete voicemail from system after it's sent to mail box.
Email (Fax/Voicemail)	Extension user's mail box, which is used for receiving fax or voicemail (you need to open the function to fax to email/voicemail), e.g. Tom@gmail.com
Web Manager	It's allowed to login Extension Management Panel to manage extension like voicemail, call recording, call transfer, etc. when you select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting
Allowing Being Spied	Check this option to allow being spied.
NAT	Check this option if extension user or the phone is located after the NAT (Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After checking this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g. Outbound Call, Internal Call, Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS

SRTP	Enable SRTP
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary.
Video Call	Check to enable video call for this extension. And select the audio codecs you need to use.
Permit IP	Set IP-Phone permitted IP to visit this IP PBX, e.g.192.168.1.77or 192.168.10.0/255.255.255.0. IP-Phone with other IPs is not allowed to visit this IP PBX.
Audio Codec	Select what audio codec you need to use.

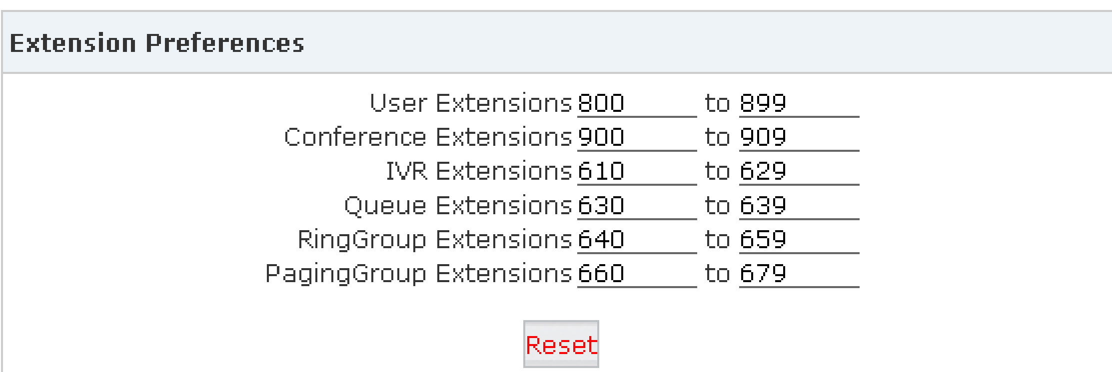
NOTE:

1. There are few default extensions which number started with "8XX", you can add or delete extension by your requirement.
2. To change the Extension
3. Maximum extensions: 100.
4. For security reason the default password is random *character / symbol / number* e.g. BB%ChH64rI, and every time when you reset to default system, it will randomly have a new password again.

Change the Extension range

To Change the range of extensions.

Go **Advanced** -> **Options** -> **General** -> **Extension Preferences**.

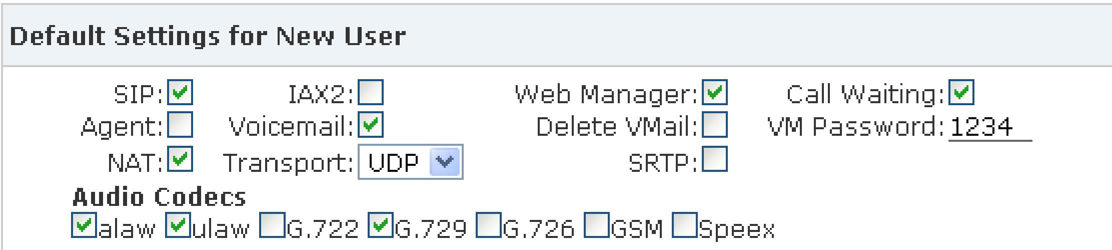


And then click **Save**.

Default settings for New User

To change default settins for New User.

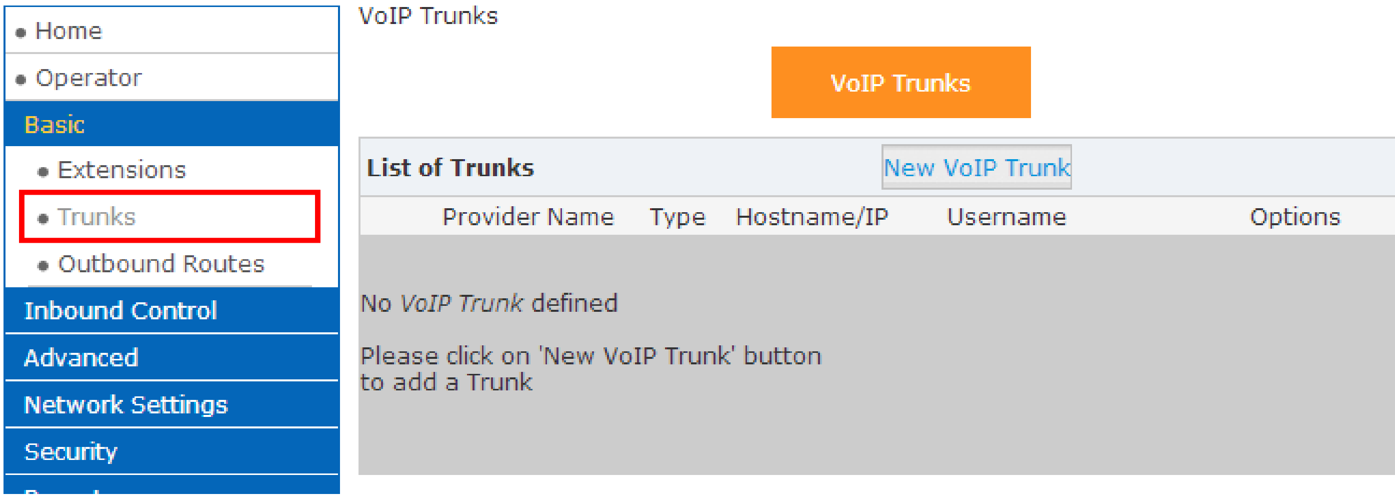
Go **Advanced** -> **Options** -> **General** -> **Default settings for New User**



And then click **Save**.

Trunks

If you want to set up outbound call to connect to PSTN (Public Switch Telephone Network) or VoIP provider, please configure on this page: **Basic** -> **Trunks**

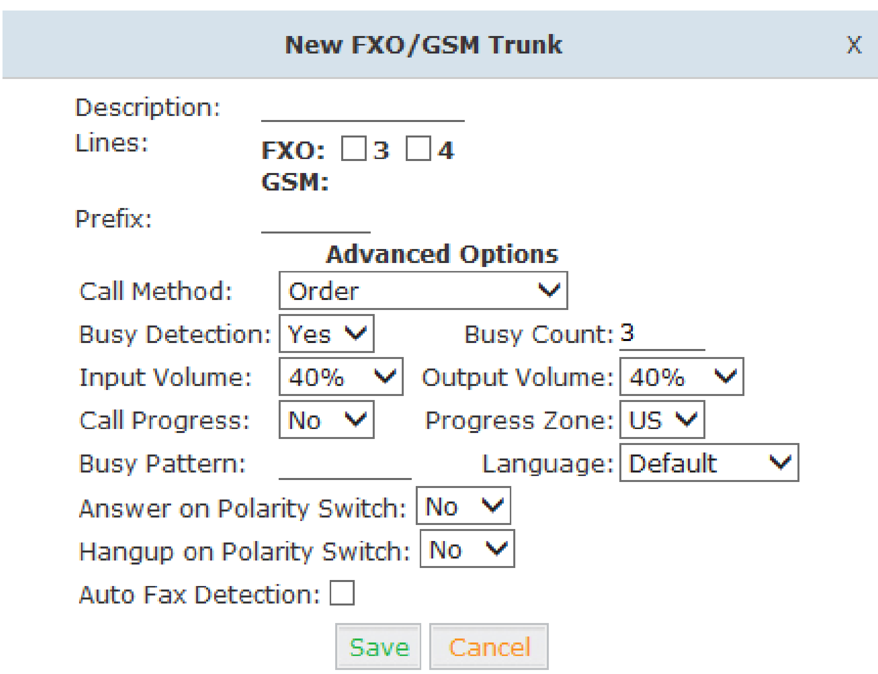


FXO / GSM Trunk

You can configure the Analog / GSM line through SIPDEX IPPBX. The same analog line can't be used in multiple trunks.

Note: *If you don't have available analog/GSM trunk Module, you can't set up trunk.*

Click **FXO/GSM Trunk** -> **New FXO/GSM Trunk**



Item	Explanation
Description	Define the description for this trunk (figure or character).
Lines	Available line
Prefix	The prefix will be added to the dialed number automatically when this trunk is in use.
Advanced Options	Advanced Options for this trunk, e.g. Call Method, Busy Detection, etc.

NOTE:

Set the available analog line for this device. The same analog line can't be used in several FXO/GSM trunks. If you don't have available analog line, you can't set up FXO/GSM trunk.

E1/T1 Trunk (PRI Trunk)

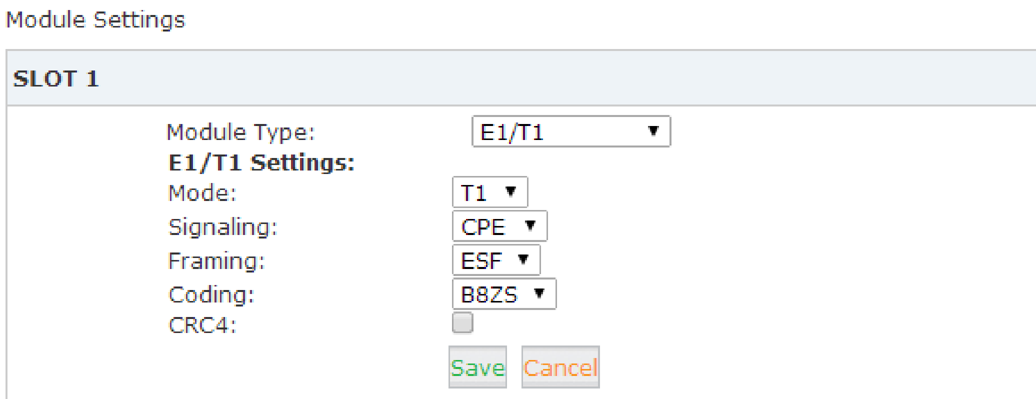
You can configure the T1 / E1 line through SIPDEX IPPBX.

Note:

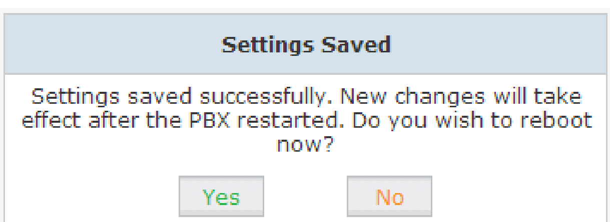
1. *If you don't have available 1PRI trunk Module, you can't set up trunk.*
2. *1PRI Module must be install in slot 1 only*

Step for configuration for the E1/T1

1. Go **System** -> **Module Settings**
2. Choose E1 / T1 connection
3. Choose your **Signaling , Framing , Coding , CRC4** settings for connect your PSTN
4. Click **Save**

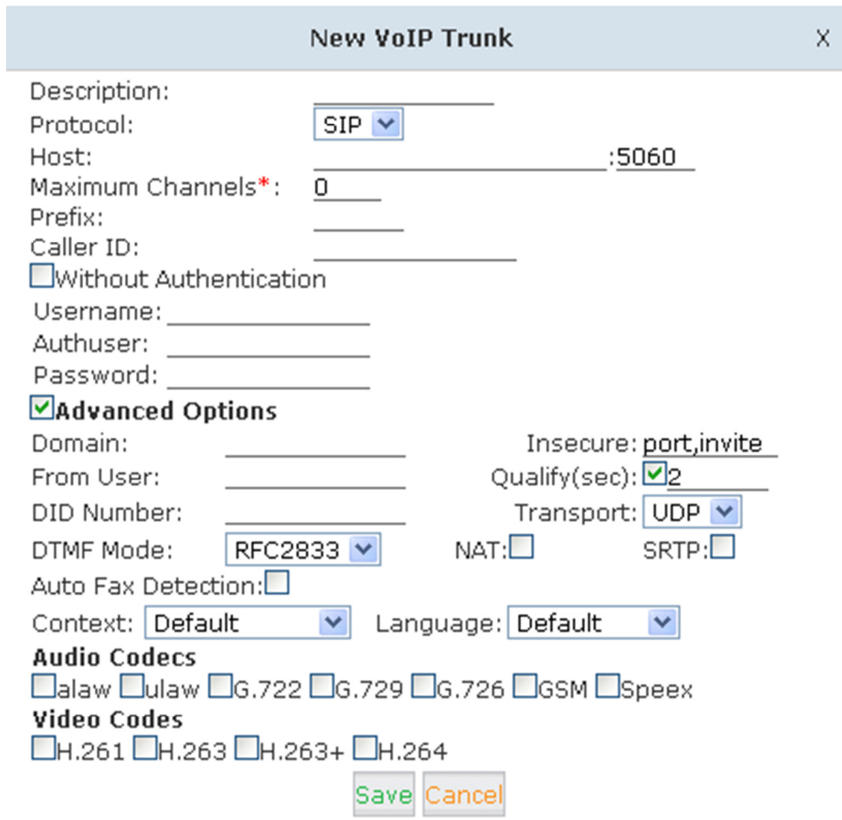


5. Click **Yes** to confirm and the system will reboot.



VoIP Trunk (SIP Trunk)

1. Click **VoIP Trunk** -> **New VoIP Trunk**

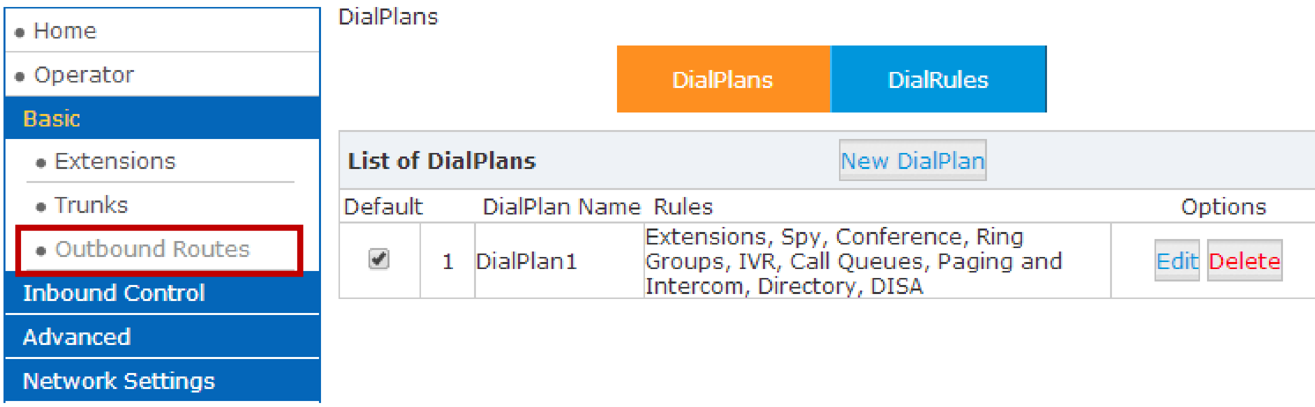


Item	Explanation
Description	Define the VoIP (figure or character).
Protocol	Select protocol for outbound route, SIP or IAX2.
Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; “0” = no limitation).
Prefix	The prefix will be added in front of your dialed number automatically when the trunk is in use.
Caller ID	This Caller ID will be displayed when user make outbound call.
Without	If you don’t need the Authentication when connecting the IP PBX, please check this option.
User Name	User Name provided by VoIP Provider.
Password	Password provided by VoIP Provider.
Advanced Options	Advanced options for this trunk, e.g. codec, dial plan, etc.

Outbound Routes

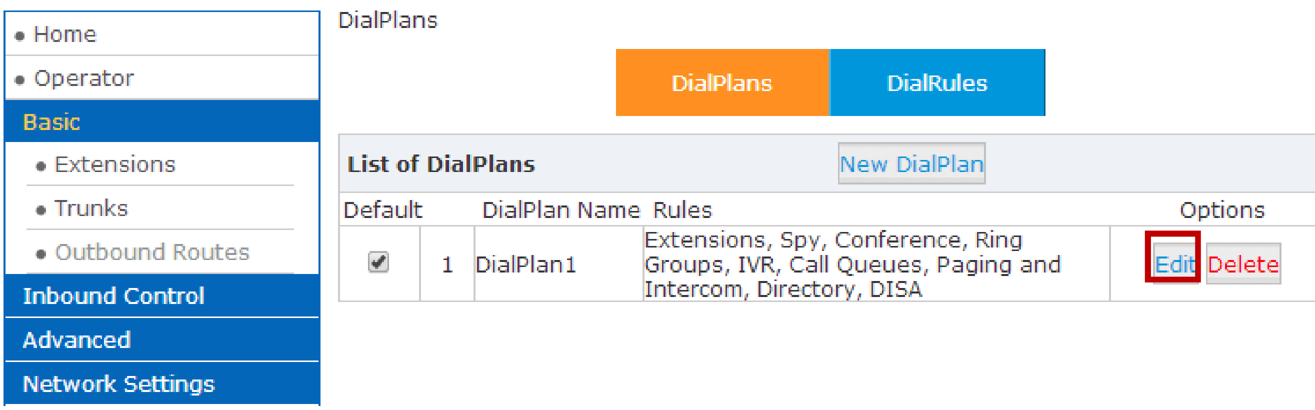
Outbound Routes is to define what trunk is used for outbound call by extension user. If user don't allow extension user to call out, please ignore this part.

Configure on this page: **Basic** -> **Outbound Routes**

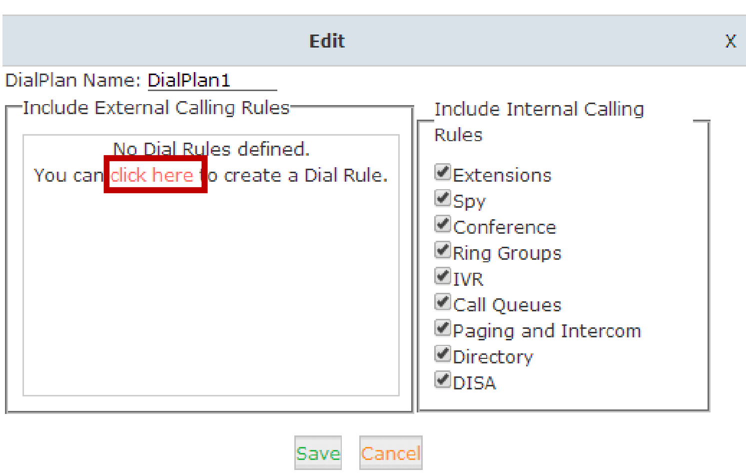


DialPlans

On **DialPlans** click **Edit** the **DialPlan1**

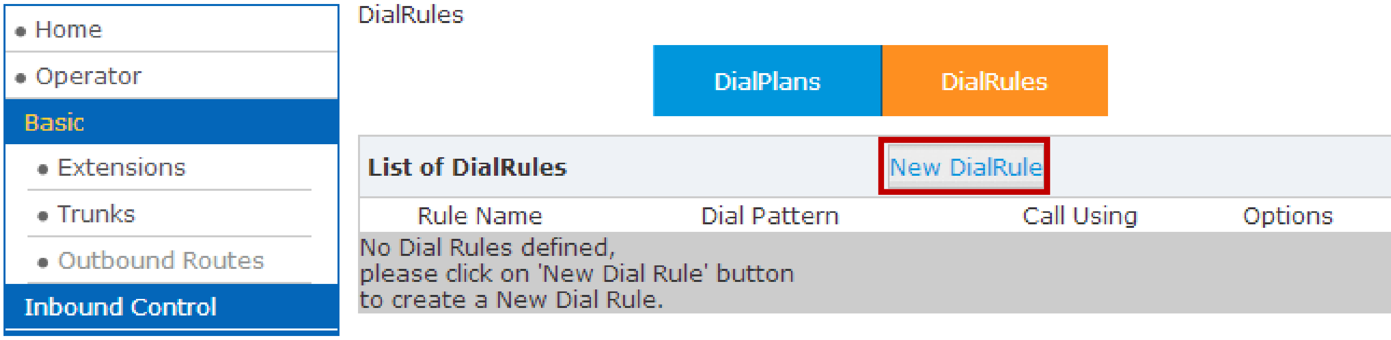


Click [**Click here**] to create the new DialRules for using the trunk to call out.

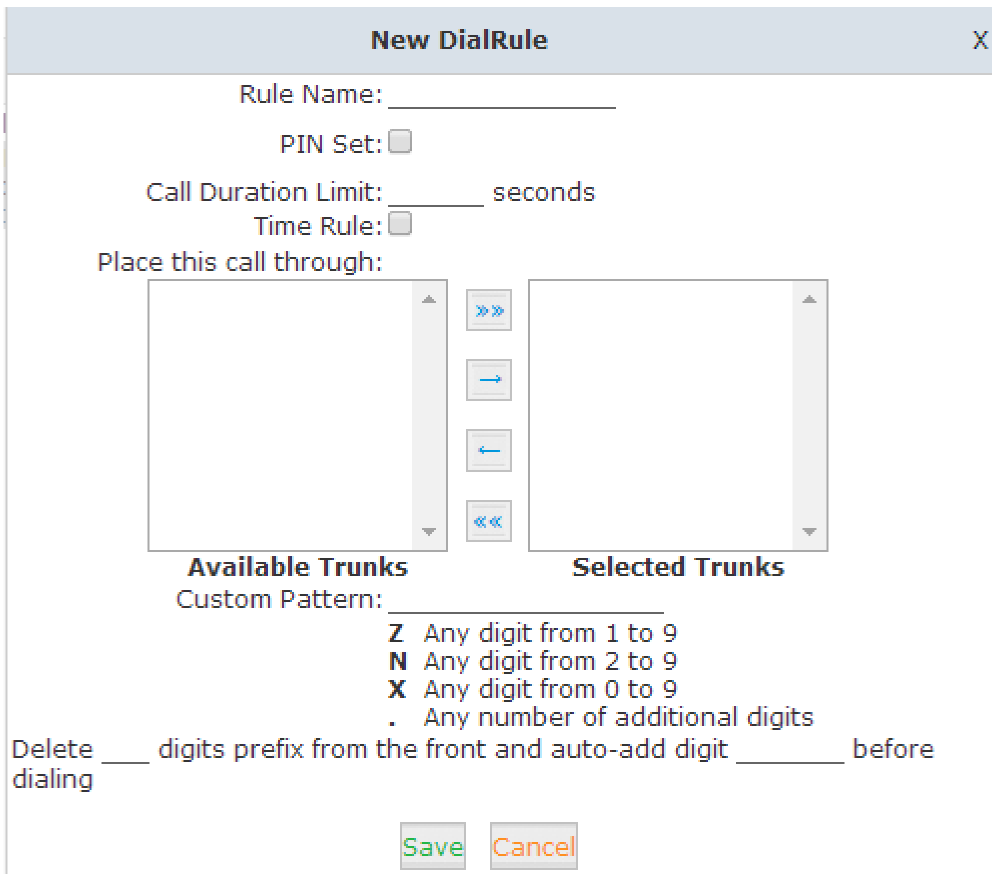


DialRules

Click New **DialRule** to create the New Dial Rule



New Dial Rule



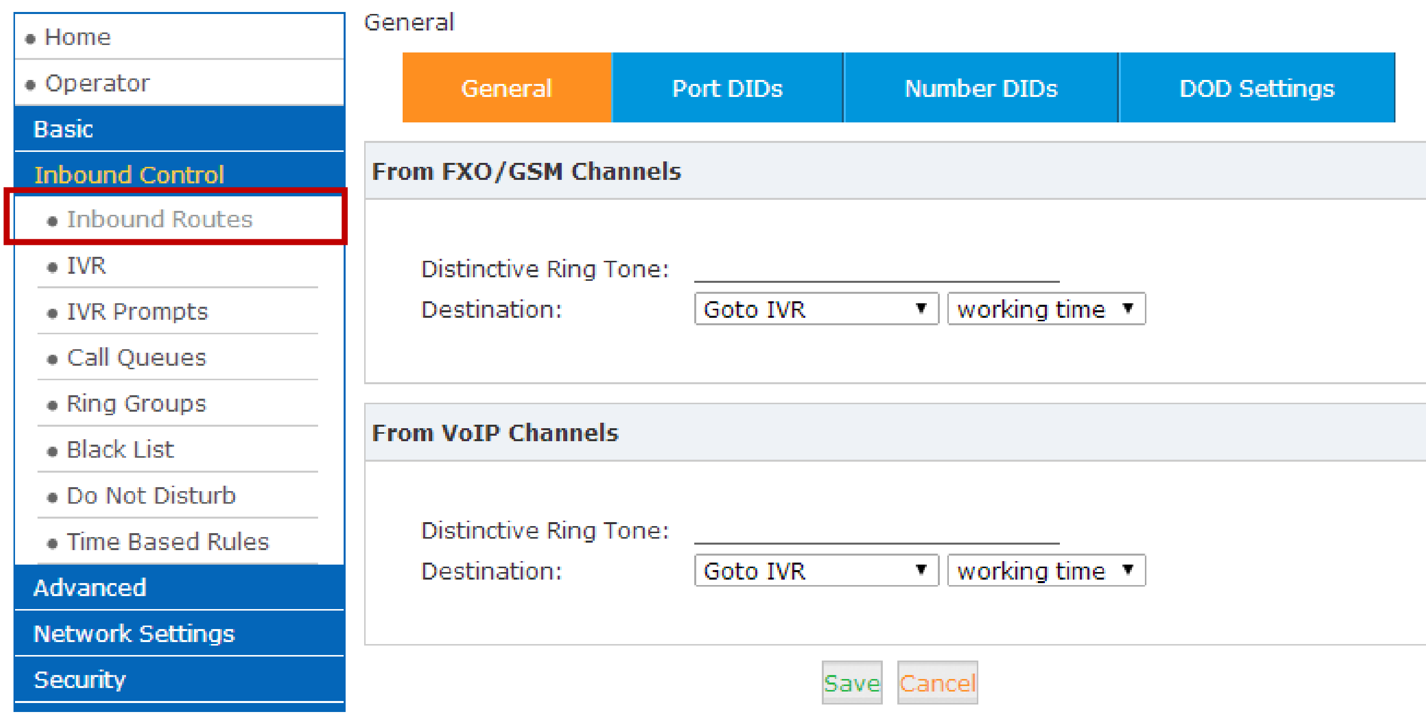
Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule.
Place this call through	Select a trunk for this dial rule
Custom Pattern	**Z** any one digit from 1 to 9 **N** any one digit from 2 to 9 **X** any one digit from 0 to 9 **.** any one digit or multi-digit figures
Delete[ ]digits prefix	If one digit prefix be deleted, when dial 12345, 2345 will be sent.
Auto-add digit[ ]	If figure “1” is added,123451 will be sent when dialing 12345

Chapter 4: Incoming Call

Inbound Routes

When a call is incoming from outside, you want to forward this call to an extension, ring group or IVR. This Chapter will introduce you how to deal with the inbound calls.

