SNR-VX50 SERIES IPPBX USER MANUAL

Basic Configuration

1 Preparation Before Operation

What kind of IP Phone can be used with SNR-VX50 IP PBX?

FXS Interface

- Analog Phone
- SIP Extension
- IP Phone which support SIP/ IAX2 protocol

2 Before Making a Call

2.1 Login IP PBX

Getting IP Address

SNR-VX50 Series IP PBX support 3 Ways to get the IP Address: Static/ DHCP/ PPPoE

Default IP And Port of WAN&LAN:

- WAN Port IP: http://192.168.1.100:9999
- LAN Port IP: http://192.168.10.100:9999
- LAN Supper IP: 169.254.1.254/255.255.0.0

Default configuration and function key

- Web GUI username: admin
- Web GUI password: admin
- **11 Play the IP Address of WAN port
- **12 Play the IP Address of LAN port
- 600 Enter into the Voicemail Box
- 900 Enter into the Meeting
- # Blind Transfer
- *2 Attended Transfer
- * Disconnect Call

Login to the system

After connecting the IP PBX to the local area network, launch the web browser on a computer which is in this local area network. Enter the IP address of the system (WAN port IP address http://192.168.1.100:9999, LAN port IP address http://192.168.10.100:9999). Enter Username and password (default username is admin, password is admin), then click “login”.

If username and password are right, this following page will be displayed:

PIC

- Network WAN/ LAN Port IP will be displayed
- Storage Total storage and used storage will be displayed
- Channels Channel information will be based on the product model
- Device Info Product Model and System Version will be displayed

Common Button

Besides of the device info in the home page, the following common buttons are displayed as well:

- ? Log out Log out GUI
- ? Reboot Reboot the IP PBX system
- ? Factory Defaults Restore all settings to factory default
- ? Activate Changes Activate the changes for your current configuration

System Menu

System Menu include the following sub menu:

- ? Home Page Display device info
- ? Basic Basic configuration on extension, trunks, etc
- ? Inbound Control Configure Inbound Route, IVR and Black List, etc
- ? Advanced Configure extension's default info, conference, etc.
- ? Status Check recording list, call logs, register status, etc here.
- ? System Configure network, time, etc; manage call logs, back up files, etc

2.2 Basic Configuration

SNR-VX50 IP PBX support SIP/IAX2 and analog extension, support "Batch Add Users".

configure extension from this page: 【Basic】---- 【Extensions】

Extension Settings

Item	Explamation
Search	Search extension precisely or fuzzily
Show all	Show all extensions
Extension	Be connected to the phone eg: "888"
Name	Extension name (English letter is supported only) eg: "Tom"
Password	Support default or random password, combined by letter and figure, eg: "12u3b6”
Caller ID	Caller's ID eg: "801"
Outbound CID	Overrides the caller id when dialing out with a trunk.
VM Password	Voicemail Password for this user, eg: "1234"
E-mail	The e-mail address for this user, eg. "Tom@gmail.com"
Analog	If this user is attached to an analog port on the system, please

Phone	choose the port number here.
Dial Plan	Please choose the Dial Plan for this user, Dial Plan is defined under the "Outbound Routes"
Voicemail	This user will have a voicemail account after choosing this option.
Can reinvite	Set up calls directly between caller and receiver, after being connected by IP PBX system. This method is known to cause problems with certain hardware, such as the common Cisco ATA 186.
SIP	Check this option if the User or Phone is using SIP or is a SIP device.
IAX2	Check this option if the User or Phone is using IAX2 or is an IAX2 device.
T.38 Fax	Enables T.38 fax (UDPTL) pass through on SIP to SIP calls
Agent	Check this option if this User or Phone is an Call Agent.
NAT	Check this option if the User or Phone is located behind a NAT (Network Address Translation) enabled gateway
Pickup Group	Select your pickup group.
Delete VMail	Voicemail will not be checkable by phone if you choose this option. Messages will be sent by email only. Note: You must configure SMTP server for this functionality
DTMF Mode	The Dual-Tone Multi-Frequency mode to be used is specified here and can be changed if necessary. The default is rfc2833.
Video Call	Enable/Disable Video call for this extension
Permit IP	IP address and network restriction. eg: "192.168.1.77" or "192.168.10.0/255.255.255.0"
Auto Provision	P PBX can work with Grandstream and Yealink IP Phone on this function. Please select the phone manufacture and input MAC address of the IP Phone.
Codecs Configure	The allowed and disallowed codecs can be selected by clicking this link. Default codecs are alaw, ulaw and G.729.

Note:

1) There are 30 default extensions which number started with "8", you can add or delete extension by your requirement.

2) Maximum extensions is 100

Upload/Download Extensions

If you wanna batch add users, please click【Upload/Download Extensions】

Please download the demo from【Download Extensions demo】, add extension files and save based on the demo, choose the extension file which you wanna upload.

You can download the extension file by click【Download Extensions(.csv)】

Extensions Log

Click 【Extensions Log】to check the extensions log, you can refresh or clear the log

2.3 Time Based Rules

You can set working time rule and after-working time rule, and deal with your inbound call based on this time rule. Please set from this page: 【Time Based Rule】--- 【New Time Rule】

New Time Rule:

Item	Explanation
Rule Name	Define the time rule name.
Time & Date Conditions	Set time segment of Month/Date/Week.
Destination	How to deal with the inbound call in different time segment eg: Inbound call will be forward to IVR in working time.

3 Outbound Call

3.1 Trunks

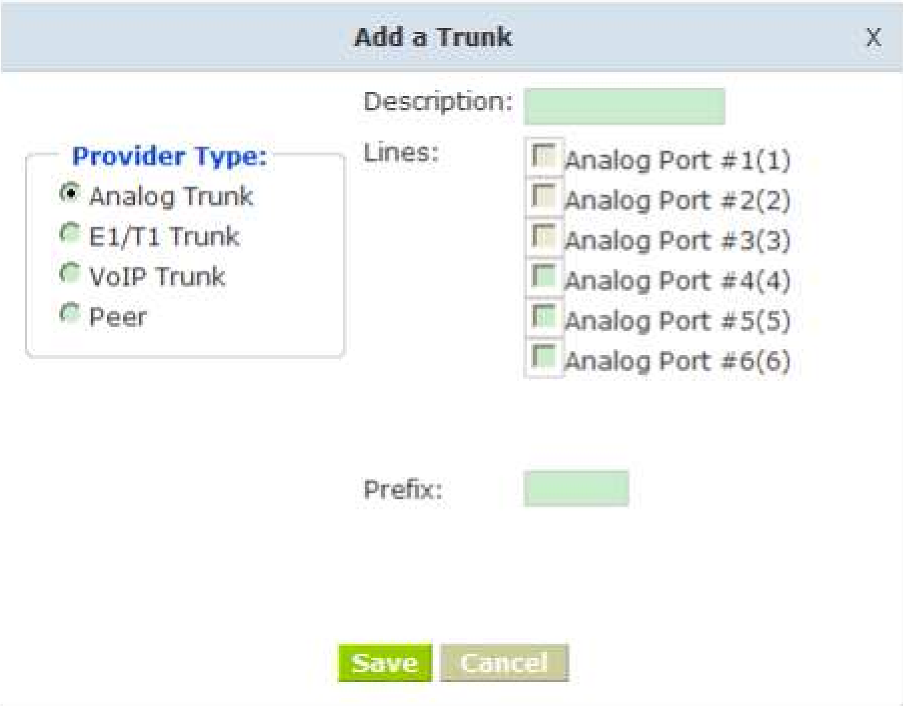
If you want to set up outbound call to connect to PSTN(Public Switch Telephone Network) or VoIP provider, please configure on this page: 【Basic】-> 【Trunks】

IP PBX support 3 kinds of trunks: Analog/GSM line, E1/T1 Line, Custom VoIP, Peer.

How to add each trunk:

1) Analog

Click 【Add a Dial Rule】-> 【Analog/GSM】



Item	Explanation
Description	Define description for the trunk.
Lines	Individual lines of the PBX eg: Analog Port #3: The third analog port of the PBX.

You can configure the Analog/GSM line through IP PBX. Same Analog line couldn't be used in multiple trunks. If you don't have available Analog/GSM trunk, you can't set up trunk.

2) E1/T1/GSM Line

Please add E1/T1/GSM Line refer to the configuration of Analog line.

3) Custom VoIP

Custom VoIP allows you to create a VoIP trunk, please configure on this page:

【Add a Trunk】->【VoIP Trunk】

Item	Explanation
Description	Description for VoIP Trunk, digit or letter is allowed.
Protocol	Choose protocol for this trunk, SIP or IAX2
Dial Plan	Choose a dial plan for this trunk, define it in the submenu named 【Outbound Routes】.
Register	Check for opening register service; otherwise register service is closed
Host	Host Address provided by VoIP Provider.
Outbound proxy	Outbound proxy is provided by VoIP Provider.
Proxy Port	Proxy Port is provided by VoIP Provider.
Prefix	The prefix will be added as default, when this trunk is used.
Without Authentication	If you don't use Authentication when connecting server, please check this option.
Username	Username provided by VoIP Provider.
Password	Password provided by VoIP Provider.

3) Peer

IP PBX will be taken as a Client when you use "Peer", it's used for outbound call

by connecting to another SNR-VX50 IP PBX.

Item	Explanation
Peer Name	Define the Peer Name, digit or letter is allowed.
Protocol	Choose protocol for this trunk, SIP or IAX2
Dial Plan	Choose a dial plan for this trunk, define it in the submenu named 【Outbound Routes】.
Host	IP Address of the other VX50 IP PBX
NAT	Check this option, extension user will be configured after NAT (Network Address Translation).
Without Authentication	If you don't use Authentication when connecting server, please check this option.
Username	Username provided by the other VX50 IP PBX.
Password	Password provided by the other VX50 IP PBX.

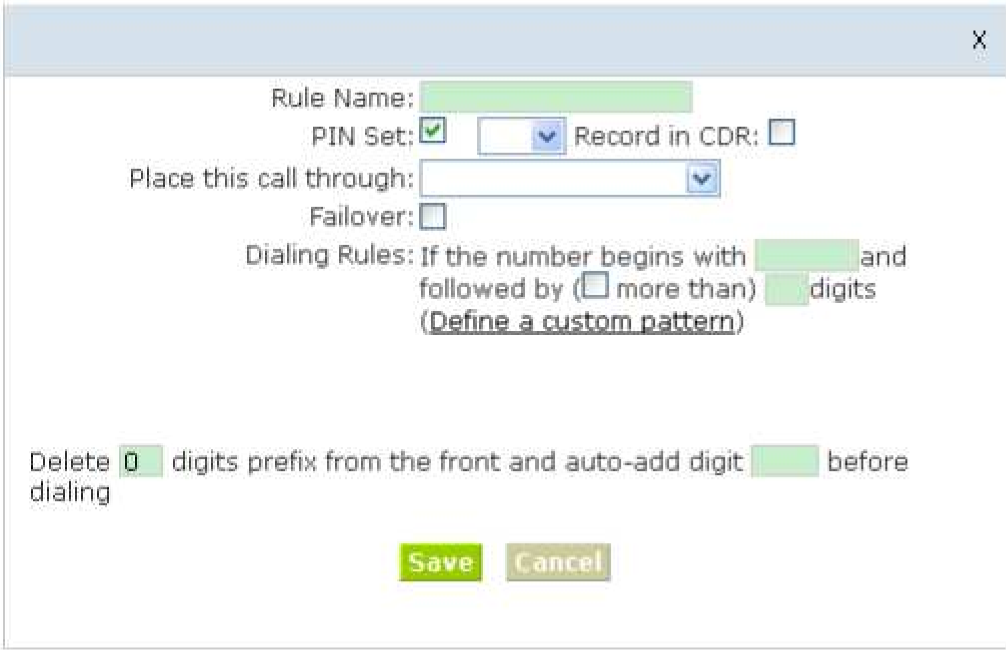
Once a trunk is added, this trunk will be displayed in the "List of Trunk". You can define the codecs, configure advanced settings or delete this trunk from the drop downs of "Option"

3.4 Outbound Routes

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

Please configure on this page: 【Basic】 -> 【Outbound Routes】

On this page, you can configure basic match pattern of outbound routes and create different dial plan. Please configure by clicking【Add a Dial Rule】



Item	Explanation
Rule Name	Set a name for this dial rule
PIN Set	Set PIN which you need input when you dial out by this rule.
Record in CDR	If you selected it, CDR will show which pin the call is outbound through
Place this call through	Choose a trunk for this rule
Failover	Choose a failover trunk for using when the above chosen trunk is not available.
Dialing Rules	Define the number match pattern for dialing.
Define a custom pattern	N digit from 2 to 9 Z digit from 1 to 9 X digit from 0 to 9 . One digit or multiple digits
Delete[ ]digits prefix	If deleted one digit prefix, when dial 12345, digit 2345 will be sent.
Auto-add digit [ ]	If added digit"1", when dial 12345, digit 123451 will be sent.

4 Inbound Call

4.1 Inbound Routes

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

Please configure on this page: 【Inbound Routes】

General

When a call from a trunk (Analog/ VoIP), it could be forwarded to an extension, call queue, conference or IVR. You can choose based on your requirement.

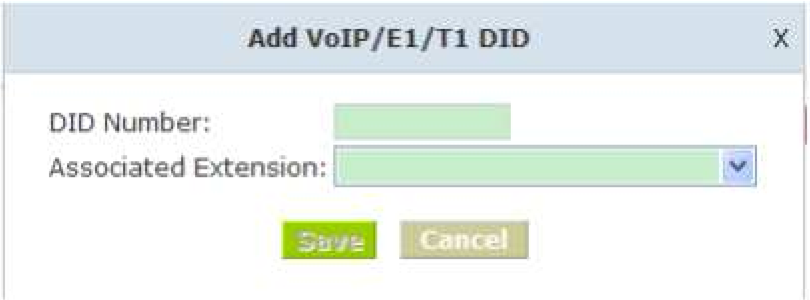
Analog Channel DIDs

If you want to direct the inbound call from a trunk (Analog) to a specified extension, call queue, conference or IVR, please configure on this page: 【Add Analog Channel】

- ·Channel Choose Analog Port of trunk
- ·Associated Extension Select Extension, call queue, conference or IVR for DID.

VoIP Channel DIDs

If you want to direct the inbound call from a VoIP trunk to a specified extension, call queue, conference or IVR, please configure on this page: 【Add VoIP Channel】



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

DODs Settings

If you want to direct the inbound call from any trunks to a specified extension, call queue, conference or IVR, please configure on this page: 【Add DOD】

- DOD Number This number is the caller's phone number, it could be called from analog channel or VoIP/GSM/E1/T1 Line.

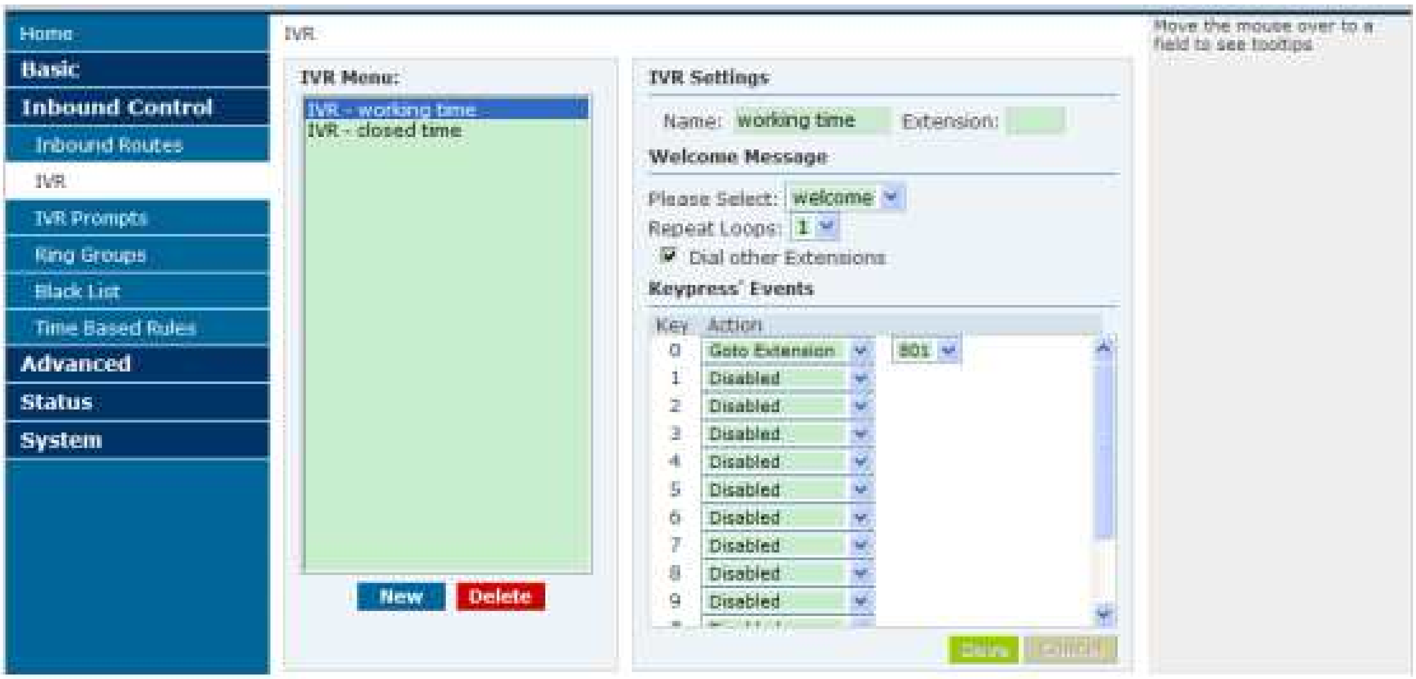
- Associated Extension Choose a specified extension, call queue, conference or

IVR to be directed to call

4.2 IVR

IVR will improve office efficiency based on your requirement.

Please configure on this page【IVR】



Item	Explanation
Name	Set a name for the IVR
Extension	If you want to listen to the IVR by dialing extension, please input an number.
Please Select	Select IVR audio file, please configure in this page: 【IVR Prompts】
Repeat Loops	loop times to repeat playing the IVR prompt.
Dial other Extensions	Allow caller to dial other extension besides of the ones listed as below.
Keypress' Events	Each digit will be related to the actions defined in the blank.

