Table of Contents

Safety Notice

Please read the following safety notices before installing or using this IP PBX. They are crucial for safe and reliable operation of the device. Failure to follow the instructions contained in this document may result in damage to your PBX and void the manufacturer's warranty.

1. Please use the external power supply which is included in the package. Other power supplies may cause damage to the device, affect the performance or induce noise.
2. Before using the external power supply in the package, please check your building power voltage. Connecting to Inaccurate power voltage may cause fire and damage.
3. Please do not damage the power cord. If the power cord or plug is impaired, do not use it. Connecting a damaged power cord may cause fire or electric shock.
4. Ensure the plug-socket combination is accessible even after the PBX is installed. In order to service the PBX it will need to be disconnected from the power source.
5. Do not drop, knock or shake the device. Rough handling can break internal circuit boards.
6. Do not install the device in places where there is direct sunlight. Also do not place the device on carpets or cushions. Doing so may cause the device to malfunction or cause a fire.
7. Avoid exposing the device to high temperature (above 40°C), low temperature (below -10°C) or high humidity. Doing so could cause damage and will void the manufacturer warranty.
8. Avoid letting the device come in contact with water or any liquid which would damage the device.
9. Do not attempt to open it. Non-expert handling to the device could cause damage and will immediately void the manufacturer warranty.
10. Consult your authorized dealer for assistance with any issues or questions you may have.
11. Do not use harsh chemicals, cleaning solvents, or strong detergents to clean the device.
12. Wipe it with soft cloth that has been slightly dampened in a mild soap and water solution.
13. If you suspect your device has been struck by lightning, do not touch the device, power plug or phone line. Call your authorized dealer for assistance to avoid the possibility of electric shock.
14. Ensure the PBX is installed in a well ventilated room to avoid overheating and damaging the device.
15. Before you work on any equipment, be aware of any hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. If you are in a situation that could cause bodily injury.

Chapter 1　Brief Introduction

1.1 Brief Introduction of LAVoice Series

The LAVoice Series IP PBXs are designed to provide SMEs (small & medium enterprises) with all the standard and advanced features that are normally only available from large, expensive, legacy PBX manufacturers. Aimed at businesses with up to 100 extensions, the LAVoice Series IP PBXs are based on SIP and OpenSource Asterisk 1.8, with whose innovative modular telephony design, that is easy to expand the PBX to meet the growing needs of your business.

LAVoice Series IP PBXs come in four sizes: 30S110 /100S / 200S / 500S.

Each model will be introduced in detail below:

LVX-100S is configured with 2 analog ports:

	FXS	FXO
LVX-30S110	1	1
	0	2

LVX-100S consists of two main parts： 100S Host and Modules. There are 2 slots in the system and the modules can be utilized as in the diagram below:

100S Slot / 100S Module	Slot 1	Slot 2
4FXS	✓	✓
4FXO	✓	✓
2FXOS	✓	✓
2GSM	✓	✓
4GSM	✓	✓
1PRI	✓	✗
4BRI	✓	✗

LVX-200S is configured with 24 analog ports:

	2FXS	2FXO	FXOS
LVX-200S	✓	✓	✓

LVX-500S consists of two main parts： 500S Host and Modules. There are 2 slots in the system and the modules can be utilized as in the diagram below:

500S Slot / 500S Module	Slot 1	Slot 2
4FXS	✓	✓
4FXO	✓	✓
2FXOS	✓	✓
2GSM	✓	✓
4GSM	✓	✓

1PRI	✓	✓
4BRI	✓	✗
4BRI	✗	✓

1.2 Main Features

1. SIP/ IAX Extension Registration
2. Video Call
3. USB Mobile Hard Disk Record (Scalable)
4. IP Phone Provisioning (Grandstream /Yealink/Fanvil IP Phone)
5. Call Record /Ring Group Record/ Call Queue Record
6. Web-based Administration and configuration
7. Web-based Extension User Management
8. Voicemail
9. Caller ID
10. Call Parking
11. Call Forward
12. Call Transfer
13. Call Waiting
14. Call Center Queues
15. Black List
16. Phonebook
17. Flexible Dial Plan
18. Virtual Fax (fax to email, and email to fax)
19. DID
20. Dial by Name
21. Speed Dial
22. Do Not Disturb
23. Callback
24. Skype for SIP
25. Ring Group
26. Conference Bridge (Three Conferences)
27. Music On Hold
28. DISA (Direct Inward System Access) /Paging And Intercom
29. Call Detail Record
30. IP Phone Feature Code
31. BLF(Busy Lamp Field)
32. Static /DHCP /PPPoE Network Access
33. DHCP Server
34. System Backup
35. T.38 Pass-through
36. Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX
37. Video Codec: H.261/ H.263 / H.263+ / H.264
38. VPN Server (L2TP / PPTP / OpenVPN, up to 10 connections for VPN clients)
39. VPN Client (L2TP / PPTP / OpenVPN / N2N)

40. SNMPv2
41. IPv4 / IPv6
42. DDNS(Dyndns.org /No-ip.com /zoneedit.com)

1.3 Modules

	
4FXS Module	4FXO Module
	
2FXOS Module	2GSM Module
	
4GSM Module	4BRI Module
	
1E1/T1 Module	32 EC Module

1.4 Hardware Interfaces

1.4.1 LVX-100S

1 * Reset Button
1 * Power Interface (*DC 12V 2A*)
1 * Ethernet Interface (10/100Mbps)
2 * Analog Ports(FXO/FXS)
1 * UMTS Port

30S110 LEDs Indication

PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	Blink	System Works
		Off	System Fails
ETH	WAN or LAN Data Status	Blink	Data Transport
G	GSM or UMTS(3G) Status	Off	Module not running
		64ms On/800ms Off	Module doesn't find network
		64ms On/3000ms Off	Module finds network
1	FXO	Red	Module Loading Success
		Blink	Channel Ringing
		Off	Module Loading Failure
2	FXS	Green	Module Loading Success
		Blink	Channel Ringing
		Off	Module Loading Failure

1.4.2 LVX-100S

1 * Reset Button

1 * Power Interface (*DC 12V 2A*)

1 * Ethernet Interface (10/100Mbps)

1 * Console Interface

1 * USB Interface

Slot 1 for Analog/GSM/PRI/BRI Module Cards

Slot 2 for Analog/GSM Module Cards Only

Indication	Function	Status	Explaination
PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	Blink	System Works
		Off	System Fails
ETH	Data Status	Blink	Data Transport
		Off	No Data Transport
USB	U-disk or UMTS(3G) Status	Off	Module not running
		On	Module Works

				Green	Module Loading Success
		FXS		Blink	Channel Ringing
				Off	Module Loading Failure
				Red	Module Loading Success
		FXO		Blink	Channel Ringing
				Off	Module Loading Failure
				Red	Module Loading Success
		GSM		Blink	Channel Ringing
				Off	Module Loading Failure
		E1/T1	L1	Red	Module Loading Success
1-4(SLOT1)	SLOT 1 Status	(PRI/		Off	Module Loading Failure
		R2)	L2	Red	CPE signal
				Green	NET signal
				Off	No signal
			L3	Red	SS7 signal
				Green	MFCR2 signal
				Off	No signal
			L4	Red	Disconnected/ Alarm
				Green	Connected/ No Alarm
		BRI		Red	TE Mode
				Green	NT Mode
				Off	Module Loading Failure
				Green	Module Loading Success
		FXS		Blink	Channel Ringing
				Off	Module Loading Failure
				Red	Module Loading Success
1-4(SLOT2)	SLOT 2 Status	FXO		Blink	Channel Ringing
				Off	Module Loading Failure
				Red	Module Loading Success
		GSM		Blink	Channel Ringing
				Off	Module Loading Failure

100S LEDs Indication

1.4.3 LVX-200S

1 * Power Interface

1 * Power Switch

2 * Ethernet Interfaces (10/100/1000Mbps)

1 * VGA Interface

2 * Audio Interfaces

2 * USB Interfaces

1 * Hardware Echo Cancellation Interfaces (onboard)

1 * UMTS Interface for 3G Data (onboard)

24 * Analog Ports (FXO/FXS)

200S LED Indication

Indication	Function	Status		Explaination
PWR	Power Status	On		Power On
		Off		Power Off
SYS	System Status	Blink		System Works
		Off		System Fails
ETH	Data Status	Blink		Data Transport
		Off		No Data Transport
1-24 SLOTS	SLOT 1-24 Status	FXS	Green	Module Loading Success
			Off	Module Loading Failure
		FXO	Red	Module Loading Success
			Off	Module Loading Failure

1.4.4 LVX-500S

1 * Reset Button

1 * Power Interface

1 * Power Switch

2 * Ethernet Interfaces (10/100 Mbps)

1 * VGA Interface

2 * USB Interfaces

2 * Audio Interfaces

SLOT 1 for any Module Cards (only not both for 4BRI)

SLOT 2 for any Module Cards (only not both for 4BRI)

500S LED Indication

Indication	Function	Status			Explaination
PWR	Power Status	On			Power On
		Off			Power Off
SYS	System Status	Blink			System Works
		Off			System Fails
ETH	Data Status	Blink			Data Transport
		Off			No Data Transport
1-4(SLOT1/2)	SLOT 1 /2 Status	FXS		Green	Module Loading Success
				Off	Module Loading Failure
		FXO		Red	Module Loading Success
				Off	Module Loading Failure
		GSM		Red	Module Loading Success
				Off	Module Loading Failure
		E1/T1	L1	Red	Module Loading Success
				Off	Module Loading Failure
			L2	Red	CPE signal
				Green	NET signal
				Off	No signal
			L3	Red	SS7 signal
				Green	MFCR2 signal
				Off	No signal
			L4	Red	Disconnected/ Alarm
				Green	Connected/ No Alarm
		BRI		Red	TE Mode
				Green	NT Mode
				Off	Module Loading Failure

1.4.5 Model Comparison Table

Items		LVX-100S	LVX-100S	LVX-200S	LVX-500S
System Capacity	Concurrent Calls	10	20	80	80
	Extension Users	30	100	200	500
	Voicemail and Recording	21,000 mins (GSM)	21,000 mins (GSM)	200,000 mins (GSM)	2,500,000 mins (GSM)
		3000 mins (wav)	3000 mins (wav)	20,000 mins (GSM)	270,000 mins (wav)
Hardware Capacity	SDRAM	128MB DDR2	256MB DDR2	1GB DDR3	2GB DDR3
	Memory (default)	4GB SD card	4GB SD card	32GB SSD	500GB HDD or 32GB SSD
Power Supply	Input	AC 100-240V	AC 100-240V	AC 100-240V	AC 100-240V
	Output	DC 12V/1A	DC 12V/2A	N/A	N/A

1.4.6 Environmental Requirements

1. Working Tempreture: 0 °C ~40 °C
2. Storage Tempreture: -20 °C ~ 55 °C
3. Humidity: 5~95% Non-Condensing

1.4.7 Packing List

LAVoice Host	1 set
Power Supply	1 piece
Ethernet Cable	1 piece
Quick Installation Guide	1 piece
Warranty Card	1 piece

Notice:

1) LAVoice Module cards will only function in LAVoice IP PBX from LAVoice;

2) Module cards for LVX-100S/500S will be packed separately but contained in the same package.