Click **Inbound Control** -> **Inbound Routes**



General

Distinctive Ring Tone: mapping the custom ring tone file, e.g. set distinctive ring tone as “External”, the phone will play this ring tone when receiving the call.

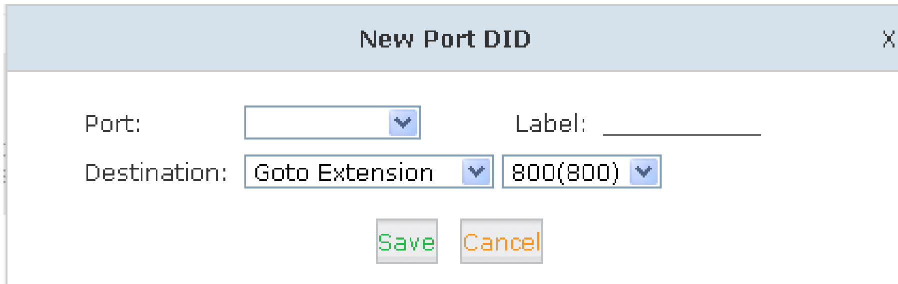
Note: The phone must support such feature as well.

When incoming calls come from PSTN (FXO/GSM, PRI, VoIP), the calls can be accessed to Extension User, Call Queue, Conference, IVR, etc. You can choose freely based on your condition.

Port DIDs

If user wants to make the incoming call from the outbound line (FXO/GSM trunk) access to the specified extension user, Ring Group, call queue, conference or IVR, please configure it here:

Click **Port DIDs**-> **New Port DIDs**:

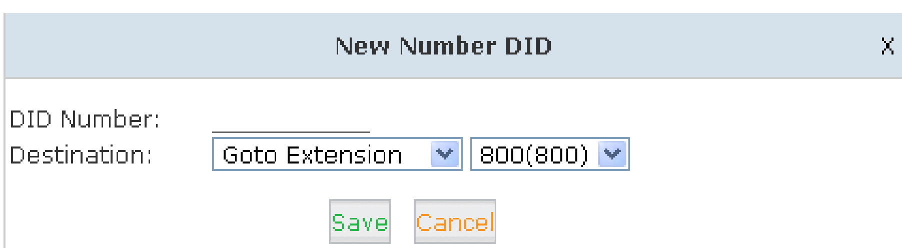


Item	Explanation
Port	Select the port for outbound line.
Label	Set a label for this port. When incoming calls are from this port, the label will be displayed.
Destination	Incoming calls will access directly to this destination (extension user, Ring Group, call queue, conference, or IVR).

Number DIDs

If you want to select the destination of inbound calls on PRI/BRI or VoIP Trunks based on the incoming DNIS (dialed number or DID). You can specify the DID and destination (user extension, queue, conference bridge, or IVR:

Click **Number DID** -> **New Number DID**

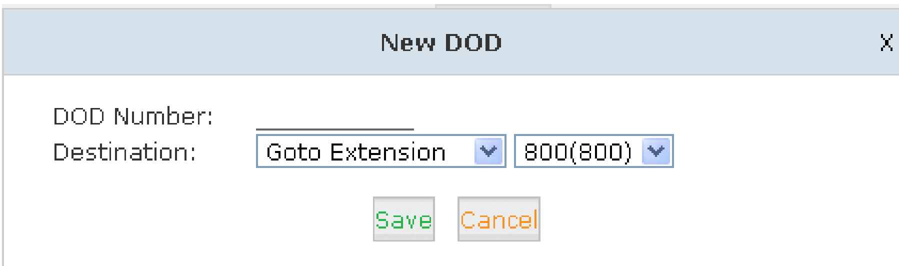


Item	Explanation
DID Number	DID number calling into VoIP (This number is configured in the advance option of VoIP trunk).
Destination	Choose a specified extension, call queue, conference or IVR to be directed to call.

DOD Settings

If user wants to make the outbound call directly to the specified extension user, Ring Group, call queue, conference, IVR, please configure it here.

Click **DOD Settings** -> **New DOD**

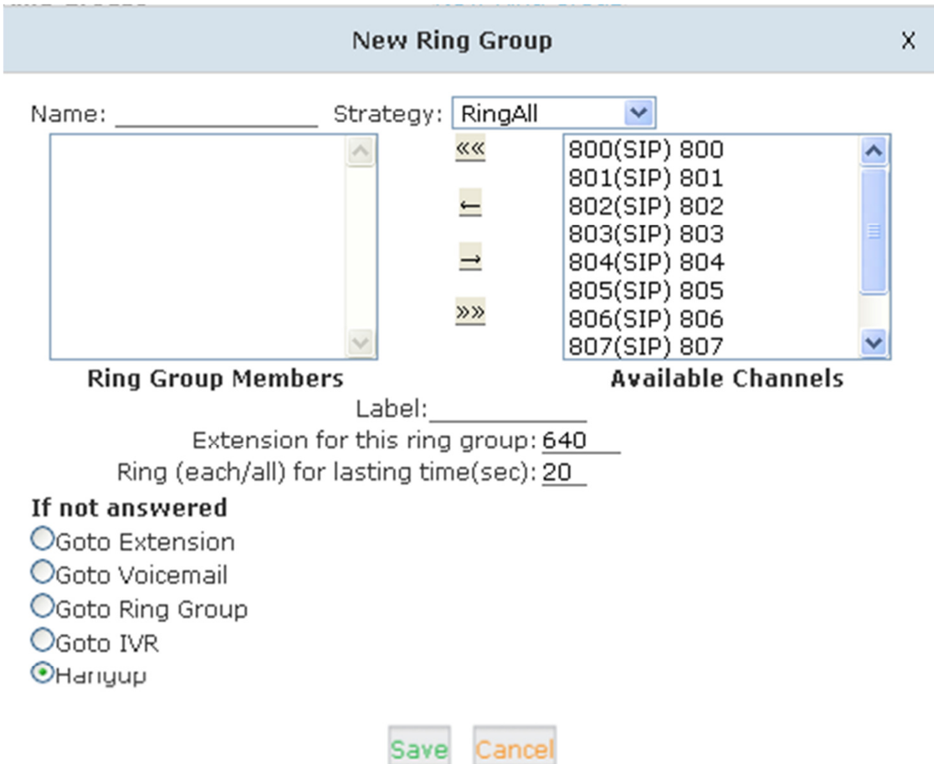


Item	Explanation
DOD Number	Set the DOD number, and use it to match the Caller ID.
Destination	Outbound calls will access directly to this destination (extension user, Ring Group, call queue, conference, or IVR).

Ring Groups

Ring Group is defines a group number that rings a group of phones simultaneously, stopping when any one of them is picked up. If ring time exceeds a defined time, the call will be directed to IVR or others based on your configuration.

Click **Inbound Control** -> **Ring Groups** -> **New Ring Group**

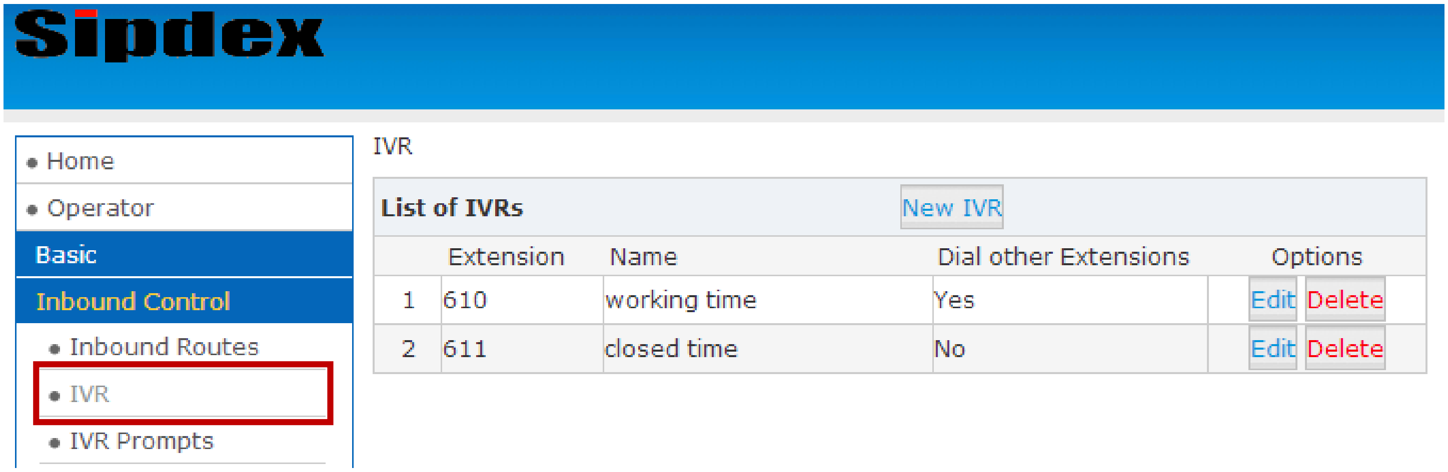


Item	Explanation
Name	Define a name for the Ring Group.
Strategy	Select “Ring All” or “Ring in order”.
Ring Group Members	Select the Ring Group Member from “the Available Channels”, click to add.
If not answered	You can choose to forward the call to extension, voicemail, ring group, IVR or hang up if not answered.

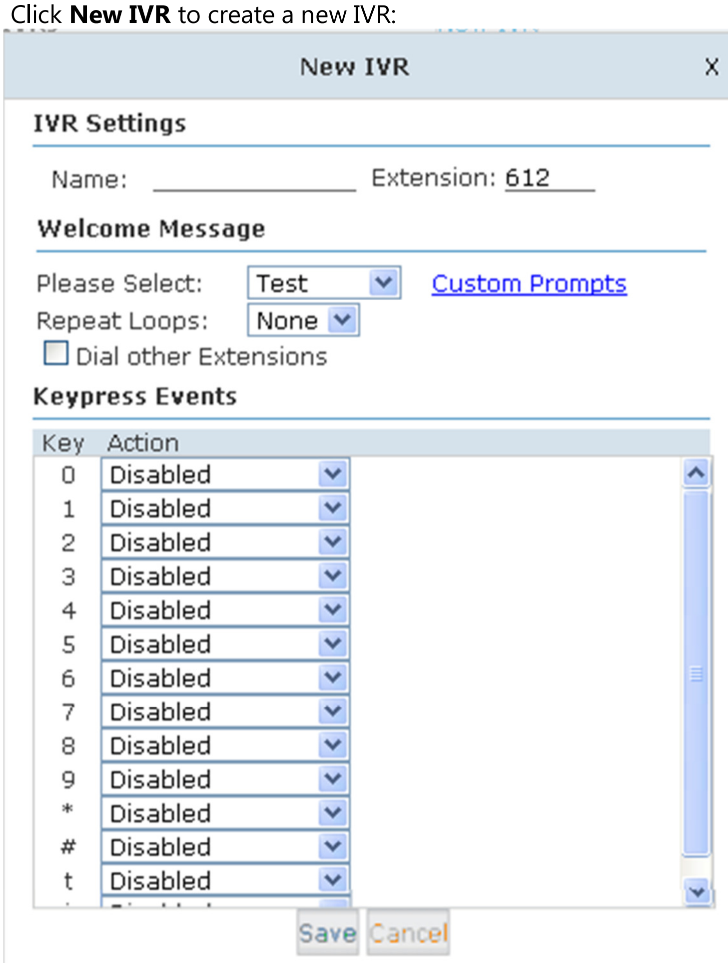
IVR (Interactive Voice Response)

IVR allows customers to interact with IPPBX via a telephone keypad, after which they can service their own inquiries by following the IVR dialogue. IVR systems can respond with prerecorded or dynamically generated audio to further direct users on how to proceed.

Please configure on this page **Inbound Control** -> **IVR**



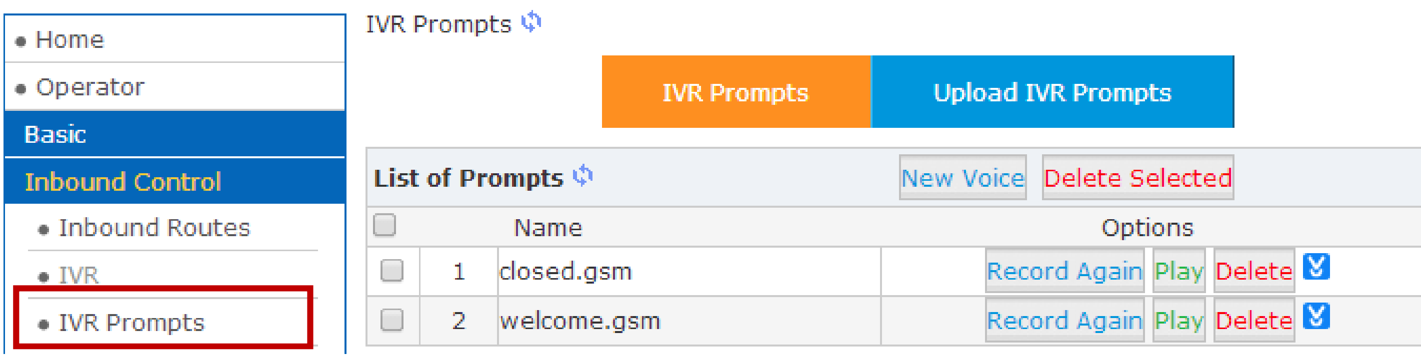
Click **New IVR** to create a new IVR:



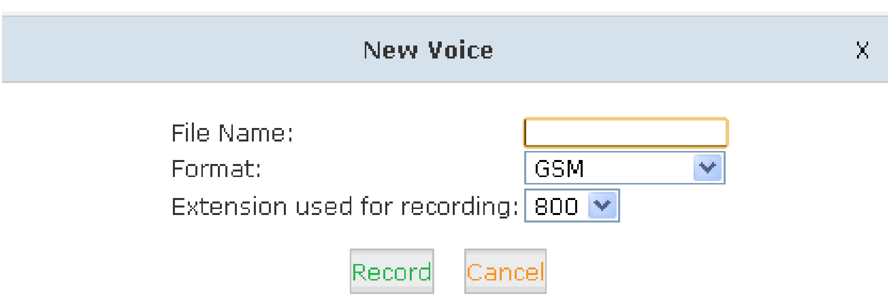
Item	Explanation
Name	Set a name for the IVR
Extension	If you want to listen to the IVR by dialing number, input an extension Number you want.
Please Select	Select IVR audio file, please configure in this page: **Inbound Control** -> **IVR Prompts**
Repeat Loops	Loop times to repeat playing the IVR prompt.
Dial Other Extensions	Allow caller to dial other extensions besides the ones listed below.
Key Press Events	Each digit will be related to the actions defined in the blank.

IVR Prompts

Record / upload or listen IVR sound from extension. Please configure on this page:

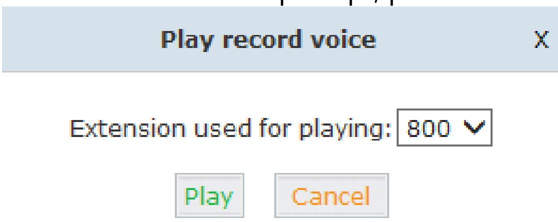


To record IVR Prompt by phone click **New Voice**



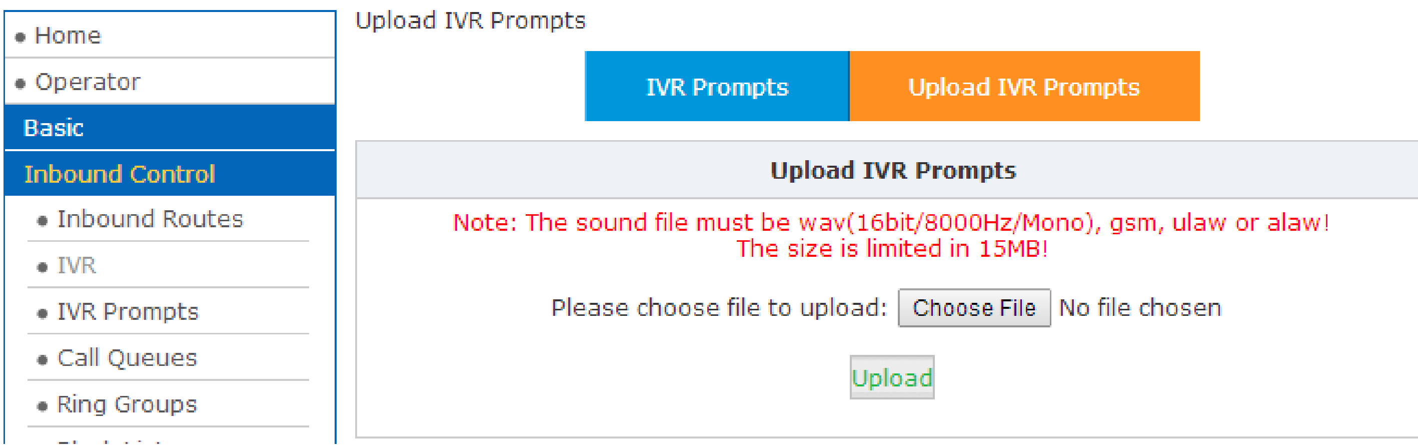
Item	Explanation
File Name	Define a name for this voice file.
Format	Select the voice format, GSM / WAV (16bit) supported only.
Extension used for recording:	Select the extension which is used for recording the IVR prompt. Click **Record**, the extension will ring, and then you can pick up the phone and record.

You want to hear the prompt, please click **Play**



Select the extension, click **Play**, the selected extension will ring, and you will hear the recorded prompt after picking up the phone

Upload IVR prompt



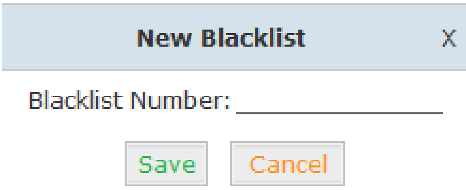
NOTE:

1. Uploading customized audio file must be wav(16bit/8000Hz/Mono), gsm, ulaw or alaw.
2. The size is limited in 15MB!

Blacklist

If some numbers need to be blocked, you can use this functionality

Click **Inbound Control** -> **Blacklist** -> **New Blacklist**

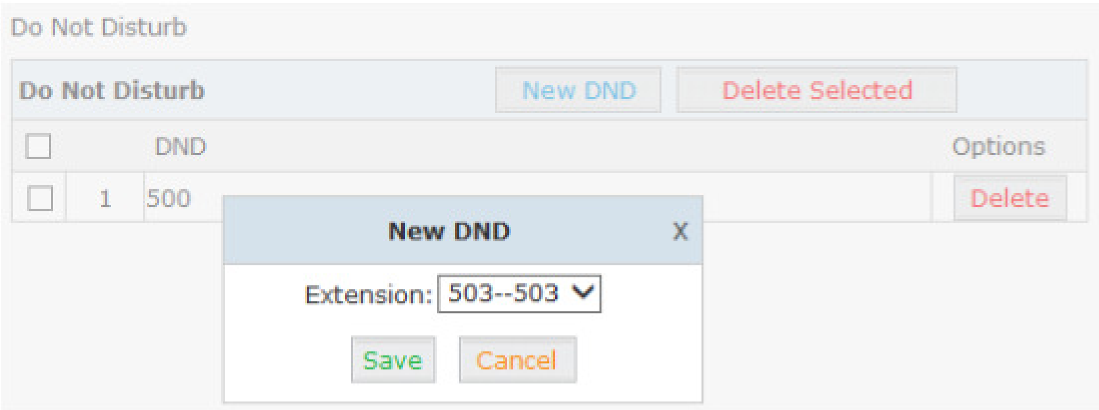


Input the caller ID in the blank and click **Save**. Next time calls from this caller ID will be blocked.

Do Not Disturb

Do Not Disturb (DND) function prevents incoming call on an extension for which DND is activated.

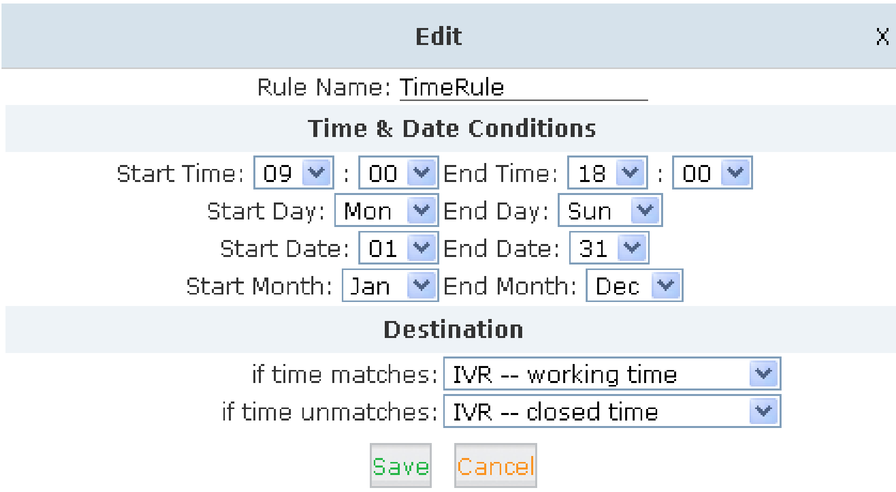
Click **Inbound Control** -> **Do Not Disturb**



Time Based Rules

When an inbound call is processed, if the current time of the PBX is within these parameters, then the “if time matches” destination will be used for the call. If the current time of the PBX is outside these parameters, then the “if time does not match” destination will be used for the call.

Please set from this page: **Time based Rule** -> **New Time Rule**



New Time Rule:

Item	Explanation
Rule Name	Define the name for this Time Rule.
Time & Date	Set time segment for Day/ Date/ Month.
Destination	How to deal with the inbound call in different time segments. For example, inbound call can be directed to operator in working time.

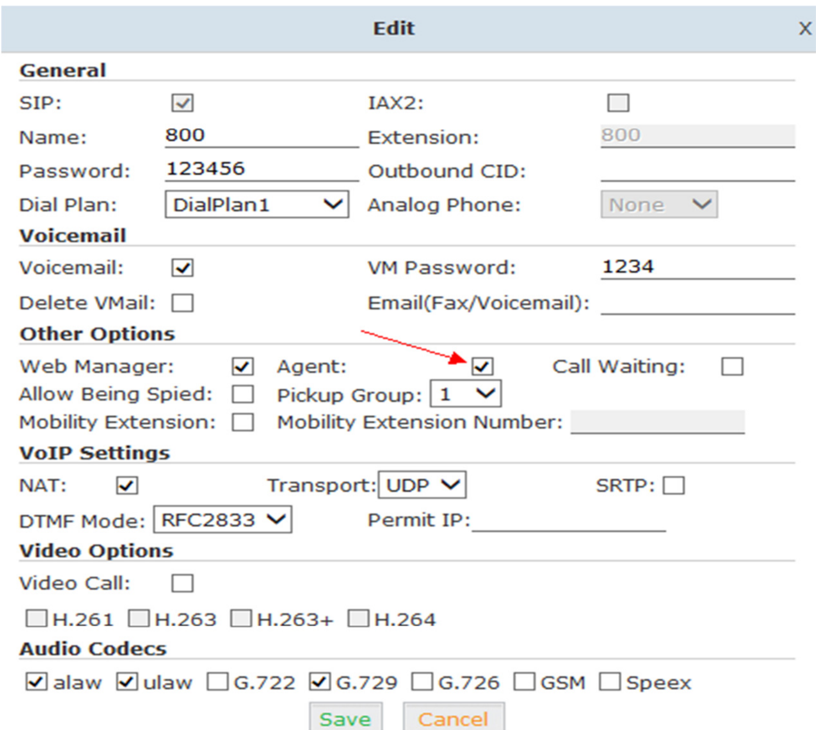
Call Queues

Call Queuing is a queuing system that allows you to accept more calls into your telephone system than you have extension's agent capable of answering them.

Create Agent

To allow a user to be considered an agent in a Call Center queue, please check the "Agent" option for that specific user extension.