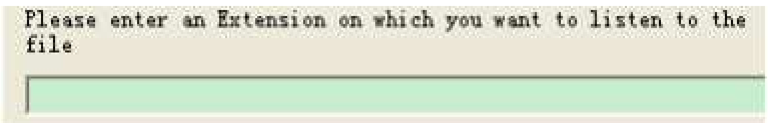
4.3 IVR Prompts

Record or play IVR music from extension. Please configure on this page:【IVR Prompts】

Click【Record a IVR Prompt】to display the diagram as below:

- File name   Define a name for the recorded IVR prompt
- Format   Define the format of the IVR Prompt, only GSM/WAV(16-bit)supported
- Extension   Select an extension for recording,click【Record】button, the selected extension will ring, then you can record IVR.

If your want to listen to the recorded IVR prompt, pls click【play】and input extension number in the following diagram, click【confirm】, the extension will ring and play the IVR prompt after hang up.



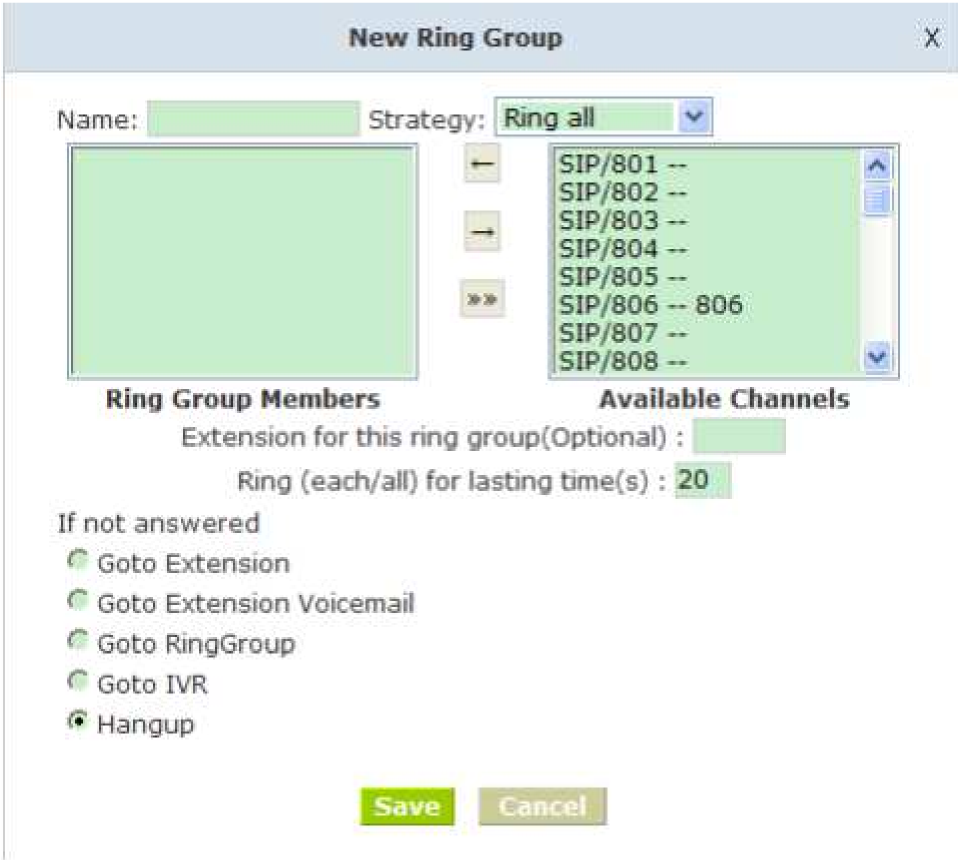
Upload Prompts

PBX prompts supports wav, gsm format, ulaw or alaw, and the size is limited in 15MB

4.4 Ring Groups

Ring Group is a collection of extensions. When a call to a ring group, all extensions in this ring group will ring in different way based on their different configuration, if ring time exceeded defined time, the call will be directed to IVR or others based on your configuration.

There isn't any data in the factory default 【Ring Groups】, please configure as below: Click【New Ring Group】to display the diagram as below:

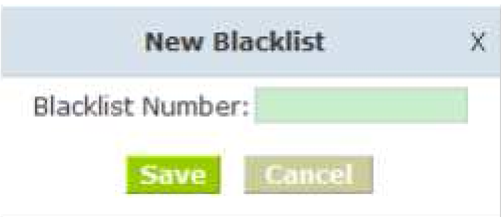


- Name Define a name for this ring group
- •Strategy Select strategy : "Ring all" or "Ring in order"
- •Ring Group Members Select ring group members in available channels, click to add
- • If not answered You can choose forward the call to extension, extension, Voicemail, RingGroup, IVR or Hangup.

5 Blacklist

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

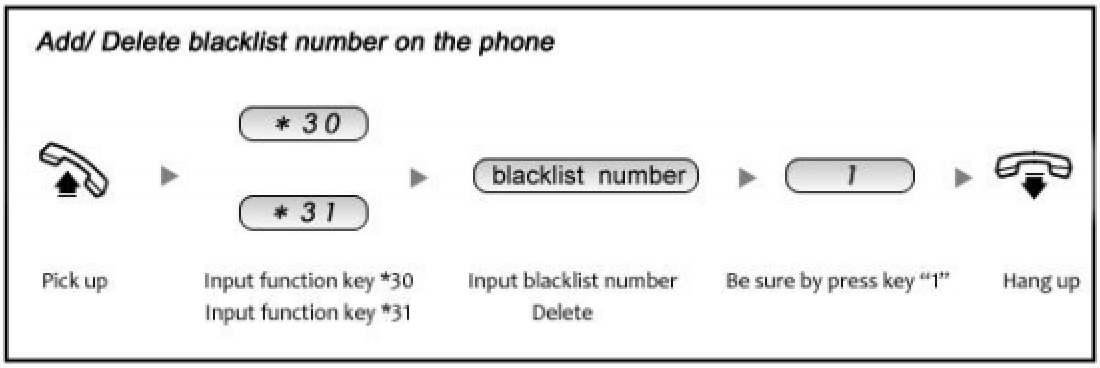
Please configure in【Blacklist】, click【New Blacklist】to display this dialog as below



Input caller's number in the blank, then this caller's number will be blocked when call again.

Meanwhile, extension user can add or delete the blacklist number by function key on the phone.

Please operate as the following diagram:

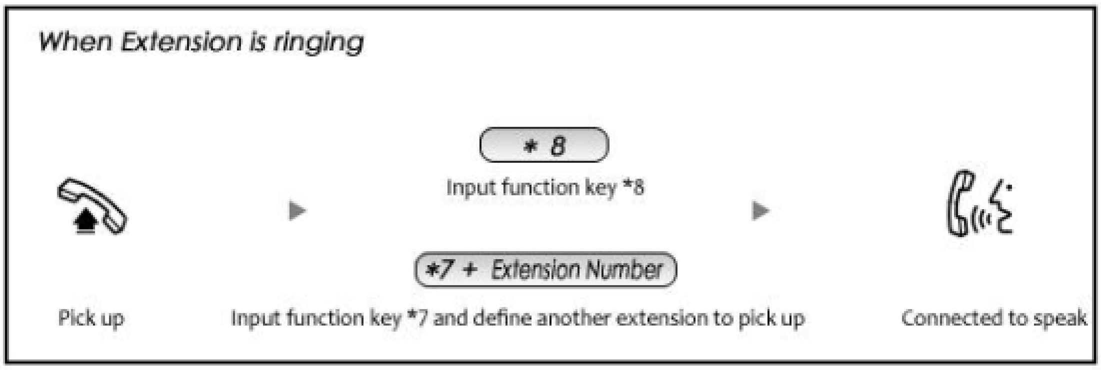


Reference Parameters and Explanation of Blacklist:

Item	Explanation
*30	When the extension user (in the system) input *30 to add a blacklist number, this number will be added to the "Black List"
*31	When the extension user input *31+ blacklist number, this number will be deleted from the "Black List".

5.1 Pickup Call

If an extension user is away from his/her desk, other extension users can pickup the call by function key on the phone. Please check the following diagram to learn:



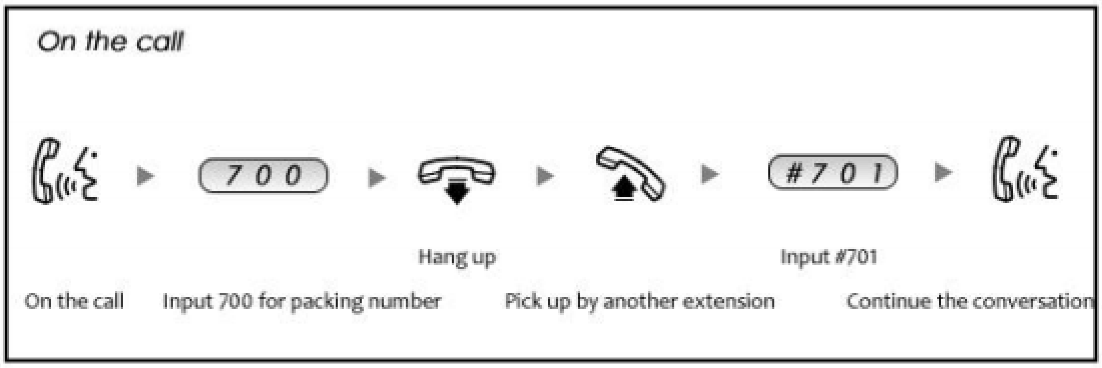
Reference Parameters and Explanation of Pickup Calls

Item	Explanation
*8	Pick up the ringing extension (in the system) at random. This can be defined in【Feature Codes】
*7	Defined extension number must be inputted after *7. This can be defined in【Feature Codes】.

6 On The Call

6.1 Call Parking

If you picked 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 diagram as below to learn:

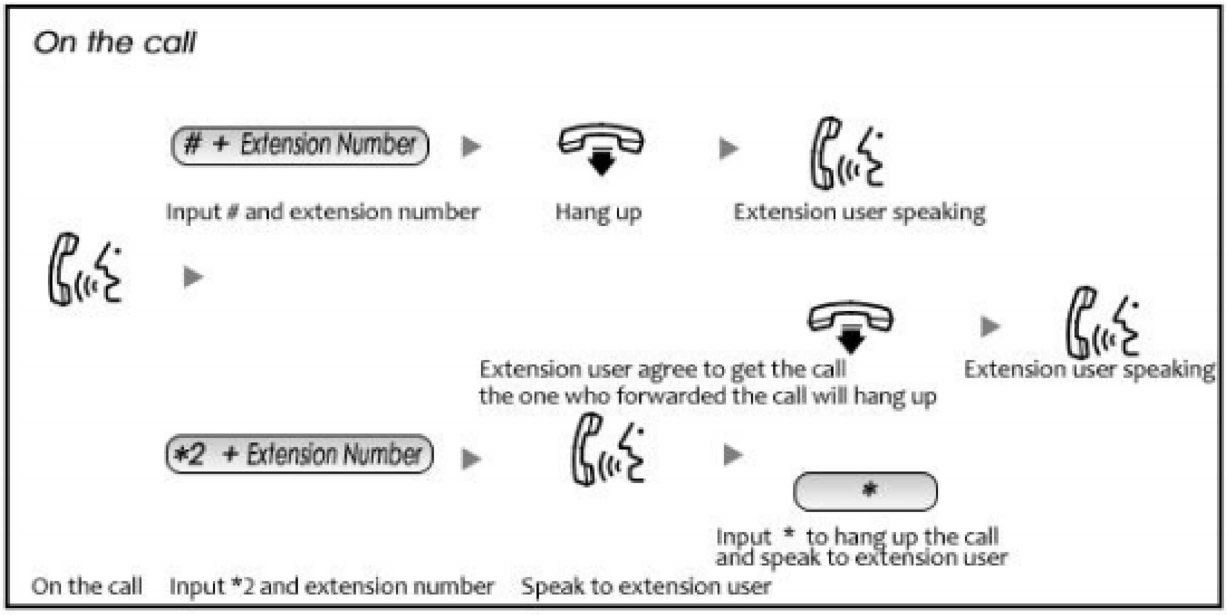


Reference Parameters and Explanation of Call Park:

Item	Explanation
Extension to Dial for Parking Calls:	Default number is 700. It can be defined in【Feature Codes】
What extension to park calls on	Default number is 701-720.It can be defined in【Feature Codes】
How many seconds a call can be parked for	Default is 45 seconds. It can be defined in【Feature Codes】

6.2 Transfer

If an incoming call asked to speak to your colleague, you can transfer the call directly to your colleague or transfer the call after agreed by your colleague. Please check the diagram as below to learn:

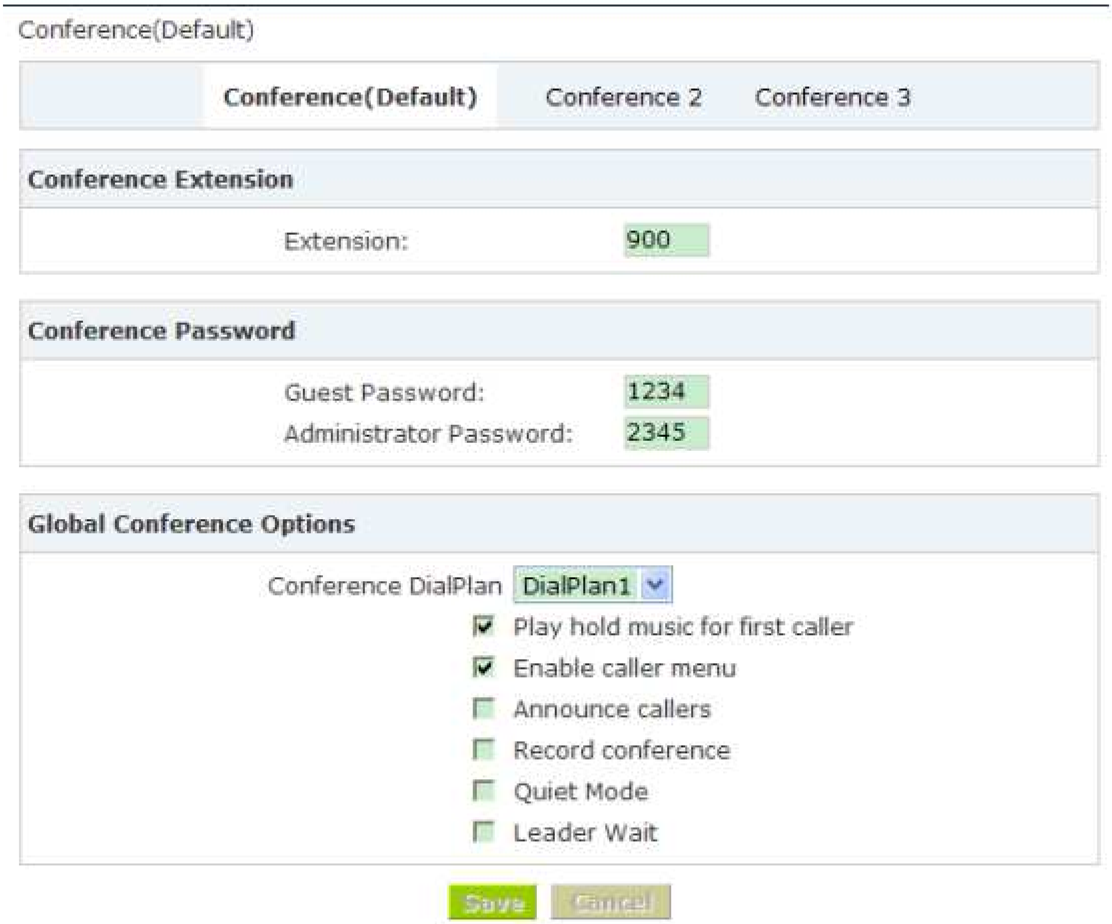


Reference Parameters and Explanation of Transfer:

Item	Explanation
Blind Transfer	Default is #, it can be defined in 【Feature Codes】
Attended Transfer	Default is *2, it can be defined in 【Feature Codes】
Disconnect Call	Default is *, it can be used after you use function key " *2 ". it can be defined in 【Feature Codes】
Timeout for answer on attended transfer	Default is 15 seconds, it can be defined in【Feature Codes】

6.3 Conference

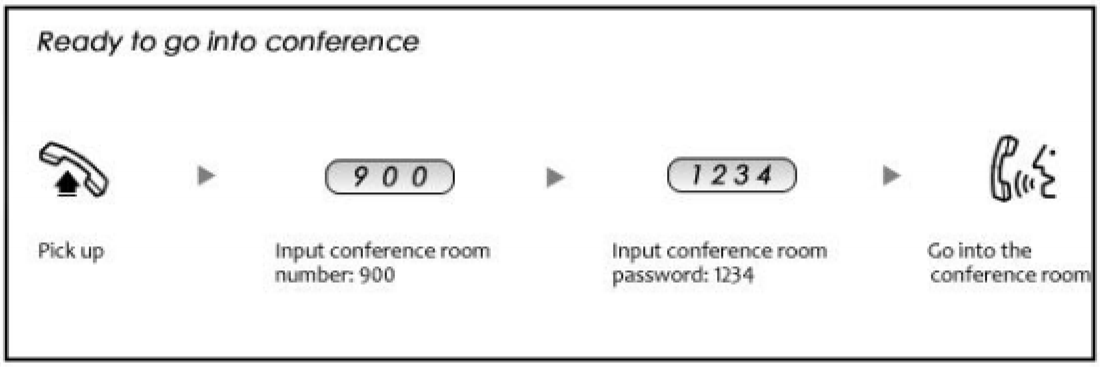
If you wanted to create a conference room for some extension users or with external lines, you can input conference room number 900, input conference room password 1234 (Admin's password is 2345), then enter into conference room. This model support 3 conference rooms. Please configure on this page【Conference】:



Item	Explanation
Extension	The number that users call in order to access the conference room, the default number is "900".
Guest Password	Guest enter the conference room by this code.
Administrator Password	Administrator enter the conference room by this code.
Conference DialPlan	Use the dialplan when you invite the other participant.
Play hold music for first caller	Check this option, Asterisk will play Hold Music to the first user in a conference, until another user has joined the same conference.
Enable caller menu	Checking this option allows a user to access the Conference Bridge menu by pressing the * key on their dialpad.
Announce callers	Checking this option announces to all Bridge participants, the joining of any other participants.
Record conference	Recording format is WAV。
Quiet Mode **Leader Wait**	If this option was checked, all users entering this conference will be marked as quiet, and will be in Wait until the conference leader (admin user) arrives before starting the conference.