Chapter 2 Getting Started

(Take LVX-100S as example for the guide)

2.1 Before Configuration

What kind of IP Phones can be used with this device?

1. FXS Interface: Analog Phone or fax machine
2. SIP Extension: IP Phones provided by LAVoice

   Any standard SIP Phone based on SIP/ IAX2 protocol
   (eg: CISCO, Grandstream, Yealink, Polycom, Siemens, Snom,etc.)

2.2 Before Making a Call

2.2.1 Login IP PBX

Getting IP Address

There are three ways to set the IP address: Static, DHCP, PPPoE.

Default IP: 192.168.1.100:9999

Notice: you have to add port number 9999 after this IP address.

Defaults and Function Key

1.	Web Panel User name:	admin
2.	Web Panel Password:	admin
3.	*60	Enter Voicemail Box
4.	900/901/902	Default three conference room numbers
5.	#	Blind Transfer
6.	*2	Attended Transfer
7.	*	Disconnect Call

Administrator Login

After connecting the LAVoice IP PBX to the local area network and setting your laptop to the 192.168.1.x subnet, launch the web browser and bring up the system login page by entering the following URL: http://192.168.1.100:9999. You will see the login interface as below:

Input username and password, press the "Login" button and you will see the configuration interface below.

1. Default username: admin and password: admin

Notice

1. Please use IE(7.0 or higher version), Chrome, Firefox web browser.
2. If you do not see the interface above after inputting default IP and port number, please check whether your computer IP address is in the same segment with your IP PBX.
3. For Security reasons, please modify the username and password after login successfully. You can modify these by selecting: 【System】---【Management】
4. With the default setting, if there is no activity on the page for more than one minute, the system will timeout and automatically log out To continue making configuration changes, you will need to login again.

---

2. Network WAN IP and MAC will be displayed
3. Storage Total storage and used storage will be displayed
4. Channels Channel information will be displayed based on the modules installed
5. Device Info Model No. And system version will be displayed

Commonly Used Buttons

On the home page, besides system info, there are other function buttons as below:

1. Logout Logout the Web panel
2. Activate Changes Activate the changes for your current configuration

System Menu

System Menu includes the following sub menu:

Home	Display device information
Operator	Extension / Trunk / Channel Status
Basic	Basic configuration on extension, trunks, etc.
Inbound Control	Configuration of Inbound Route, IVR and Black List, etc.
Advanced	Configuration of extension's default information, Conference Call, Call Transfer, Function Key, etc.
Network Settings	Configuration of Routing, Network, VPN, DHCP and other related network parameters
Security	Configuration of Firewall, SSH, FTP
Report	Record List, Call Logs and System Logs
System	Time Settings, Management, Back Up and Upgrade, etc.

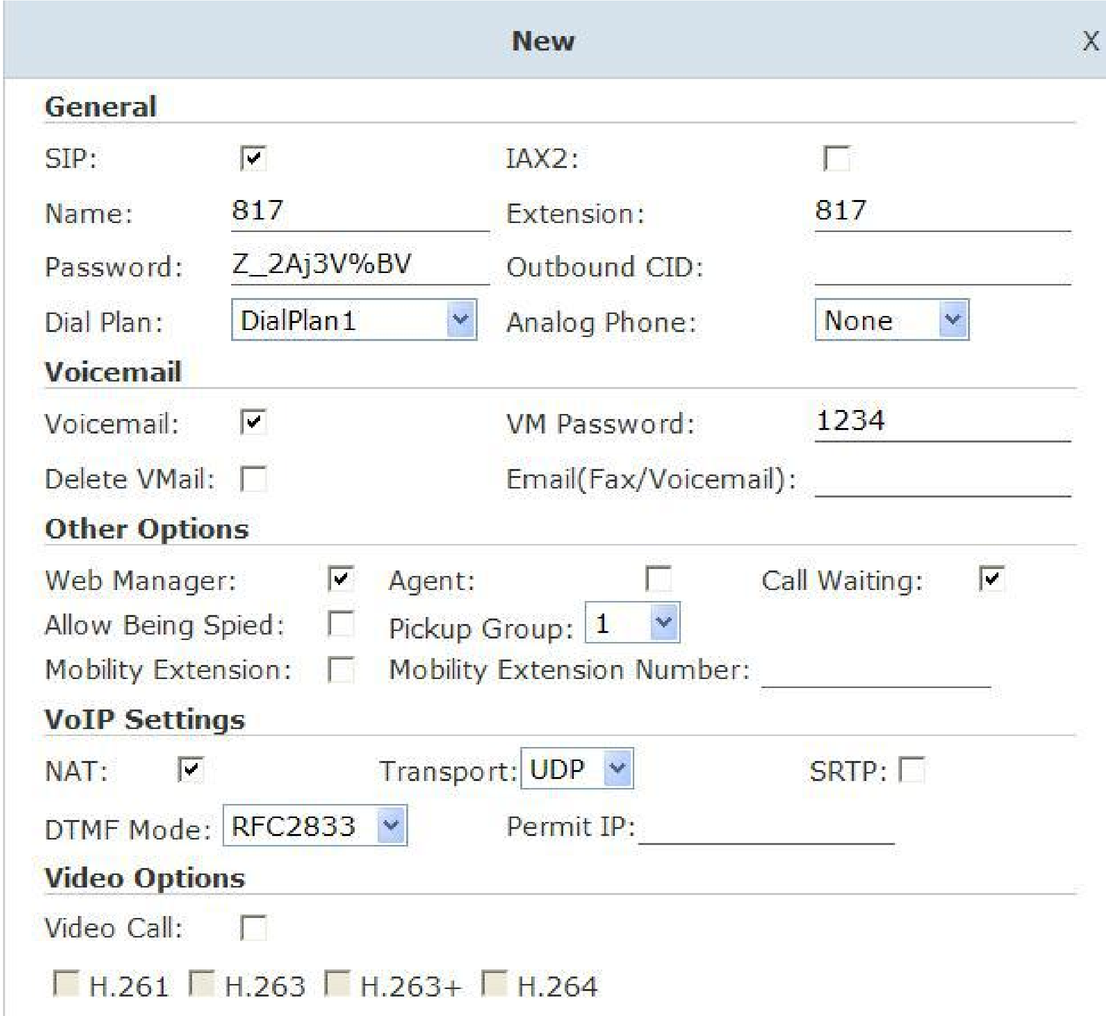
2.2.2 Basic Configuration

Extension Configuration

LAVoice 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 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	Same password as voicemail. (4-16 digits, e.g.:123456)
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 choose the relative FXS port for your analog phone.
Voicemail	Check this option to enable the voicemail account.
VM Password	Set password for Voicemail, for security reasons, do not use the extension number or any easy combination like "1234"
Delete VMail	Check this option to delete voicemail from the PBX after it’s sent by email.
Email (FAX/Voicemail)	Extension user’s email address to receive email messages with attached fax or voicemail (you need configure the fax to email/voicemail options), e.g.: Tom@gmail.com
Web Manager	Allow this user to login to the Extension Management Panel to manage extension options including 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 this extension to be monitored (listened to or “spied”).
NAT	Check this option if extension user or the phone is located outside the NAT(Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After check 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 (Secure Real-time Transport Protocol)
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary..
Video Call	Check to enable video calling for this extension. And select the video codecs you need to use.
Permit IP	Set device ip address or subnet permitted to register this extension with the IP PBX, e.g.:192.168.1.77 or 192.168.10.0/255.255.255.0. Devices with other IP addresses are not allowed to register this extension with the IP PBX.
Audio Codecs	Select what audio codecs you need to use.

Notice:

1. There are 30 default extensions which number started with “8”*; you can add or delete extension by your requirement.
2. Maximum extensions: 100

Upload/Download Extensions

Click【Upload/Download Extensions】to batch add extensions as below:

Download the extension template from the【Download Extensions Template】,

open the template using an editor or application like Microsoft Excel and carefully add extension information based on the template format and save. Select the extension file to upload from【Upload Extensions】

Download current extensions information from【Download Extensions(.csv)】

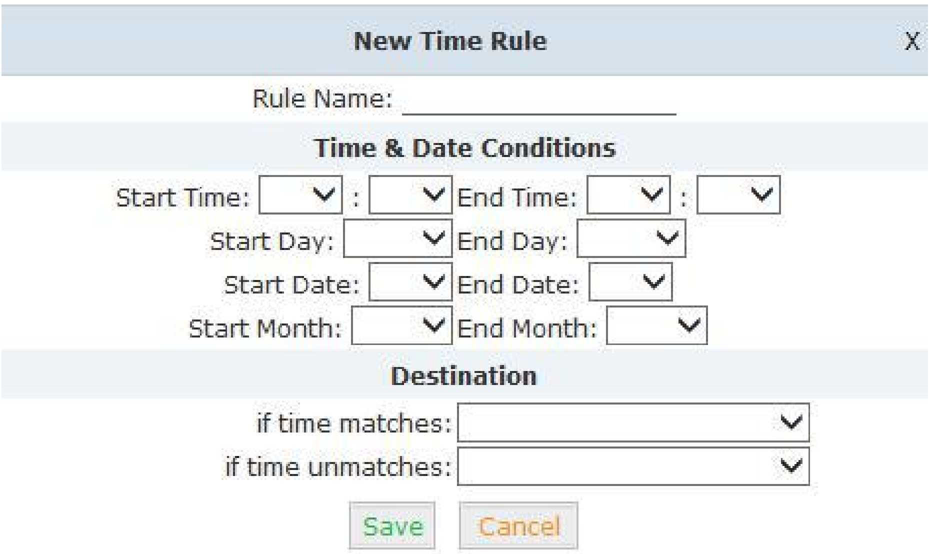
2.2.3 Time Based Rules

Create a Time Rule. For example, BusinessHours.