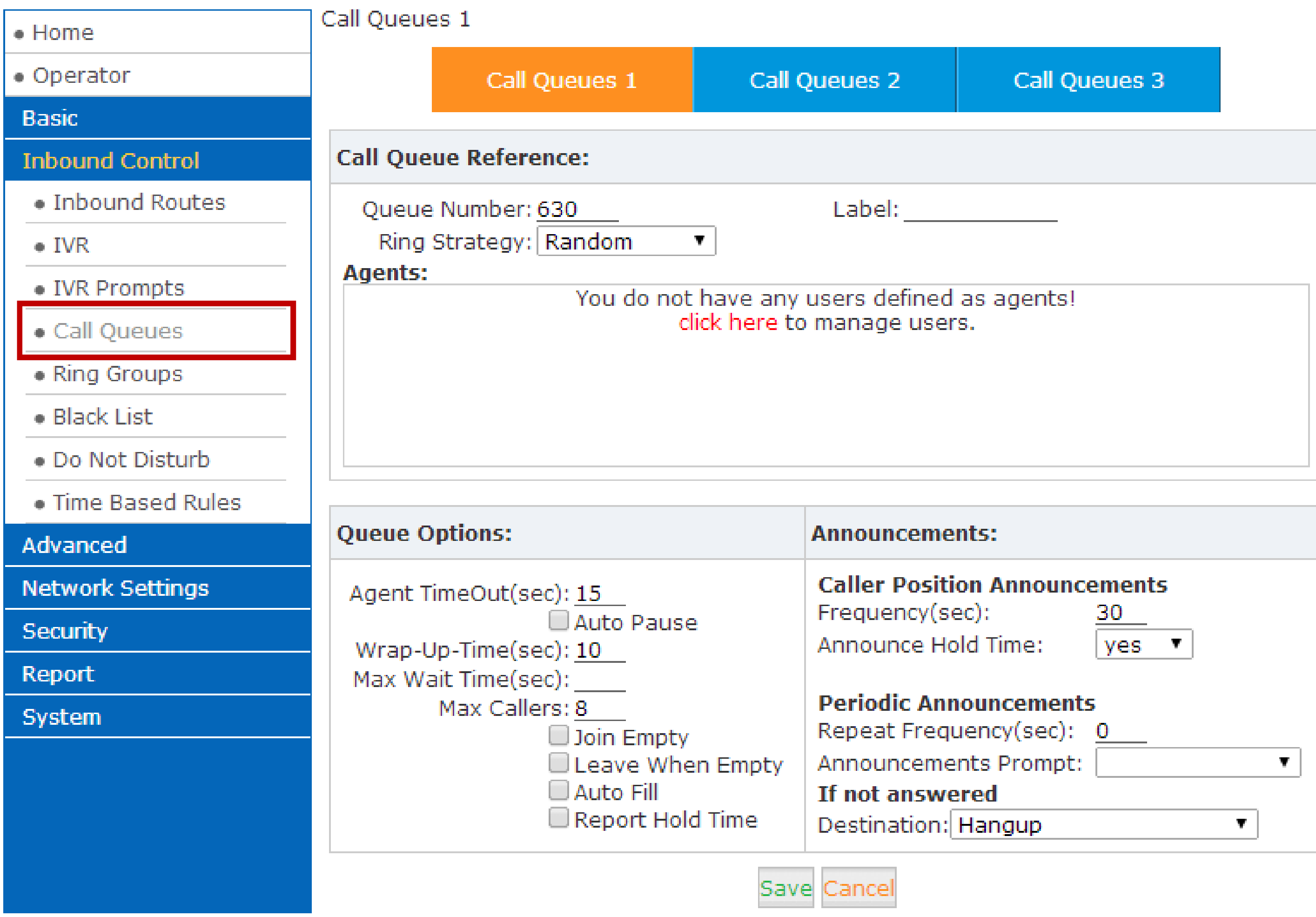
Check **Agent** and **Save**



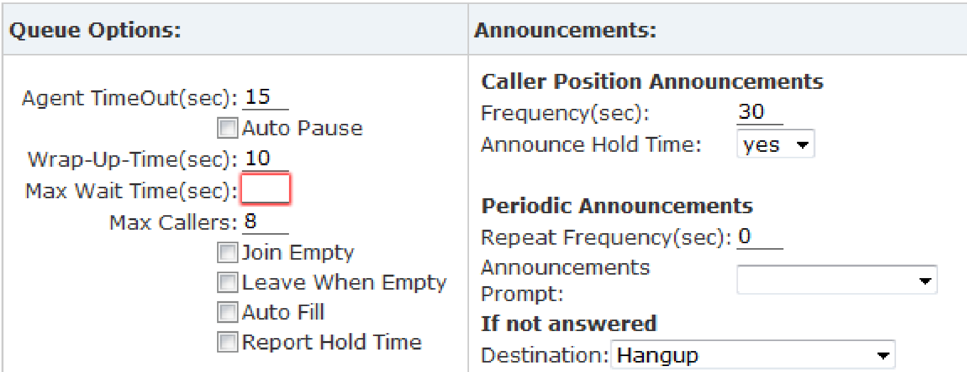
Call Queues Configuration

Total have 3 Call Queue you can use in this system.

Click **Inbound Control** -> **Call Queues**



Item	Explanation
Queue Number	Define an extension number for the queue.
Label	Define the label for the queue.
Ring Strategy	RingAll -- Ring all available agents until one answers (default) RoundRobin -- Every available agent will take turns to ring. LeastRecent -- Agent with the least calls rings FewestCalls -- Agent with the fewest completed calls rings. Random -- Agent
Agent	Every extension defined as Agent will be listed here. Selected agent will be a member of the current Queue.



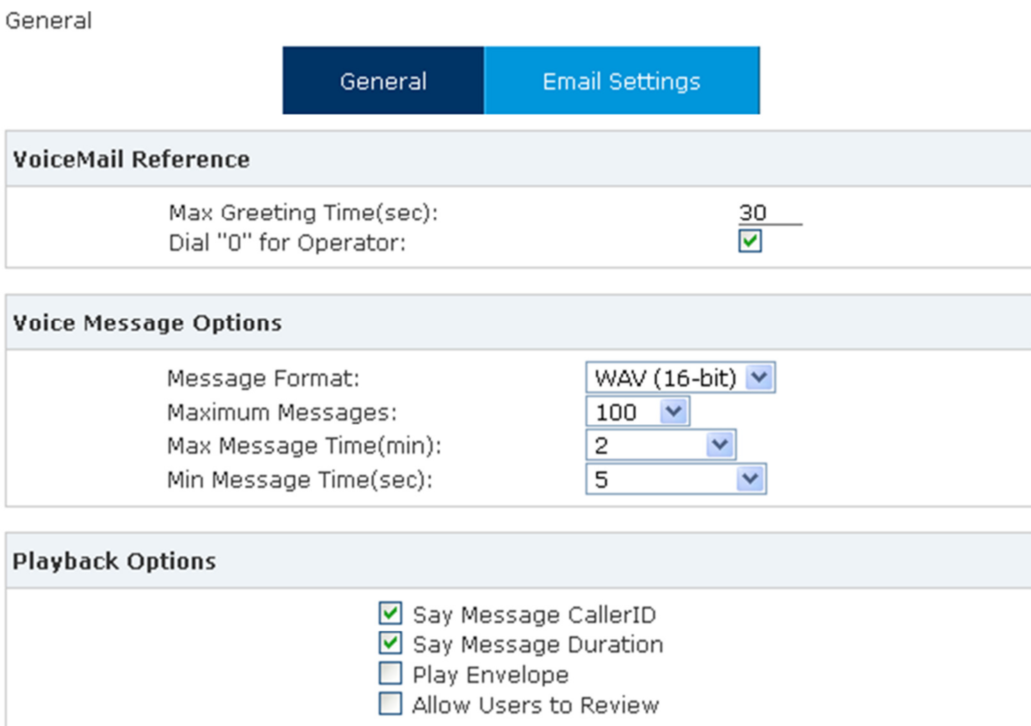
Item	Explanation
Agent TimeOut (sec)	The next Agent will ring after this time.
Auto Pause	Pause the Agent when it fails to answer the first call.
Wrap-Up-Time (sec)	Wrap-up time between the first answer and second answer. (Default is 0, which means no wrap-up time.)
Max Wait Time (sec)	Maximum wait time for callers in the queue.
Max Callers	Maximum number of callers who are allowed to wait in the queue.
Join Empty	Allow callers to enter the Queue when no Agents are available. If this option is not defined, callers will not be able to enter Queues with no available agents.
Leave When Empty	All callers in the Queue will be moved out when new caller cannot enter the Queue. This option cannot be used with Join Empty simultaneously.
Auto Fill	Callers will be distributed to Agent automatically.
Report Hold Time	Report the hold time of the next caller for Agent when the Agent is answering the call.
Frequency(sec)	Repeat frequency to announce the hold time for callers in the Queue. (“0” means no announcement).
Announce Hold Time	Announce the hold time. Announce (yes), not announce (no) or announce once (once), I will not be announced when the hold time is less than 1 minute.
Repeat Frequency(sec)	Interval time to play the voice menu for callers. (“0” mean not to play).
Announcement Prompt	Select a prompt as the Announcements Prompt from the IVR Prompts.

Chapter 5: System Feature

Voicemail

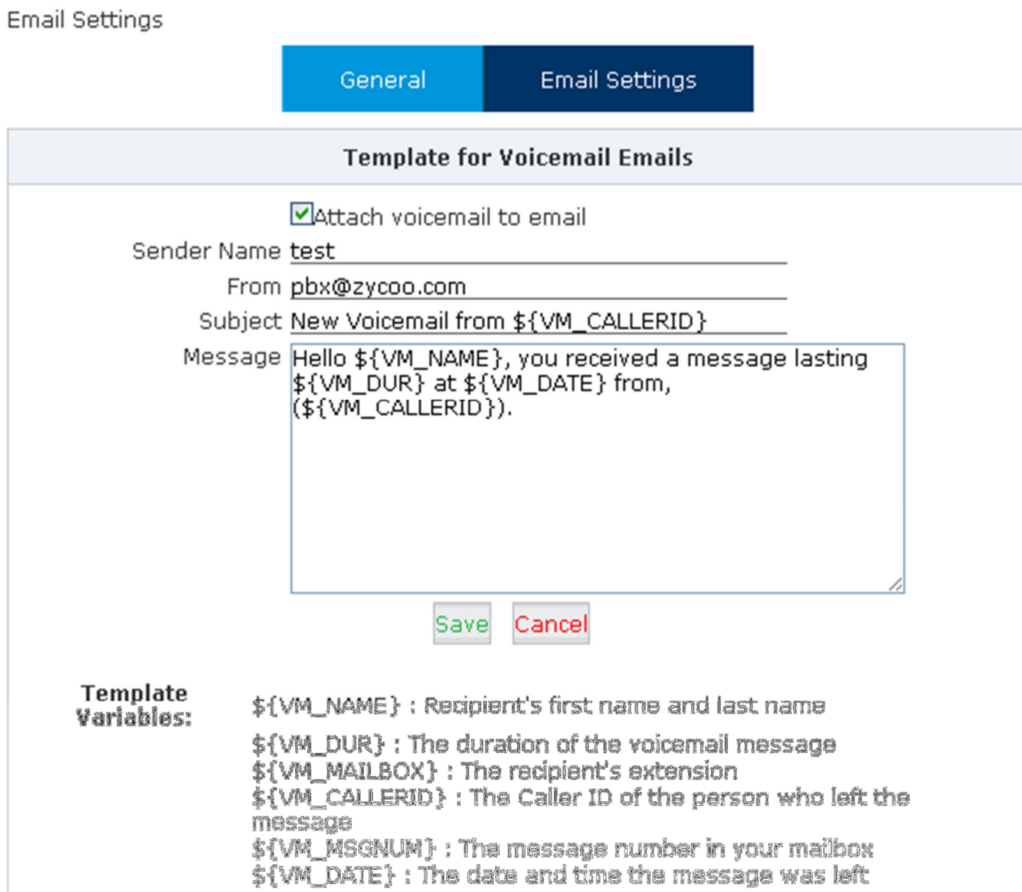
Details configuration on Voicemail: Voicemail Reference/ Voice Message Options/ Playback Options. If you need to send message by mail to your defined mailbox, you must configure SMTP and Email model.

Click **Voicemail** to display the dialog as shown below



Item	Explanation
Max Greeting Time(sec)	Maximum Greeting Time
Dial "0" for Operator	Dial "0" to cancel the voicemail and forward to Operator.
Message Format	Save the voice message as this format, WAV (16-bit) or Raw GSM.
Maximum Messages	Maximum messages to be allowed to leave.
Max Message Time(min)	Maximum Time for each message to be allowed to leave.
Min Message Time(sec)	Minimum Time for each message. The message will be deleted automatically if the time is less than the minimum message time.
Say Message Caller ID	Checking this option, Caller ID will be played when user login email to receive the voice message.
Say Message Duration	Checking this option, the message duration will be played before playing the voice message.
Play Envelop	Envelop includes date, time and caller ID.
Allow Users to Review	Check this option to allow users to review the voice message.

Voicemail to Email Settings

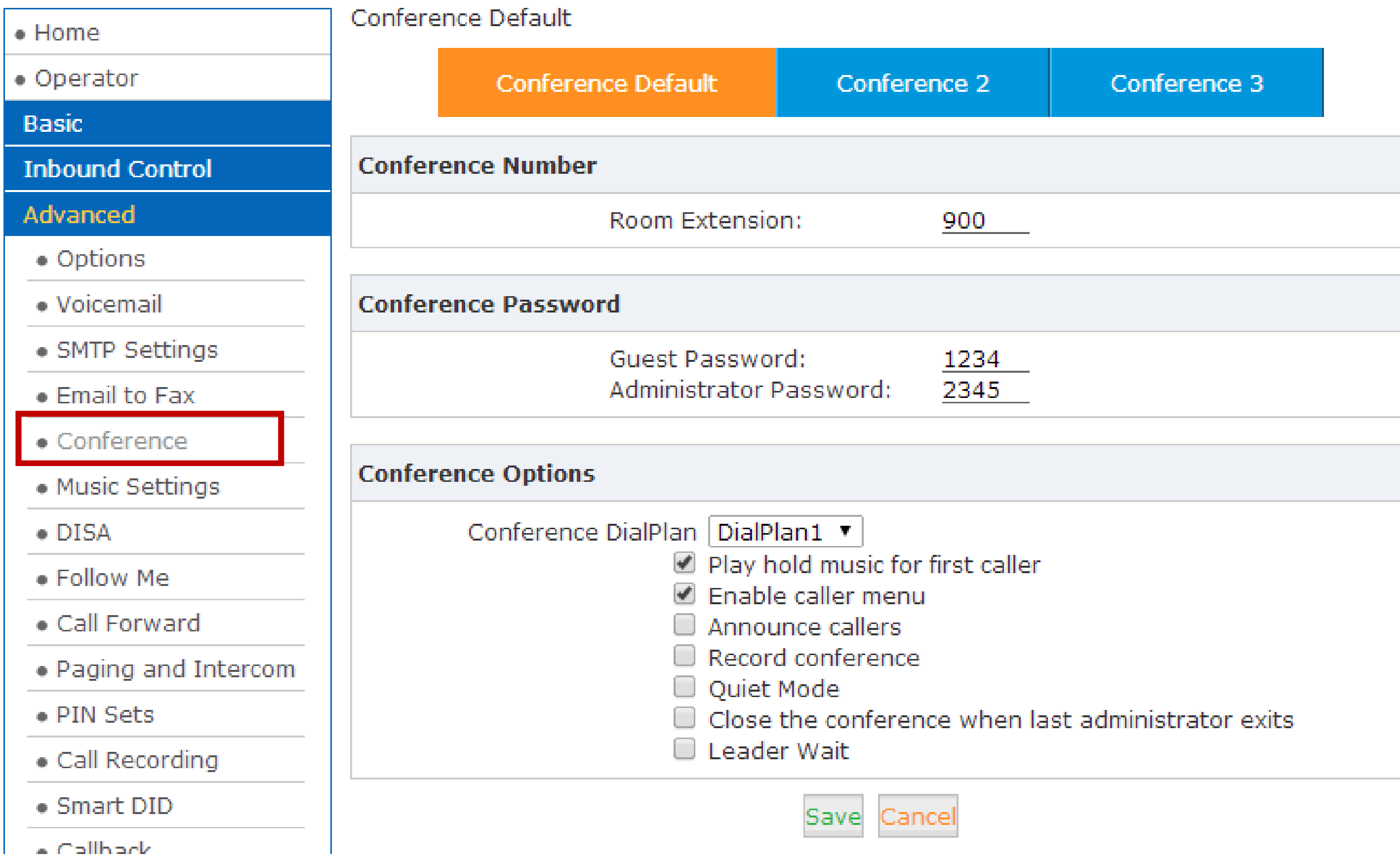


Item	Explanation
Attach voicemail to Email	The voicemail will be sent as attachment to the user’s Email.
Sender Name	The sender’s name will be displayed when you receive the Email.
From	Mailbox to send email
Subject	Subject of the Email.
Message	Input the Email template.

Conference Bridge

A conference bridge allows a group of people to participate in phone call. The most common form of bridge allows participants dial into a virtual meeting room from their own phone. Meeting rooms can hold dozens or even hundreds of participants.

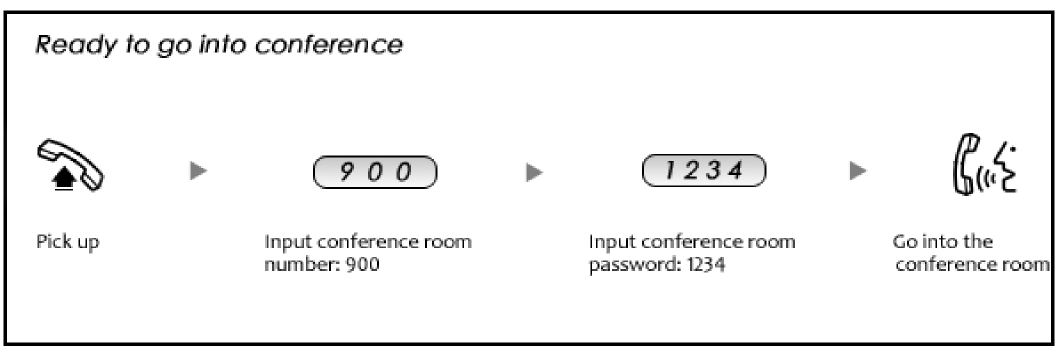
This M-200 supports 3 conference rooms. Please configure it on this page



Item	Explanation
Conference Number	The number that users call in order to access the conference room; the default number is “900”.
Conference Password	Password for users to access the conference, e.g."1234".
Administrator	Password for administrator to access the conference.
Conference DialPlan	Use this dial plan to invite other participants.
Play hold music for the first participant	Check this option to play the hold music for the first participant in the conference until another participant enters this conference.
Enable caller menu	Check this option to allow the participant to access the Conference Bridge menu by pressing “*” on the dialpad.
Announce callers	Check this option to announce to all Bridge participants that a new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If this option is checked, all the participants in the conference can hear only, but it is not allowed to speak.
Leader Wait	Wait until the conference leader (administrator) enters the conference before starting the conference.

Please check the following diagram to learn:

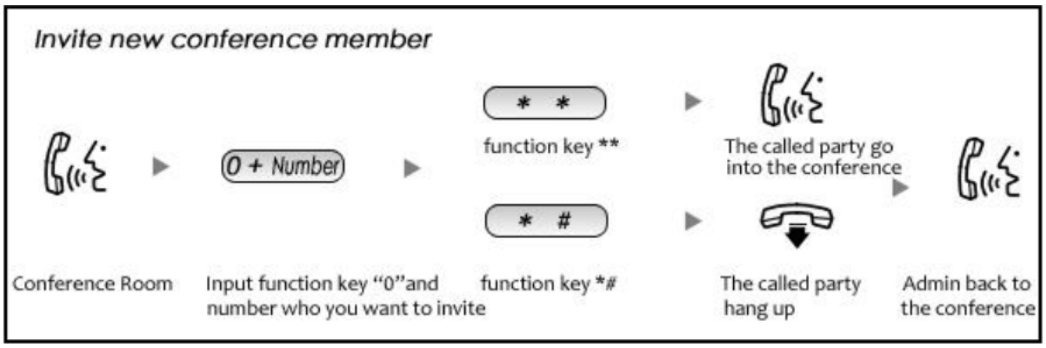
Go to conference:



In the conference, admin can add new participant (extension user or external number) to the conference.

In the conference, the administrator can invite new guest (extension user or external number) to the conference. (Default password for admin is 2345)

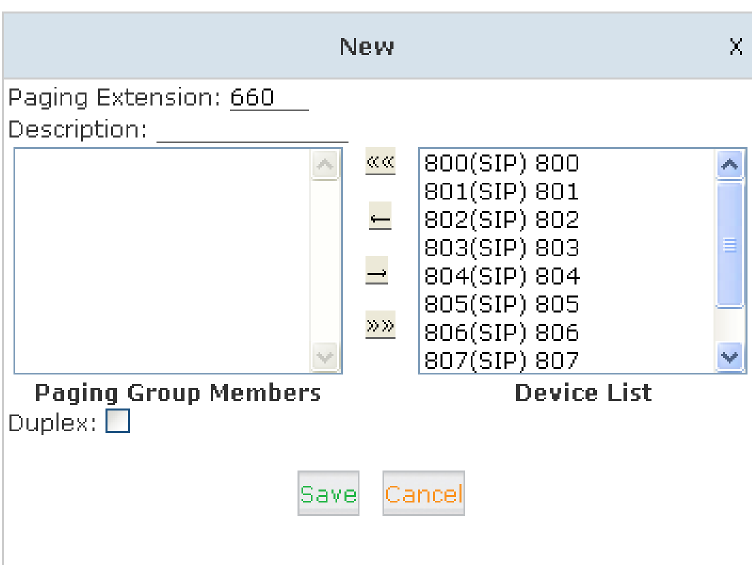
Learn how to invite new guest to the conference as the diagram is shown below:



Paging and Intercom

Paging and Intercom is used for calling a paging extension; all terminals which support this function will be picked up automatically and listen; meanwhile, it supports duplex.

Click **Advanced** -> **Paging and Intercom** -> **New Paging Group**



Item	Explanation
Paging Extension	The number users will dial to page this group.
Description	Provide a descriptive title for this Page Group.
Paging Group Members	Selected device(s) on this page
Device List	Select Device(s) to page.
Duplex	Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference".

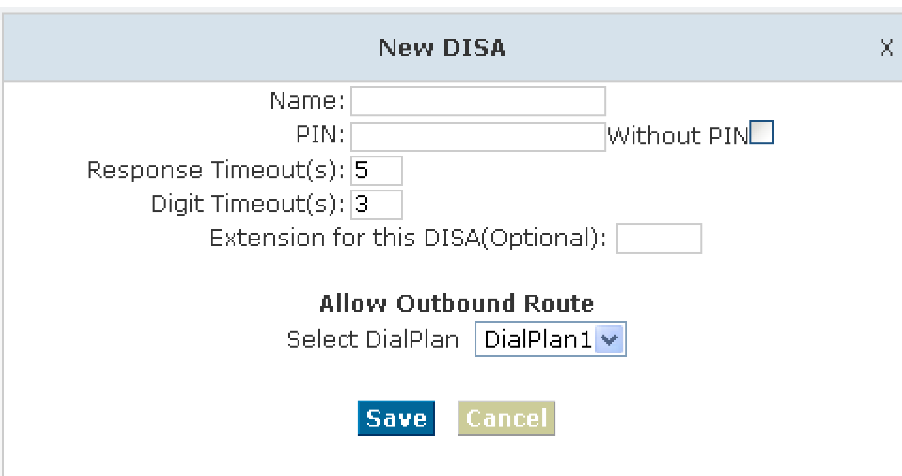
NOTE:

For Paging/Intercom function extension (IP phone), support Auto Answer

DISA

A trunk call is made to the PBX, and call is made to another trunk through outbound route of the PBX. This trunk can make international calls. You are out of the office and want to contact your customer in a foreign country. Now you can dial DISA number after PIN authentication. You are now connected to your customer, and you can speak to your customer now.

Click **DISA** -> **New DISA** to display the dialog as shown below

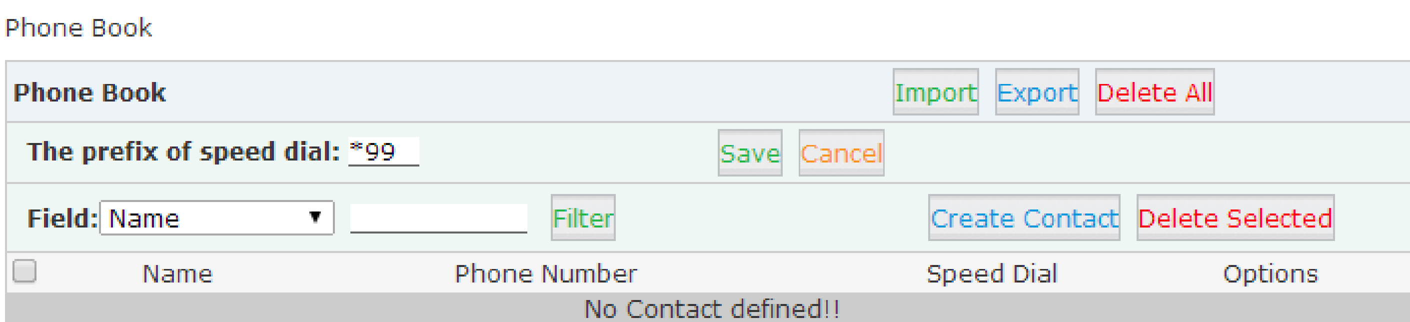


Item	Explanation
Name	Define a name for DISA.
PIN Set	User will be prompted to input this number when PIN Authentication is needed.
Record in CDR	Check to record.
Response Timeout(sec)	The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds
Digit Timeout(sec)	The maximum interval time between digits when typing extension number is 5 seconds by default.
Extension for this DISA(Optional)	If you want to access DISA by dialing an extension, you can define an extension number for this DISA.
Select Dial Plan	Select the Dial Plan for this DISA.

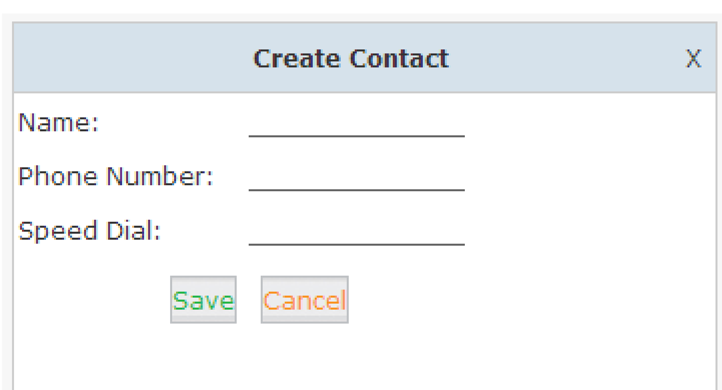
Phone book

Manage your Phone book and set the Phone book Contact for speed Dial.