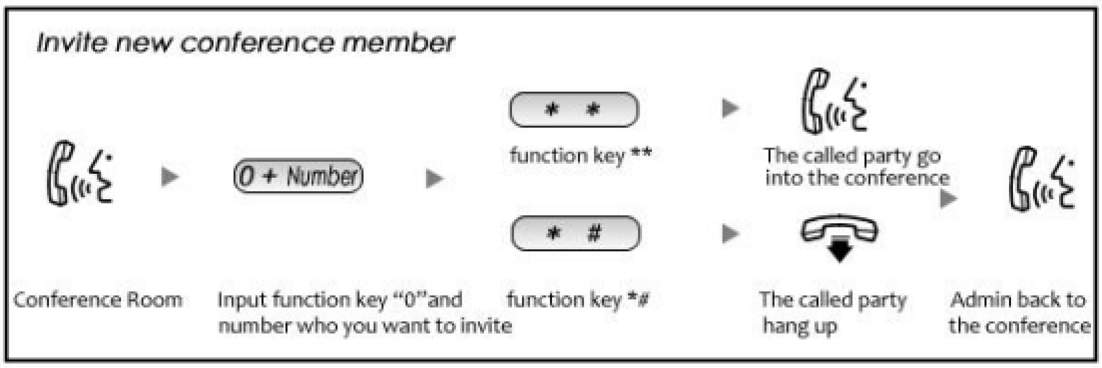
Please check the following diagram to learn:

Go to conference:



In the conference, admin can invite new guest (extension user or external number) into the conference.

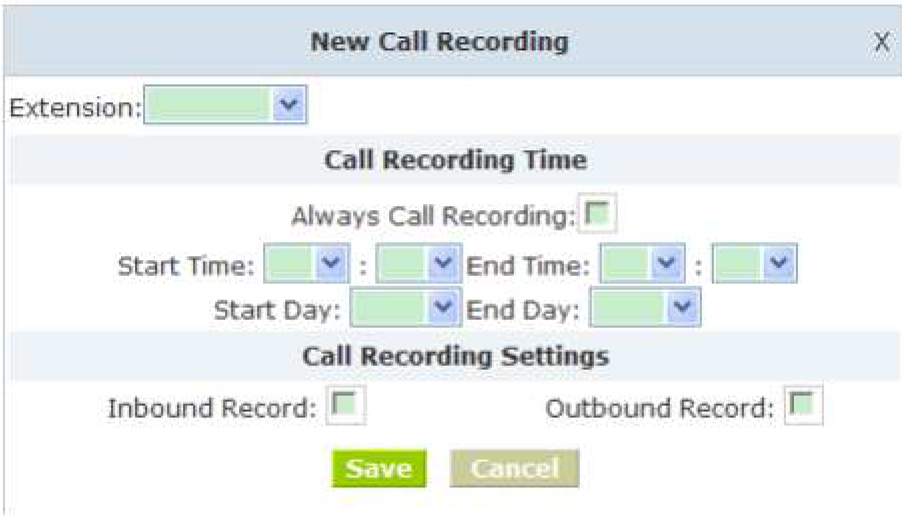
Invite new guest:



6.4 Call Recording

Record the specified extension, also you can record in different time.

Please click【Call Recording】--【New Recording】to configure:



Item	Explanation
Extension	Select an extension which need to be monitored
Call Recording Time	Always monitor or monitor in different time.
Call Recording Settings	Set inbound record and outbound record.

7 Settings before leaving office

7.1 Follow Me

If you don't want to lose any call, you can use this function.

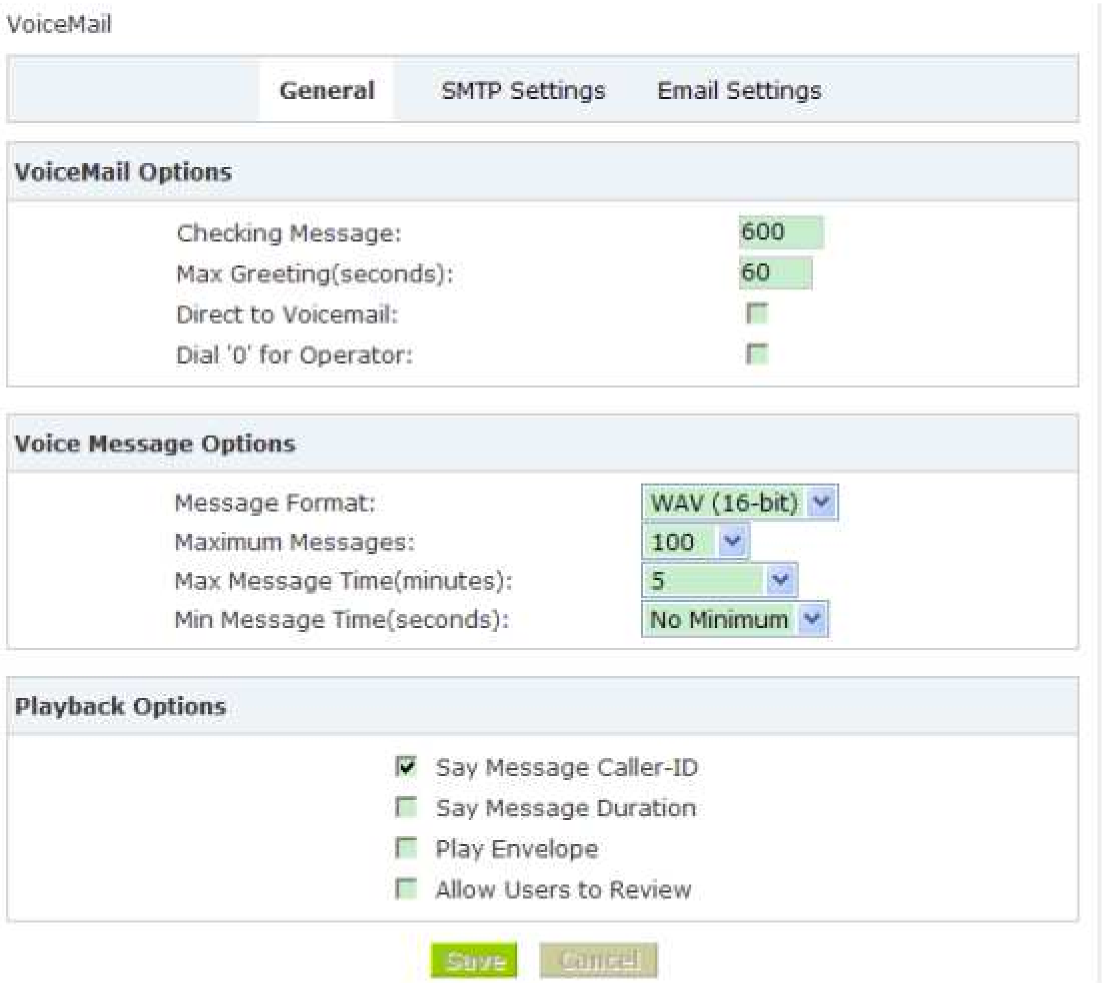
Please click【Follow Me】---【New Follow Me】

Item		Explanation
Extension		Choose an extension
Status	Always	All incoming calls will be forwarded
	Busy	Forward when extension is busy
	No answer	Forward when extension not answer
Ring lasting for(s)		Default is 20 seconds, you can define it by yourself.
Set your Follow Me number	Forward to an Internal Extension	Incoming call will be forwarded to internal extension.
	Forward to an External Extension	Incoming call will be forwarded to external number or mobile number.
Set Internal Extension		Set an internal extension to pick up the call.
Select DialPlan		Select DialPlan when forward the call to external number.
Set External Number		Set external number, like Mobile number.

7.2 VoiceMail

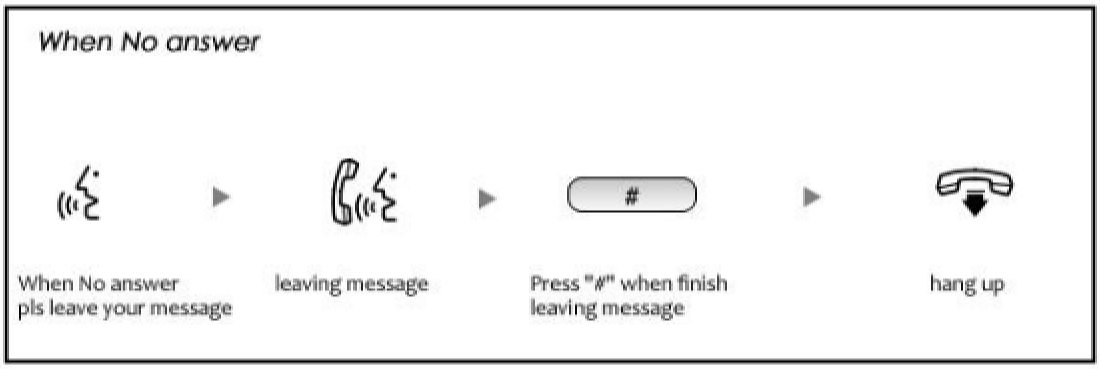
If you don't want to configure "Follow Me", you can record the message of incoming call, and email the message to your defined mailbox.

Click【Extension】---【Extension Settings】

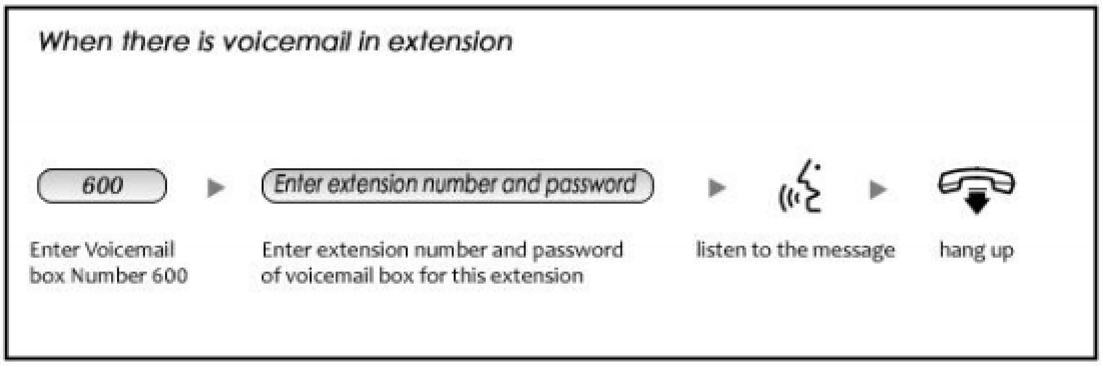


【VoiceMail】must be opened and【VM Password】must be configured before using "VoiceMail"。If no answer, when default ring time is over, the system will play and ask you to leave your message, press # to end recording. If you configured email, your voice message will be sent to your defined email.

Leave a message:



Listen to the message



Note:

1) If you would like using this function, you must write correct email address in "extension settings"

2) You need configure SMTP and Email model in【VoiceMail】, please check the details in the above chapter【VoiceMail】

8 Call Queue

8.1 Create Agent

Check agent in the 【Extension Settings】---【Advanced Options】, then assign agent and

Ring Strategy in 【Call Queue】, please learn from the following configuration interface:

Item	Explanation
Queue Number	This option defines the extension number that may be dialed to reach this Queue.
Queue Name	This option defines a name for this Queue, eg. "Sales"
Ring Strategy	RingAll -- Ring All available Agents until one answers(default). RoundRobin -- Take turns ringing each available Agent. LeastRecent -- Ring the Agent which was called least recently. FewestCalls -- Ring the Agent with the fewest completed calls. Random -- Ring a Random Agent. RRmemory --RoundRobin with Memory, and remember where it left off in the last ring pass.
Agents	All the users who is defined as Agent will be shown here. Selected agent will be a member of the current Queue.

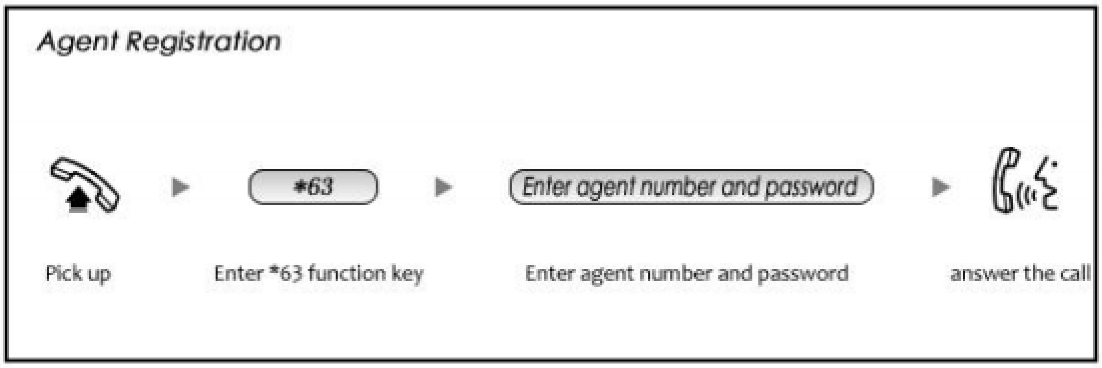
Note:Each agent needs to login to the queue using the login extension defined in Feature Codes.

Item	Explanation
Agent TimeOut(s)	This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, eg. "15"
Auto Pause	Pause an Agent if they fail to answer a call.
Wrap-Up-Time(s)	After a successful call, how many seconds needed to wait before sending another call to a potentially free agent (Default is 0, which means No Delay).
Max Wait Time(s)	The maximum number of seconds a caller can wait in a queue before being pulled out(empty for unlimited).
Max Callers	This option sets the maximum number of callers that may wait in a Queue(Default is 0, Unlimited).
Join Empty	Defining this option allows 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	Defining this option forces all callers to exit the Queue if New Callers are also not able to Enter the Queue. This option should generally be set in concert with the "Join Empty" option.
Auto Fill	Defining this option causes the Queue, when multiple calls are in it at the same time, to push them to Agents simultaneously. Thus, instead of completing one call to an Agent at a time, the Queue will complete as many calls simultaneously to the available Agents.
Report Hold Time	Check this option if you wish to report the caller's hold time to the agent member before they are connected to the caller.
Frequency(s)	How often to announce queue position and estimated holdtime(0 to Disable Announcements).
Announce Hold Time	Should we include estimated hold time in position announcements? Either yes, no, or only once; hold time will not be announced if <1 minute.
Repeat Frequency(s)	How often to announce a voice menu to the caller(0 to Disable Announcements).
Announcements Prompt	Select the 'Announcements Prompt' from IVR Prompts

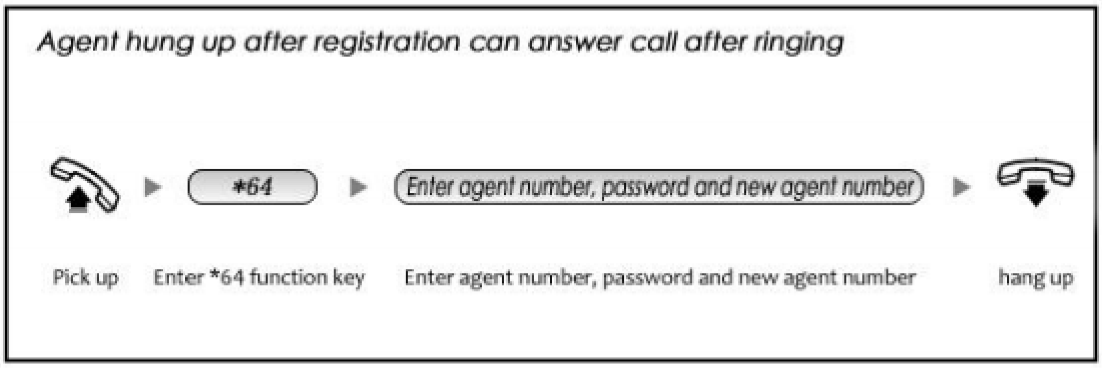
8.2 Agent Registration

You need register for using after creating agents.

Agent Registration when hook off



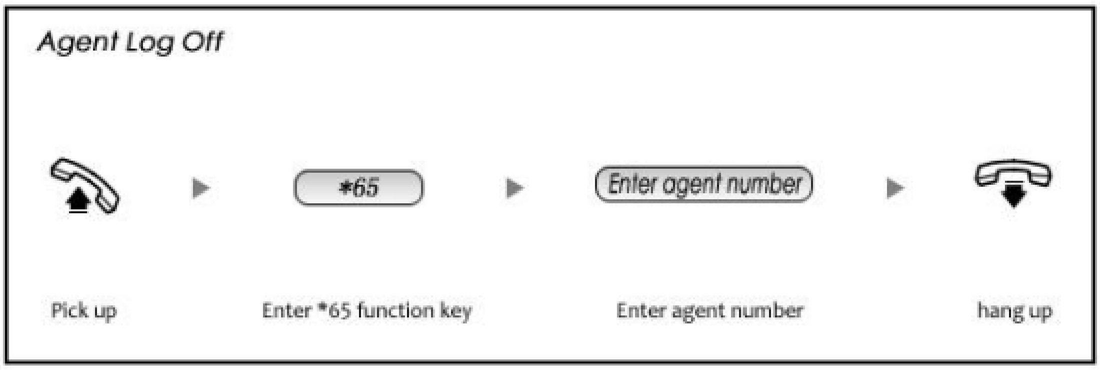
Agent Registration when hook on



8.3 Agent Log Off

If agent would leave and log off, none of agent will answer calls then.