Select the starting and ending time, starting and ending days of the week, specific start and end dates and/or start and ending month of the year.

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 Conditions	Set parameters for Time/Day/ Date/ Month.
Destination	Select destination if time matches or does not match the conditions set. For example for BusinessHours, “if time matches”, select operator extension during BusinessHours. If outside business hours, select “if time does not match” destination of Operator voicemail

2.3 Outbound Call

2.3.1 Trunks

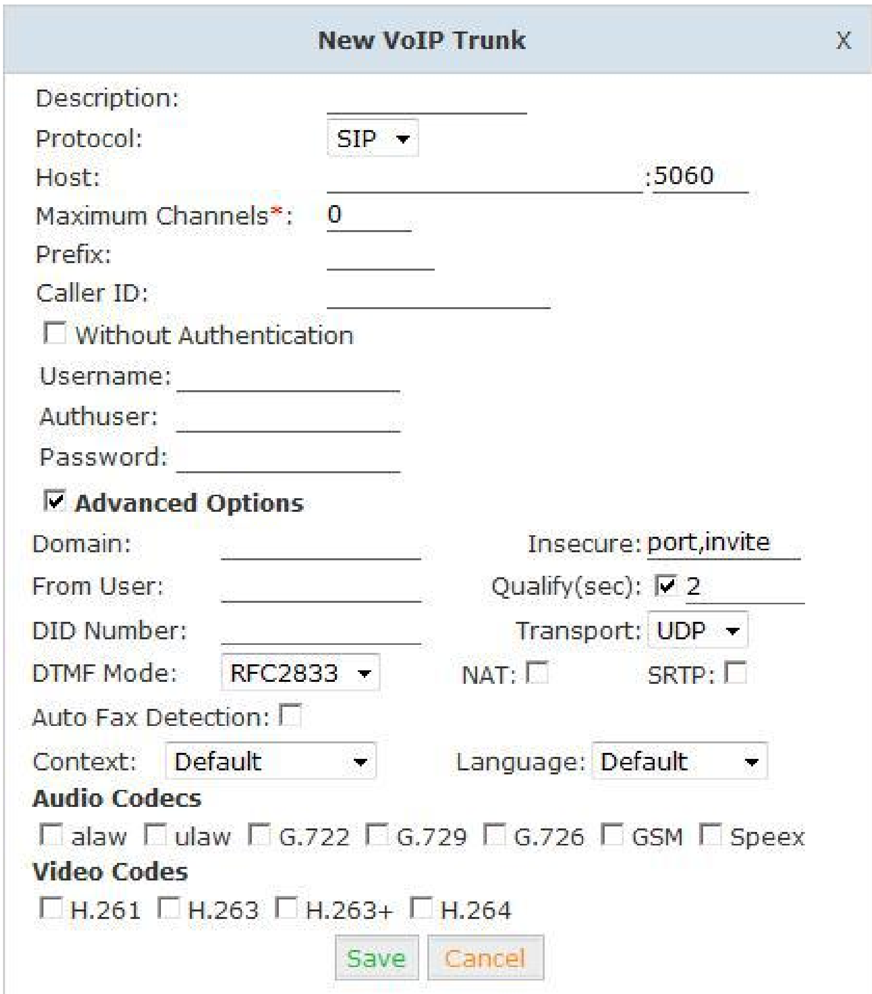
If you want to set up outbound route connected to PSTN (Public Switch Telephone Network) or VoIP provider, please configure on this page:【Basic】->【Trunks】

LAVoice supports two kinds of trunks for your choice: VoIP or SIP Trunk and FXO/GSM/PRI/BRI Trunk.

How to add each trunk:

VoIP Trunks

Click【VoIP Trunk】->【New VoIP Trunk】:



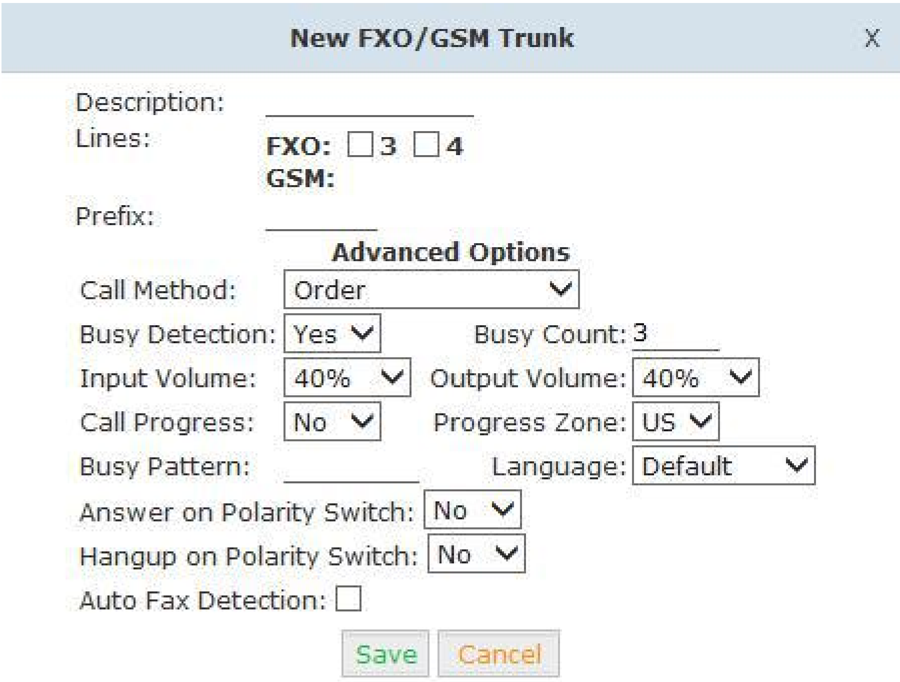
VoIP Trunks Reference:

Item	Explanation
Description	Description of SIP trunk.
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. Note: This function must be supported by local provider.
Without Authentification	If your trunk is static IP based and does not require a registration string when connecting the LAVoice IP PBX, check this option.
Username	Username provided by VoIP Provider.
Password	Password provided by VoIP Provider.
Advanced Options	Advanced options for this trunk, e.g.: codecs, dialplan, etc.

The outbound trunk will be in the list of VoIP Trunk when the trunk is added successfully. **FXO/GSM Trunk**

Click【FXO/GSM Trunk】->【New FXO/GSM Trunk】:



FXO/GSM Trunk Reference:

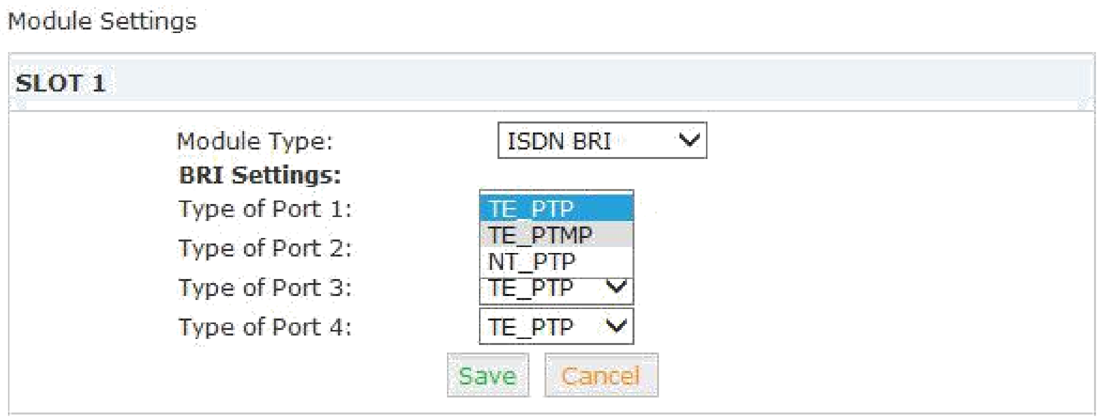
Item	Explanation
Description	Description for this trunk.
Lines	Check one or more channels (FXO or GSM) to be included in this trunk group
Prefix	The prefix will be added to the dialed number automatically when this trunk is in use.
Advanced Options	Advanced Options for this trunk, e.g.: Call Method, Busy Detection, etc.

Select one or more of the available channels to be used for this trunk group.

Note: each channel can only be included in one trunk group. If no channels appear then all available channels are already defined.

BRI / PRI Trunk

Please set up BRI/PRI trunk similarly to the FXO/GSM trunk settings above.