Click **Advanced** -> **Phone book**



Create Contact



Using Phone book for Speed Dial

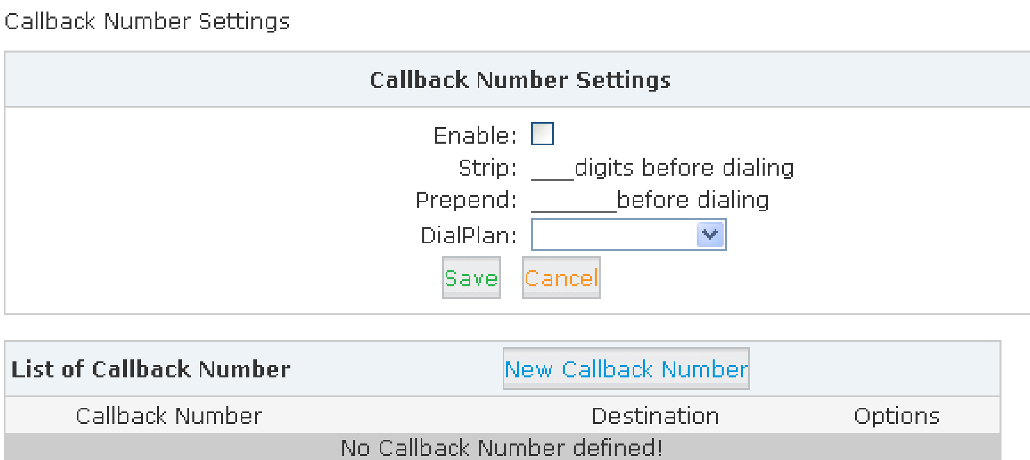
E.g. prefix is *99, speed number is 00, destination telephone number is 2345678. When dialing *9900,

The call is going to 2345678 automatically.

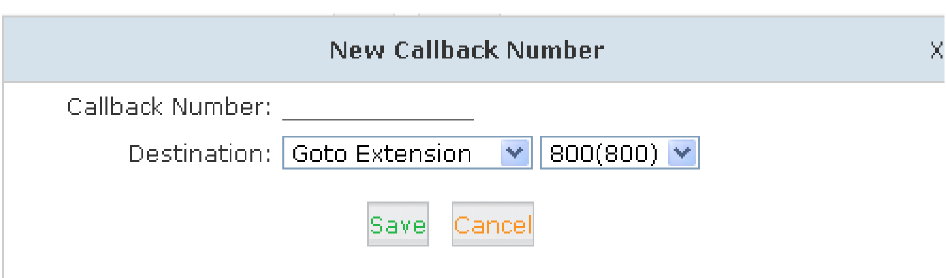
Callback

When user makes calls by the callback number to M-200 IPPBX, the call will be hung up automatically. Then the PBX will call back this number and forwarded to define destination after the call is connected. Please configure it as shown below

Click **Advanced** -> **Callback**



At first, enable this function. Select Dial Plan, and define the callback rule (strip digits or prepend prefix). Click New Callback Number to add callback number

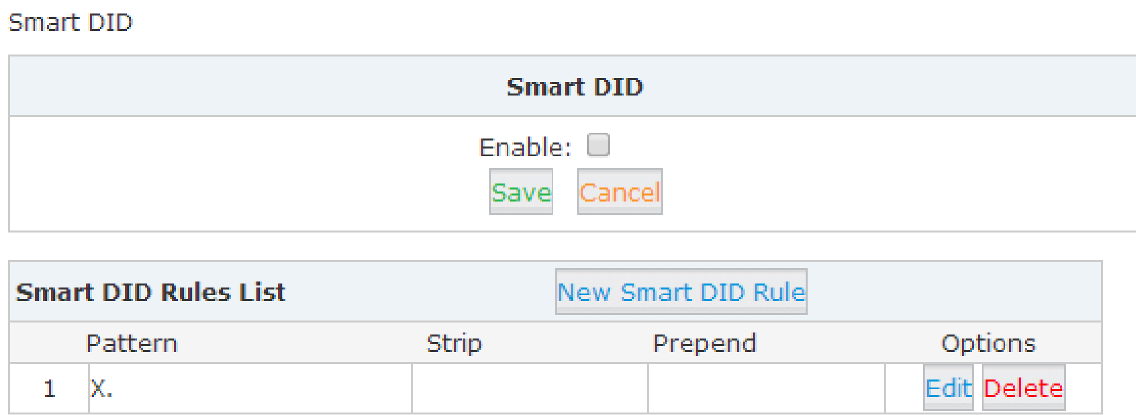


Input callback number and define the destination

Smart DID

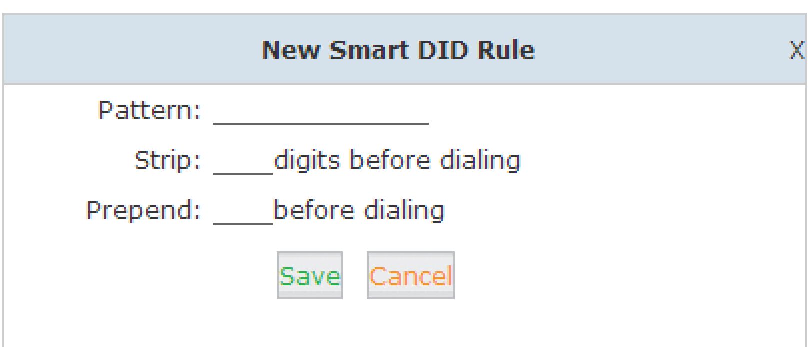
Smart DID: After extension user makes an outbound call, the call is ringing back to IP PBX, and directed to the one who made the last call. Please configure it as shown below

Click **Advanced** -> **Smart DID**



Check “Enable” and “Save” to make this function activates.

Click **New Smart DID Rule** to display the following diagram:

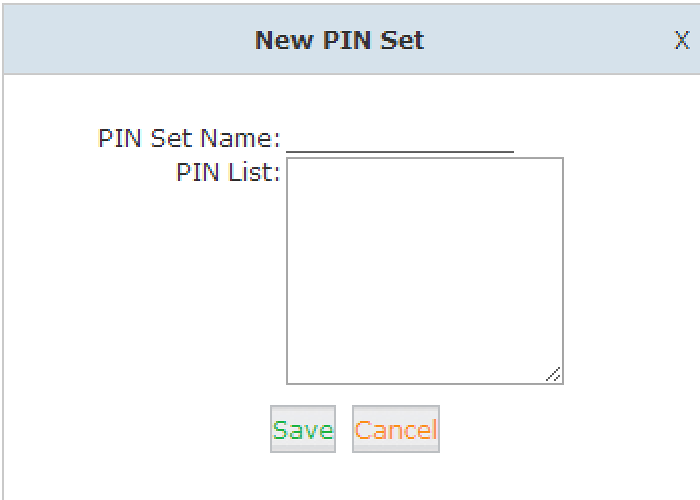


Input the pattern and define how many digits need to be striped or prepend, and then click “Save”--“Activate”.

Pin Sets

That Pin can used for DialRules

Click **PIN Sets** -> **New PIN Set** to display the dialog below:



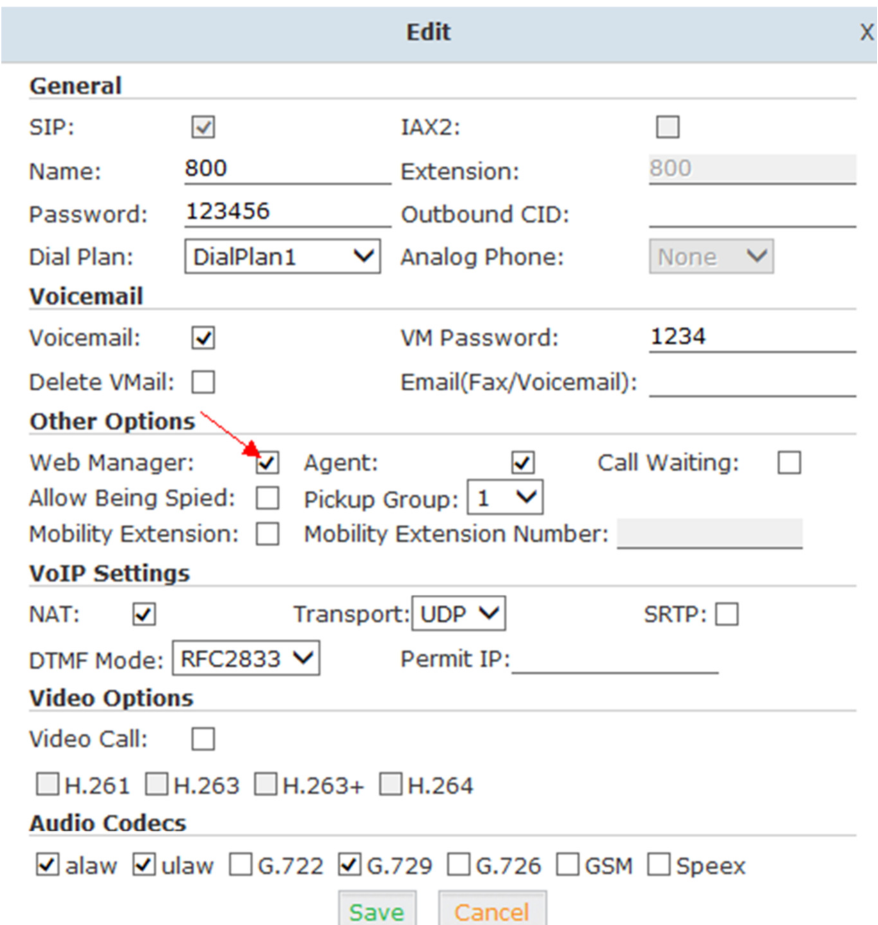
Item	Explanation
PIN Set Name	Define the name for this PIN Set.
PIN List	Define PIN codes is this list.

Chapter 6: Extension User Web Portal

Enable Extension User Web Portal

Allowed access to the User Web Portal.

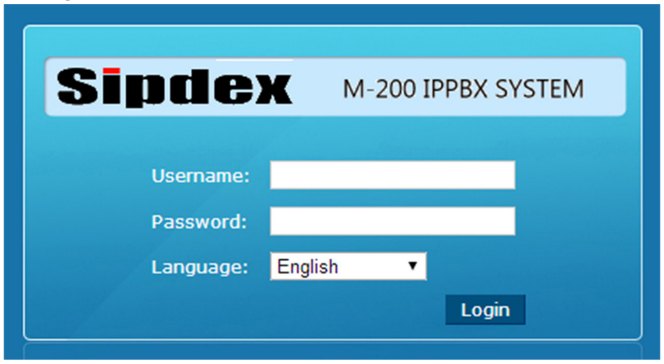
Basic-> **Extension** -> **Edit** the extension you want to configure



Select the "Web Manager" box under "Other Options", then check **Web Manager** and click **Save**.

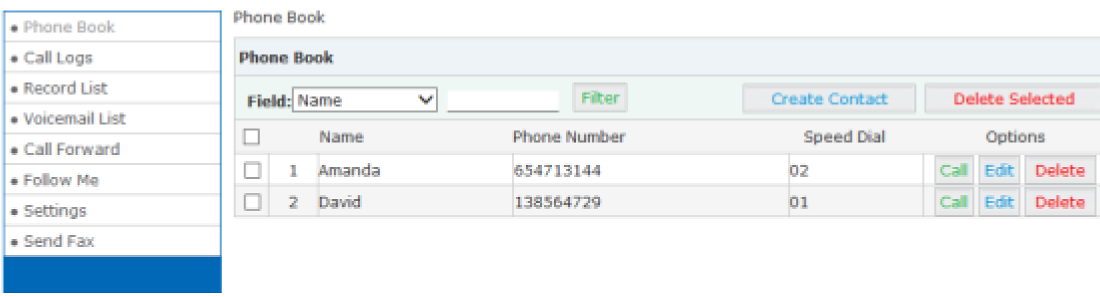
Login to the Extension Web Panel

Using *Extension Number for the Username & Voicemail Password* to login Web GUI.



Phone Book

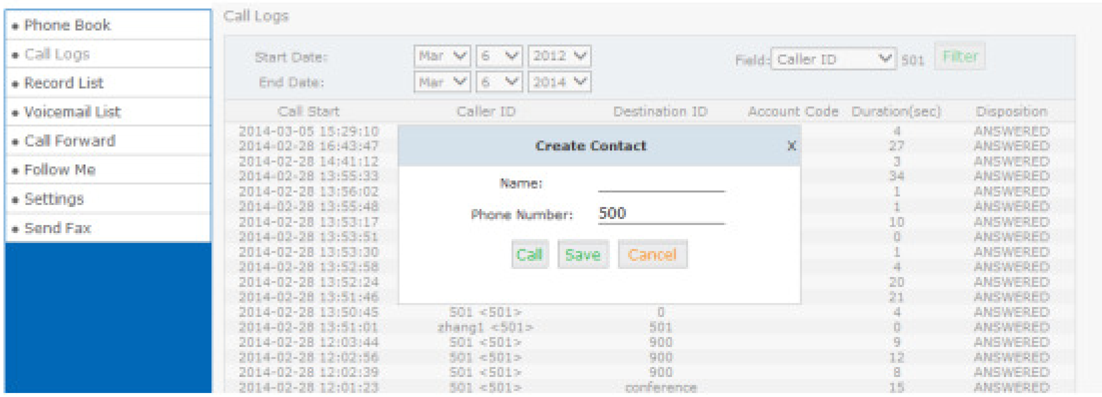
Mange the Phone Book for Speed Dial



Item	Explanation
Filter	Search contacts, by name, phone number or speed dial
Create Contact	Create a contact
Delete Contact	Delete a selected contact
Call	Click to call the number directly

Call Logs

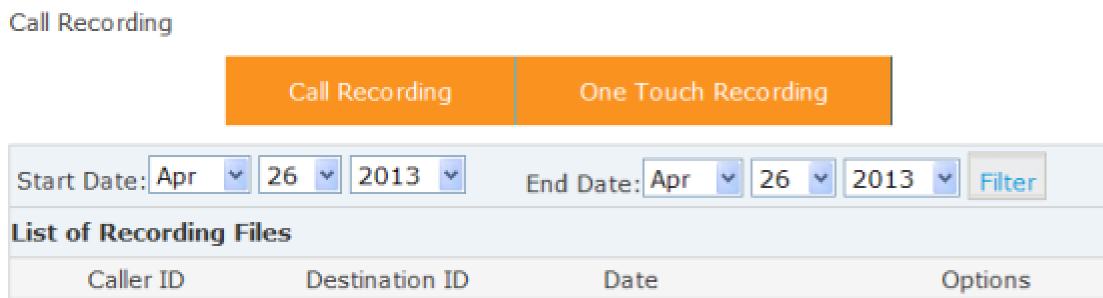
Report your Call-in / Call-out record.



Select the Start Date & End Date to filter your call log.

Recording List

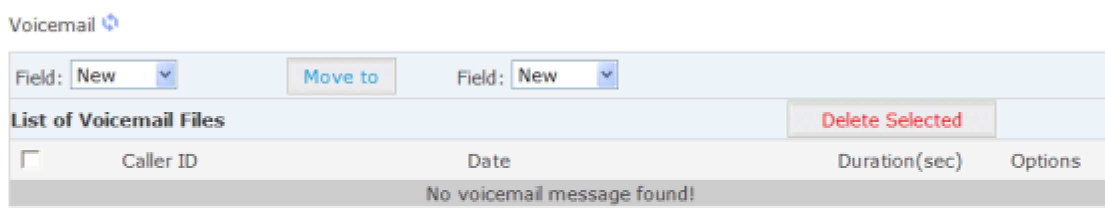
To download and manage your Voice log.



Select the Start Date & End Date to filter your voice log.

Voicemail List

To manage your voicemail.

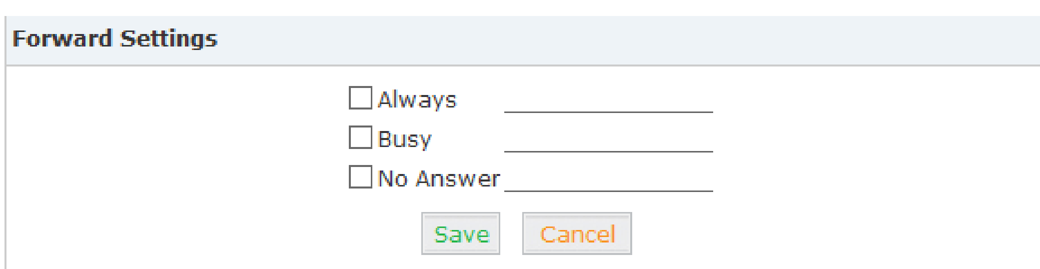


Click **Move to** to move the voicemail to another field.

Check one voicemail file, click **Delete Selected** to delete the selected voicemail file; or click **Delete** after the voicemail file to delete the voicemail.

Call Forward

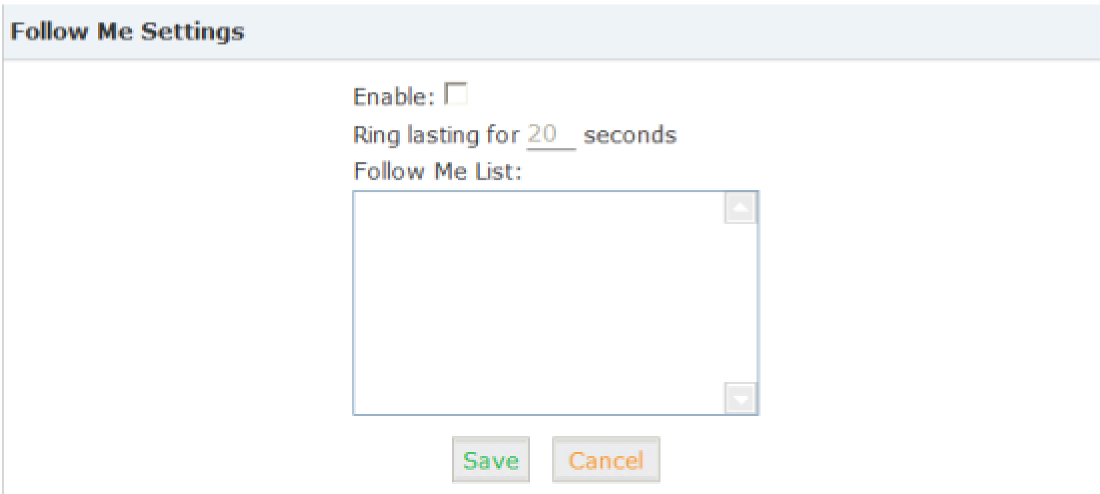
Redirects a telephone call to another destination status in Always / Busy / No Answer.



Item	Explanation
Always	All incoming calls will be forwarded.
Busy	Forward when extension is busy.
No Answer	Forward when no answer from extension.

Follow Me

If no answer from extension, when the ring times out, the calls will be forwarded one by one to the number listed in <Follow Me List>.

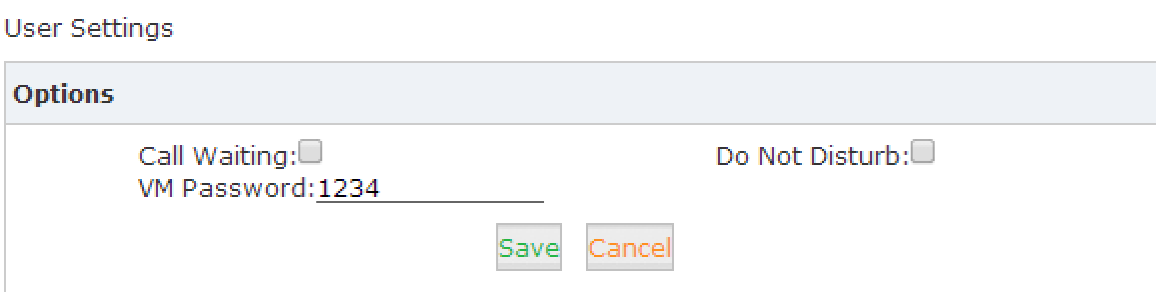


Item	Explanation
Enable	Check the Enable box to enable this function.
Ring lasting for _ seconds	The seconds ringing, the call will be forwarded from your extension
Follow Me List	The destination will be forwarded

Settings

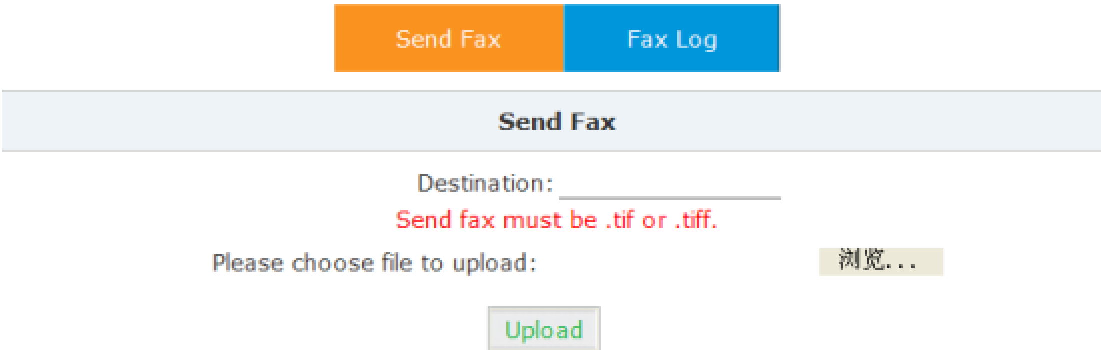
User settings for Enable/Disable Call Waiting, Do Not Disturb.

And Change VM Password



Send Fax

The fax can be sent by WEB and Email. Fax format must be .tif or .tiff.



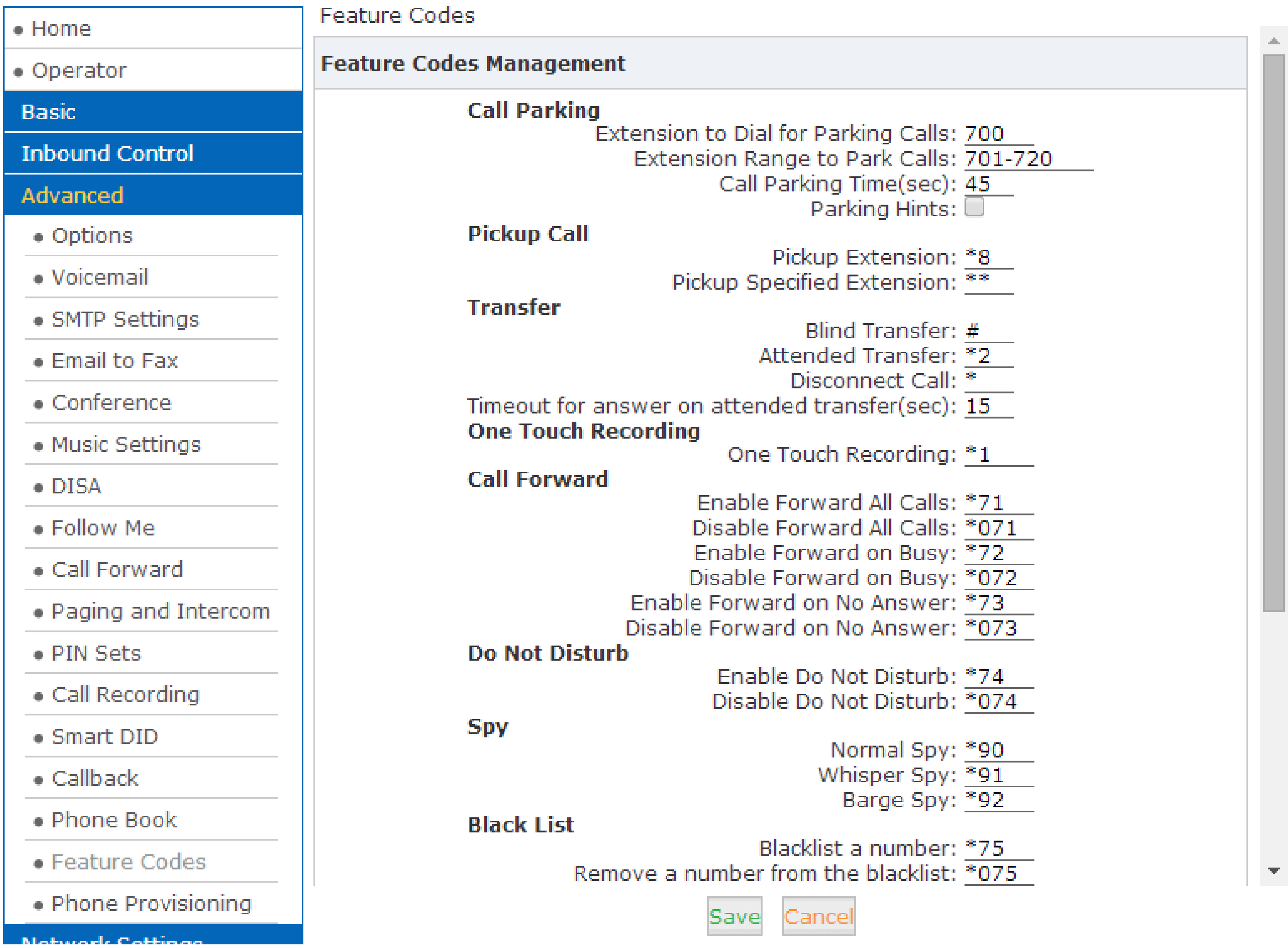
Enter the receiver’ s fax number in **Destination**, click **browse** to select the fax file, then **Upload**.

Chapter 7: Feature Code

To change the default Feature Code

Go **Advanced** -> **Feature Codes**

Change the codes and then press **save.**

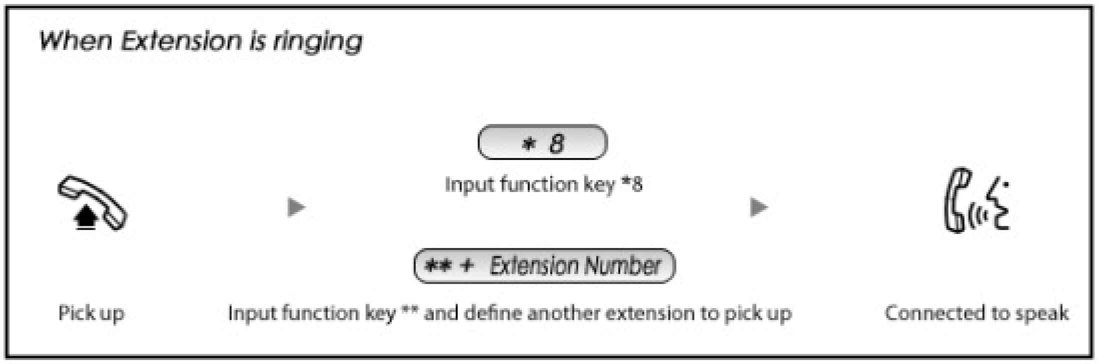


Do Not Disturb

Item	Explanation
*74	Enable Do Not Disturb.
*074	Disable Do Not Disturb.