Agent Log Off:



Advanced

1 Options

Options

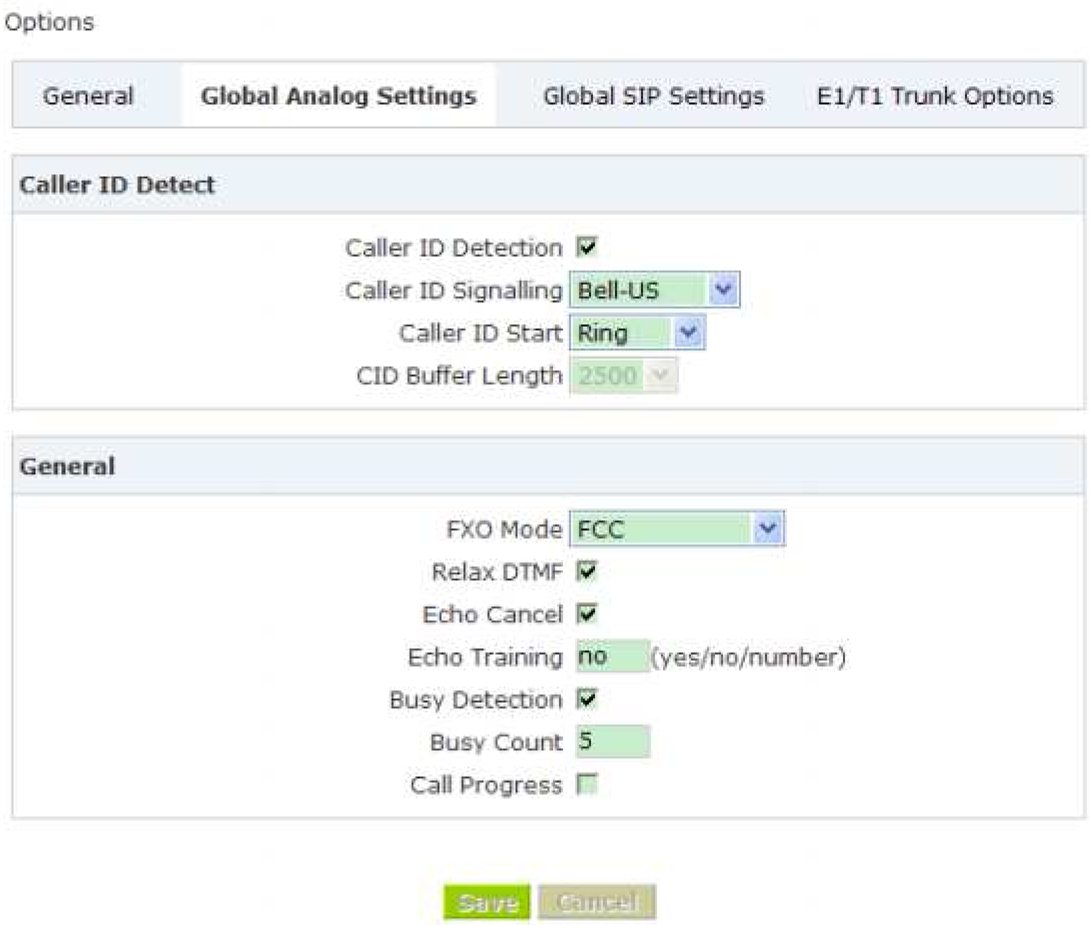
Options Include local extension settings and new extension default settings.

Click【Option】to display the dialog

Item	Explanation
Local Extensions	Set up the digit of local extensions
Operator Extension	Set up Operator Extension.
Global Ring Time Set(s)	Set Ring Time for each extension.
Enable Transfer	Enable transfer feature key.
Enable Music On Ringback	Enable music on ringback.
Allow multiple extensions to be assigned to one analog phone	Allow multiple extensions to be assigned to one analog phone.
Allow extensions to be Alpha Numeric (SIP/IAX users)	If extension is Alpha, outside line can't call in, but extension can call out.
SIP	Enable this option if the User or Phone is using SIP or is a SIP device.
IAX2	Enable this option if the User or Phone is using IAX2 or is an IAX2 device.
Agent	Enable this option if the User or Phone is an Call Agent.
NAT	Enable this option if the User or Phone is located behind a NAT (Network Address Translation) enabled gateway.
VM Password	Voicemail Password for this user, eg: "1234".
Delete VMessage	Voicemail will not be checkable by phone if you chose this option. Messages will be sent by e-mail only. Note:you must configure SMTP server for this functionality.

Global Analog Settings

Click 【Options】---【Global Analog Settings】to see the following diagram:


Options
General
Global Analog Settings
Global SIP Settings
E1/T1 Trunk Options
Caller ID Detect
Caller ID Detection
Caller ID Signalling
Bell-US
Caller ID Start
Ring
CID Buffer Length
2500
General
FXO Mode
FCC
Relax DTMF
Echo Cancel
Echo Training
no
(yes/no/number)
Busy Detection
Busy Count
5
Call Progress
Save
Cancel

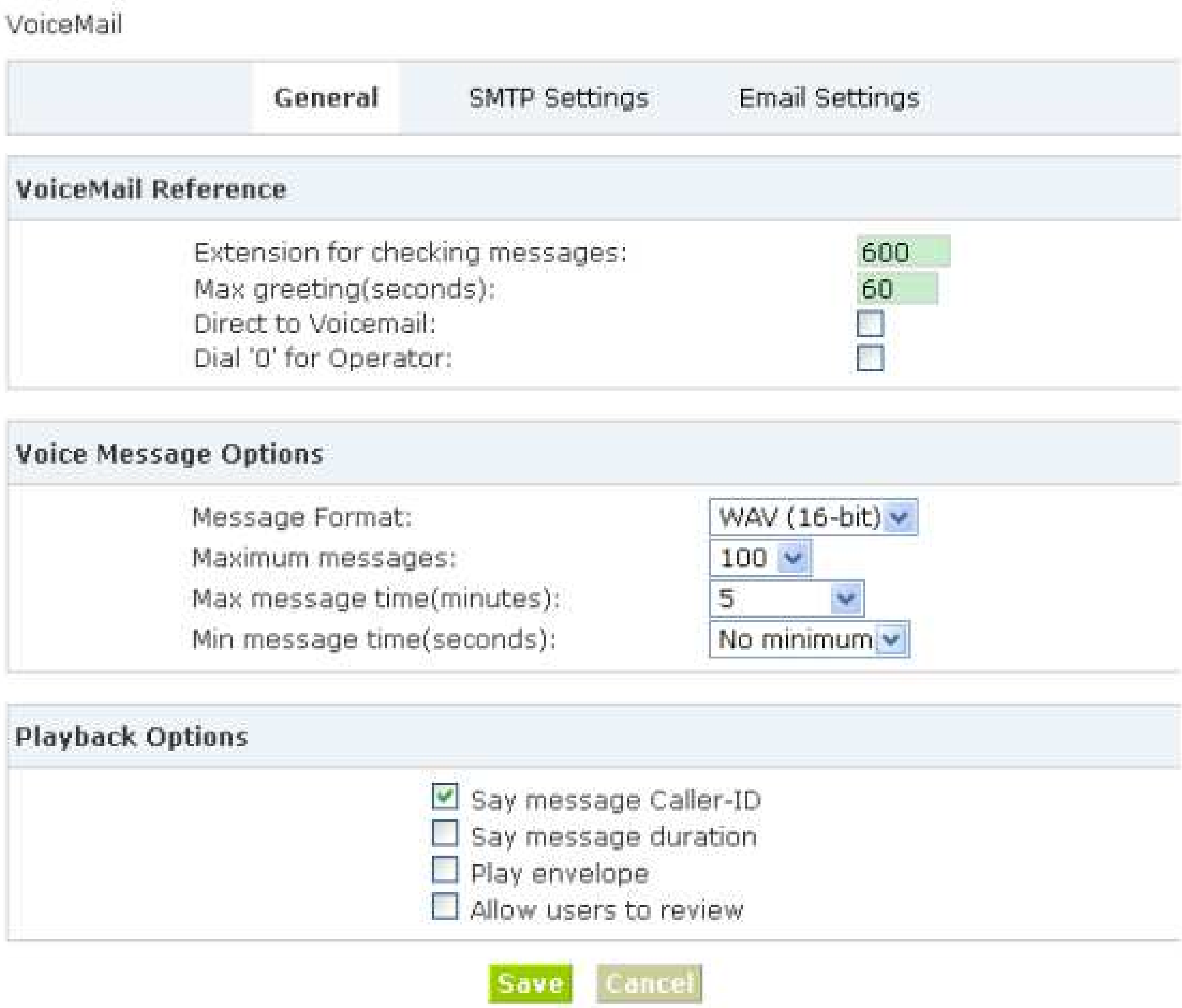
Item	Explaination
Caller ID Detection	For FXO trunk lines,this option causes PBX to look for Caller ID on incoming calls
Caller ID Signalling	This option allows you to choose the type of Caller ID signalling to use. Bell-US-- Used in the United States; DTMF-- Used for CallerID under DTMF mode.(eg: Denmark, Sweden and Netherlands etc); V23-- used in the UK; V23-Japan-- used in Japan
Caller ID Start	This option allows you to define the start of a Caller ID signal: Ring-- to start when a ring is received. Polarity-- to start when a polarity reversal is started.
CID Buffer Length	Default CID Buffer Length
FXO Mode	Select FXO Mode here
Relax DTMF	If you met trouble with DTMF detection, you can relax the DTMF detection.
Echo Cancel	Enable/Disable the Echo Cancel function.
Echo Training	Enabling echo training will cause the PBX system to briefly mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo. Value may be "yes", "no", or a number of milliseconds to delay before training (default = 400)
Busy Detection	Used for detecting far end hangup or a busy signal.
Busy Count	If Busy Detection is enabled,it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results can be achieved if set to 6 or even 8. Mind that the higher the number, the more time that will be needed to hangup a channel, but lower the probability that a false detection may occur.
Call Progress	If turned on, call progress attempts to determine answer, busy, and ringing on phone lines.

Global SIP Settings

【Global SIP Settings】 is appropriated for operating by professional engineer or technician, if you need modification, please contact with our technician support.

2 VoiceMail

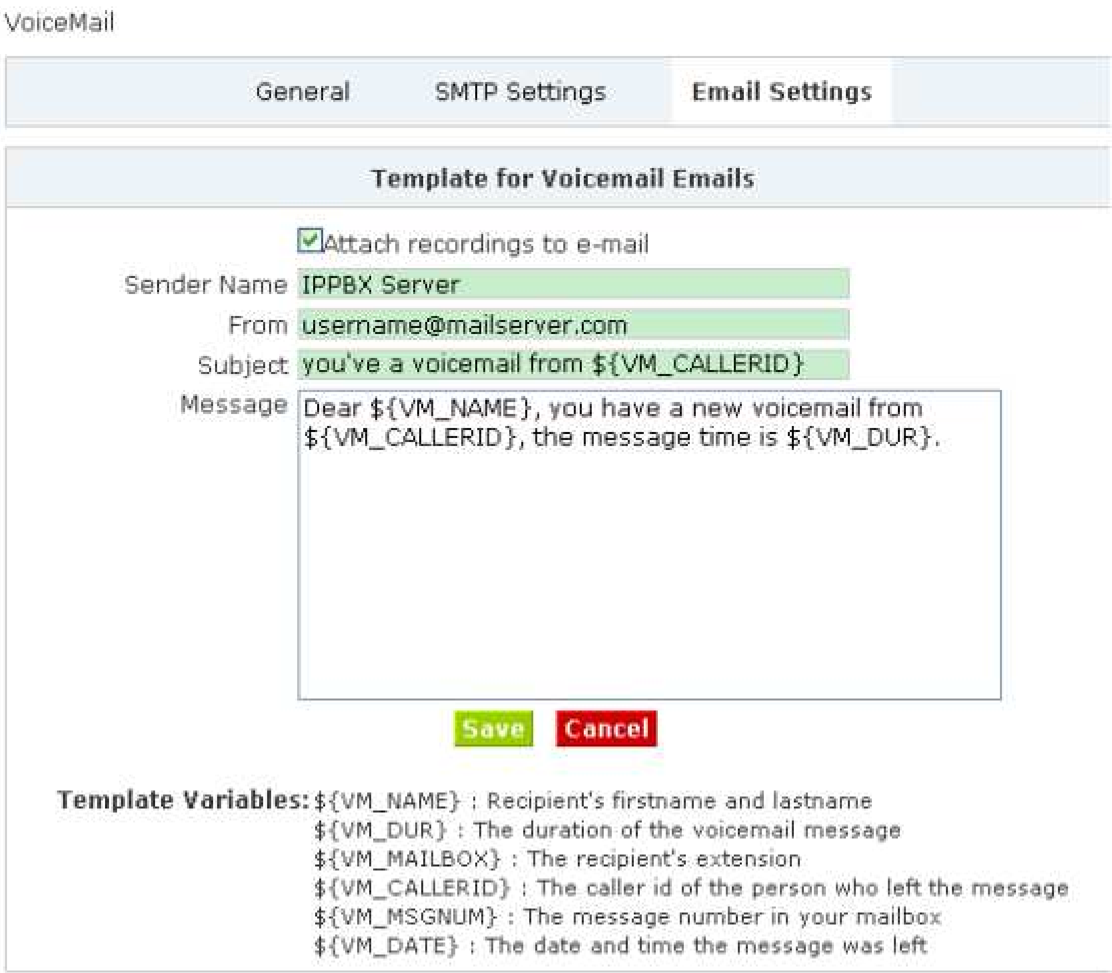
Details configuration on VoiceMail: VoiceMail Reference/ Voice Message Options/ Playback Options. If you need send message by mail to your defined mailbox, you must configure SMTP and Email model. Click【Voicemail】to display the dialog as below:


VoiceMail
General
SMTP Settings
Email Settings
VoiceMail Reference
Extension for checking messages:
600
Max greeting(seconds):
60
Direct to Voicemail:
Dial '0' for Operator:
Voice Message Options
Message Format:
WAV (16-bit)
Maximum messages:
100
Max message time(minutes):
5
Min message time(seconds):
No minimum
Playback Options
Say message Caller-ID
Say message duration
Play envelope
Allow users to review
Save
Cancel

Item	Explanation
Extension for checking messages	The number that users call in order to access their voicemail accounts, the default number is "600".
Max greeting(seconds)	Defining this option to set a maximum time for the greeting message.
Direct to Voicemail	Defining this option to go to voicemail box directly.
Dial "0" for Operator	Callers entering the voicemail application can leave for Operator by dialing "0".
Message Format	Choose the format of the voicemail messages in this selection box.
Maximum Messages	Choose the maximum number of messages in this selection box.
Maximum message time (min)	Choose the maximum duration of a voicemail message. Message recording will be stopped when it's timeout.
Minimum message time (s)	Choose the minimum duration of a voicemail message in this selection box. Message time below this threshold will be deleted automatically.
Say message Caller-ID	Choose this option to play Caller's ID before voicemail message is played.
Say message duration	Choose this option to play the duration of message before the voicemail message is played.
Play envelope	Choose this option to play envelop (including date, time and caller ID).
Allow users to review	Choosing this option, the caller leaving the voicemail can review their recorded message before it's submitted.

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 your IP PBX may connect to. eg: mail.yourcompany.com
Port	The port number which the SMTP server running is generally port 25. If SSL is encrypted, please use port 465 instead.
SSL/TSL	Enable use 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.
Username	Input username of your email box.
Password	Input password of your email box.

Email Settings



Item	Explanation
Attach recordings to e-mail	This option defines whether or not voicemails are sent to the Users' e-mail addresses as attachments.
Sender Name	Display the Sender name when you receive a voicemail.
From	Sender's email address
Subject	Subject of the mail
Message	The message pattern

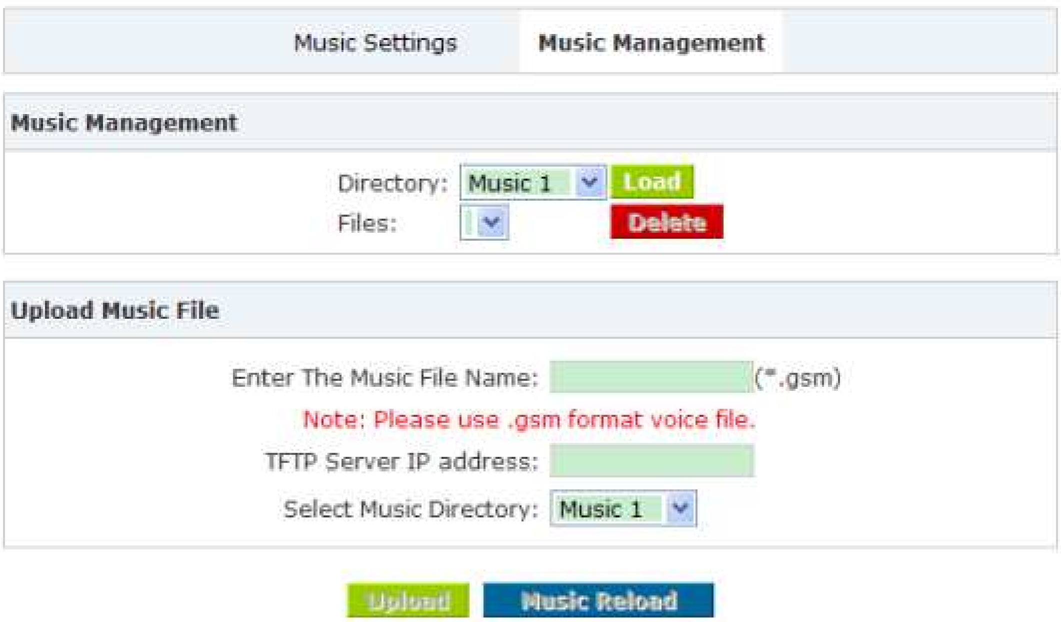
3 Music Settings

Management for music on hold, music on ringback, music on call queue.

Click【Music Settings】 to display the dialog as below:

Please define different music file for different music folders.

Music Management:



Item	Explanation
Directory	Load music in the music file.
Files	Display music in the music file, or you can delete it.
Enter The Music File Name	Input music file name which you want to upload.(GSM/ WAV format, If it's WAV, it must be accord with PCM 16 bits, 8000HZ format)
TFTP Server IP address	Please enter your TFTP server IP address.
Select Music Directory	Select directory where the uploaded music file will be saved.