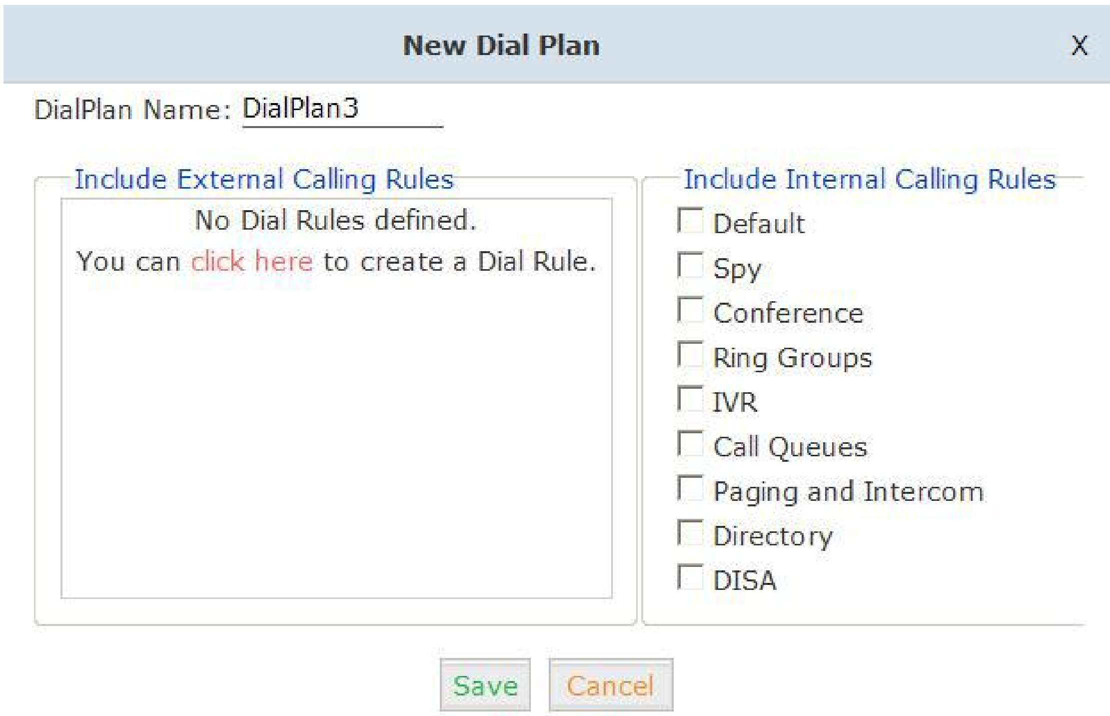
2.3.2 Outbound Routes

Outbound Routes are used to define which trunk groups are used by a specific extension when placing outbound calls. If you don't allow an extension user to place external calls, please ignore this part.

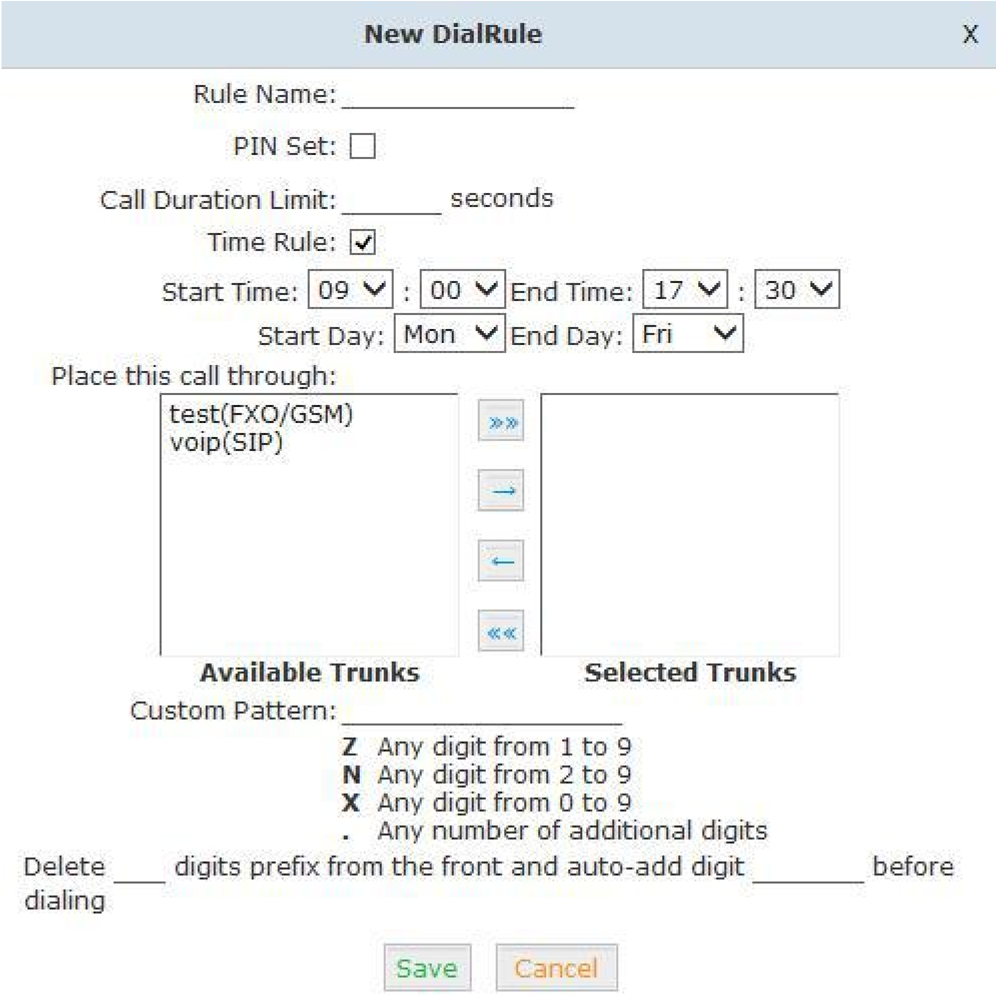
Please configure on this page: 【Basic】->【Outbound Routes】

You can configure the basic match pattern of outbound routes and create different dial plan on this page. Create as many different dial plans as you need to determine how you need extensions to be allowed to make calls. For example, create “InternalDialPlan” to include all Internal Calling Rules but do not select any outbound dial rules. Select “InternalDialPlan” for all extension users that do not need the ability to make external calls.

Click【DialPlans】->【New DialPlan】:



You can create one or more DialRules for DialPlans from this page:



Reference:

Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule.
Call Duration Limit	Set the duration limit for a call, beyond which the call will be auto hung up
Time Rule	Set the time interval for this DialRule, beyond which the call based on this DialRule won't work
Place this call through	Select one of the trunk groups that have been set up to use for this dial rule
Custom Pattern	N any digit from 2 to 9 Z any digit from 1 to 9 X any digit from 0 to 9 . One or more digits
Delete[ ]digits prefix	How many digits will be deleted from what the user dialed to what is actually sent over the trunk. For example, user dialed 94166445775 and you selected to delete 1 digit, then 4166445775 is sent out the trunk.
Auto-add digit[ ]	If add digit "9", when dial 12345, 912345 will be sent.

2.4 Inbound Call

2.4.1 Inbound Routes

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.

Select all calls coming in on a specific port (FXO/GSM/VOIP) and select which destination (Extension User, IVR, Queue, Conference Bridge, IVR, etc) should answer those calls. Setting the label will assign this label to be displayed.

Port DIDs

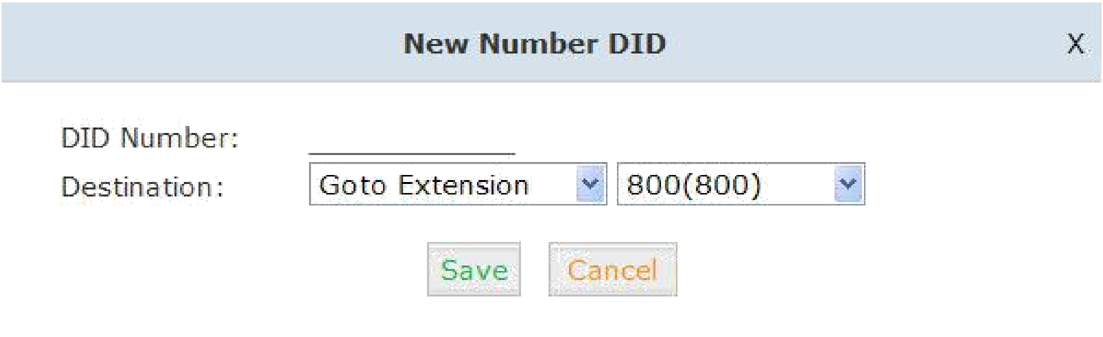
To have incoming calls from a PSTN trunk port (FXO/GSM trunk) answered by a specific extension user, call queue, conference bridge, or IVR, please configure here: Click【Port DIDs】->【New Port DIDs】:

1. Port — Select the trunk group port
2. Label — Set a label for this port. Incoming calls from this port will display the specified label.
3. Destination — Incoming calls will be answered by the specified destination (extension user, call queue, conference bridge, 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】:



4. DID Number Set DID Number
5. Destination Select the extension for access directly(Extension User/ Call Queue/ conference/ IVR)

DOD Settings

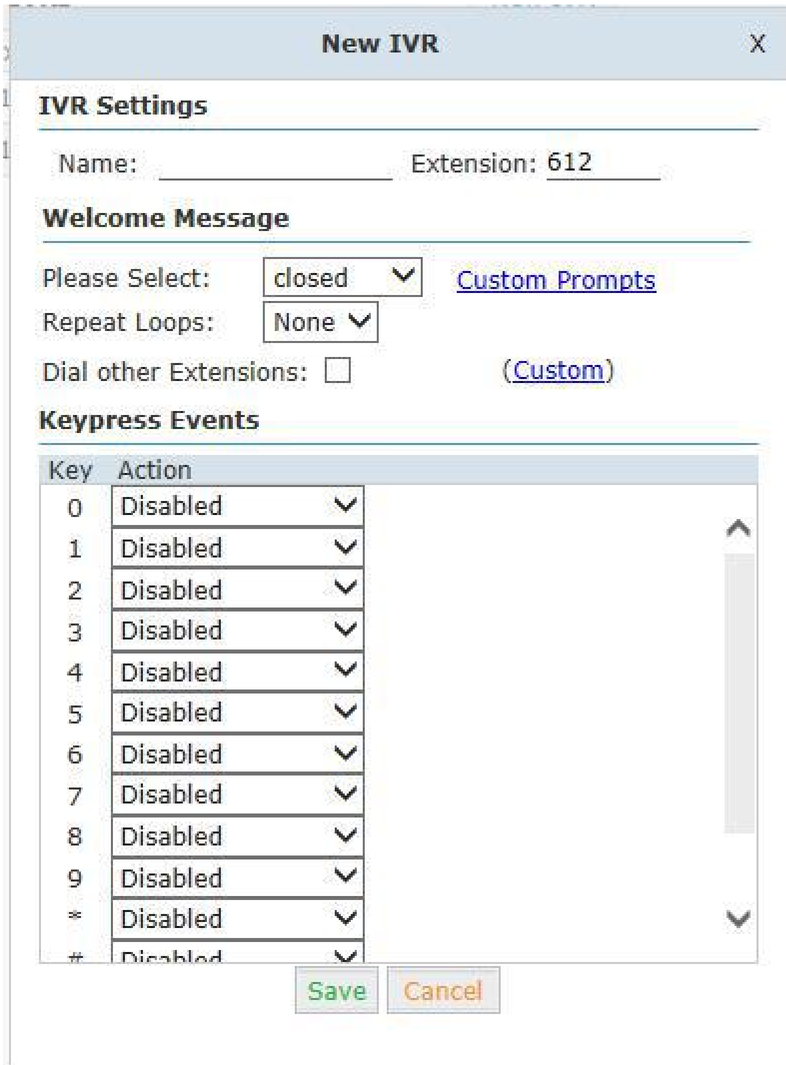
To configure outbound calls from user extensions to answer with specified destinations (user extension, queue, conference bridge, IVR), please click 【DOD Settings】->【New DOD】