Call Pickup

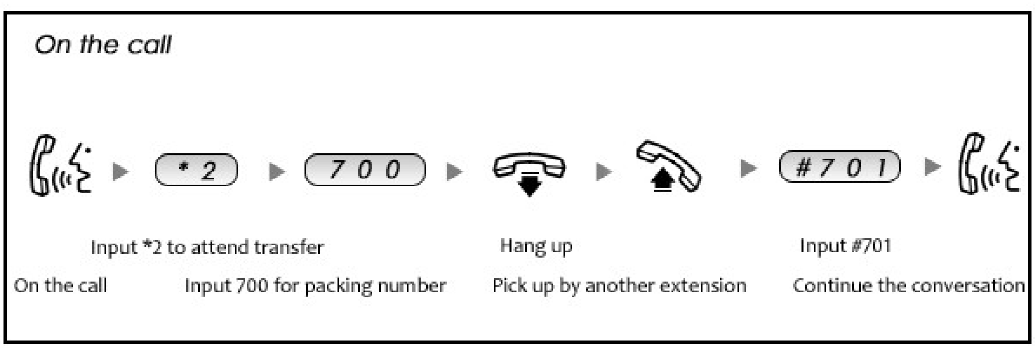
This feature allows users to answer a call that is ringing on another user's extension by pressing the selected feature code on their own phone as shown in the diagram below.



Item	Explanation
*8	Input function key *8 to pick up the registered extension which is in the same pickup group
** + Extension Number	Input function key ** and define another extension to pick up. This can be defined in **Feature Codes**

Call Parking

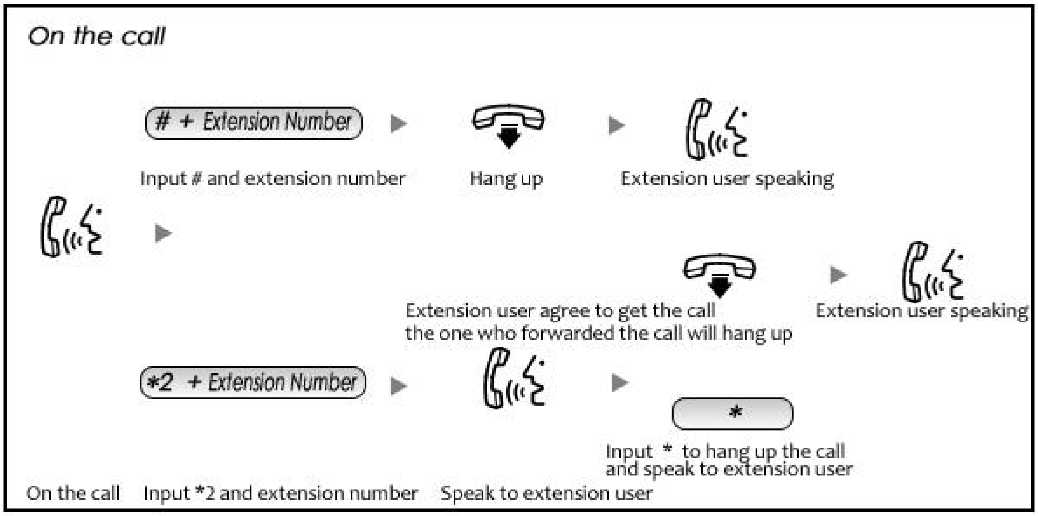
If you pick up a call at your seat, but it's not convenient to talk in public, you need go to the conference room to talk secretly. At this time, you can input 700 to park this call. The system will tell you a parking number 701 which you can input for continuing conversation when you go to the conference room. Please check the following diagram to learn more:



Item	Explanation
Extension to Dial for Parking Calls	Default Number: 700, Define in **Feature Codes**
What Extension to park calls on	Default Number: 701 - 720. Define in **Feature Codes**
How many seconds a call can be parked for	Default is 45 seconds. Define in **Feature Codes** .

Call Transfer

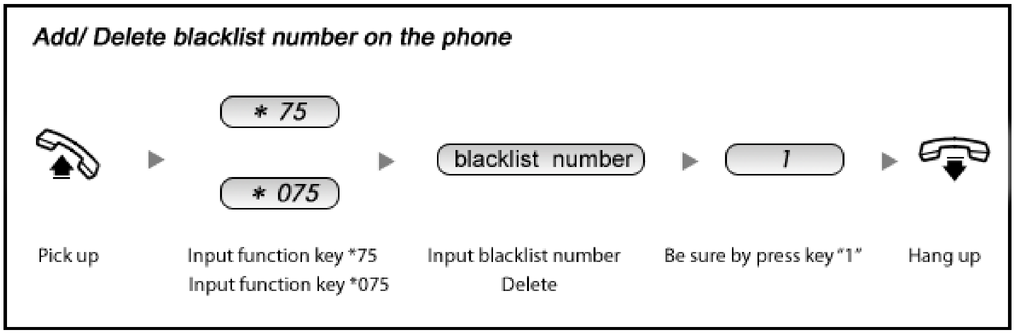
This feature allows an incoming call that is answered on one extension to be sent to another user's extension. Refer to the diagram as below:



Item	Explanation
Blind Transfer	Default is #t. Define in **Feature Codes**
Attended Transfer	Default is *2. Define in **Feature Codes**
Disconnect Call	Default is *, it can be used when you use *2. Define in **Feature Code**
Timeout for answer on attended transfer	Default is 15 seconds. Define in **Feature Codes**

Blacklist

To maintain this list of blocked numbers, see the instructions in the following diagram:



Reference Parameters and Explanation of the Blacklist:

Item	Explanation
*75	When the registered extension user inputs *75 + blacklisted number, this number will be added in the list of Blacklist Number.
*075	When the registered extension user inputs *075 + blacklist number, this number will be deleted in the list of Blacklisted Number.

Chapter 8: Fax

Fax to Email

The capabilities receive faxes as TIF attachments into your email inbox.

Step for Create the Extension for receipt the Fax:

1. Go **Basic** -> **Extensions** -> **New User**
2. Check **SIP ,** input the Extension number in the **Extension**
3. Fill in the email address in **Email(Fax/Voicemail)**
4. Click **Save**. (See Figure 8.1)
5. Go **Inbound Control** -> **Inbound Routes**
6. Al the Destination fill can be select Goto Fax, and then select Fax2Email – [your extension number] (See Figure 8.2)

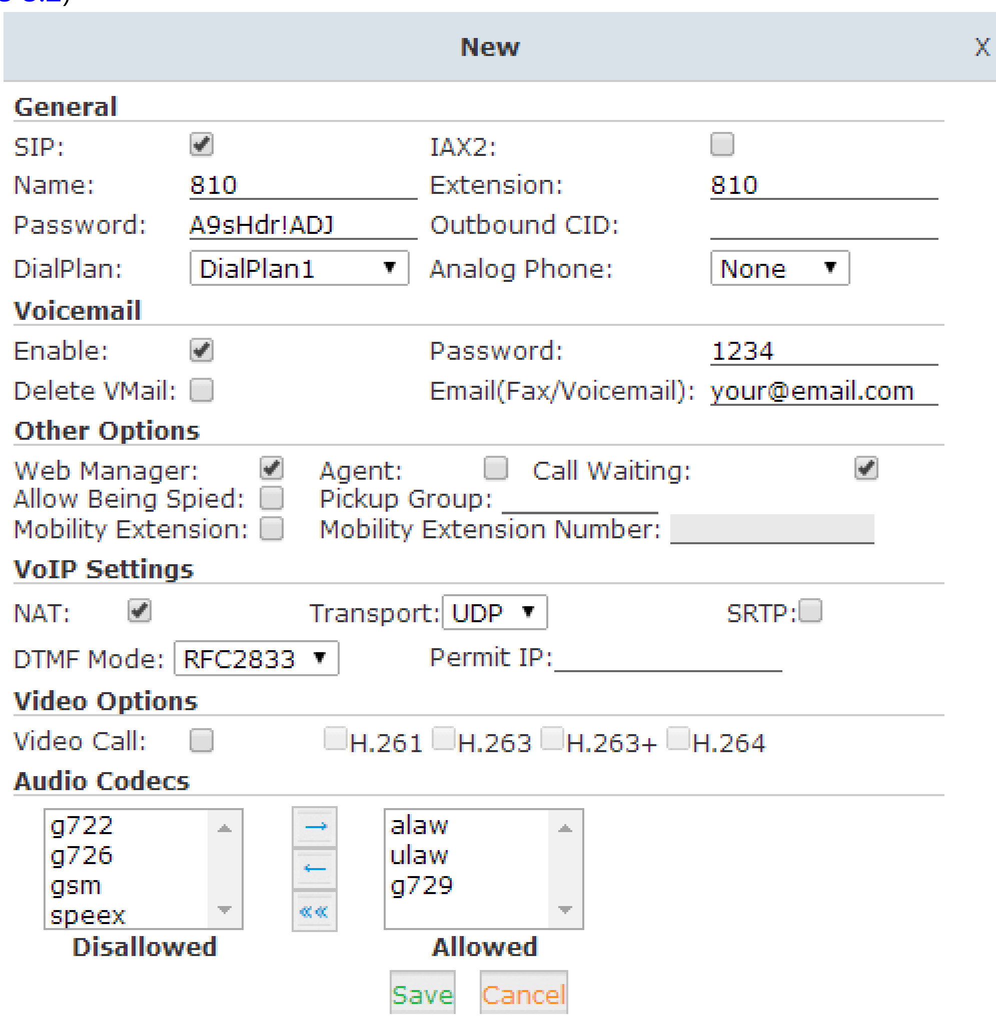


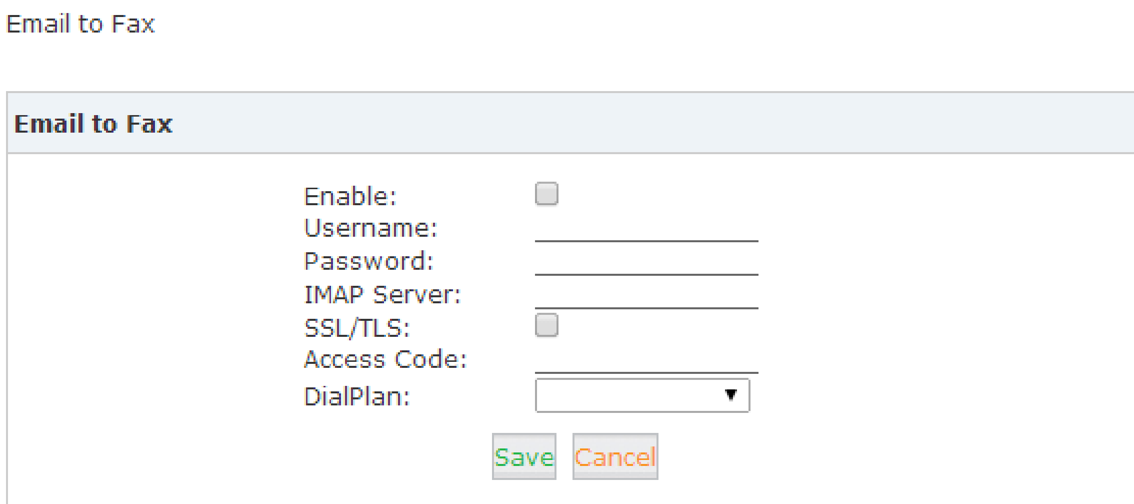
Figure 8.1

Figure 8.2

Email to Fax

Users can send fax by Email. Please configure as shown below.

Click **Advanced** -> **Email to Fax**



Check **Enable**, input user name, password and IMAP Server, select the DialPlan and then “Save” and “Activate”.

Practical Case:

Send a fax to telephone number 12345678: In Dial Plan 1, there is prefix “9” before the telephone number; you need to input the Access Code : 912345678 and take it as the subject when sending Email. Then the fax will be sent by Email as attachment.

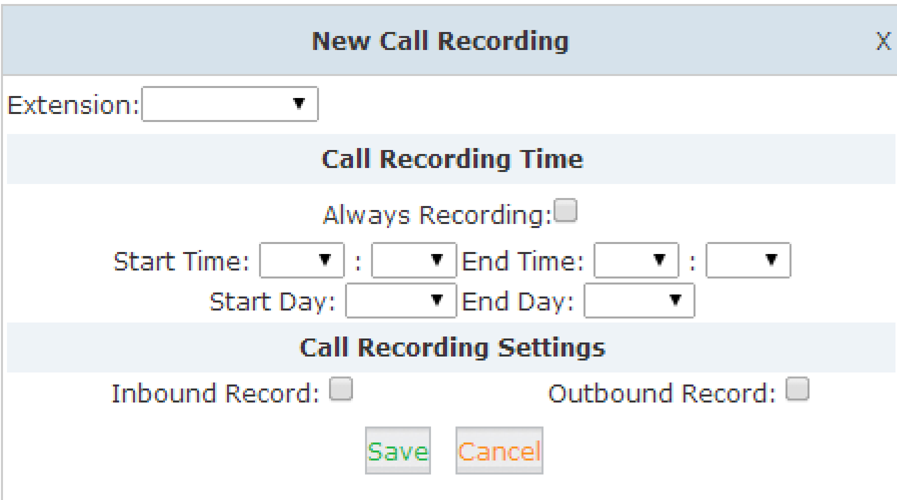
If you need to dial the extension when sending fax, e.g. fax number: 12345678 ext.888, you need to use the Access Code : 912345678-888 as subject.

Chapter 9: Call Recording

Recording on Specified Extension

Call Recording is used for recording extension. Please configure it as shown below:

Click **Advanced** -> **Call Recording** -> **New Call Recording**



Reference:

Item	Explanation
Extension	Define an extension for recording.
Call Recording Time	Set the time to record.
Inbound Record	Check to record inbound calls.
Outbound Record	Check to record outbound calls.

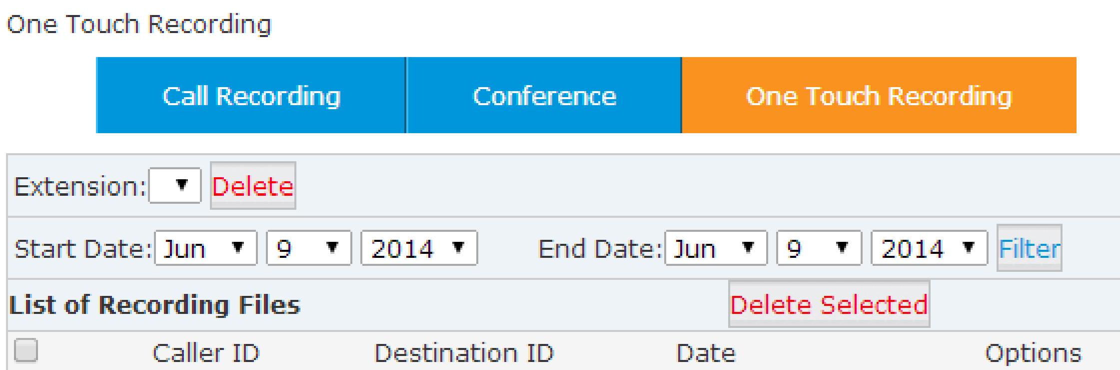
One Touch Recording

Easily record conversations on demand.

NOTE:

The Extension is on the line, press *1 to start the recording.

Click **One Touch Recording**, select the Start Date & End Date for filter your voice log.

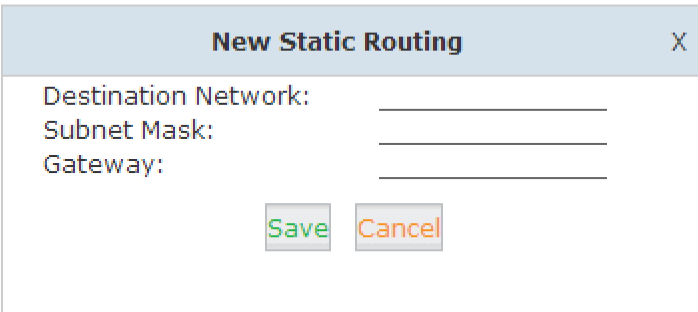


Chapter 10: Networking

Static Route

Provide additional routing information to your IPPBX. Under normal circumstances, the router has adequate routing information after it has been configured for Internet access, and you do not need to configure additional static routes. You must configure static routes only for unusual cases such as multiple routers or multiple IP subnets located on your network.

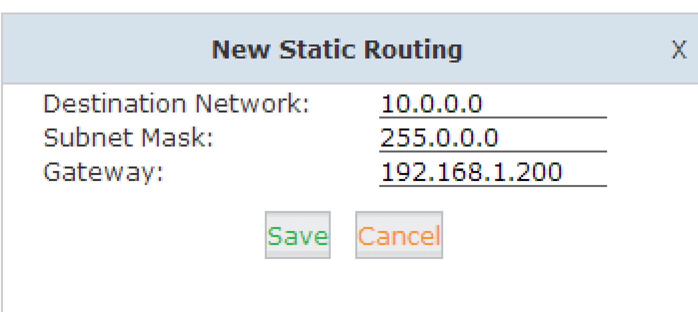
Click **Static Routing** -> **New Static Routing**



Item	Explanation
Destination Network	Input destination network
Subnet Mask	Input subnet mask
Gateway	Input gateway

Example:

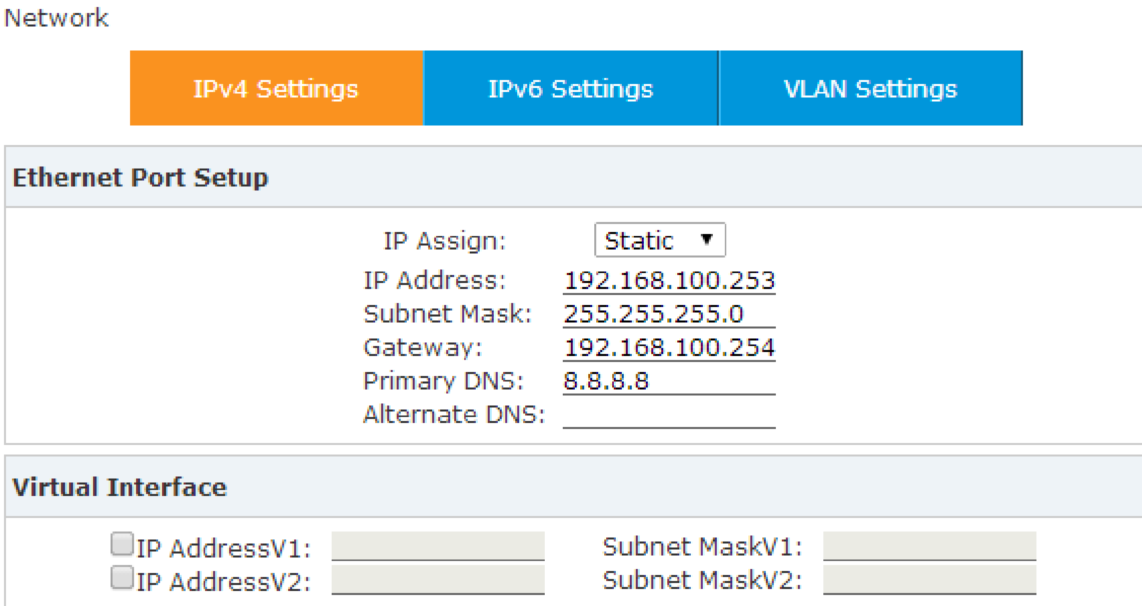
M-200 need connect SIP Trunk provider device 10.0.0.1/8 through SIP Trunk gateway 192.168.1.200, the setting will be such as following figure:



Virtual Interface

You can configure the Virtual interface

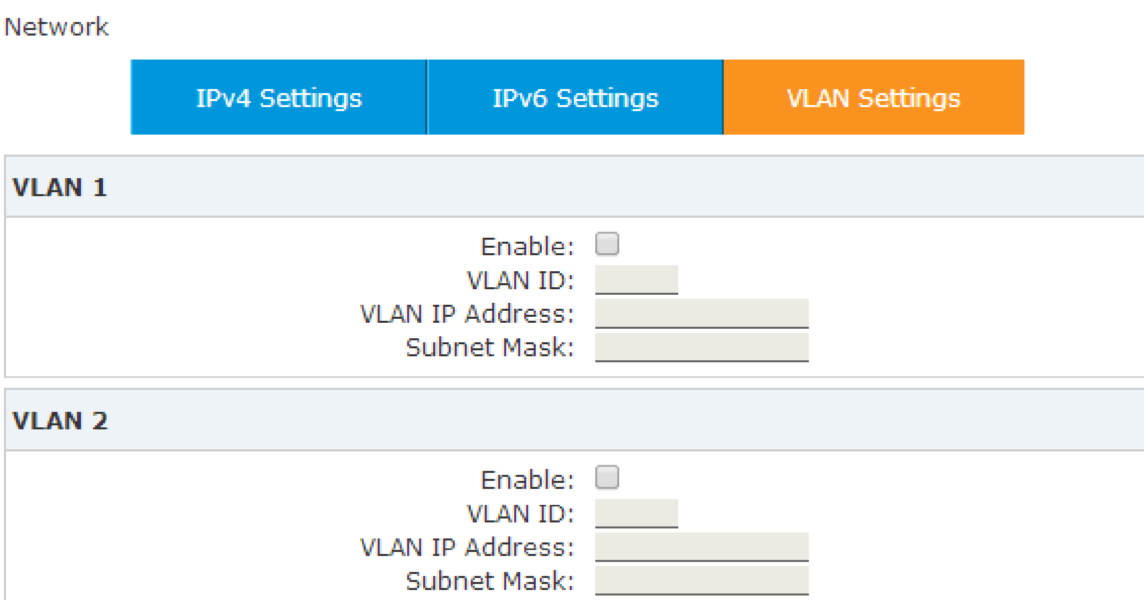
Click **Network Settings** -> **Network IPv4 Setting**



VLAN Settings

Defend VLAN on network environment can be increase security or management from other VLAN.

Click **Network Settings** -> **Network** -> **VLAN Settings**

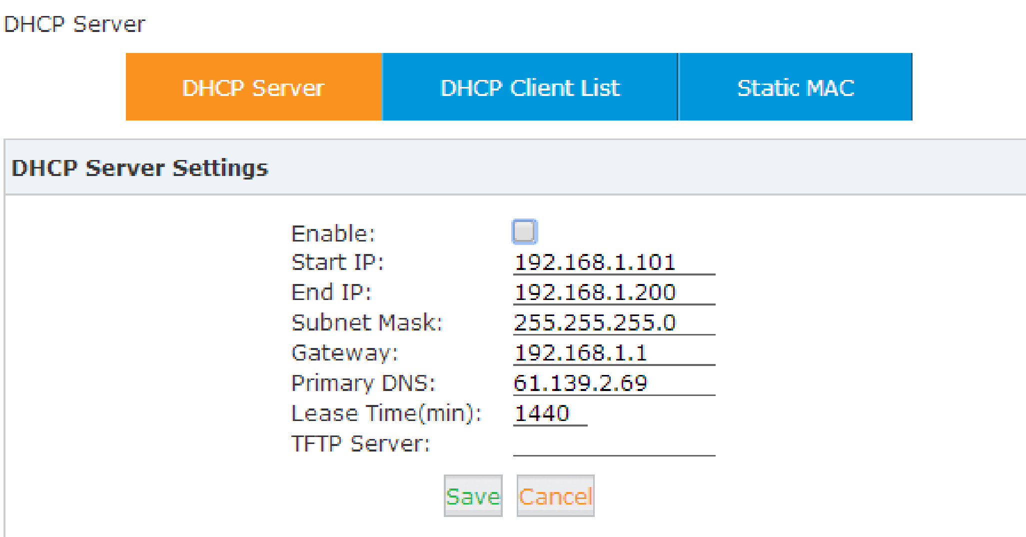


Item	Explanation
Enable	Enable / Disable VLAN
VLAN ID	Input VLAN ID, range between 1- 4094
VLAN IP Address	Input VLAN IP for specific network
Subnet Mask	Input VLAN subnet

DHCP Server

DHCP is a network protocol that enables a server to automatically assign an IP address to a computer from a defined range of numbers.

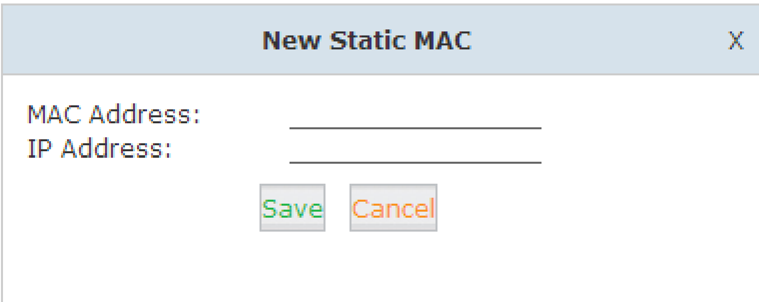
Click **Network Settings** -> **DHCP Server**



Item	Explanation
Enable	Enable / Disable DHCP server
Start IP	Define the first IP for DHCP
End IP	Define the last IP for DHCP pool
Subnet Mask	Define subnet mask
Gateway	Define Gateway IP
Primary DNS	Define Primary DNS
Lease Time(min)	Define lease time to release IP
TFTP Server	Define TFTP server for using Phone provisioning

When DHCP Server distributes address, the Client's MAC address is associated with the IP address, and then the device will get the same IP address every time.