4 DISA

A trunk call into the PBX, and call to another trunk through outbound route of the PBX. Eg: This trunk can make international call, you are out of the office and want to contact with your customer in foreign country, now you can dial DISA number, after PIN authentication, you are connected to your customer, and you can speak to your customer now.

Click【DISA】---【New DISA】to display the dialog as below:

Item	Explanation
Name	Give this DISA a brief name to help you identify it.
PIN	The user will be prompted for this number
Response Timeout(s)	The maximum amount of time it will wait before hanging up if the user has dialed an incomplete or invalid number. Default is10 seconds.
Digit Timeout(s)	The maximum amount of time permitted between digits when the user is typing in an extension. Default is 5 seconds.
Extension for this DISA (Optional)	If you want this DISA to be accessible by dialing an extension, you can define an extension number for this DISA.
Select DialPlan	Set the DialPlan that calls will originate from.

5 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.

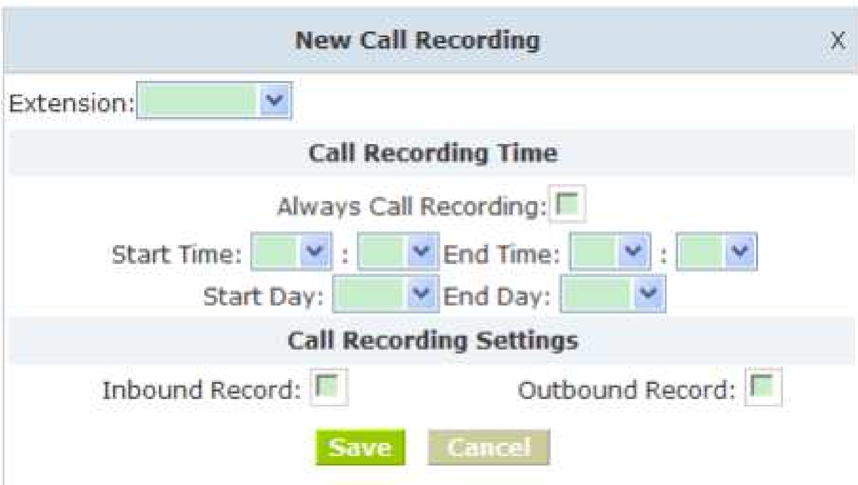
Click【Paging And Intercom】---【Add Paging Group】to display the dialog as below:

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) in 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".

6 Call Recording

Call Recording is used for recording the defined extensions.

Click【Call Recording】---【New Call Recording】to display the dialog as below：



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

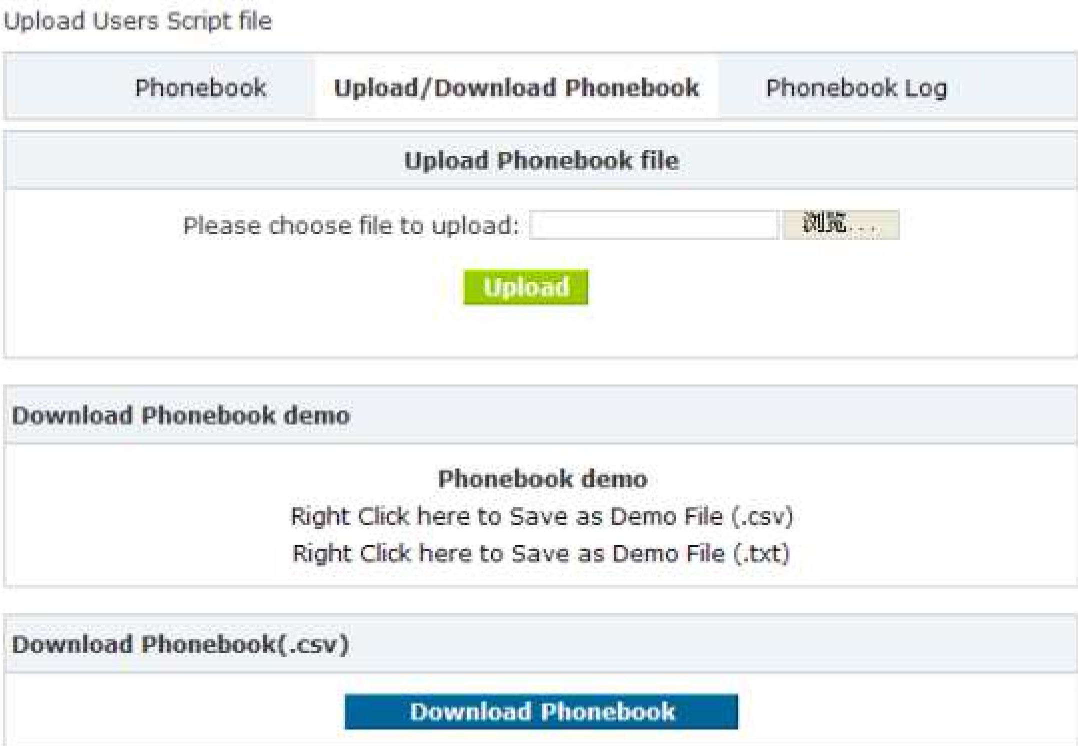
7 Phone Book

If incoming call was matched with the number in the phone book, the incoming call will display the name of matched number.

Click【Phone Book】to display the dialog as below:

- Name   Add contact's name, Alphabetic or numeric only.
- Number   Add contact's number, international phone number is supported.

SNR-VX50 IP PBX also support "Batch Add Users", Click【Advanced】->【Upload/Download Phonebook】to display the following diagram:



Download a phonebook demo from【Download Phonebook demo】, add and save information refer to the demo content, choose the file what you want to uploaded from【Upload Phonebook file】You can download the phonebook file from【Download Phonebook(.csv)】

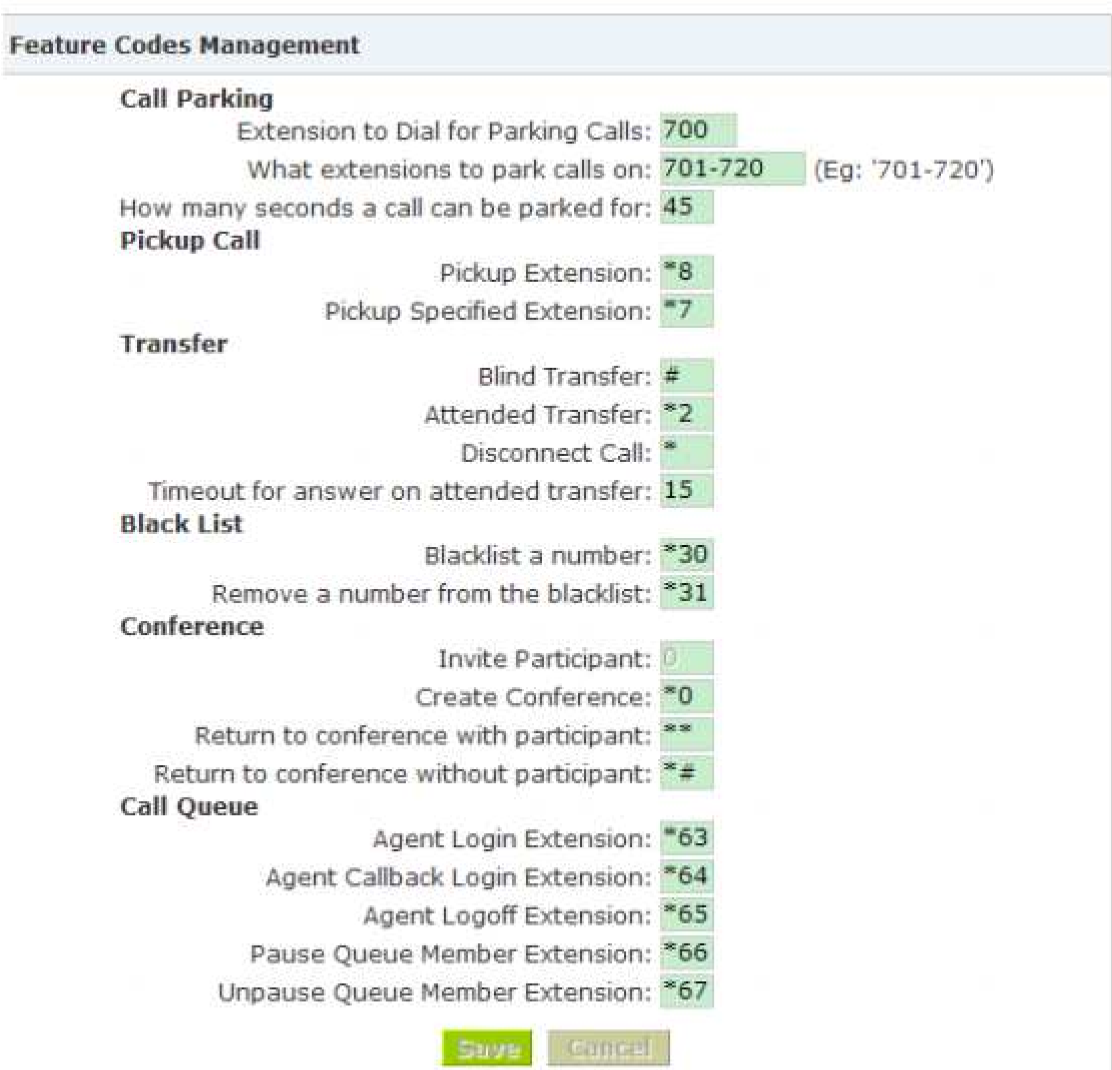
8 PIN Set

PIN Set will distribute one PIN Code to different extension user, if you selected PIN Set on the Dial rule page in Outbound menu, the extension user who has the PIN code can dial long distance call. Click【Pin Set】to show the dialog as below:

- PIN Set Name Set the PIN Sets Name
- PIN List Enter a list of one or more PINs. One PIN per line.

9 Feature Codes

Click【 Feature Codes】to display the dialog as below, you can define relevant parameter.



Item	Explanation
Extension to Dial for Parking Calls	Set Call Parking number.
What extensions to park calls on	What extensions to park calls on, eg: (701-720)
How many seconds a call can be parked for	Set the call time by second, if it's time out, system will call the previous extension again.
Pickup Extension	Set Pickup Extension.
Pickup Specified Extension	Set Pickup Specified Extension, default: dial *7+extension to pickup the extension.
Blind Transfer	Allow unattended or blind transfers. It works like this: While on a conversation with A, you dial the blind transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number(B's number) and A is put through to B immediately. Your line is off. The caller ID displayed to B is exactly the same as the caller ID presented to you.
Attended Transfer	Allow attended transfer or supervised transfer. It works like this: While on conversation with A, you dial the Attended Transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number(B's number) and talk with B to introduce the call, then you can hang up and A will be connected with the B. In case B does not want to answer the call, he/she simply hangs up and you will be back to your original conversation.
Disconnect Call	Disconnect the current transfer call(for Attended transfer).
Timeout for answer on attended transfer	Set the answer timeout value.
Blacklist a number	Add a black list number.
Remove a number from the black list	Remove a black list number.
Invite Participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dialtone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself.
Create Conference	While you speak with another party you can press *0, you and the callee are immediately transferred to conference.

Return to conference with participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dialtone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself
Return to conference without participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dialtone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself.
Agent Login Extension	Logs the current caller into the queue as a call agent. Once logged in, the agent can take calls with the phone off-hook; each call is preceded by a warning tone. Calls are ended by pressing the "*" key.
Agent Callback Login Extension	Extension to be dialed for the Agents to Login to the Specific Queue. Same as Agent Login Extension, except you do not have to remain on the line.
Agent Logoff Extension	Agent logoff from the queue.
Pause Queue Member Extension	'Pauses' a queue member. so that the member can not receive calls.
Unpause Queue Member Extension	'Unpause' a queue member who is 'paused' previously. so that the member can receive calls again.

10 Phone Provisioning

When you need many IP Phone for using, please record the MAC, extension number, and username of each phone according to the format (please take reference of the auto provision script file model for details) , then, import the format file, once the phone is connected to the local network, it will get the extension number and password automatically.

There are two operation methods to fulfill this function, please see details as below:

Enable DHCP service

Click【System】->【Network Advanced】, enable DHCP Server in the dialog as below:

Method 1：

Click【 Extension】->【Create New User】， select the relative IP Phone manufacture, and input relative MAC in the part of Auto Provision, Save and Activate.

Method 2：

Click【Phone Provisioning】 to download auto provision script file model, this script file

model support csv and txt format, Mac, Extension, Fullname must be filled,<password>,<IP Phone version> could be optional. Save it in your local PC after you fill based on the model format, select the relative manufacture on this page and upload.

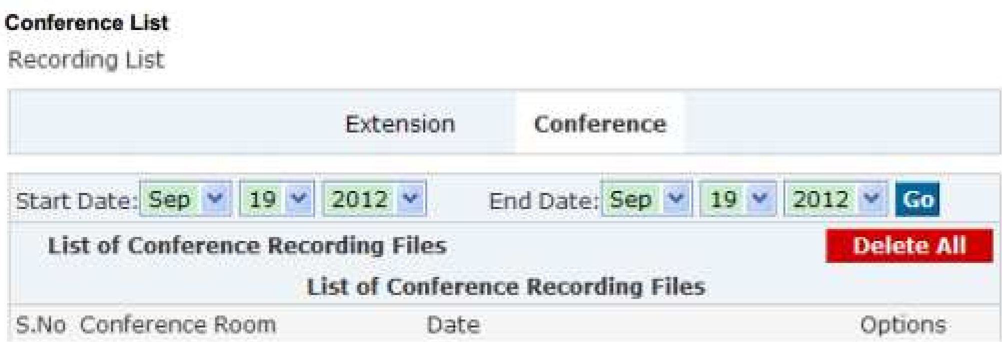
Status

This chapter will introduce you the status of record list, call logs, system info, register status etc.

1 Recording List

Check the record list of defined extension or conference, you can delete the record list.

Click【Recording List】---【Extension】and【Conference】will be displayed as below:



2 Call Logs

Check call logs of extension by caller ID or callee ID. Click【Call Logs】 to display the dialog as below:

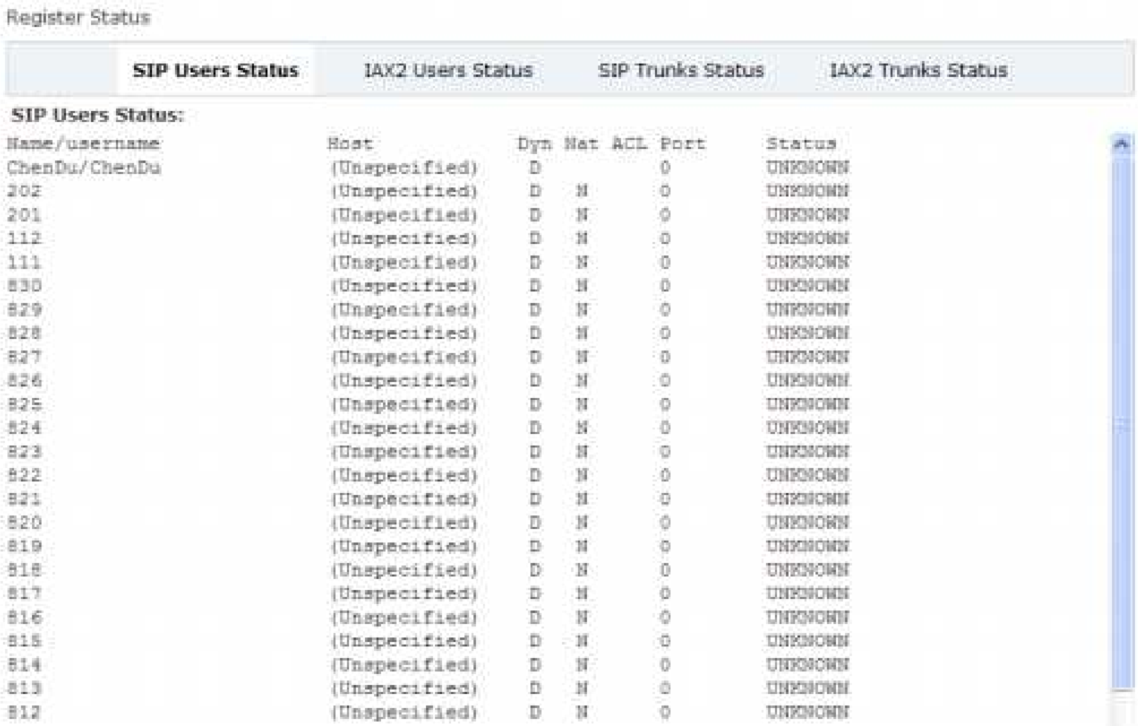
Call Logs Interface

Note:

Duration in the call logs is not real charged duration, if you need billing, PSTN must support polarity reversal function, meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for the IP PBX.

3 Register Status

Check SIP/ IAX2 User, and SIP/IAX2 Trunk status. Click【Register Status】to display the dialog as below:



4 System Info

Check OS version, firmware version and memory, etc from here.

Click【System Info】to display the dialog as below:

System

This chapter will introduce you how to configure the system of SNR IP PBX.

1 Network And Country

Configure WAN/ LAN IP, and tone zone.

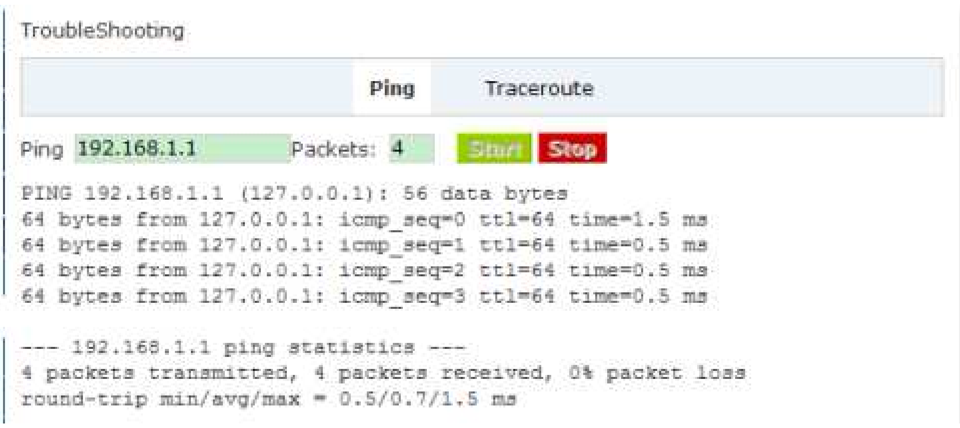
Click【Network And Country】to display the dialog as below:

IP Assign	Static, DHCP, PPPoE are supported
HTTP Port	Set the http server port, default is 9999
Remote Administration	Enable/ Disable Access GUI through WAN port.
Tone Zone Settings	Define the tone zone for home country or place.