6. DOD Number Set the DOD (direct outbound dial) number, and use it to match the Caller ID.
7. Destination Outbound calls will access directly to this destination (user extension, queue, conference bridge, or IVR)

2.4.2 IVR

IVR (Interactive Voice Response) or Automated Attendant will allow callers to select from a specific set of options by pressing the selected digit on their telephone dial pad. Click 【Inbound Control】->【IVR】:

Click【New IVR】to create a new IVR:



Item	Explanation
Name	Enter a descriptive name for the IVR
Extension	Enter a unique extension or IVR number. This number is used to access the IVR from an internal extension
Custom	Click “Custom” to choose a DialPlan for IVR
Please Select	Select the IVR prompt that will provide the caller with instructions on what options are available. To configure the prompt in this page:【IVR Prompt】
Repeat Loops	Loop times to repeat playing the IVR prompt if the caller does not select an option
Dial Other Extension	Allow user to dial other extensions besides of the listed options
Keypress Event	Select the available options beside the designated digit

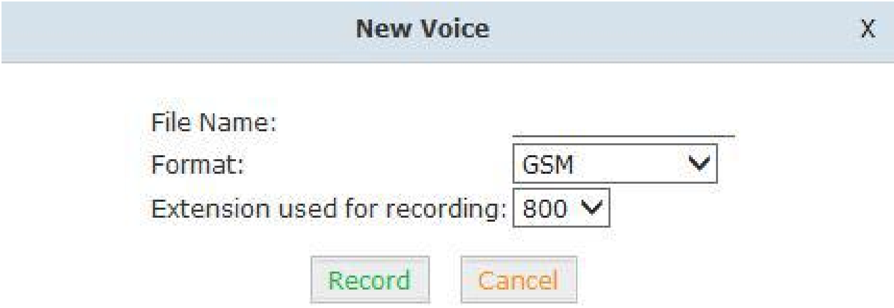
2.4.3 IVR Prompts

IVR prompts can be recorded by using any extension registered to the PBX or they can be uploaded from the “Upload IVR Prompt” section below.

IVR Prompts

【IVR Prompts】

Click【IVR Prompts】----【New Voice】to create new IVR prompt:



1. File Name Define a name for this voice file.
2. Format Select the voice format,GSM/WAV(16bit) supported only.
3. Extension used for recording: Select the extension which is used for recording the IVR prompt.

Click【Record】, the extension will ring, and the prompt can be recorded after picking up the phone.

To hear the existing recording, 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

【Upload IVR prompt】

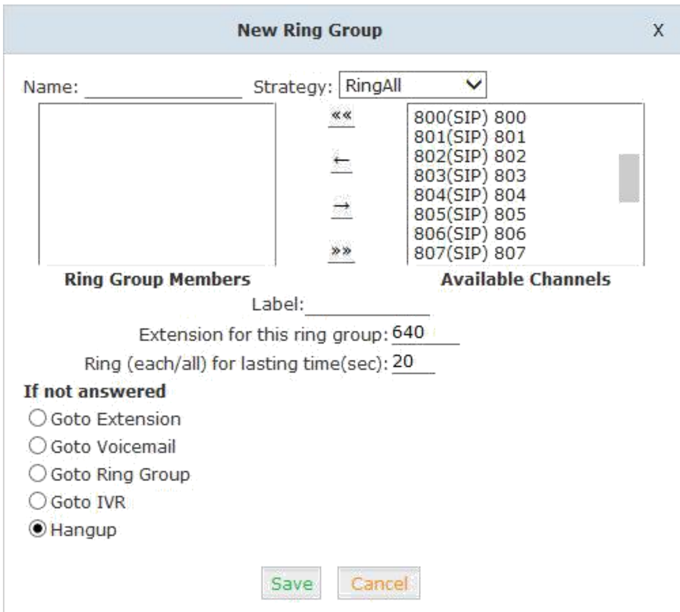
Notice:

LAVoice supports custom audio file with wav,gsm,ulaw,alaw format. Recordings must be smaller than 15MB.

2.4.4 Ring Groups

A Ring Group (sometimes called a Hunt Group) is a way to ring a collection of extensions by dialing a single extension number. The methodology used to ring that collection of extensions is called the ring strategy. Once the timeout (number of seconds) is reached, the call will then be directed to the “if not answered” or failover destination.

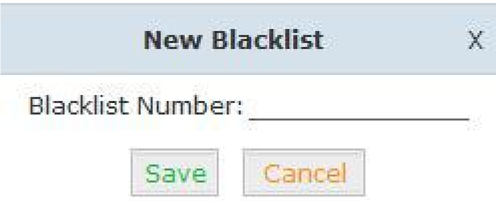
To configure a Ring Group Click【Inbound Control】->【Ring Groups】->【New Ring Group】:



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

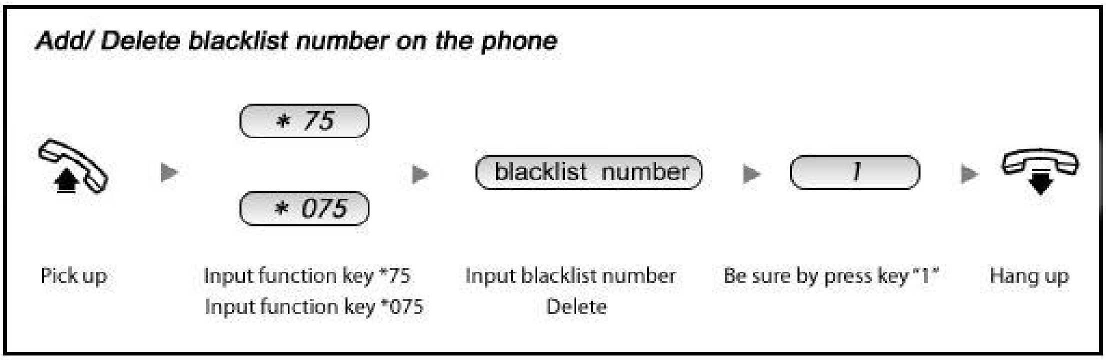
2.4.5 Blacklist

The Blacklist feature allows the blocking of specific phone numbers by Callerid.
Click【Inbound Control】->【Blacklist】->【New Blacklist】



Input the caller ID in the space provided. Once configured, future calls from this caller ID will be blocked.

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



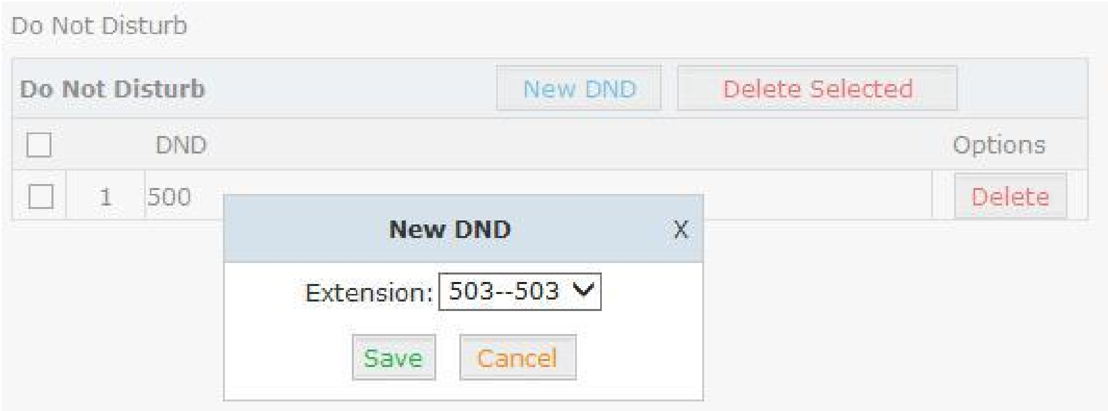
Reference:

Item	Explanation
*75	When the registered extension user inputs *75 + blacklist 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 Blacklist Number.