Click **Network Settings** -> **DHCP Server** -> **Static MAC** -> **New Static MAC**

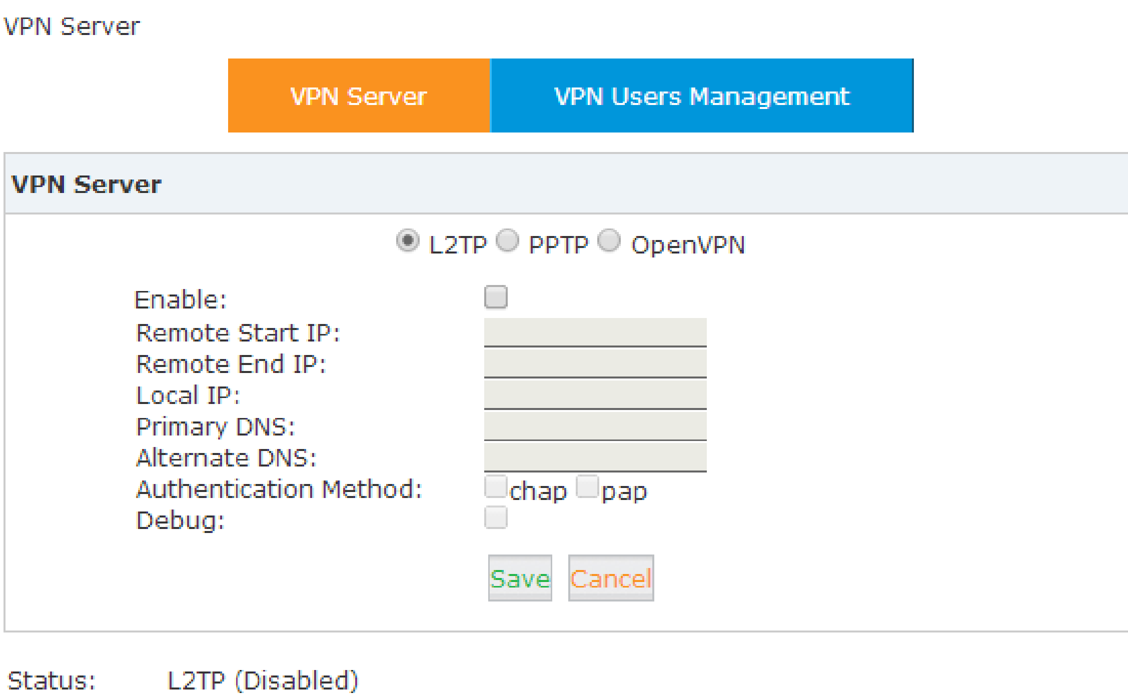


Item	Explanation
MAC Address	Input MAC Address that device need set static IP by M200
IP Address	Input IP address for specific device

VPN Server

SIPDEX IP PBX supports three kinds of VPN servers: L2TP, PPTP and OpenVPN.

Click **Network Settings** -> **VPN Server**



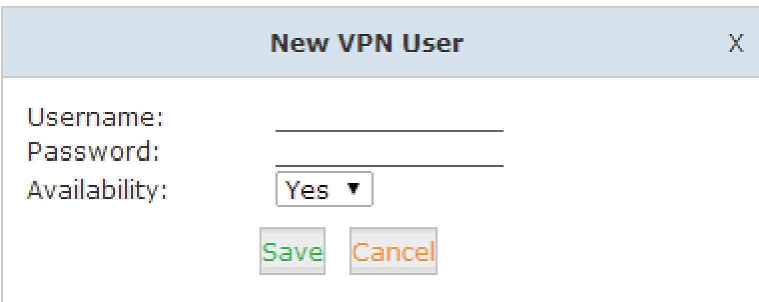
Reference:

Item	Explanation
VPN Server Mode	L2TP, PPTP and OpenVPN (Only one mode can be enabled as the same time)
Enable	Enable / Disable VPN Server

Create PPTP and L2TP Account

If you choose PPTP or L2TP VPN method, you need create VPN user for PPTP.

Click **Network Settings** -> **VPN Server** -> **VPN Users**

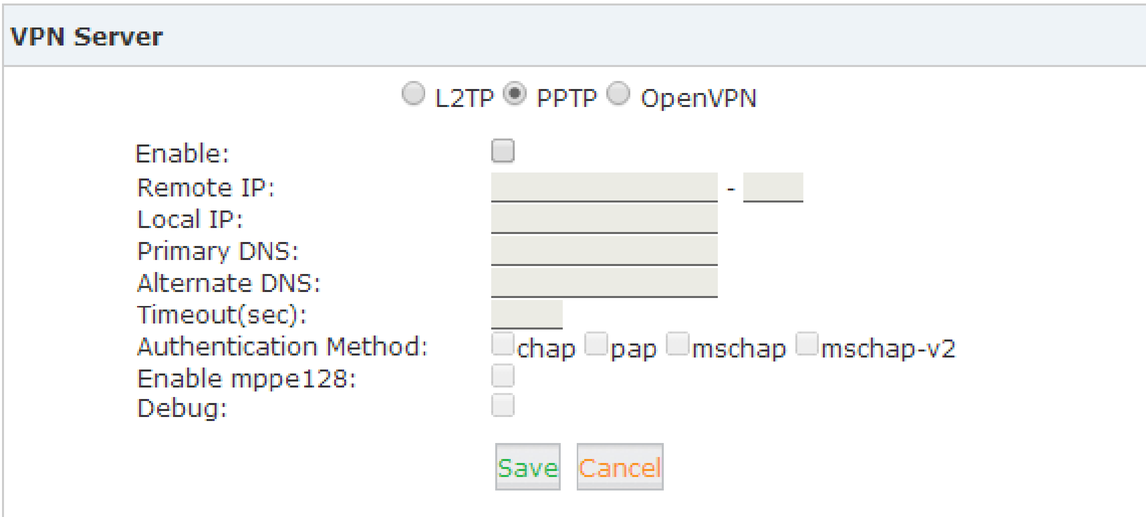


Reference:

Item	Explanation
Username	VPN client login name
Password	VPN client login password
Availability	Yes = Enable this account, No = Disable this account

PPTP

Remember, after enable PPTP VPN server, you need open TCP port 1723 on your Router.

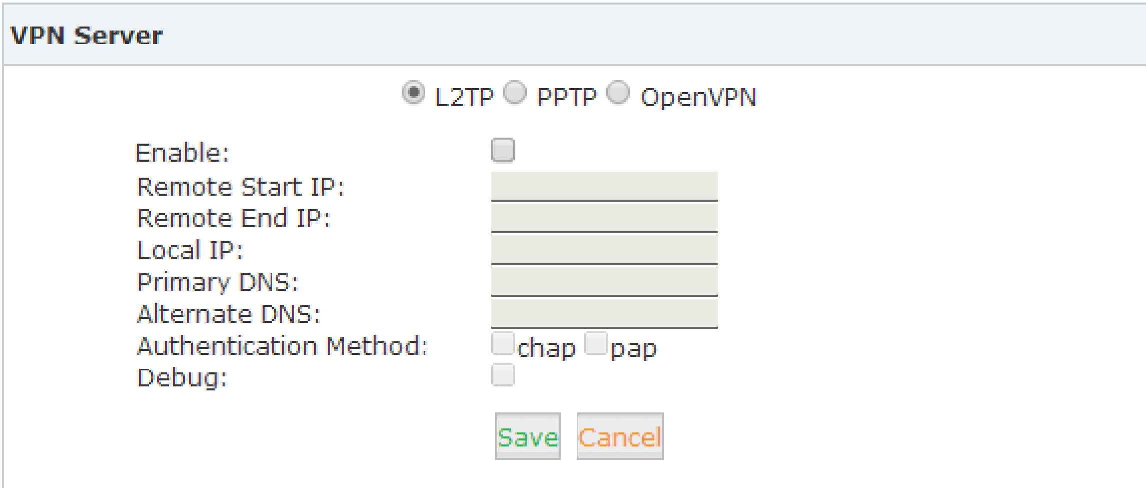


Reference:

Item	Explanation
Enable	Enable / Disable PPTP VPN server
Remote IP	Define IP for VPN client
Local IP	Define Gateway IP (normally use router IP)
Primary DNS	Define Primary DNS
Alternate DNS	Define Second DNS
Timeout(sec)	Define VPN connection time out value
Authentication Method	Define authentication method
Enable mppe128	Enable / Disable mppe128
Debug	Enable / Disable debug report. When Enable, report will store in **Report** -> **System Log**

L2TP

Remember, after enable L2TP VPN server, you need open UDP port 1701 on your Router.



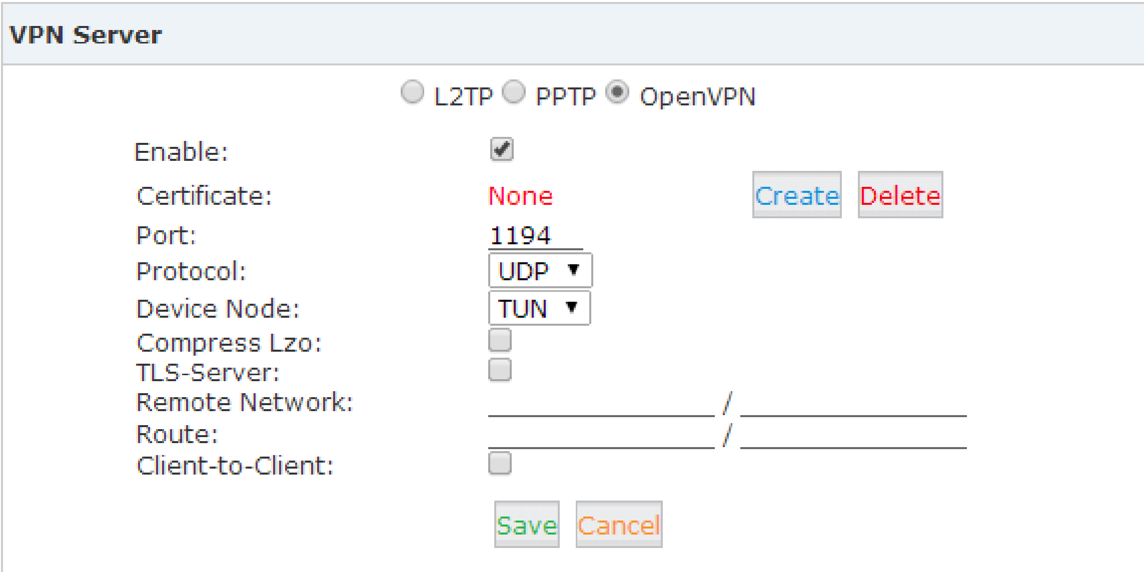
Reference:

Item	Explanation
Enable	Enable / Disable L2TP VPN server
Remote Start IP	Define first IP for L2TP IP pool
Remote End IP	Define last IP for L2TP IP pool
Local IP	Define Gateway IP (normally use router IP)
Primary DNS	Define Primary DNS
Alternate DNS	Define Second DNS
Authentication Method	Define authentication method
Debug	Enable / Disable debug report. When Enable, report will store in **Report** -> **System Log**

OpenVPN

Remember, after enable OpenVPN server, you need open UDP port 1194 on your Router.

When the mode is OpenVPN server, click **Network Settings** -> **VPN Server**

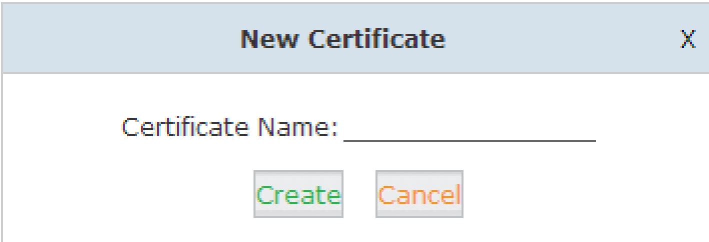


Reference:

Item	Explanation
Enable	Enable / Disable OpenVPN server
Certificate	Press Create to build certificate, Press Delete to remove certificate which store in M200
Port	Default OpenVPN port is 1194
Protocol	OpenVPN packet transfer method, default is UDP
Device Node	Connect method TUN (Layer 2), TAP (Layer 3)
Compress Lzo	Compress packet
TLS-Server	Enable / Disable OpenVPN server TLS
Remote Network	Define Remote network
Route	Define Route
Client-to-Client	Enable clients can access each other

After setup OpenVPN server, you need download OpenVPN client certificate and install on you OpenVPN client.

Click **VPN Server** -> **OpenVPN Certificate Download -> New Certificate**



Reference:

Item	Explanation
Certificate Name	Define certificate name

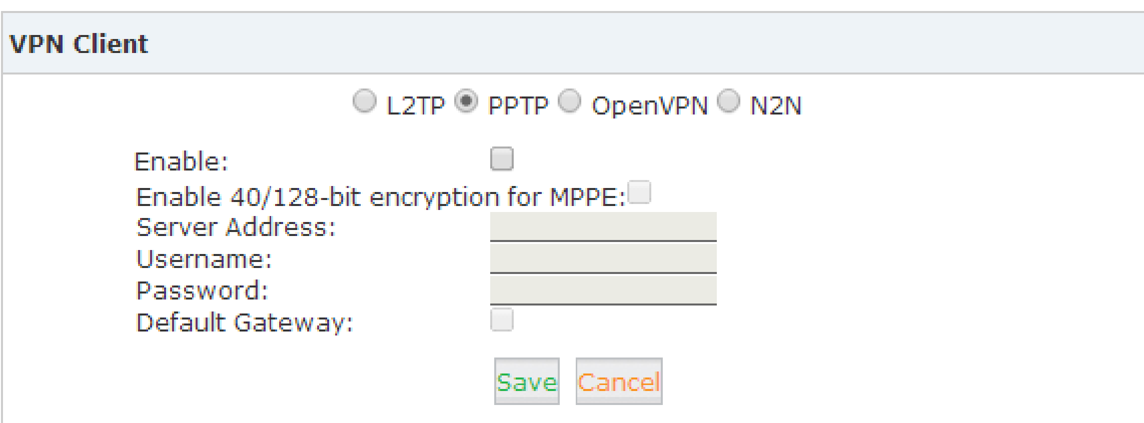
VPN Client

SPIDEX IP PBX supports four kinds of VPN Clients: L2TP, PPTP, OpenVPN and N2N.

Click **Network Settings** -> **VPN Client**

PPTP

Setup PPTP client.



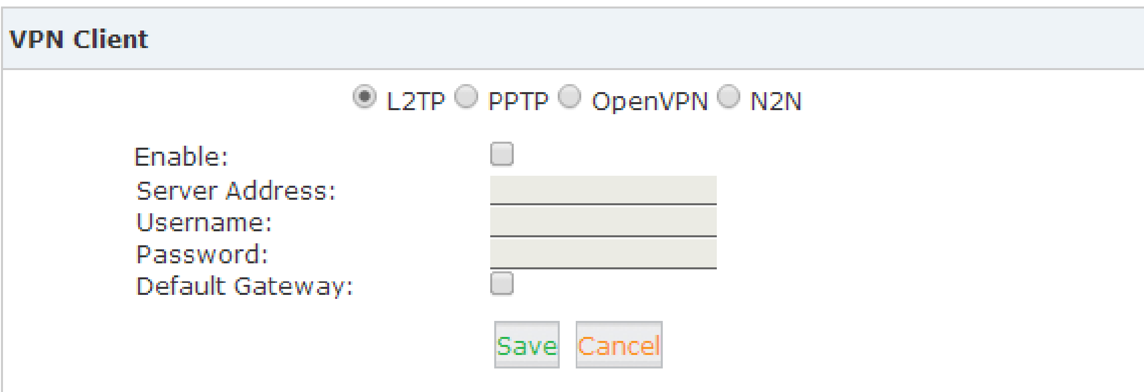
Reference:

Item	Explanation
Enable	Enable / Disable PPTPVPN server

Enable 40/128-bit encryption for MPPE	Enable / Disable 40/128-bit encryption for MMPE
Server Address	Input PPTP IP address
Username	Input PPTP username (Login name)
Password	Input PPTP login password
Default gateway	Enable / Disable default gateway

L2TP

Setup L2TP client

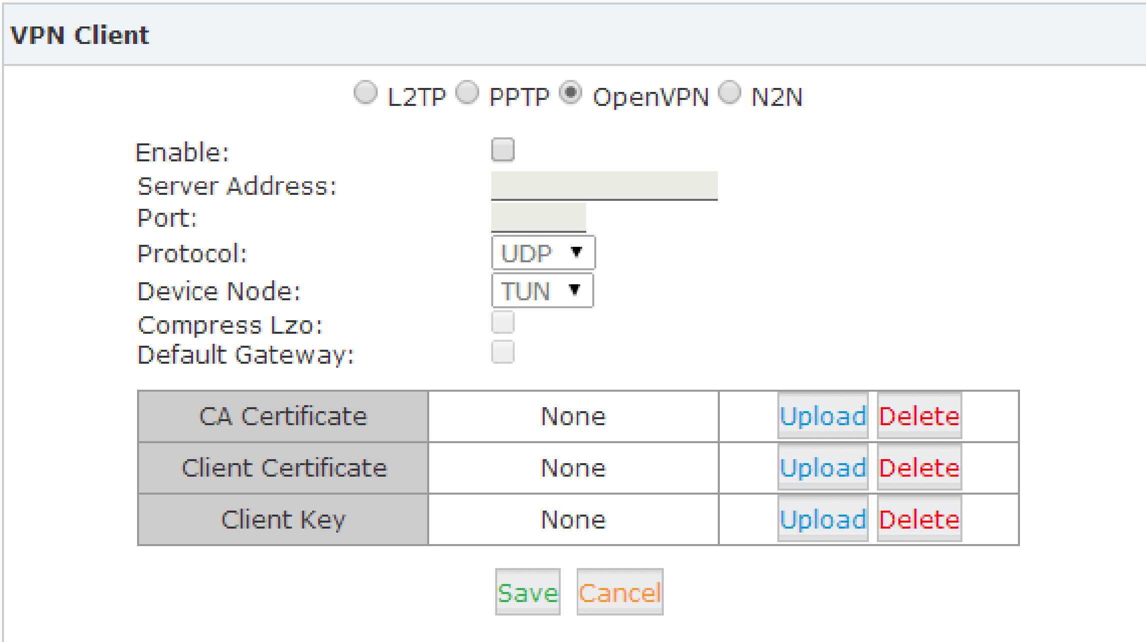


Reference:

Item	Explanation
Enable	Enable / Disable L2TPVPN server
Server Address	Input L2TP IP address
Username	Input L2TP username (Login name)
Password	Input L2TP login password
Default gateway	Enable / Disable default gateway

OpenVPN

Setup OpenVPN client



Reference:

Item	Explanation
Enable	Enable / Disable OpenVPN server
Server Address	Input OpenVPN server IP address
Port	This value need same as OpenVPN server port, default is 1194
Protocol	This value need same as OpenVPN server, default is UDP
Device Node	This value need same as OpenVPN server, default is TUN
Compress Lzo	This value need same as OpenVPN server, default is uncheck
Default Gateway	Enable / Disable default gateway
CA Certificate	Press Upload to add CA certificate which created by OpenVPN server
Client Certificate	Press Upload to add client certificate which created by OpenVPN server
Client Key	Press Upload to add client key which created by OpenVPN server

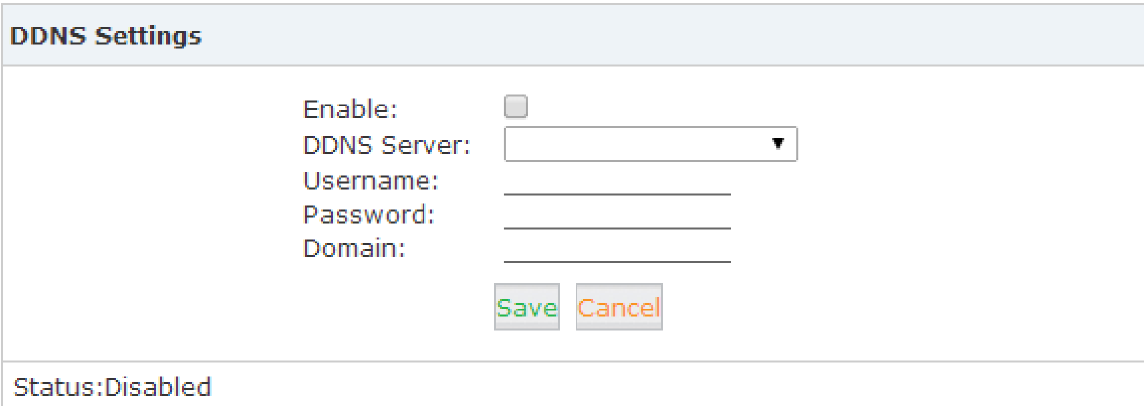
N2N

<TODO>: Insert description text here... And don't forget to add keyword for this topic

DDNS Settings

After setting DDNS (Dynamic Domain Network Server), SIPDEX IP PBX settings will be visited remotely.

Click **Network Settings** -> **DDNS Settings**



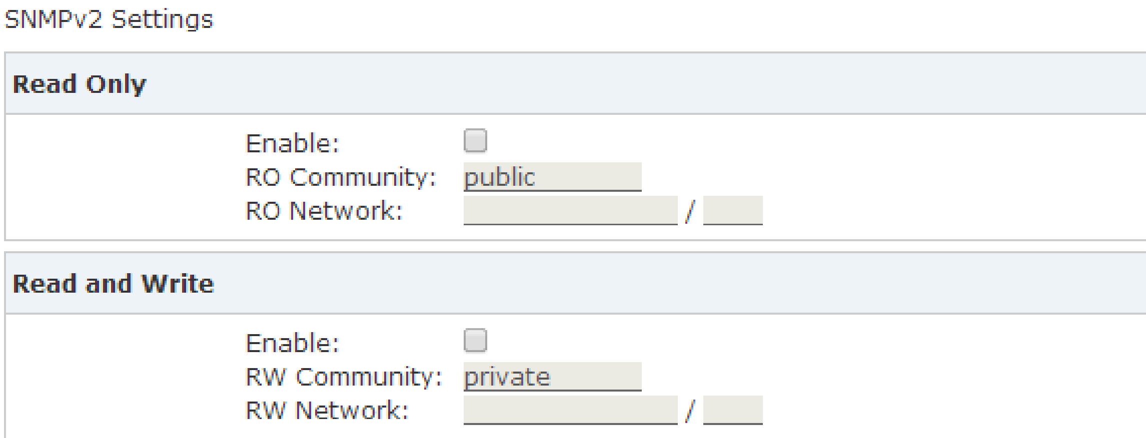
Reference:

Item	Explanation
Enable	Enable / Disable DDNS
DDNS Server	Choose DDNS Server
Username	Input DDNS username
Password	Input DDNS password
Domain	Input DDNS domain

SNMPV2 Settings

SNMP (Simple Network Management Protocol) is used for remote management.

Click **Network Settings** -> **SNMPv2 Settings**

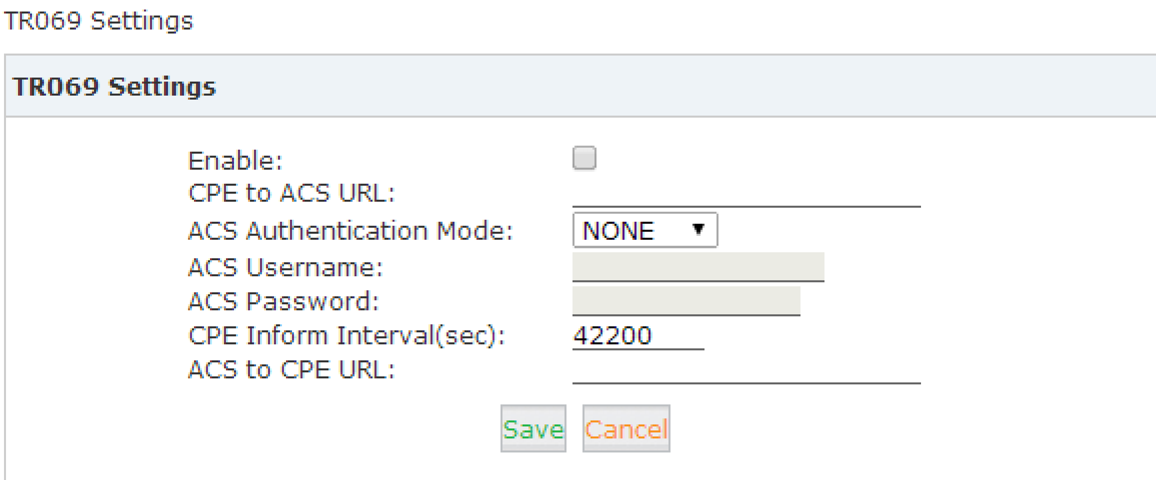


Reference:

Item	Explanation
Enable	Enable / Disable Read Only of SNMP or Read and Write SNMP
RO Community	Define the name of RO Community of SNMP
RO Network	Define network of RO
RW Community	Define the name of RW community of SNMP
RW Network	Define network of RW

TR069

TR069 is used for remote management

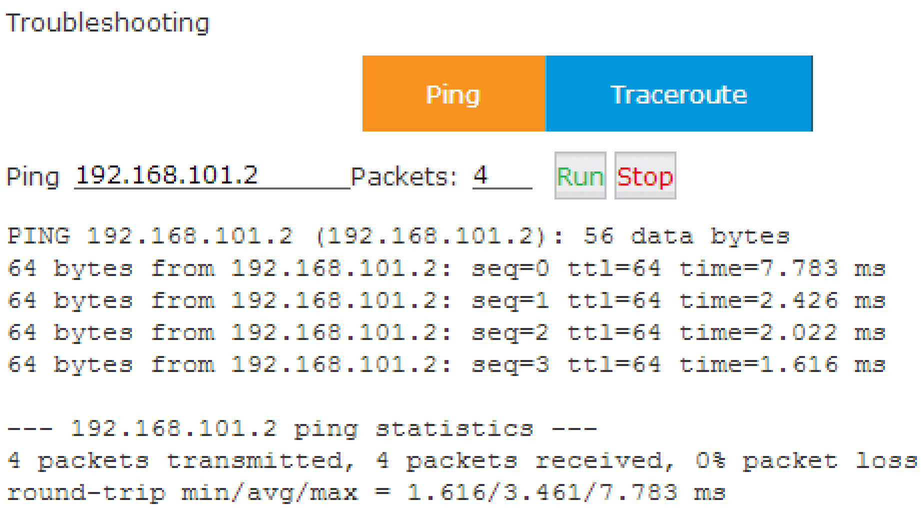


Item	Explanation
Enable	Enable/Disabled TR069 functionEnable/Disabled TR069 function
CPE to ACS URL	Set the CPE to ACS URL. This parameter MUST be in the form of a valid HTTP URL
ACS Authentication Mode	Select the ACS Authentication Mode for CPE.
ACS Username	Set the ACS Username. The username used to authenticate the CPE when making a connection to the ACS.
ACS Password:	Set the ACS Password. The password used to authenticate the CPE when making a connecting to the ACS.
CPE Inform Interval (sec)	The duration in seconds of the interval for which the CPE MUST attempt to connect with the ACS and send the inform.
ACS to CPE URL	Set the ACS to CPE URL. This parameter MUST be in the form of a valid HTTP

Trouble Shootings

You can ping or traceroute other network devices through SIPDEX IP PBX and track network routing by command "Traceroute" .