2 TroubleShooting

You can ping other network device through SNR IP PBX and track network route by command "Traceroute" .

Click【TroubleShooting】to display the dialog as below:



3 Network Advanced

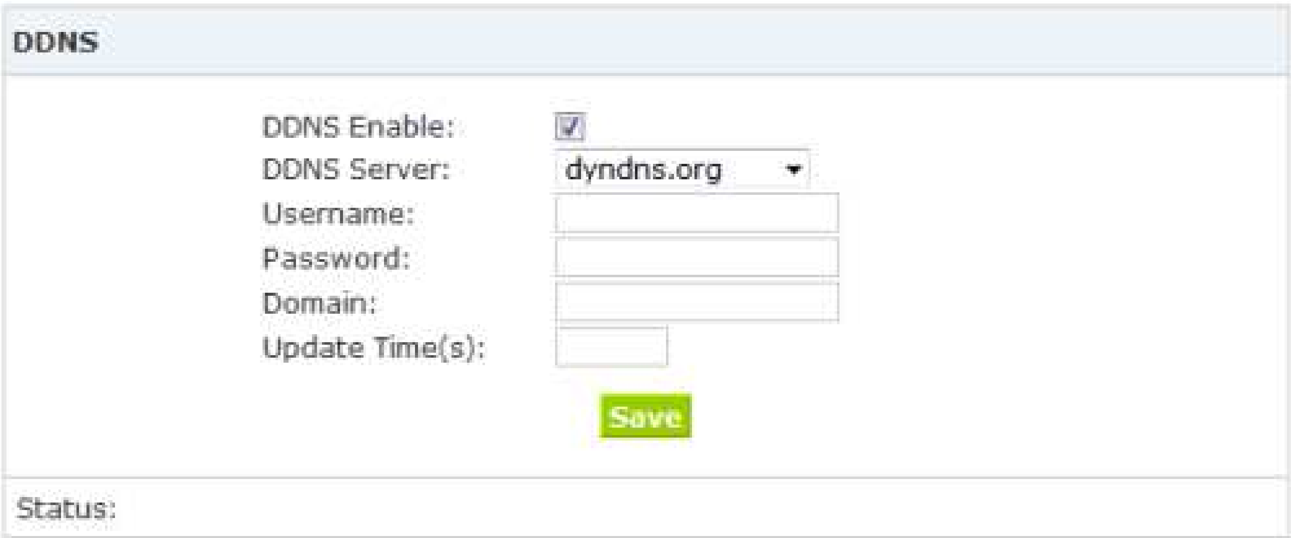
DHCP Server Settings

VX50 Series IP PBX support DHCP , Click【Network Advanced】->【DHCP Server Settings】 to show the following diagram:

DDNS Settings

After configure DDNS, you can visit by domain remotely.

Click【DDNS Settings】to display the dialog as below:



VPN Settings:

A virtual private network (VPN) is a method of computer networking---typically using the public internet---that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. VX50 Series IP PBX support N2N and L2TP.

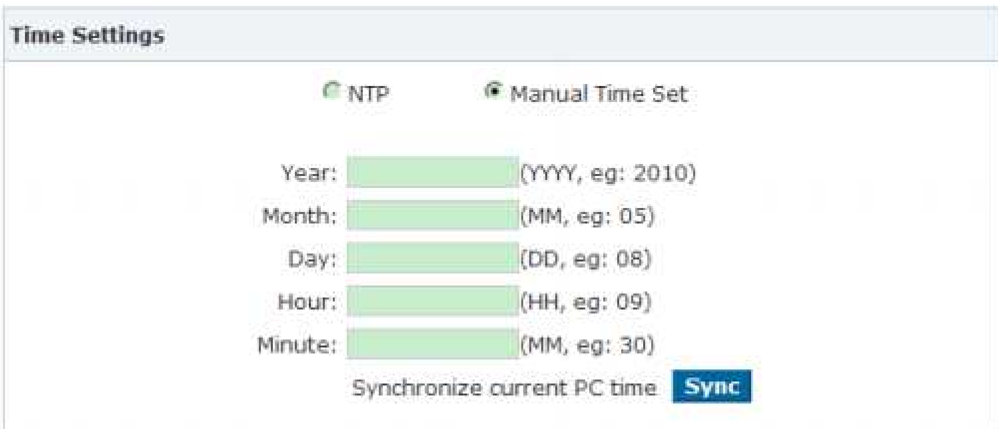
Note:

1) DDNS supports the domain provided by Dyndns.org/ No-ip.com only.

2) VPN supports N2N/L2TP only.

4 Time Settings

Click【Time Settings】to display the diagram as below:

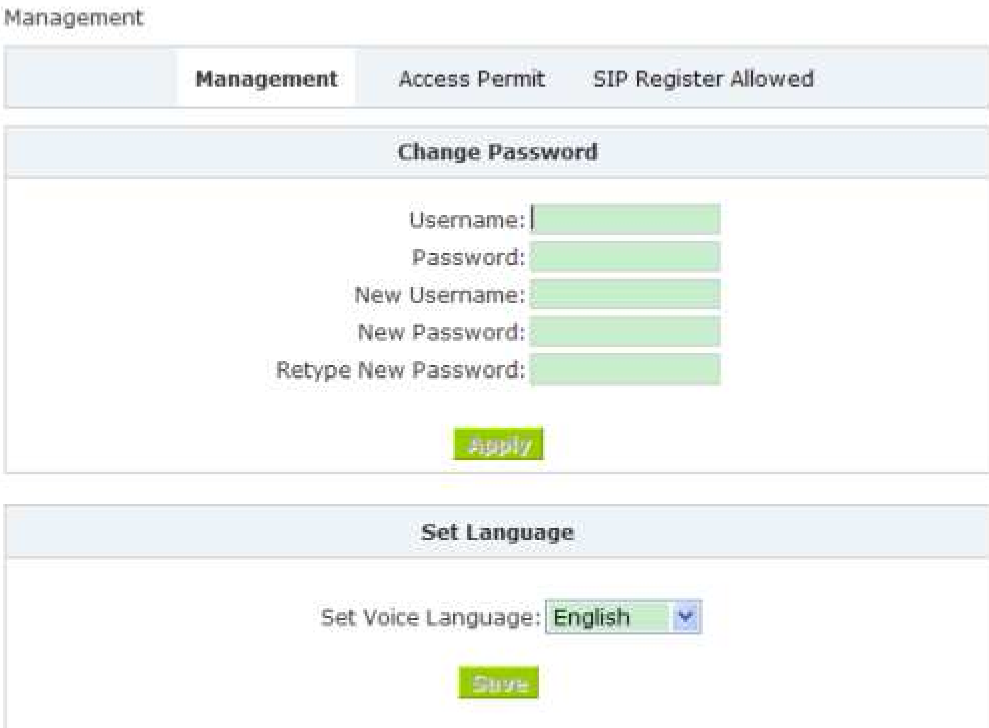


Item	Explanation
NTP Server	Specify the NTP server that you wish to use. You may type either the domain name or the IP address of the server, and it may be either remote or local. The default server is pool.ntp.org. Be aware that the PBX needs to be able to connect to a NTP server for perfect function.
Time Zone	Select your time zone so that the system will set time based on the time zone.
Synchronize with current PC time	Click the button to synchronize the PBX time with the current PC time.

5 Management

Management

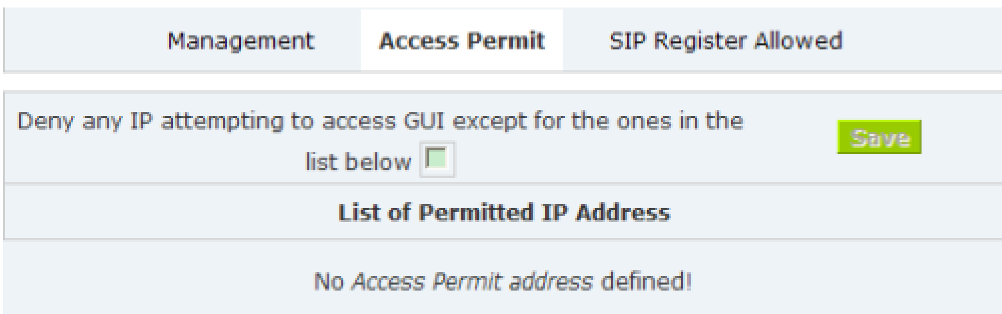
Click【Management】to display the diagram as below:



- Change Password: You can change the password of admin here (default password is admin)
- Set Language: Set voice language of the system. And you can set the SIP & Analog channel here by clicking "Show Advanced Options"

Access Permit

Click【Access Permit】to display the diagram as below:

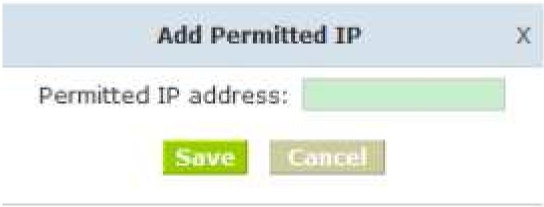


Note:

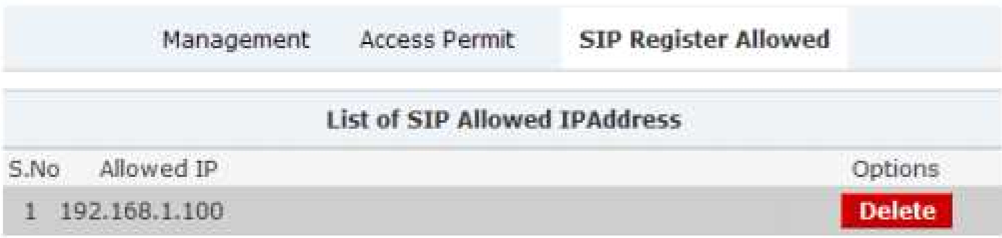
After you added a permitted IP, you can only login the system by this IP, other IP address isn't effective to login the system.

SIP Registered Allowed

Click【SIP Registered Allowed】---【Add Permitted IP】to define the allowed SIP user. Input the permitted IP address---IP address and network restriction.eg: "192.168.1.77" or "192.168.10.0/255.255.255.0"



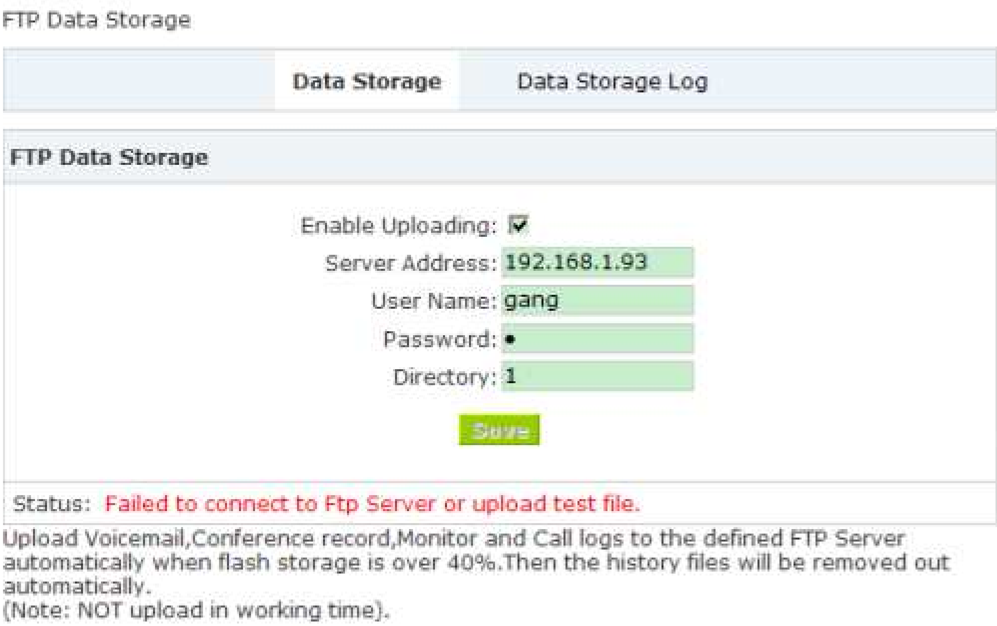
In the following diagram, 192.168.1.100 is the allowed IP registered by SIP.



6 Data Storage

Upload the voicemail, call recording, conference, call logs, etc to the defined FTP server for storage.

Click【Data Storage】to display the diagram as below:



Item	Explanation
Enable Uploading	Enable periodical FTP uploading.
Server Address	Set FTP Server address(IP address or Domain).
User Name	FTP account name.
Password	FTP account password.
Directory	Define a directory on the FTP server.

Note:

1) Upload Voicemail, Conference record, Monitor and Call logs to the defined FTP Server automatically when flash storage is over 40%. Then the history files will be removed out automatically.

2) NOT upload in working time by default.

7 Backup

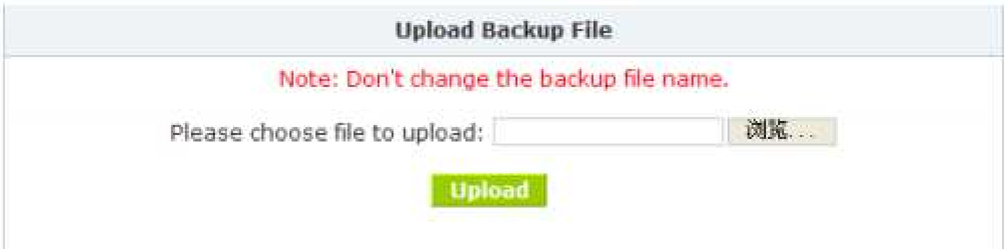
Backup

Backup all the settings. Click【Backup】to display the diagram as below:

- Restore　Restore your selected backup file to system.
- Delete　Delete your selected backup file.
- Download your selected backup file to your PC. (Note: Please don't change the backup file name.)

Upload Backup File

Click【Upload Backup File】to display the diagram as below:

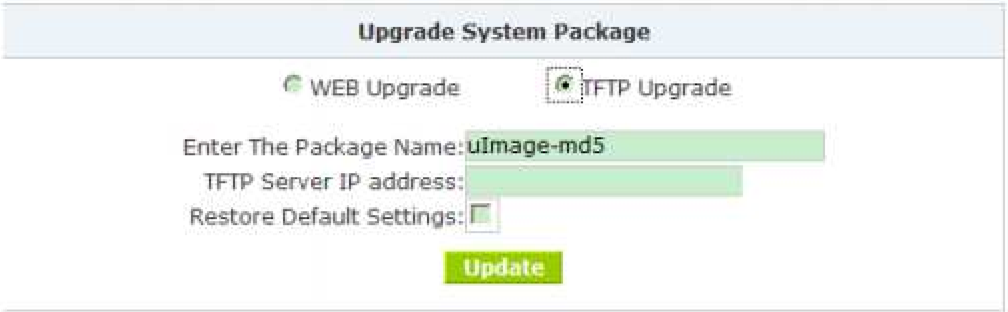


8 Upgrade

- Click【WEB Upgrade】to upgrade as below

Choose the file to upload. If you enabled Restore Default Settings, the system will be restored to default after upgrading.

- Click【TFTP Upgrade】to upgrade as below:



Extract the downloaded firmware package which includes one TFTP server and one upgrading file.

Run TFTP server

Go into the "update" page, and upload firmware;

Enter the package name;

Enter TFTP Server IP address,

Click Update button to finish upgrading system package after entering the TFTP Server IP.

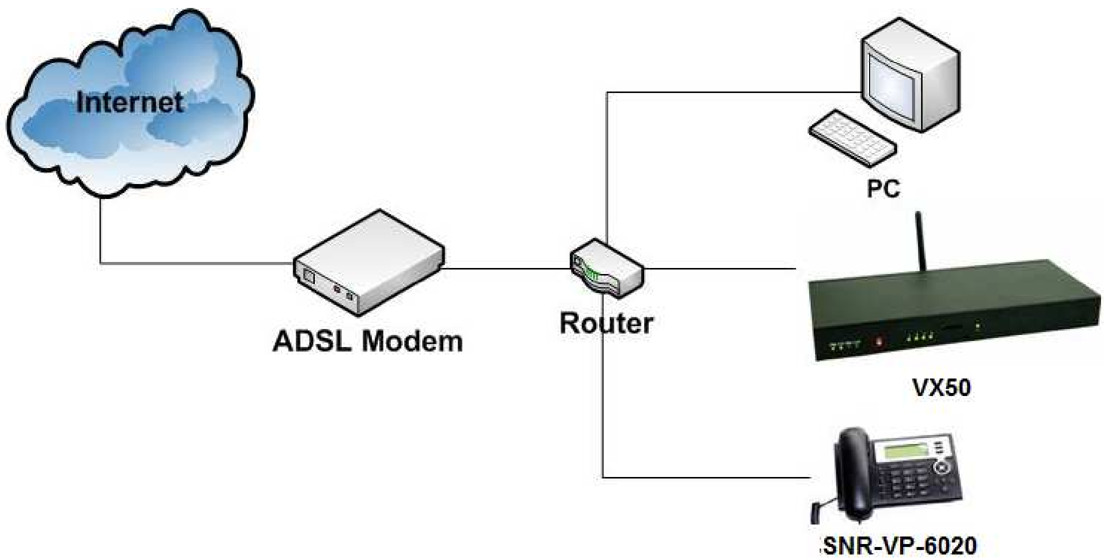
Then system will reboot automatically.