2.4.6 Do Not Disturb

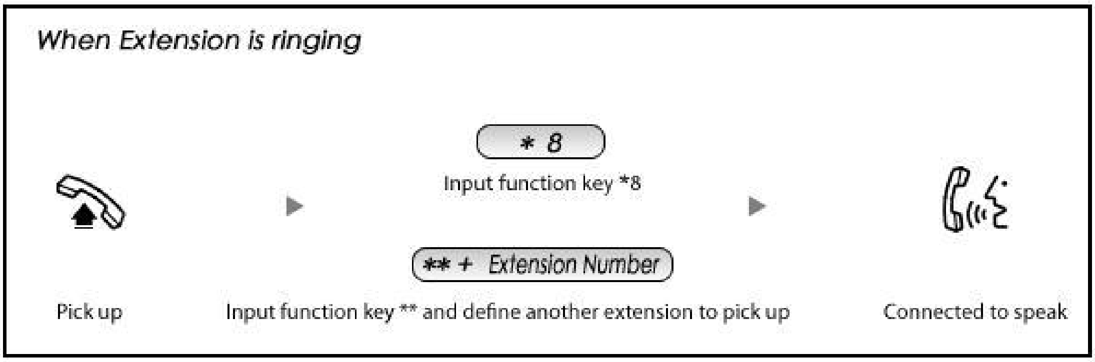
The administrator can config DND for extensions on this page:

Click【Inbound Control】->【Do Not Disturb】:



2.4.7 Call Pickup

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



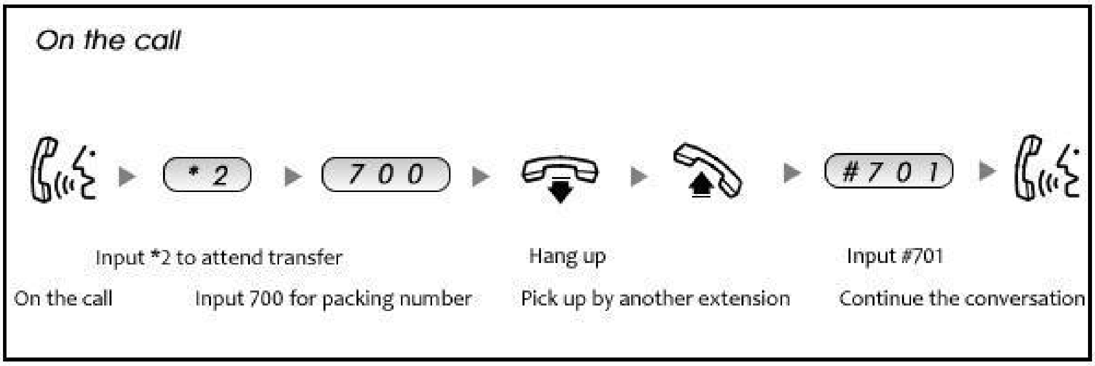
Reference:

Item	Explanation
*8	Input function key *8 to pick up the registered extension which is in the ring at random. This can be defined in 【Feature Codes】
**	Input function key ** and define another extension to pick up. This can be defined in 【Feature Codes】.

2.5 During a Call

2.5.1 Call Parking

This feature allows a call to be placed on hold (system will play the parked number, e.g. 701) and then retrieved from any other extension by entering the parked number. After answering the call, to park the call press *2 700 on the telephone dialpad (to transfer the call to the parking lot 700). This will park the call and the system will play the parking space (e.g. 701). To retrieve the call from the parking lot, anyone can pick up any registered extension and dial the parking space number (e.g. 701) and will be connected with the parked caller. Refer to the diagram below:

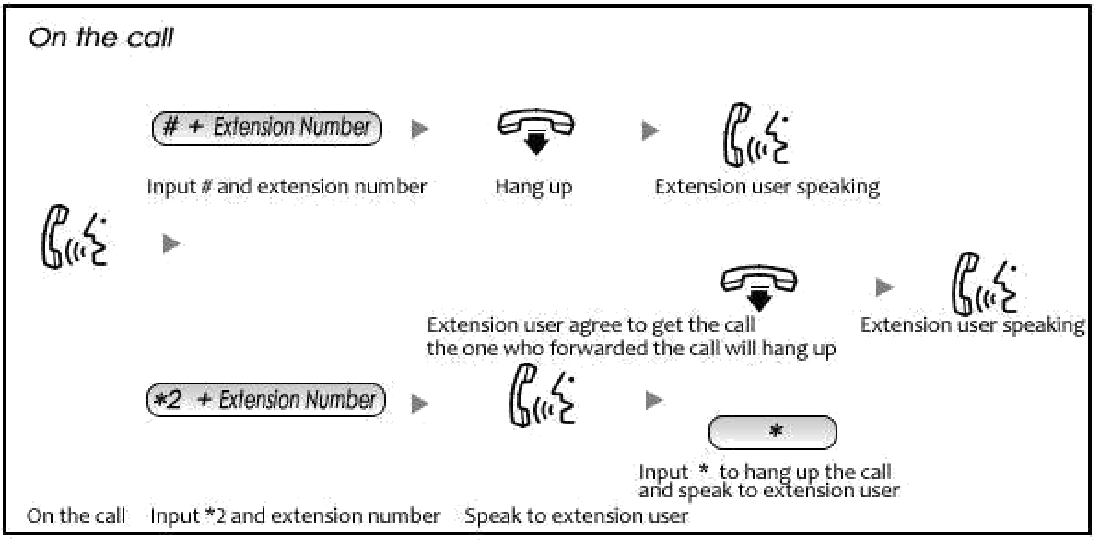


Reference:

Item	Explanation
Extension to Dial for Parking Calls:	Default Number: 700, Define in【Feature Codes】
What Parking space or 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】.

2.5.2 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:



Reference:

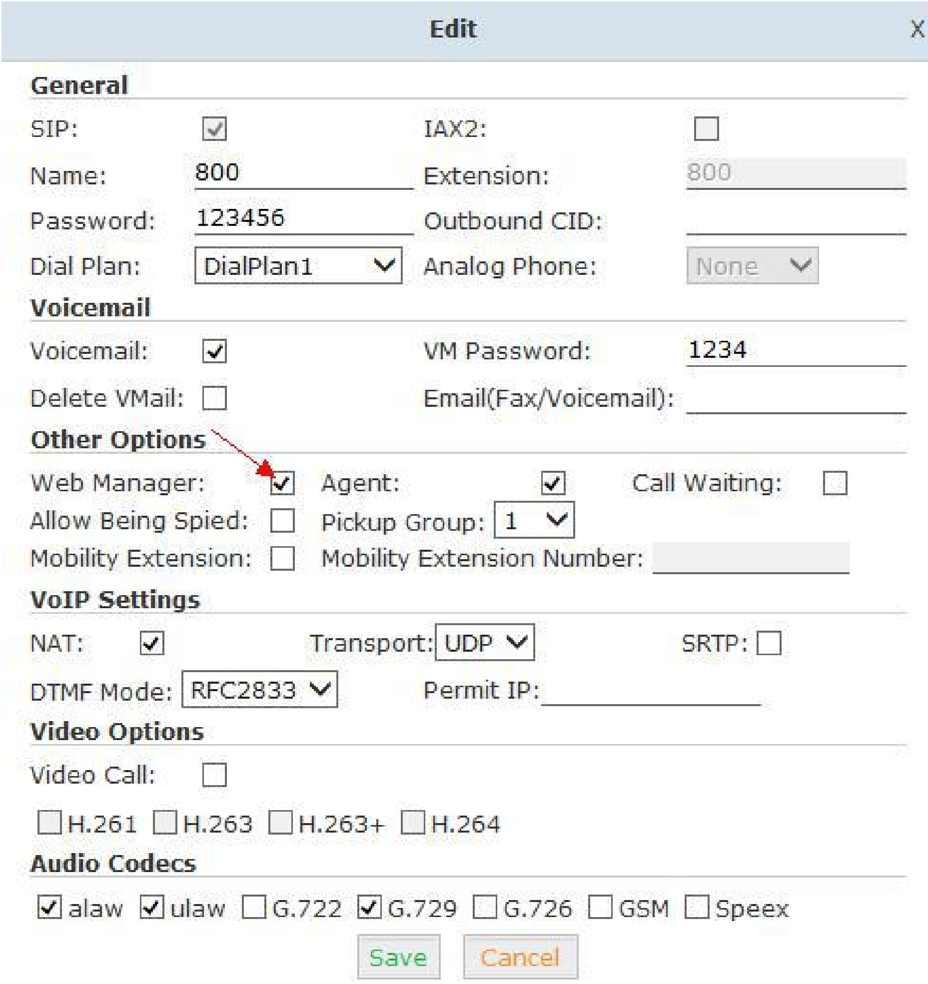
Item	Explanation
Blind Transfer	Default is #. Define in 【Feature Codes】
Attended Transfer	Default is *2. Define in【Feature Codes】
Complete Attended Transfer	Default is *, it can be used when you use *2. Define in 【Feature Code】
Timeout for answer on attended transfer	Default is 15 seconds. Define in【Feature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

2.6 User Extension Settings

2.6.1 Follow Me Settings

This feature allows a call to an extension to be automatically forward to one or more internal extensions or external phone numbers. To allow the user to configure these settings, first the user must be allowed access to the User Web Portal. To do this, select the "Web Manager" box under "Other Options".

Click【Basic】->【Extension】->【Edit】the extension you want to configure.



Check【Web Manager】 and【Save】

Then login the Extension Web Panel:

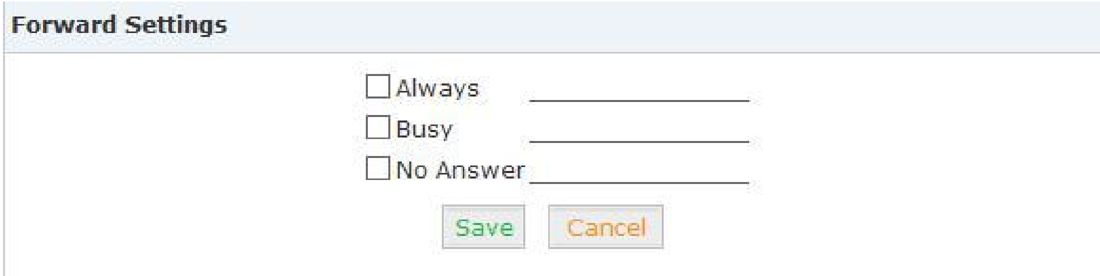
2.6.2 Call Recording

This feature allows users to access calls they have recorded. To configure this setting, please see the diagram below.