Click **Network Settings** -> **Troubleshooting**

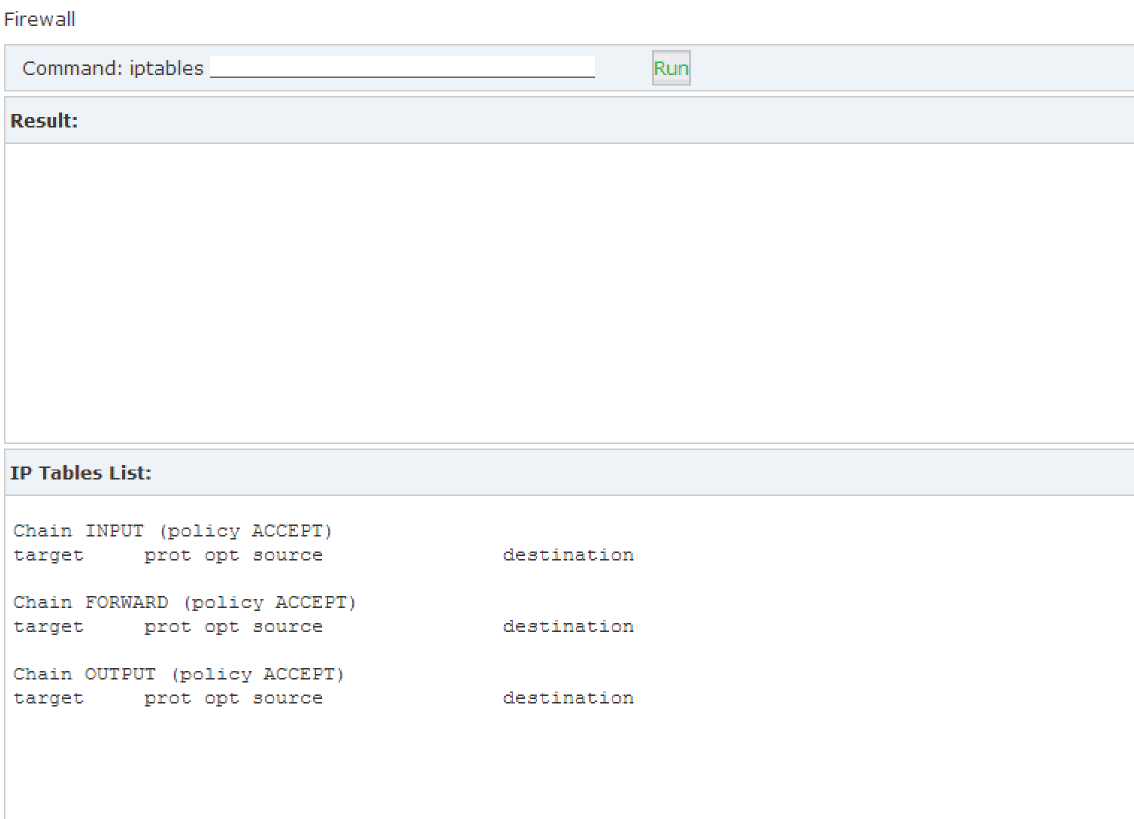


Chapter 11: Security

Iptables Firewall

Iptables is tool used to configure firewall rules. Which sit on top of and use Iptables commands.

Click **Security** -> **Firewall**



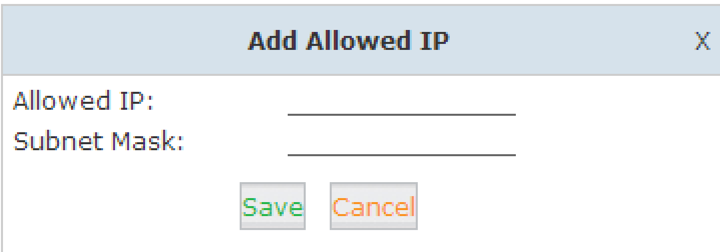
Reference:

Iptables Command	Explanation
Check iptables list	iptables -L -n
Clear iptables list	iptables -F
Deny an IP 192.168.0.3	iptables -A INPUT -s 192.168.0.3 -j DROP
Deny every IP to access 80 port	iptables -A INPUT -p tcp --dport 80 -j DROP
Deny IP 192.168.0.3 to access port 80	iptables -A INPUT -s 192.168.0.3 -p tcp --dport 80-j DROP

SIP Allowed Address

Define trusted device to connect M-200 IPPBX

Click **Security** -> **SIP Allowed Address** -> **Add Allowed IP**



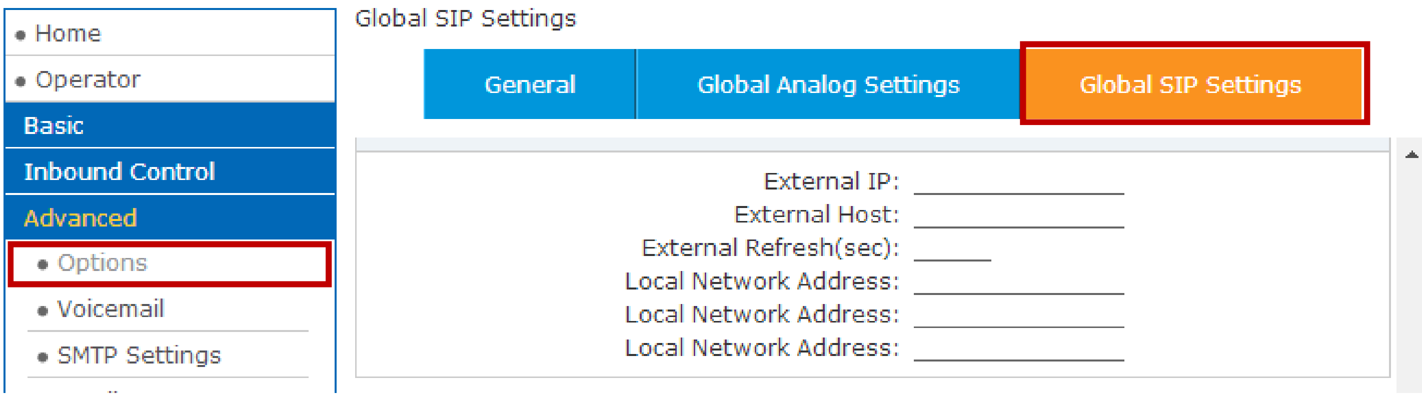
Reference:

Item	Explanation
Allowed IP	Input allowed IP or network segment, e.g.: 192.168.100.1 or 192.168.1.0
Subnet Mask	Define subnet of trust network

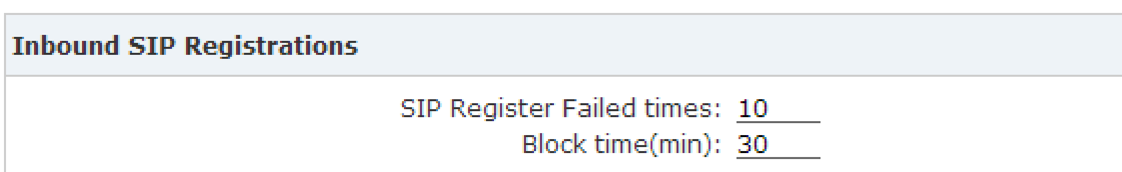
Block SIP Register Failed

This function is for Block IP address when the same IP address is going to SIP register too many password failures in the specify period.

Go Advanced -> Options -> Global SIP settings



At Inbound SIP Registrations

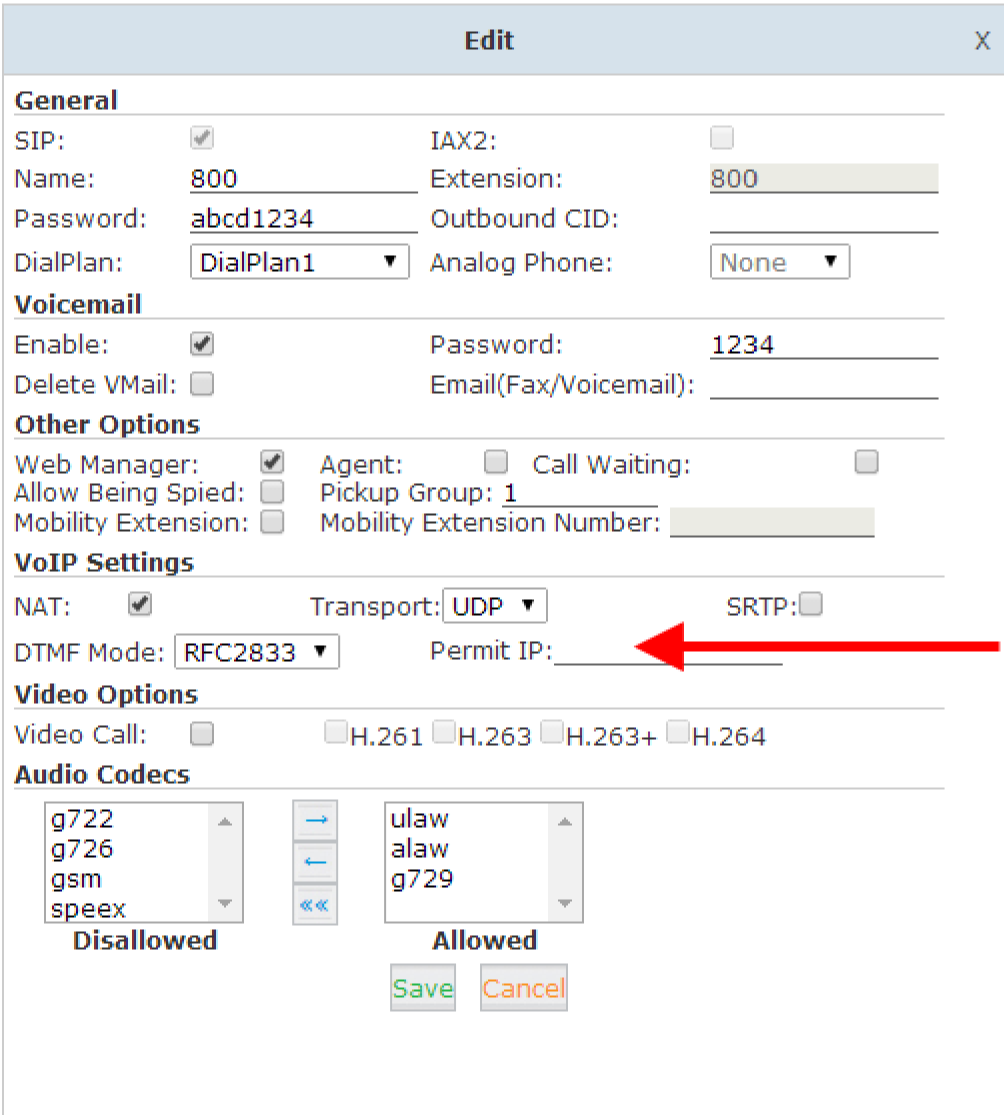


NOTE:

For this setting when the same IP address use SIP register failed at 10 times it will block the IP address 30 mins.

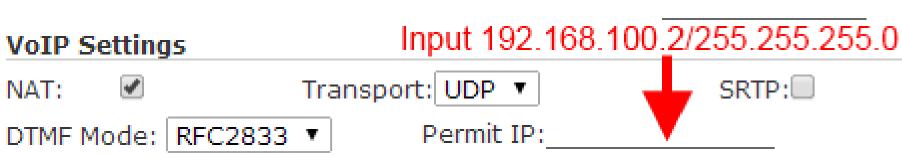
Permit specific IP for Extensions

Click **Basic** -> **Extensions** -> choose extension which only permit specific IP connect-> **VoIP Settings** -> **Permit IP**



Example

Only IP 192.168.100.2 can use Extension 800

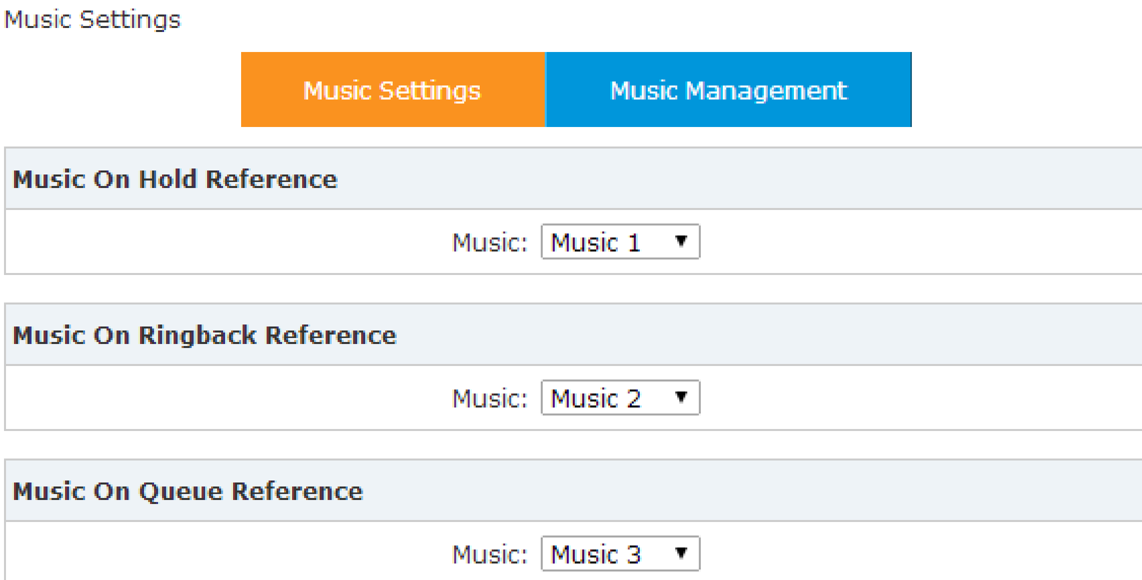


Chapter 12: System Settings

Music Settings

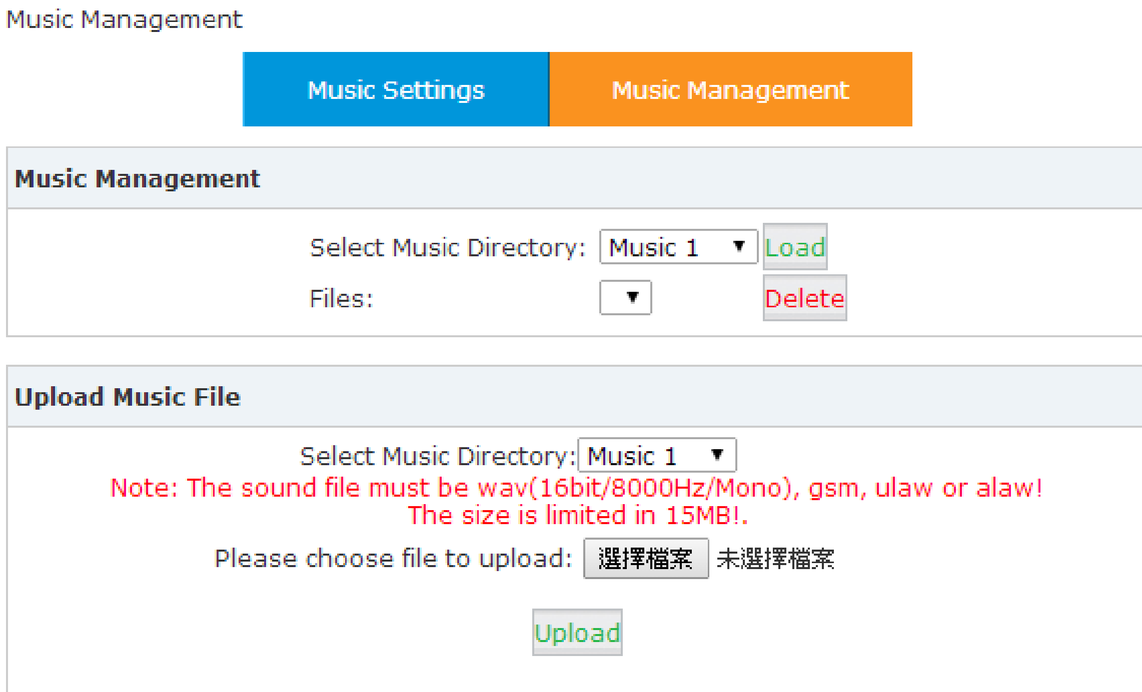
By default, System had store 10 Music, let you change Music of Music on hold, Music on Ring back, Music on Queue

Click **Advanced** -> **Music Settings**



Also you can add your own Music, remember custom music must be wav (16bit/8000Hz/Mono), gms, ulaw or alaw format and max file size is 15MB.

Click **Advanced** -> **Music Settings** -> **Music Management**

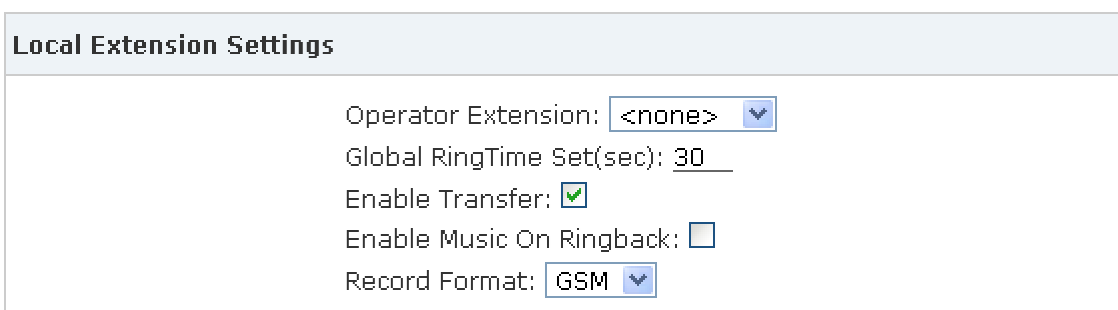


System Options

General

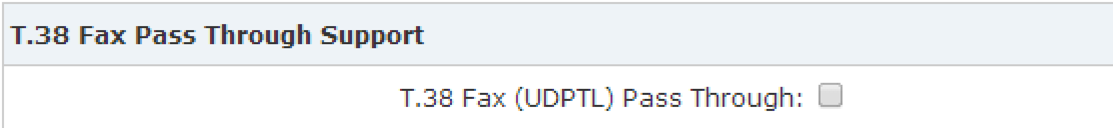
To change the General Options

Go **Advanced** -> **Options** -> **General**



Item	Explanation
Operator Extension	Set extension number for Operator.
Global Ring Time Set	Set Ring time for every extension.
Enable Transfer	Check to enable Transfer.
Enable Music On Ring back	Check to enable Music On Ring back.
Record Format	Set the format for recording files. (GSM/WAV only)

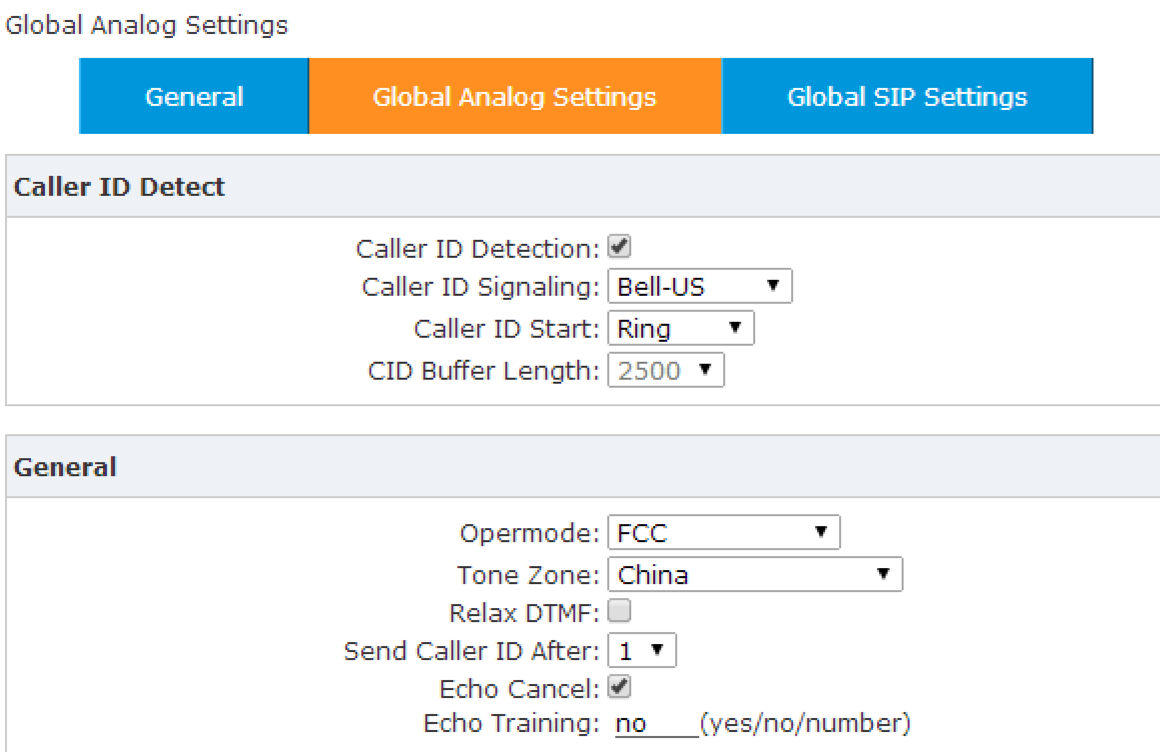
T.38 Fax Pass Through



Item	Explanation
T.38 fax (UDPTL) Passthrough	Enables T.38 fax (UDPTL) passthrough on SIP to SIP calls

Global Analog Settings

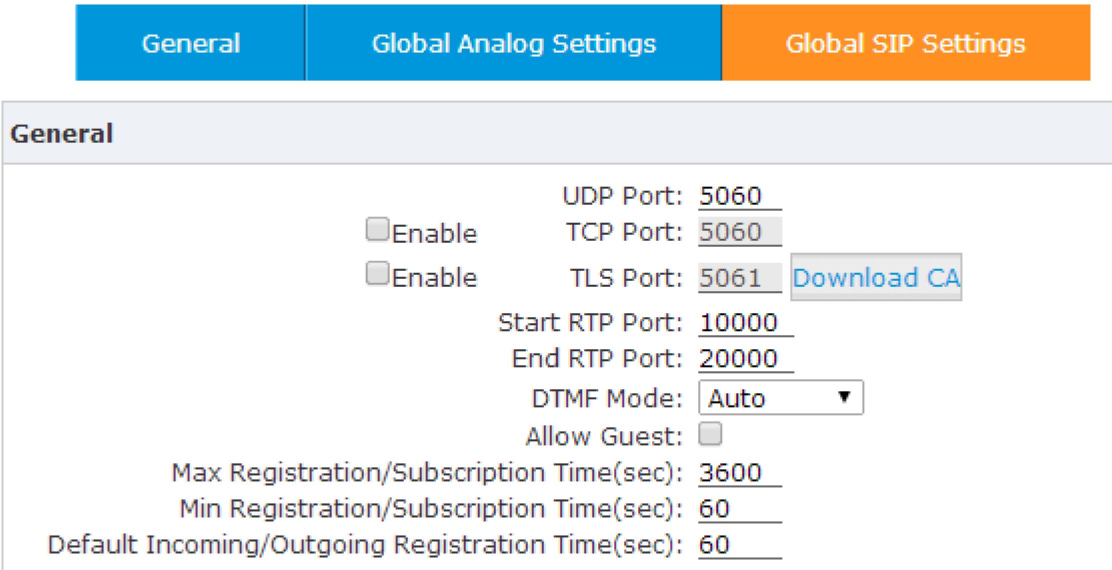
Go **Advanced** -> **Options** -> **Global Analog Settings**



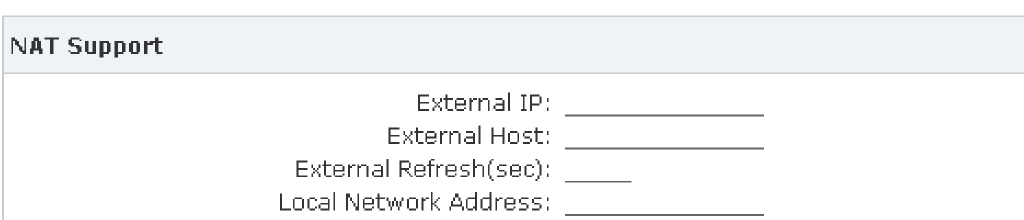
Item	Explanation
Caller ID Detection	Enable/Disable Caller ID Detection
Caller ID Signaling	Select the mode of Caller ID Signaling.
Caller ID Start	Ring--Caller ID start before ring.
CID Buffer Length	Default CID Buffer Length
Opermode	Set the Opermode for FXO/GSM Ports.
ToneZone	Select the ToneZone in your country.
Relax DTMF	Enable/Disable Relax DTMF inspection.
Echo Cancel	Enable/Disable Echo Cancel
Echo Training	Set Echo Training (default unit: ms)
Busy Detection	Enable/Disable Busy Detection.
Busy Count	Count the Busy Detection. It will be active when enabling

Global SIP Settings

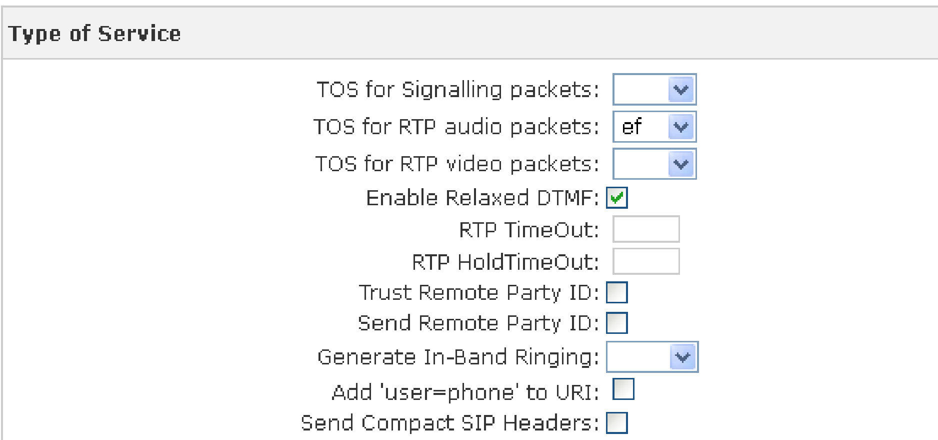
Global SIP Settings is appropriate for professionals. If anything needs to be modified, please contact our tech-support people.