Operating Instruction

1 How to connect the SNR-VX50 IP PBX to the Internet

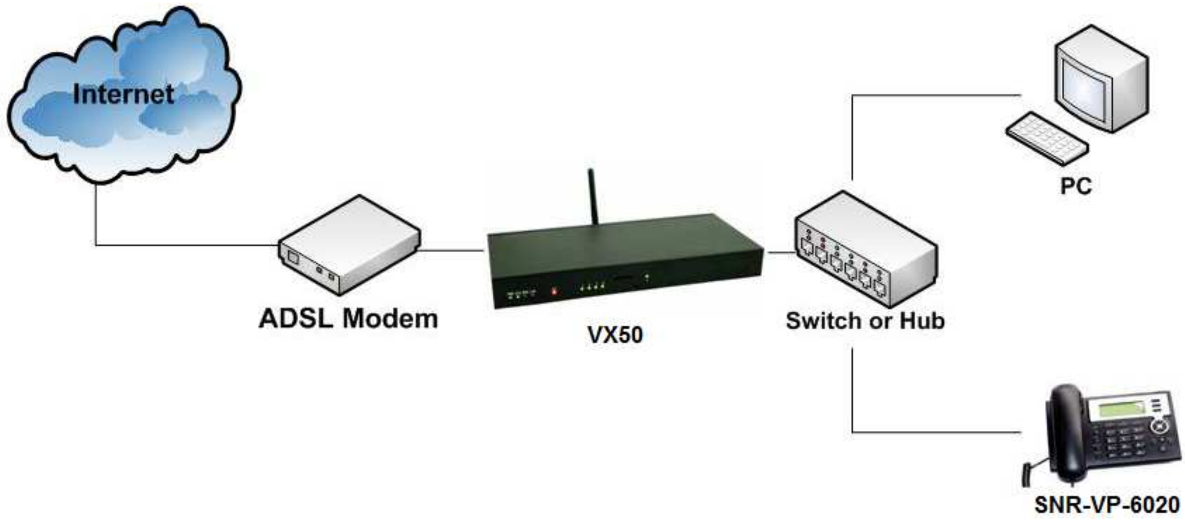
1.1 IP PBX behind the Router

If your office access the public network through router, you can put the IP PBX behind the router. You should connect the WAN port of the IP PBX to the LAN ports of the router, and you can also connect HUB or Switch to the LAN port of the IP PBX to enable some PC or IP Phone to access the public network..



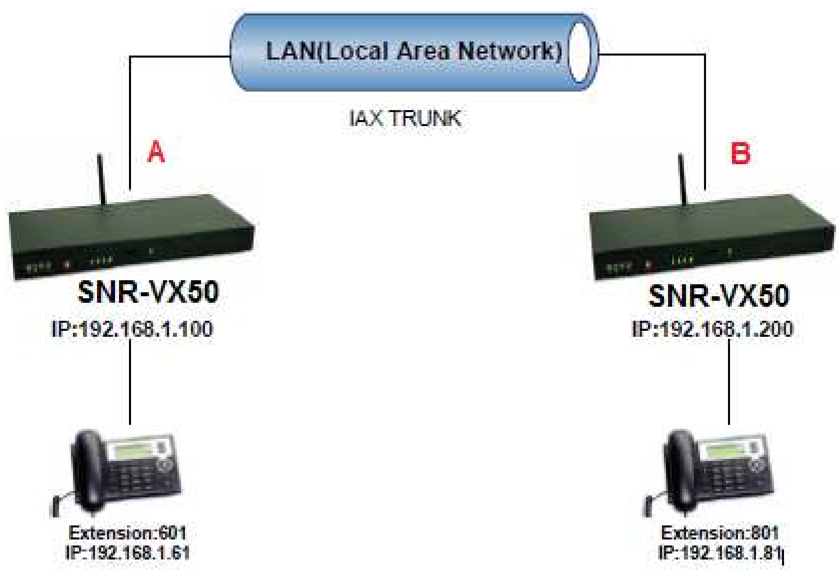
1.2 IP PBX behind the Modem

If you have the public IP and want to enable the IP PBX access the public network directly without router, then you should connect the Wan port of the IP PBX to the public network and connect HUB or Switch to the LAN ports of the IP PBX to enable your PC access the public network.(If you want to access the public network through Modem, then you should use the PPPoE function of the IP PBX and make the IP PBX dial-up to connect to the public network)



2 How to combine two SNR-VX50 IP PBX in the same network

We start combining two IP PBX in the same network and then try to expand to different network. Below is the structure of how to combine two IP PBX in the same LAN:



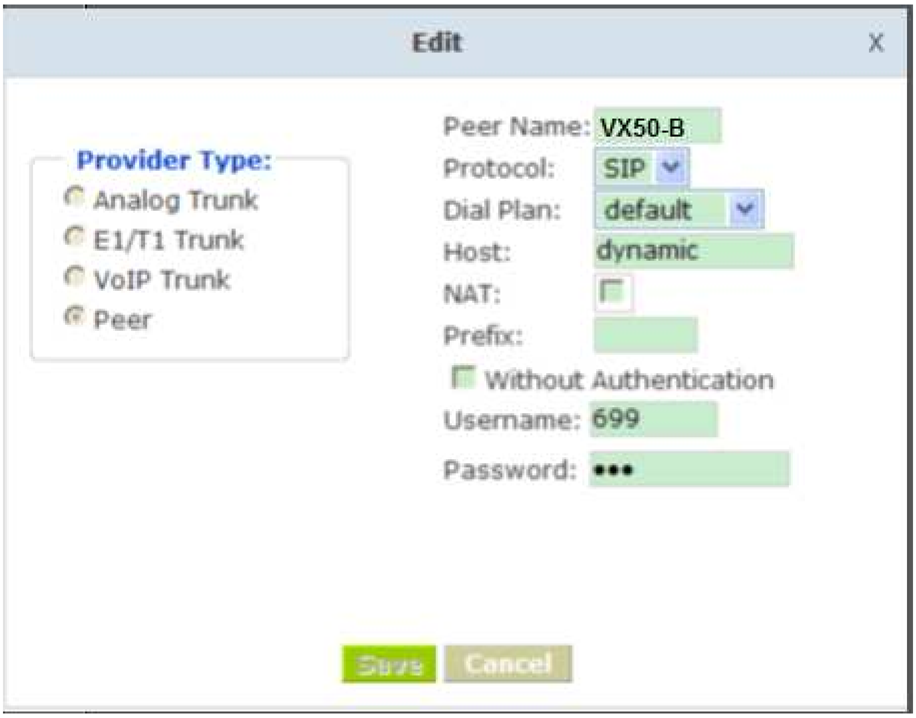
Register the SNR-VX50-A as an peer in SNR-VX50-B(via SIP trunk),so the extensions in SNR-VX50-A can make calls to SNR-VX50-B's extensions via this "special" trunk. In above structure:

1. ZP302A registers to SNR-VX50-A as extension 601.
2. ZP302B registers to SNR-VX50-B as extension 801.
3. All the extensions under SNR-VX50-A are in the format 6XX.
4. All the extensions under SNR-VX50-B are in the format 8XX
5. Extensions under SNR-VX50-A can make calls to extension under SNR-VX50-B with format 8XX.

6. Extensions under SNR-VX50-B can make calls to extension underSNR-VX50-A with format 6XX.

Step 1: Set up a peer 699 in SNR-VX50-A

In the page Trunks --> Add a Trunk



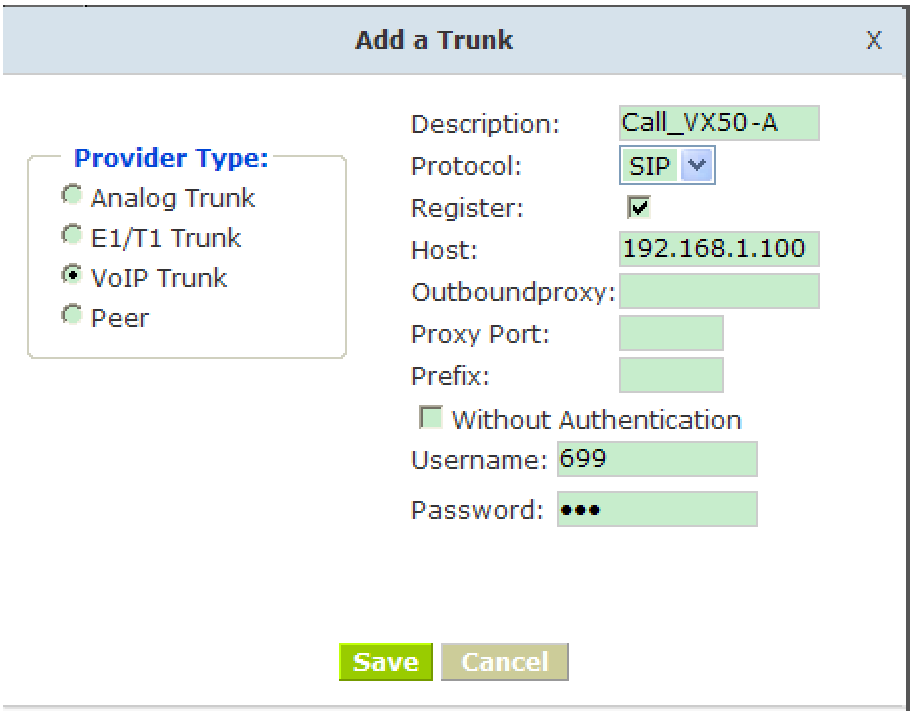
Peer Name: SNR-VX50B ;

Peer Username: 699 Account of this

Peer Password: 699 SIP Log on password

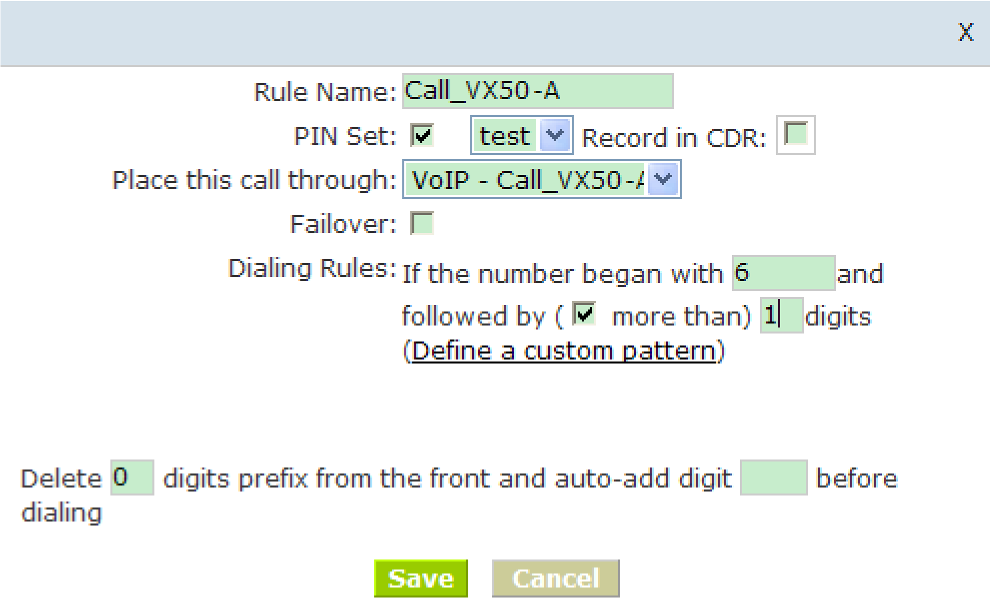
Advance Options: Select SIP protocol

Step 2: Set up an SIP trunk in SNR-VX50-B to connect to SNR-VX50-A via this SNR-VX50B Peer. In the page Trunks--> Add a Trunk

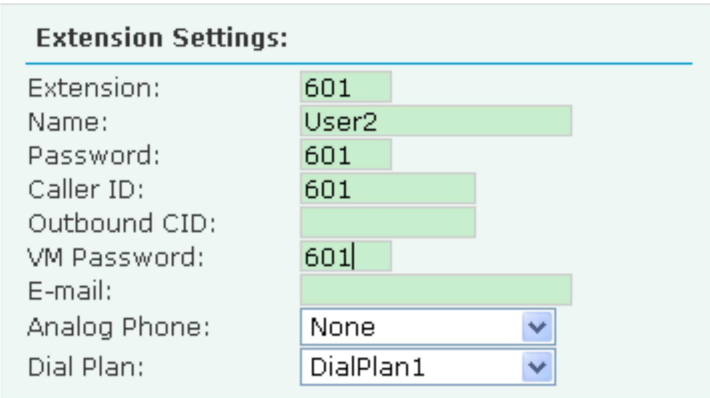


Step 3: Set Dial Rule in SNR-VX50-B, all calls starting with 6 will be sent to SNR-VX50-A.

In the page: Outbound Routes --> Add a Dial Rule



Step 4: Set the user 601 and Dial Plan in SNR-VX50-A. In the page: Extensions → Dial Plan



Activate the change and apply the test:

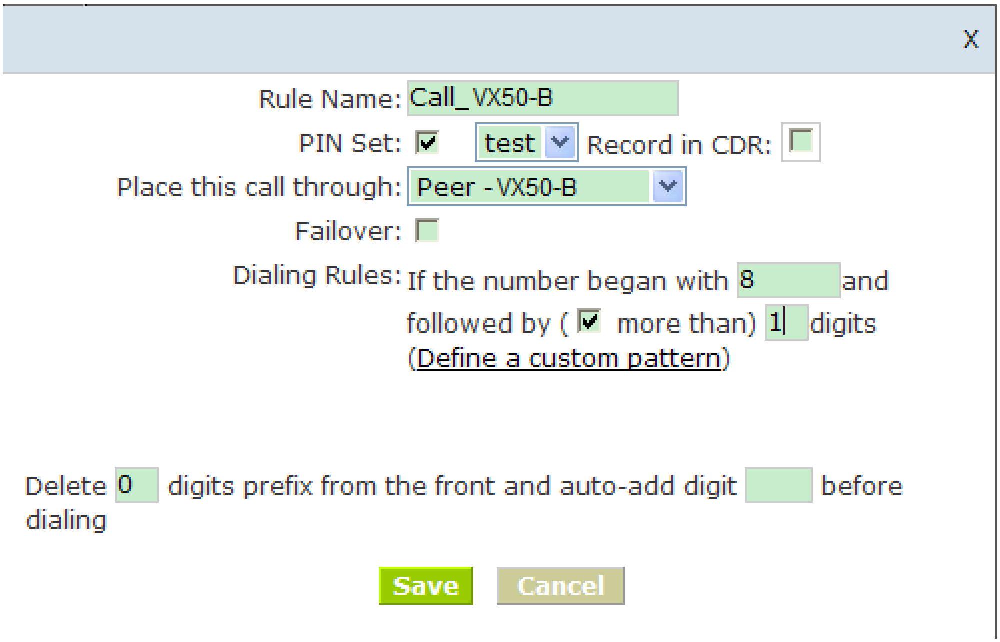
1. Register an IP phone B to SNR-VX50-B with 801 extension.

2. Register an IP phone A to SNR-VX50-A with 601 extension.

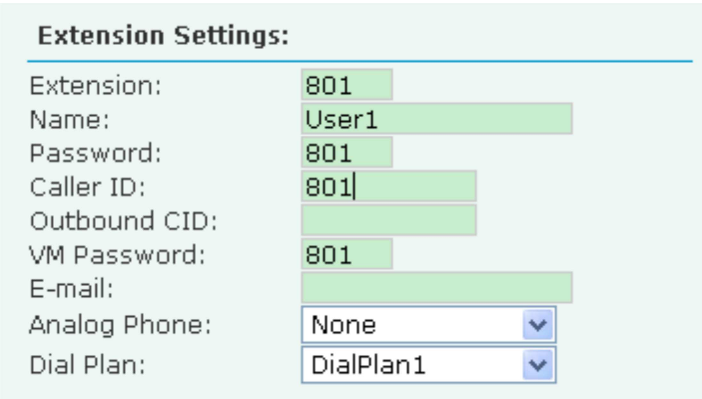
3. 801 call 601. And you can see 601 will ring and you can pick up the call. Above is the way to route SNR-VX50-B's call to SNR-VX50-A

Accordingly, if you want to c**all** from SNR-VX50-A to SNR-VX50-B, continue as below:

Step 5: Set Dial Rule in SNR-VX50-A all calls starting with 8 will be sent to SNR-VX50-B.