2.6.3 Call Forward

This feature allows calls to an extension to be automatically forwarded to a specific internal extensions or external phone number. To configure this setting, please see below:

Click【Call Forward】:



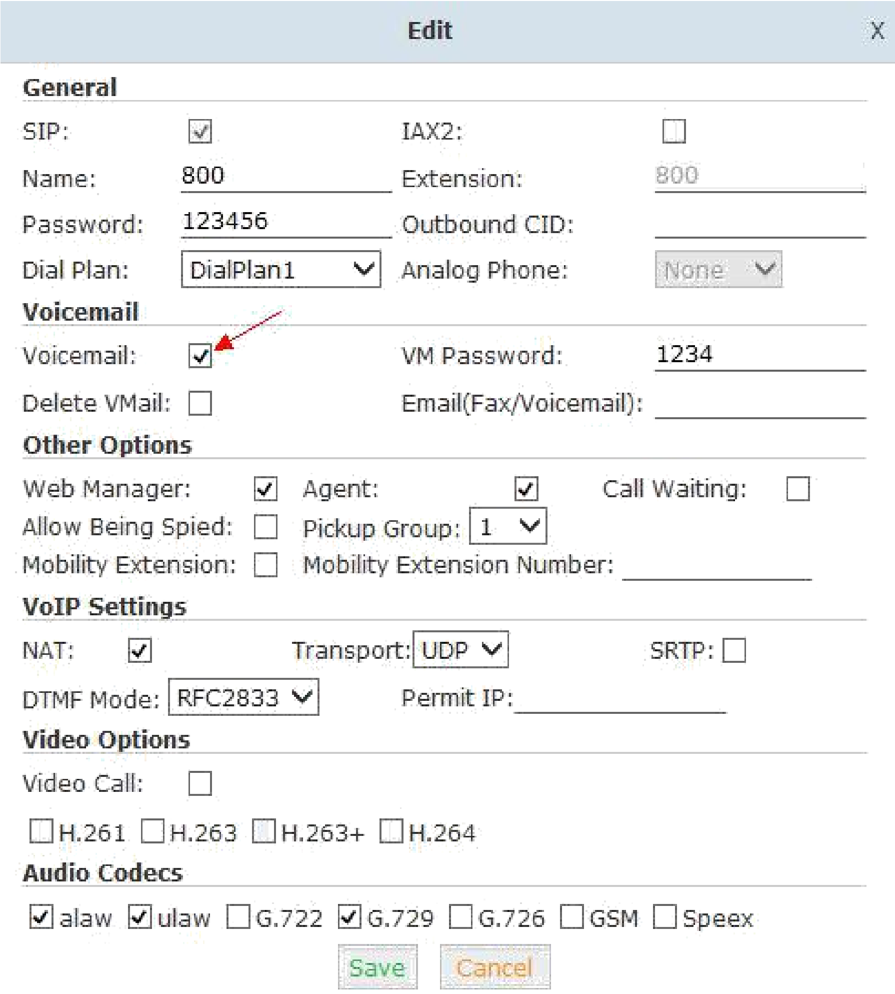
Reference

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

2.6.4 Voicemail

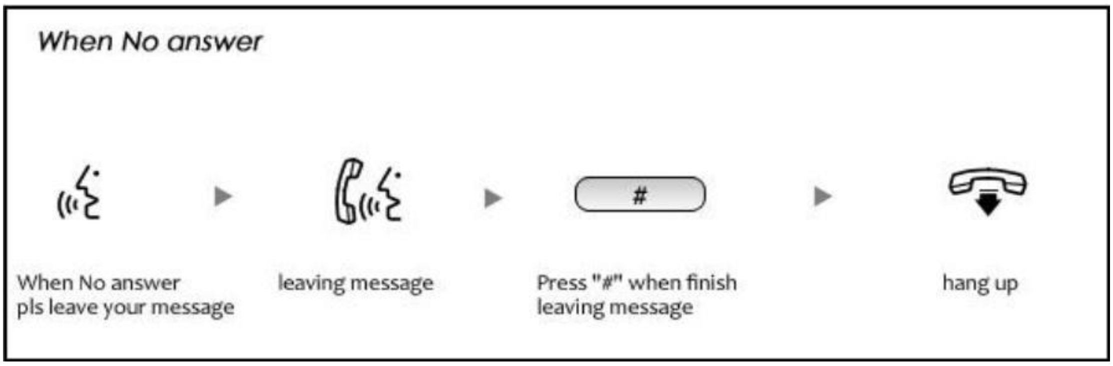
Calls that are not answered have the option to be sent to a voicemail account so the caller can leave a recorded message. Optionally, these recorded messages may be sent to a user's email account.

Click【Basic】->【Extension】->【Edit】the extension you want to configure.

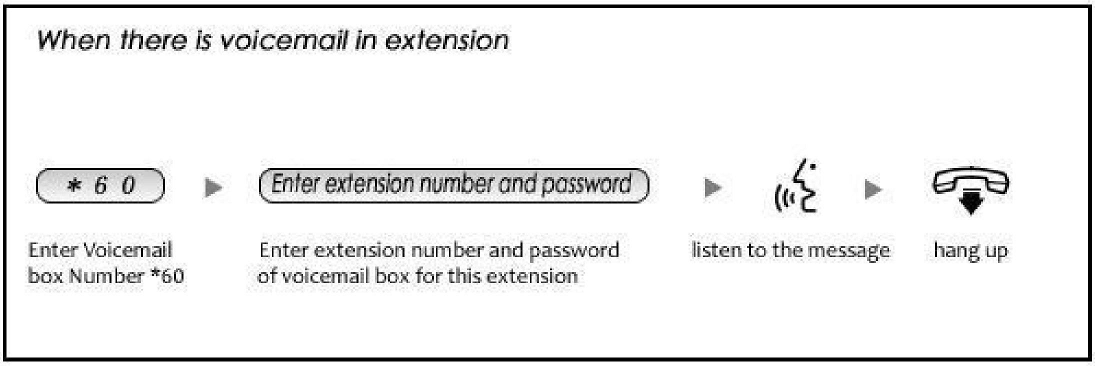


Please enable 【Voicemail】 before configuration, and configure 【VM Password】 and 【Email】. If incoming calls are not answered, when the default ring time is over, the system will play: “please leave your message and press the “#”key ”. Then voicemail will be sent to the specified mailbox by email.

To Leave a Message



To Listen to the message using the users desk phone



Notice:

1. Proper Email address is necessary to receive voicemail via email.

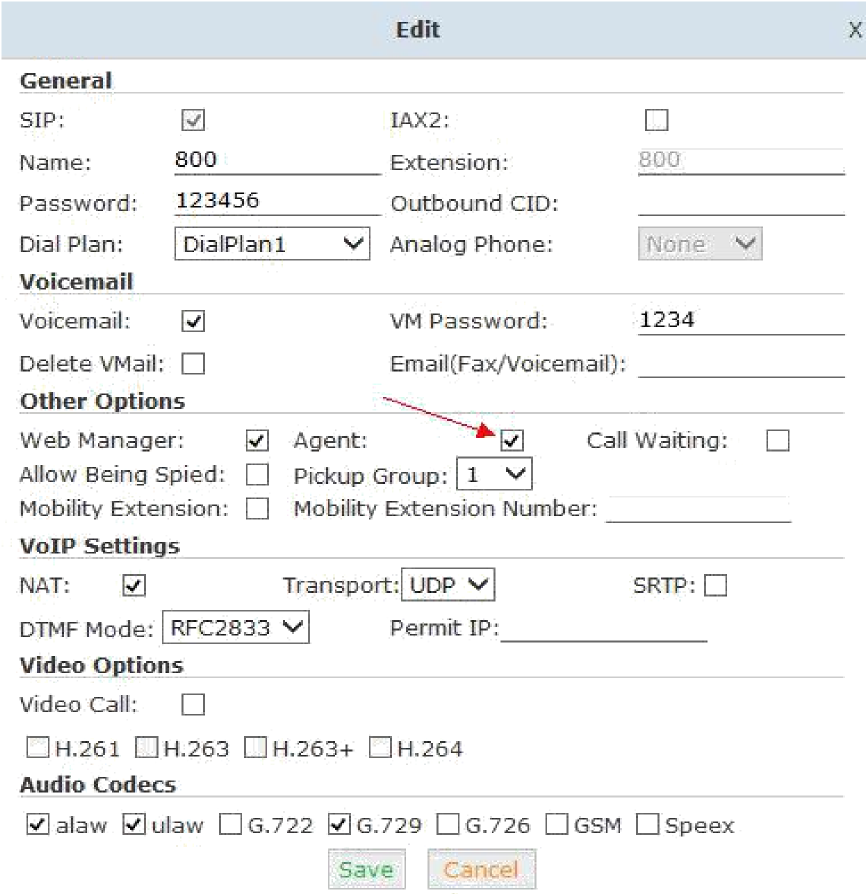
2. You must configure the SMTP and Email template. For detail settings, please see the detail configuration guide【Voicemail】in Chapter 3.

2.7 Call Center (Call Queues)

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.
Click【Basic】->【Extension】->【Edit】the extension you want to configure:

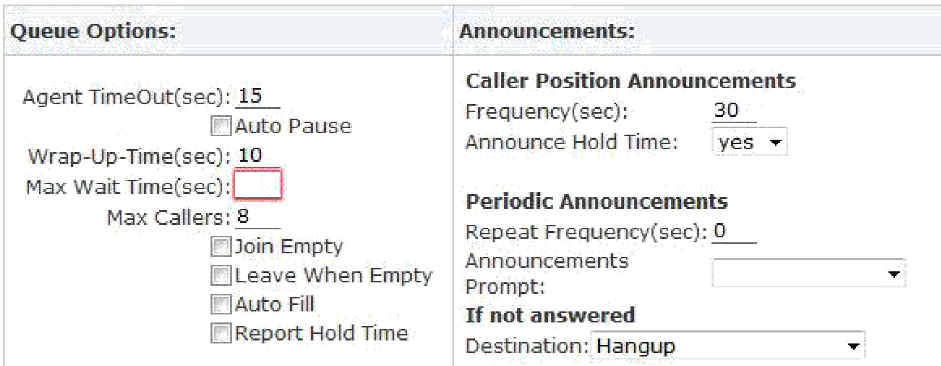
Step1: Check【Agent】and【Save】



Step2: Click【Inbound Control】->【Call Queues】

Reference

Item	Explanation
Queue Number	Define an extension number to identify the queue.
Label	Define the label for the queue.
Ring Strategy	RingAll--Ring all available agents until one answers(default) RoundRobin – Starting with the first agent, ring the extension of each agent in turn until the call is answered. LeastRecent – ring the extension of the Agent who has least recently received a call FewestCalls – ring the extension of the Agent who has taken the fewest number of calls. Random – ring the extension of a random Agent. RRmemory -- RoundRobin with Memory, like RoundRobin above, except instead of the next call starting with the first agent, the system remembers which extension was was called last and begins the round robin with the next agent .
Agent	Check each agent that is to be a member of this specific Call Center Queue.