Item	Explanation
UDP Port to bind to	SIP standard port is 5060
TCP Port	Default TCP port is 5060
TLS Port	Default TLS port is 5061
Start RTP Port	RTP port range
End RTP Port	RTP port range
DTMF Mode	Set default DTMF mode for sending DTMF, support auto,
Max Registration/Subscription	Maximum duration (in seconds) of incoming
Min Registration/Subscription	Minimum duration (in seconds) of
Default Incoming/Outgoing	Default duration (in seconds) of incoming/outgoing



Item	Explanation
External IP	Address that we're going to put in outbound SIP
External Host	Alternatively, you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use external IP instead
External Refresh	How often to refresh external host if used. You may
Local Network Address	192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks, '10.0.0.0/255.0.0.0' : Also RFC1918, '172.16.0.0/12' : Another RFC1918 with CIDR notation,'169.254.0.0/255.255.0.0' : Zero conf local network

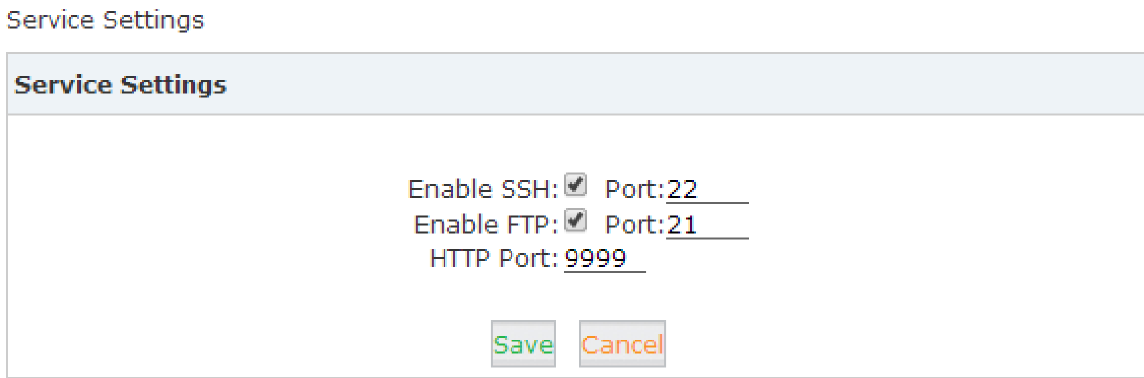


Item	Explanation
TOS for Signaling packets	Sets Type of Service for SIP packets
TOS for RTP audio packets	Sets Type of Service for RTP audio packets
TOS for RTP video packets	Sets Type of Service for RTP video packets
Enable Relaxed DTMF	Relax DTMF handling
RTP Time Out	Terminate call if 60 seconds of no RTP activity when we're not on hold
RTP Hold Time Out	Terminate call if 300 seconds of no RTP activity when we're on hold (must be > RTP time out)
Trust Remote Party ID	If Remote-Party-ID should be trusted
Send Remote Party ID	If Remote-Party-ID should be sent
Generate In-Band Ringing	If we should generate in-band ringing always, use 'never' to never use in-band signaling, even in cases where some buggy devices might not render it. Default: never
Add 'user=phone' to URI	If checked, 'user=phone' is added to URI that contains a valid phone number
Send Compact SIP Headers	Send compact sip headers

Settings of SSH / FTP / HTTP Port

You can enable SSH, FTP service on M-200 or change Web GUI management port.

Click **Security** -> **Service**



USB External Storage

External Storage for store the Voice log and Call Detail.

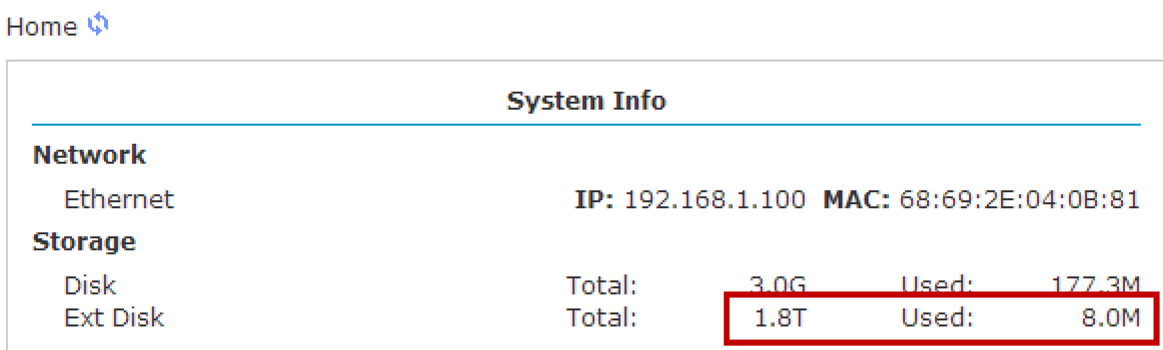
NOTE:

Before connect your disk to M-200 IPPBX Make sure your USB Storage in FAT32.

Step to connect USB Storage

1. Unplug the power supply.
2. Plug the USB Disk to USB port.

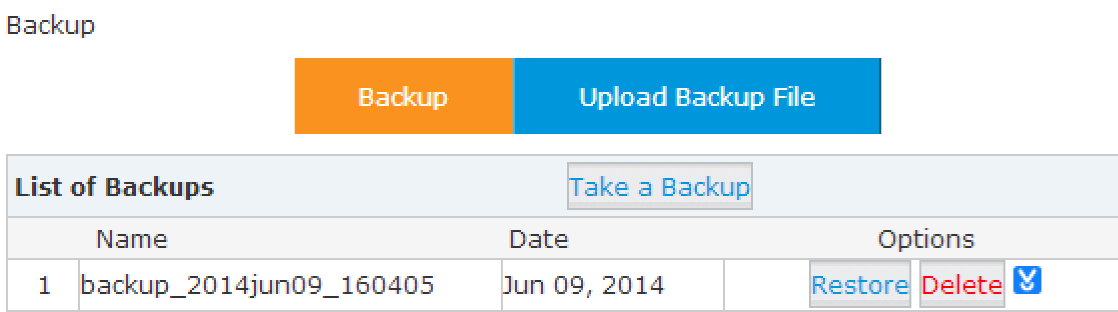
3. Connect the power supply.
4. Login in to the M-200 IPPBX
5. If connection success In **Home** -> **System Info** will show your capacity



Backup and Restore

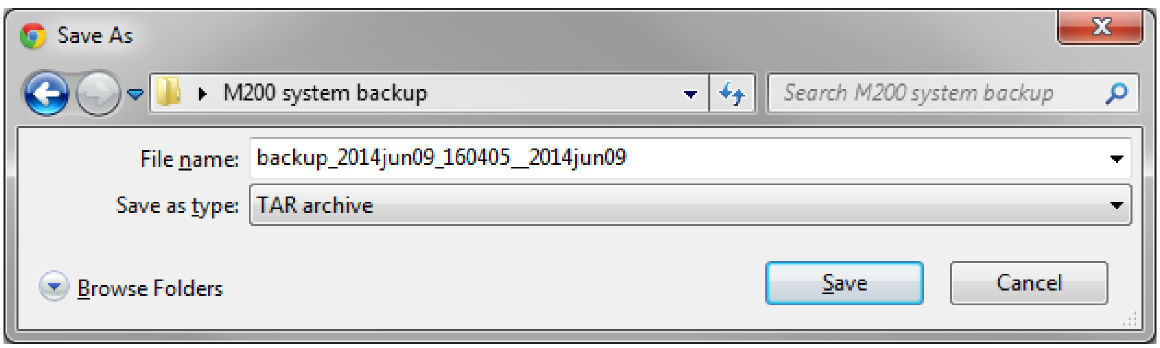
Backup system can decrease the loss when system failure.

Click **System** -> **Backup** -> **Take a Backup**



Download backup file from SIPDEX PBX

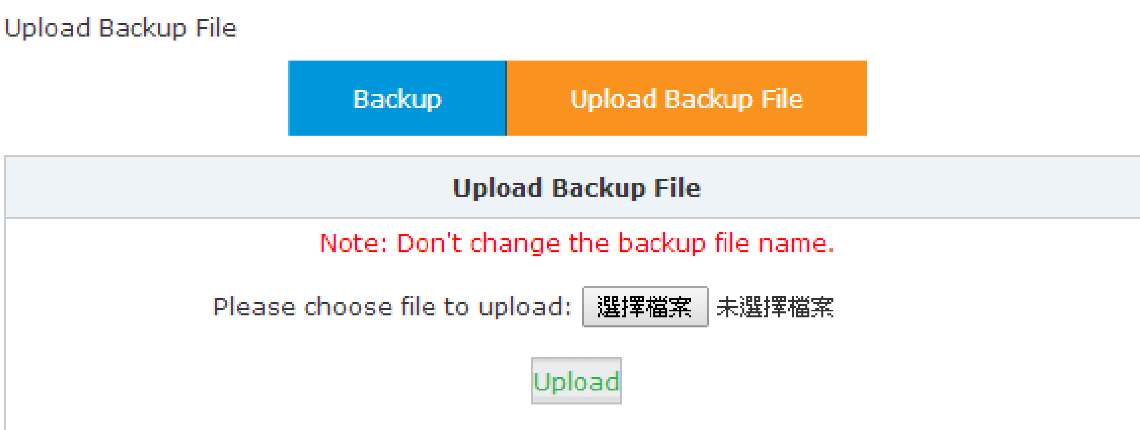
Click can download backup file



* Don't change file name.

Upload backup file to SIPDEX IP PBX

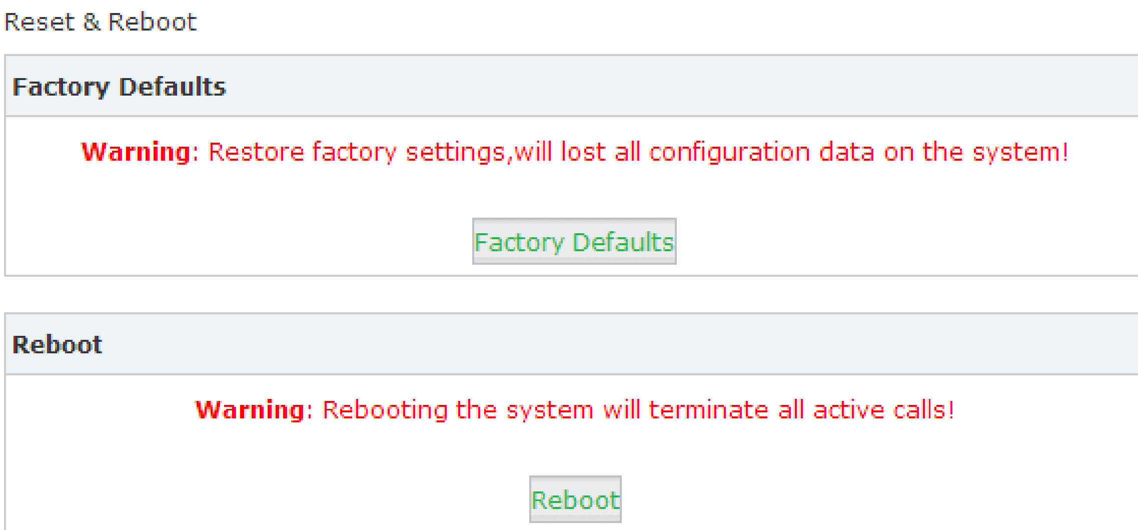
Click **System** -> **Backup** -> **Upload Backup File** -> **choose backup file** -> **Upload**



Reset & Reboot

In this session, you can reboot system or restore system to factory state.

Click **System** -> **Reset & Reboot**



Reference:

Item	Explanation
Factory Defaults	Reset whole system to default factory state (**All setting will be cleared**)
Reboot	Reboot system

Upgrade

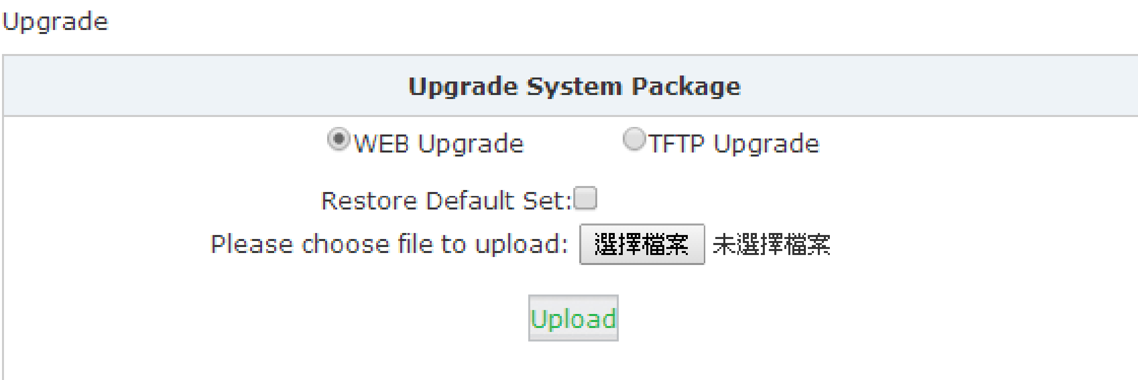
A software image upgrade can be necessary for these reasons:

●You want to implement new features in your IPPBX that are available in the later software release.

●You want to install a new module card that the current software version does not support.

●A known bug has affected your IPPBX. The later software release resolves the bug.

Click **System** -> **Upgrade**



Reference:

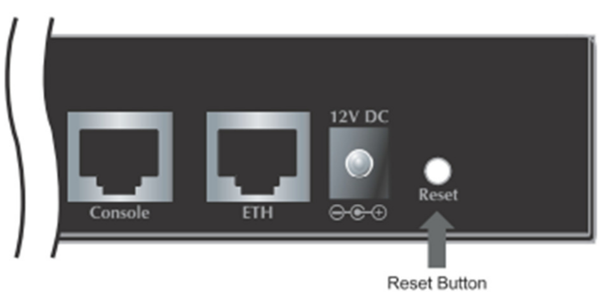
Item	Explanation
WEB Upgrade	Through WEB GUI to upgrade SIPDEX IPPBX system
TFTP Upgrade	Through TFTP Server to upgrade SIPDEX IPPBX system
Restore Default Set	If checked, after upgrade success system will be restore to factory default state

Reset Button

To reset the IP address to the default IP address “192.168.0.100”(ETH) or reset the login & password to default value, press the reset button on the panel for **more than 6 seconds**. After the device is rebooted

Button	Action	Description
Reset	Press less than 6 sec	System reboot.
	Press over 6 sec	Reset to Factory Default

* Please be reminded to reset to factory default. Uploaded music setting (on hold music) and backup file will not be removed.

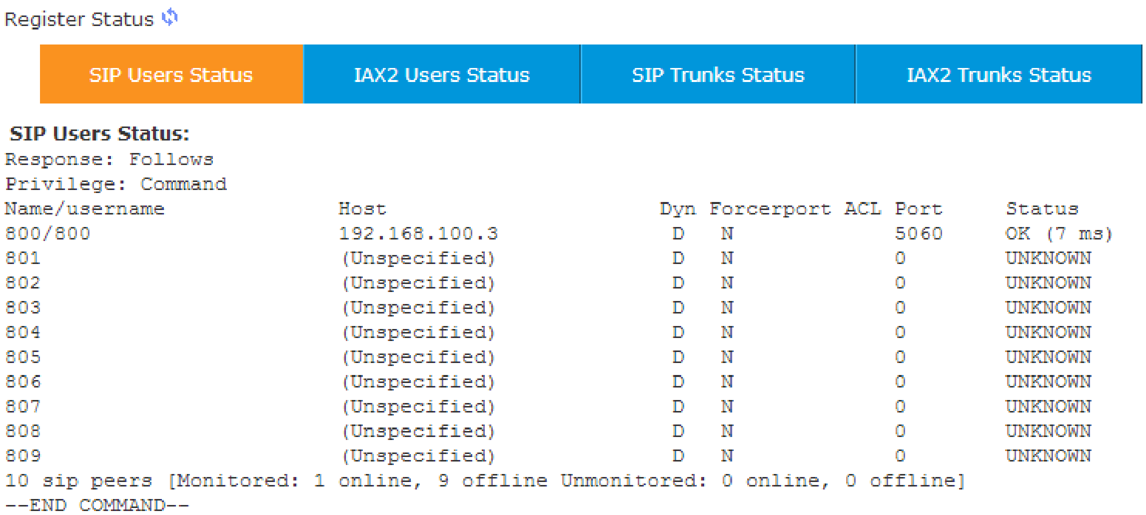


Chapter 13: Reporting

Register Status

On Register Status, you can find which SIP, IAX2, SIP trunks, IAX2 Trunks have been registered.

Click **Report** -> **Register Status**



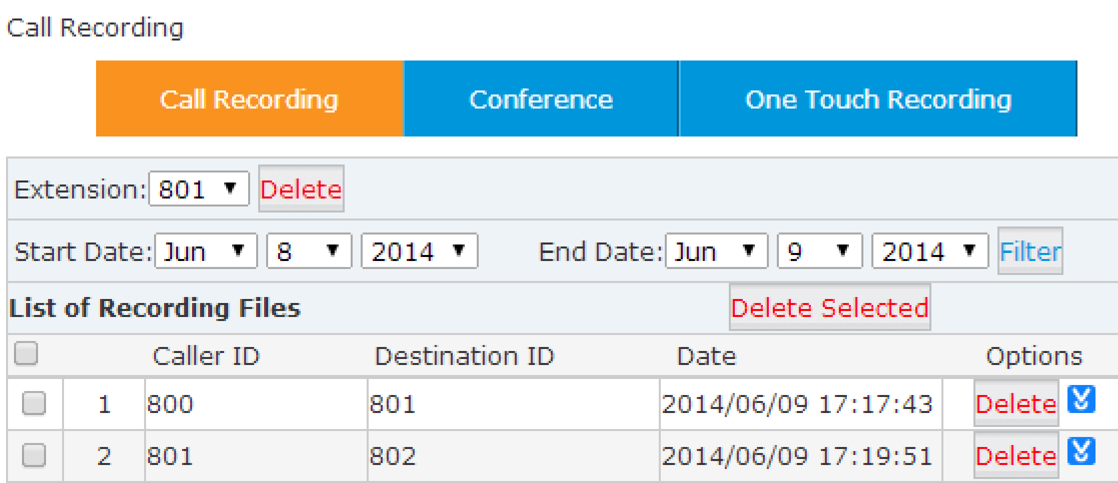
SIP Users Status:

```
Response: Follows
Privilege: Command
Name/username             Host                     Dyn Forceport ACL Port     Status
800/800                   192.168.100.3             D   N            5060     OK (7 ms)
801                       (Unspecified)             D   N            0        UNKNOWN
802                       (Unspecified)             D   N            0        UNKNOWN
803                       (Unspecified)             D   N            0        UNKNOWN
804                       (Unspecified)             D   N            0        UNKNOWN
805                       (Unspecified)             D   N            0        UNKNOWN
806                       (Unspecified)             D   N            0        UNKNOWN
807                       (Unspecified)             D   N            0        UNKNOWN
808                       (Unspecified)             D   N            0        UNKNOWN
809                       (Unspecified)             D   N            0        UNKNOWN
10 sip peers [Monitored: 1 online, 9 offline Unmonitored: 0 online, 0 offline]
--END COMMAND--
```

Record List

When enable **Call Recording (**for detail please read Chapter 8 Recording on Specified Extension**)** function, system will be record specific extension when Inbound call or outgoing call.

Click **Report** -> **Record List**



		Caller ID	Destination ID	Date	Options
	1	800	801	2014/06/09 17:17:43	Delete
	2	801	802	2014/06/09 17:19:51	Delete

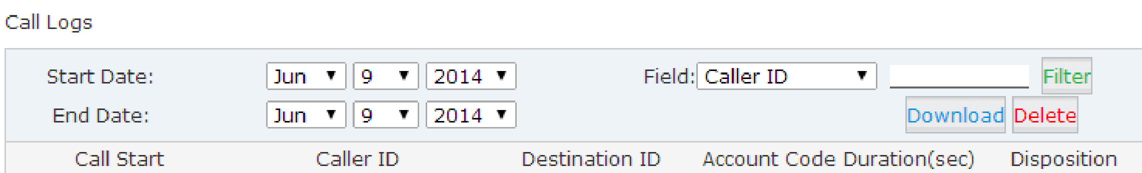
Reference:

Item	Explanation
Delete (near Extension)	Remove all specific extension call record.
	Download Record data.
Delete (under column **Options**)	Delete specific call record.

Call Logs

You can search all or specific extensions call logs is the session.

Click **Report** -> **Call Logs**



Reference:

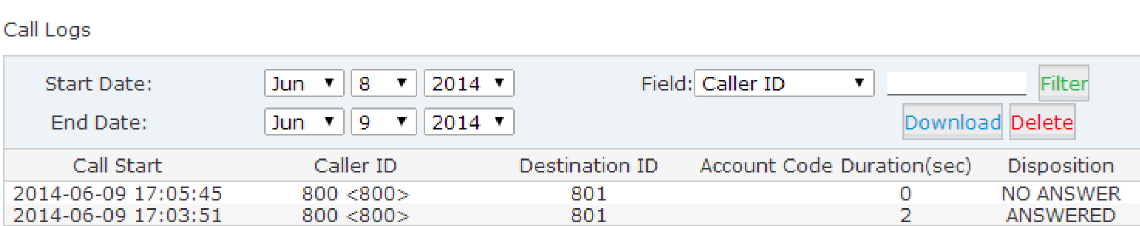
Item	Explanation
Start Date / End Date	Set searching date between Start date and End date
Field	Choose search method, by caller ID, by destination ID or by Account code
Download	Export search result
Delete	Delete all call logs record

Example:

I want to list call log between Jun 8 2014 and Jun 9 2014

1) Start Date set Jun 8 2014 and End Date set Jun 9 2014

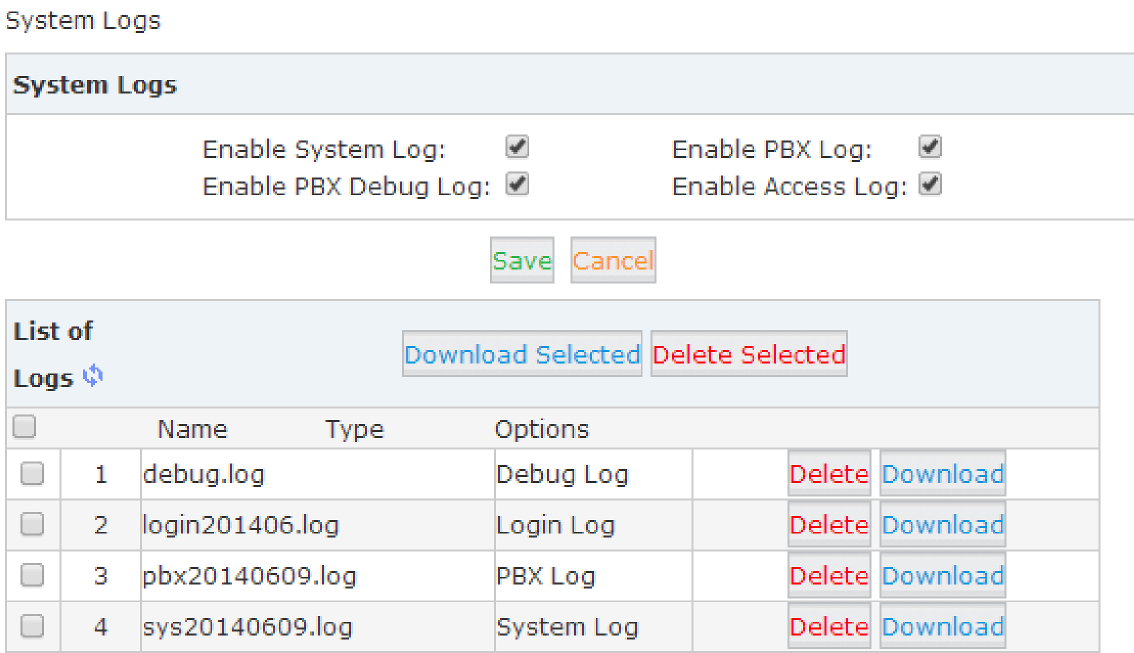
2) Click **Filter**



System Logs

Export IP PBX logs such as System logs, PBX log, PBX debug and Access log

Click **Report** -> **System Logs** -> check which log(s) you want to export -> Press **Save**



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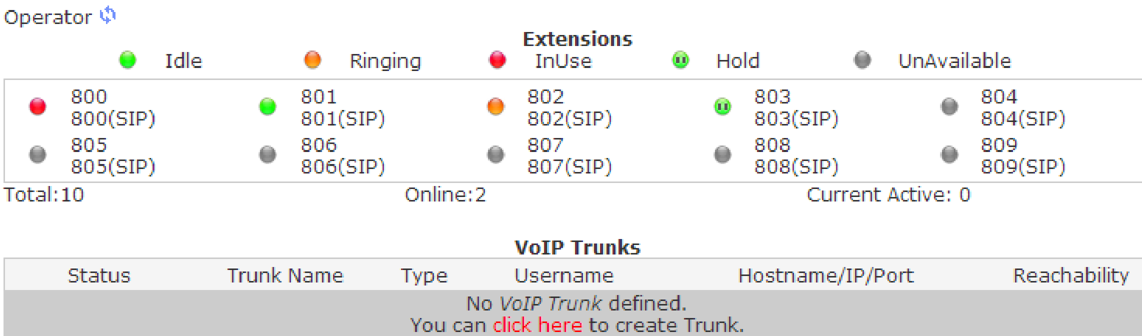
Operator Panel

Click **Operator**



Item	Explanation
Idle	This extension is standby.
Ringing	This extension has incoming call.
InUse	This extension is using
Hold	This extension is using Hold function.
UnAvailable	This extension has not registered.