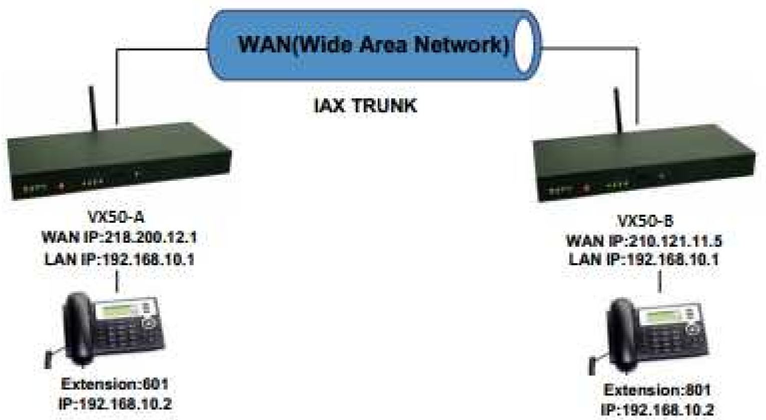
Step 6: Set the user 801 and Dial Plan in SNR-VX50-B



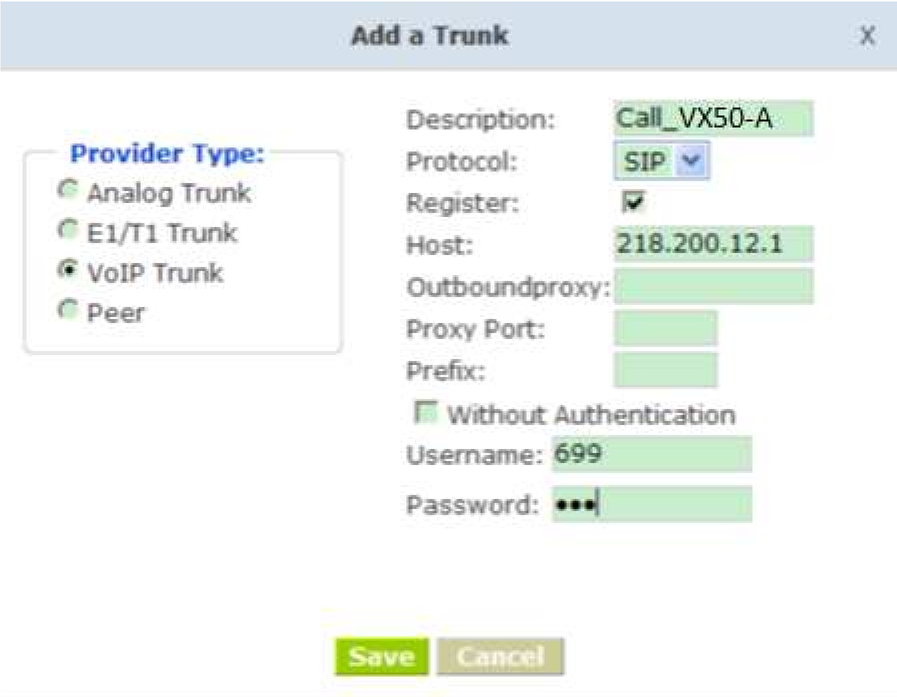
Activate the change and apply the test: 601 call 801, and 801 will ring and you can pick up the call

3 How to combine two IPPBX in different network

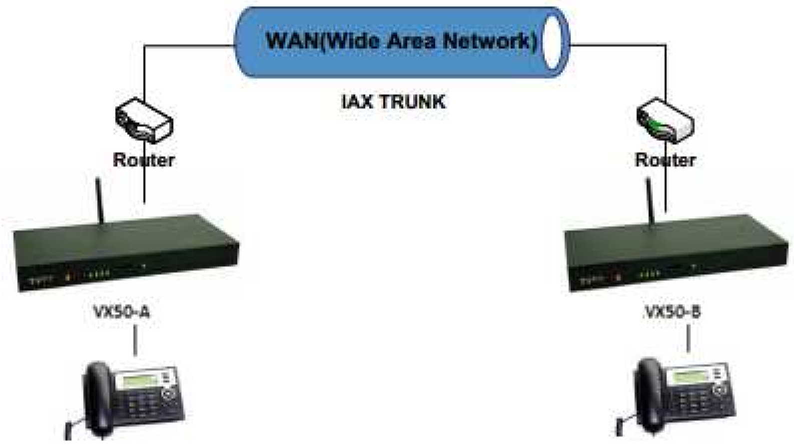
The general environment for two SNR-VX50 in different locations is: Both SNR-VX50 IP PBX are in the Internet and using the public IP.



The configuration is same as above guide (2 Combine two SNR-VX50 IP PBX in the same network) , but use the public IP address as the "HOST" settings, set as below: In the page Trunks of SNR-VX50-B--> Add a Trunk

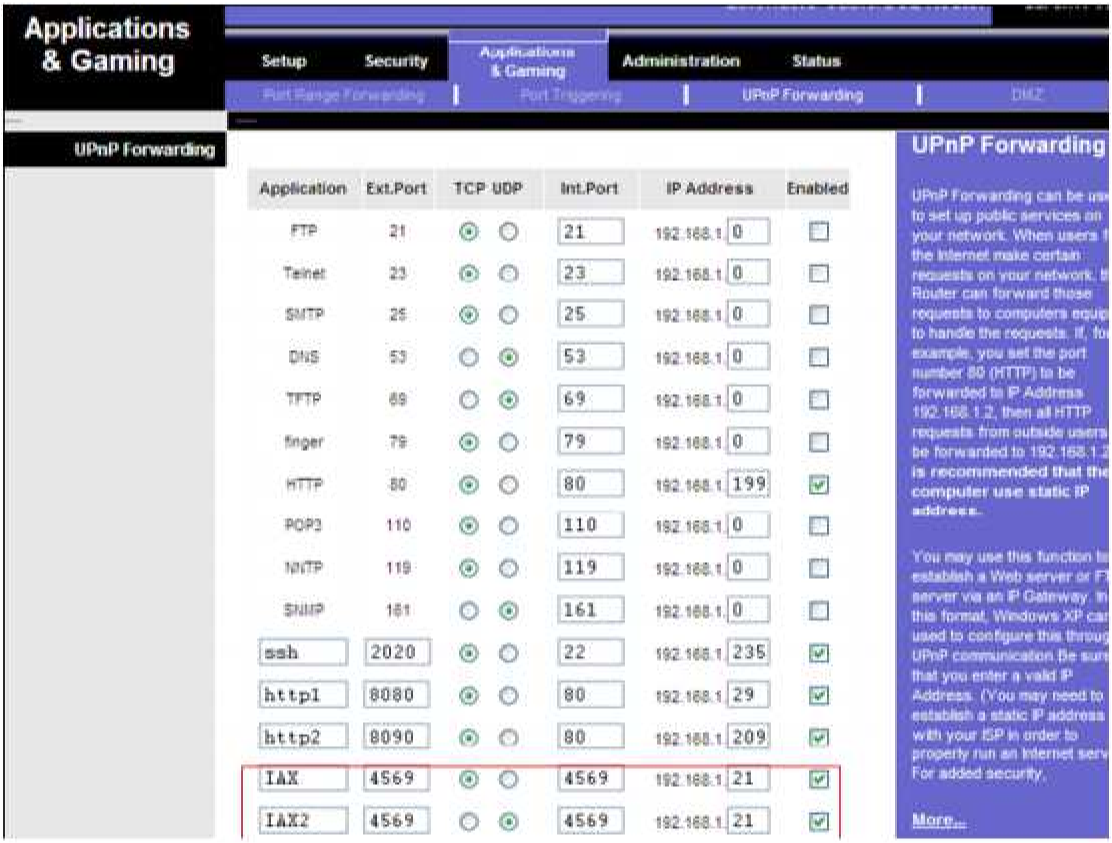


The general environment for two ZPX50 IP PBX in different location and one or both two are behind router and using the private IP. So we need to make port forwarding in the router and make SNR-VX50 IP PBX reach to each other.



Step 1: Set port forwarding in the router for SNR-VX50-A

For the SNR-VX50-A is behind the router, you need forward the IAX2 port in your router, so all the packets received on the router WAN port (210.11.25.127:4569) will be forwarded to the SNR-VX50-A (192.168.1.21:4569). Below is the setting page in a linksys router:



Step 2: Set up the Provider Host in SNR-VX50-B

Set up the service provider and calling rule in SNR-VX50-B to make it register to SNR-VX50-A. This method is almost the same as above, EXCEPT you need to use the 210.11.25.127 as the service provider instead of 192.168.1.21.

Step 3: Set port forwarding in the router for SNR-VX50-B

Use the same method as Step 1 to do port forwarding in router-B for SNR-VX50-B as above.

Step 4: Combine two SNR-VX50 and make calls

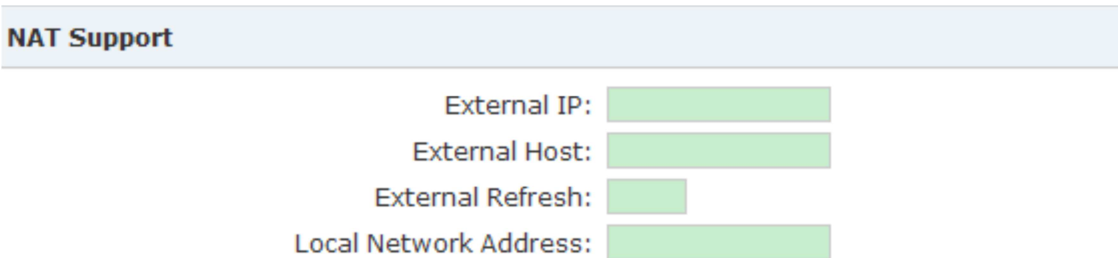
Accordingly, set the 601 users in SNR-VX50-A and 801 users in SNR-VX50-B, and build the correct dial rules as above, you can make calls between two the SNR-VX50 IP PBX.

Note: You can also apply a DDNS to get one fixed domain for both SNR-VX50 IP PBX and connect to each other rather than using the Port Forwarding in the router.

4 How to resolve problems about hearing on one side only

If your IP PBX is behind the Router, you should build an IP Address Map to resolve this problem as below:

【Advance】----【Options】 ----【Global SIP Settings】 ---【NAT Support】



- External IP: Replace your external IP address as your public IP or domain

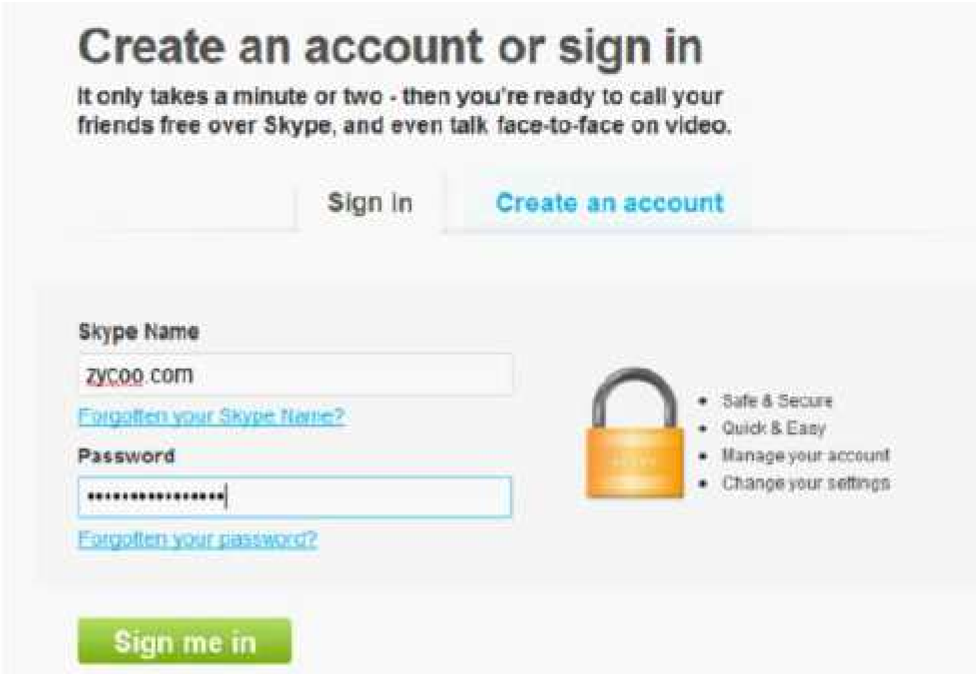
- External Host: Replace your external IP address as your public IP or domain
- External Refresh: Set time for refresh, default is 10
- Local Network Address: Replace your local network address and mask

How to use Skype account in SNR-VX50

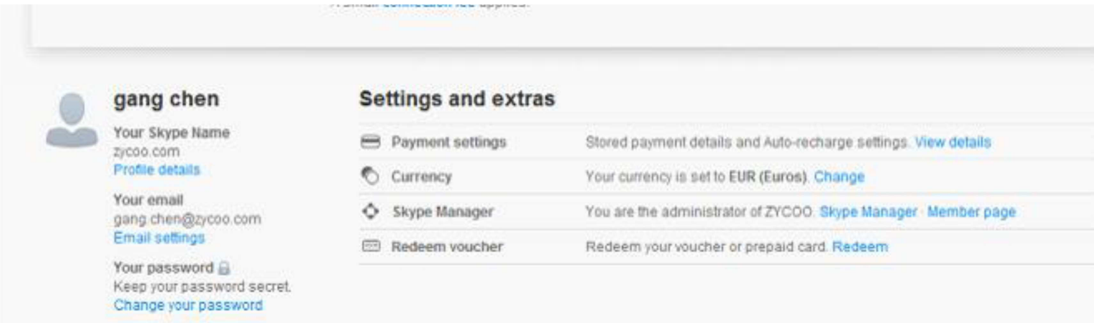
Notice: The fee of your business account must be more than €50 when you use the account first time.

1. Sign in with the business account on this page:

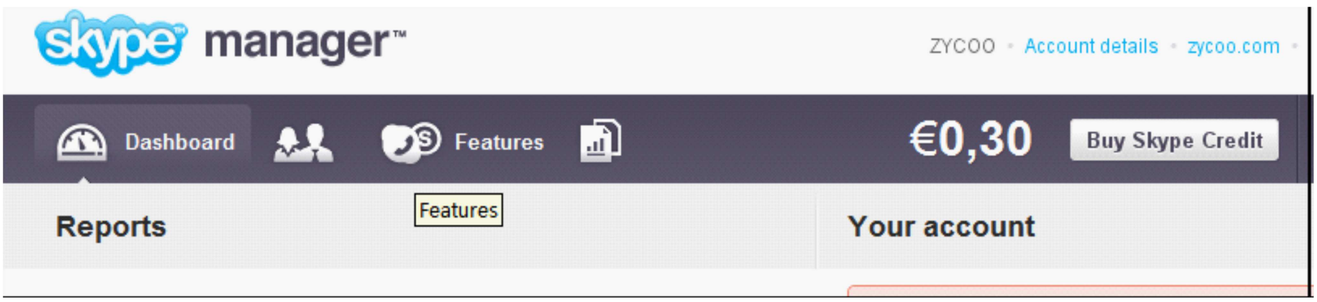
https://login.skype.com/account/login-form?intcmp=sign-in&return_url=https://secure.skype.com/account/login



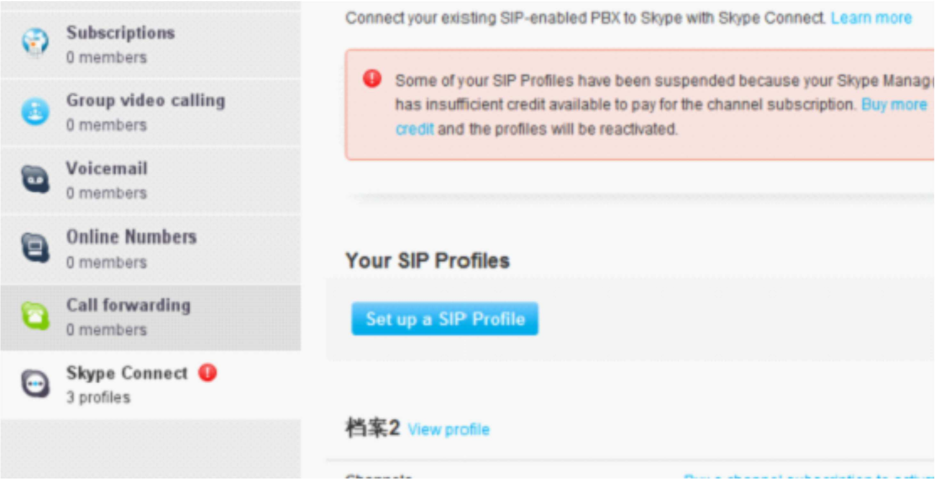
2.When you have signed in, please click **Skype Manager** at the end of this page.



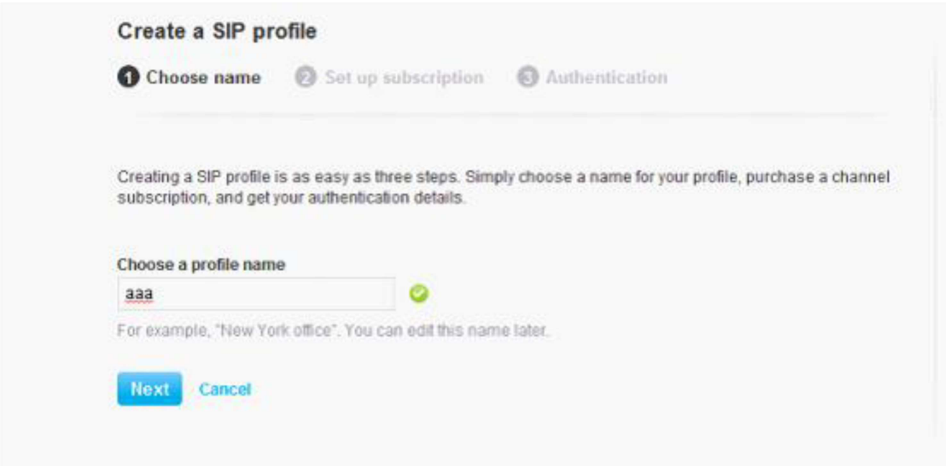
3.Please click the button **Features**.



4. Please click the **Skype connect**

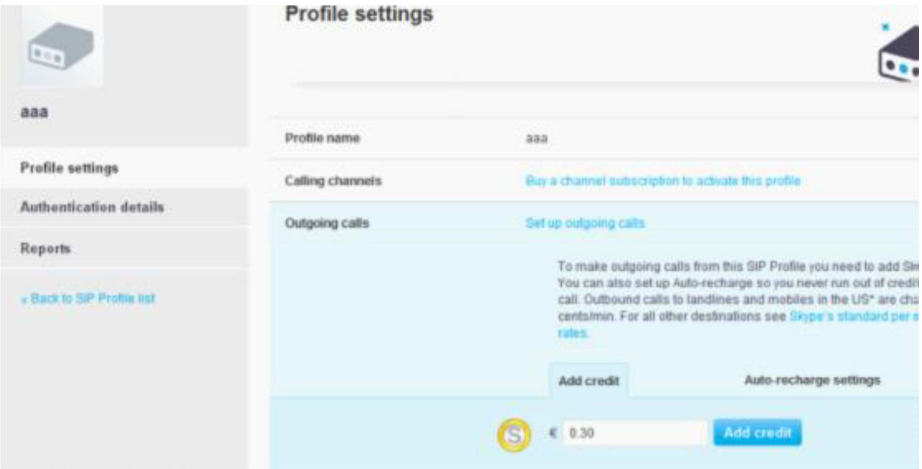


5. Create a SIP profile



Then you can create one sip account, you need pay for € 4.95 for one channel as monthly rent and you need input the register information to our VoIP trunk blank, then you can register to skype server. And you need assign money for outgoing calls, then you can call out.

Note: Skype Channel belongs to VoIP channel, so any calls from Skype will be directed to the same destination of VoIP



Then you can see the sip account information by clicking **Authentications details**.