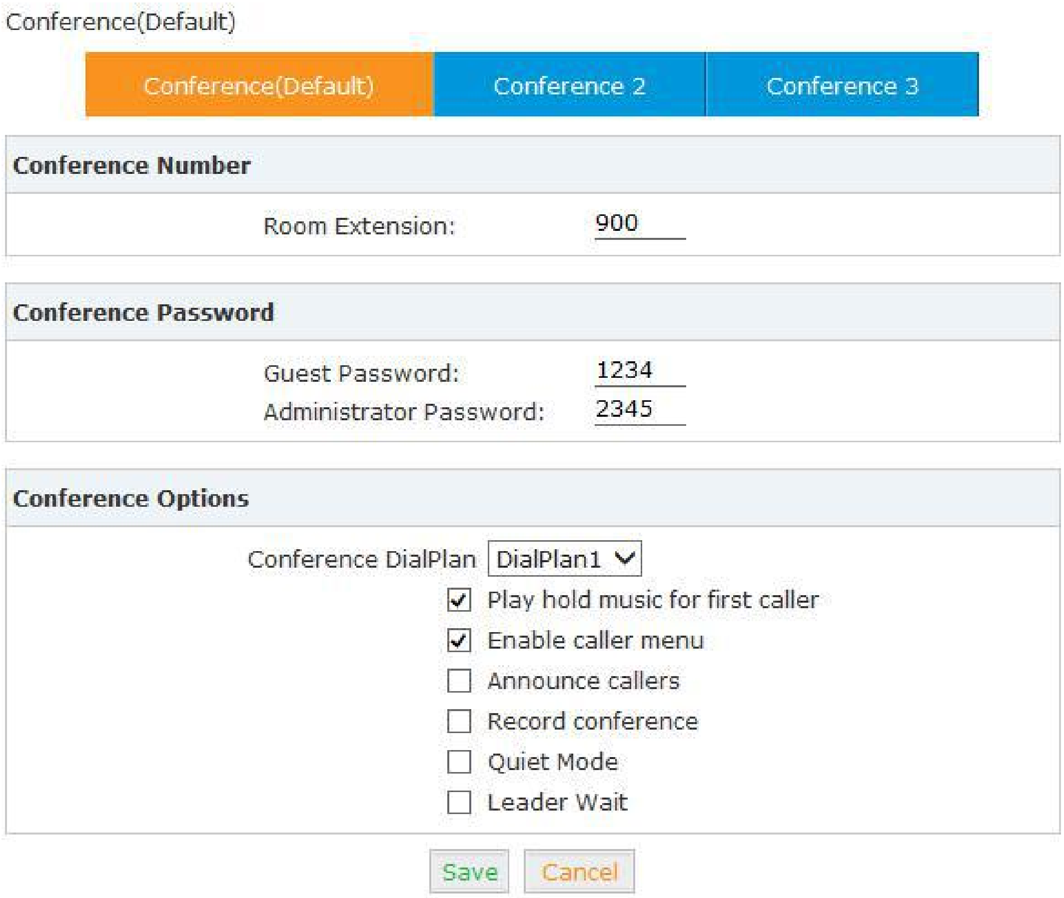


Reference:

Item	Explanation
Agent TimeOut(sec)	Specify the number of seconds to rin an agent's extension before sending the call to the next Agent (based on Ring Strategy).
Auto Pause	If an Agent's extension rings and the Agent fails to answer the call, automatically pause that agent so the stop receiving calls from the queue.
Wrap-Up-Time(sec)	This is the amount of time in seconds that an agent has to complete work on a call after the call is disconnected. (Default is 0, which means no wrap-up time.)
Max Wait Time(sec)	Calls that have been waiting in the queue for this number of seconds will be sent to the ""If not answered" destination.
Max Callers	Max number of the callers who are allowed to wait in the queue. (Default is 0, which means no limitation.). With this number of callers in the queue already, subsequent callers will be sent to the ""If not answered" destination.
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 - callers will be sent to the "If not answered" destination.
Leave When Empty	If this option is selected and calls are still in the queue when the last agent logs out, the remaining callers in the Queue will be transferred to "If not answered" destination. 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), do not announce (no) or announce once (once), it 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.

2.8 Conference Bridge

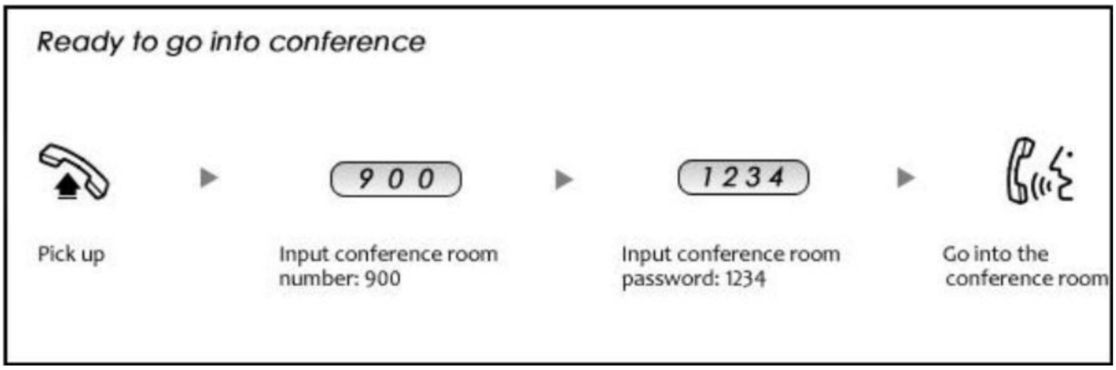
A conference bridge is a virtual meeting room that allows multiple callers to hear and speak to each other. The conference bridge can be protected with a password so only callers with the password can access the conference. The software supports up to three conference rooms. To configure a conference bridge, go to 【Advanced】->【Conference】:



Reference:

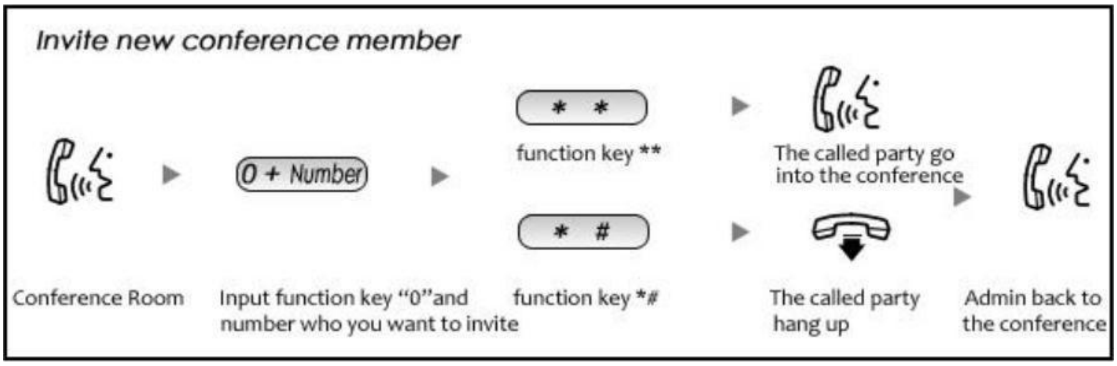
Item	Explanation
Conference Number	The number that internal callers use to access the conference room, the default number is "900".
Conference Password	Password for users to access the conference, e.g.:"1234".
Administrator Password	Password for administrator to access the conference.
Conference DialPlan	Use this dialplan 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 in 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 new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If check this option, all the participants in the conference can hear only, but it is not allowed to speak.
Leader Wait	Wait until the conference leader(administrator) entering the conference before starting the conference.

To join a conference, refer to the diagram as below:



While in a conference, the administrator can invite new guest (extension user or external number) into the conference. (Default password for admin is 2345)

As an administrator, to invite a new guest to the conference, refer to the diagram as below:



Chapter 3 Advanced

3.1 Options

General

Default settings for local extension and new extension.

Click【Advanced】->【Options】->【General】:

Reference

Item	Explanation
Operator Extension	Set extension number for Operator.
Global RingTime Set	Set RingTime for every extension.
Enable Transfer	Check to enable Transfer.
Enable Music On Ringback	Check to enable Music On Ringback.
Record Format	Set the format for recording files. (GSM/WAV only)
Defaut Setting for New User	Check to enable the default settings.
Extension Preferences	Set the rule for extensions.

Global Analog Settings

Click【Advance】->【Options】->【Global Analog Settings】:

Reference:

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. Polarity--Caller ID start when polarity reversal starts.
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 enable Busy Detection.

Global SIP Settings

【Global SIP Settings 】is appropriate for advanced administrators. Please contact our technical support department before modifying anything in this section.

3.2 Voicemail

Click【Advanced】->【Voicemail】->【General】:

Reference

Item	Explanation
MaxGreeting Time(sec)	Maximum recording length for voicemail greetings
Dial "0" for Operator	Select this option to allow callers to press Dial "0" to transfer out of voicemail to the Operator.
Message Format	Save the voice message as this format, WAV(16-bit) or Raw GSM.
Maximum Messages	Maximum voicemail messages to be allowed to leave.
Max Message Time(min)	Maximum Time for each message to be allowed to leave.
Min Message Time(sec)	MinimumTime for each message. The message will be deleted automatically if the time is less than the min message time.
Say Message CallerID	Play the Caller ID of the caller before playing the voice message.
Say Message Duration	Play the message duration before playing the voice message.
Play Envelope	Play the date, time and caller ID for the voicemail message.
Allow Users to Review	Check this option to allow users to review the voice message.

Click【Advance】->【Voicemail】->【Email Settings】

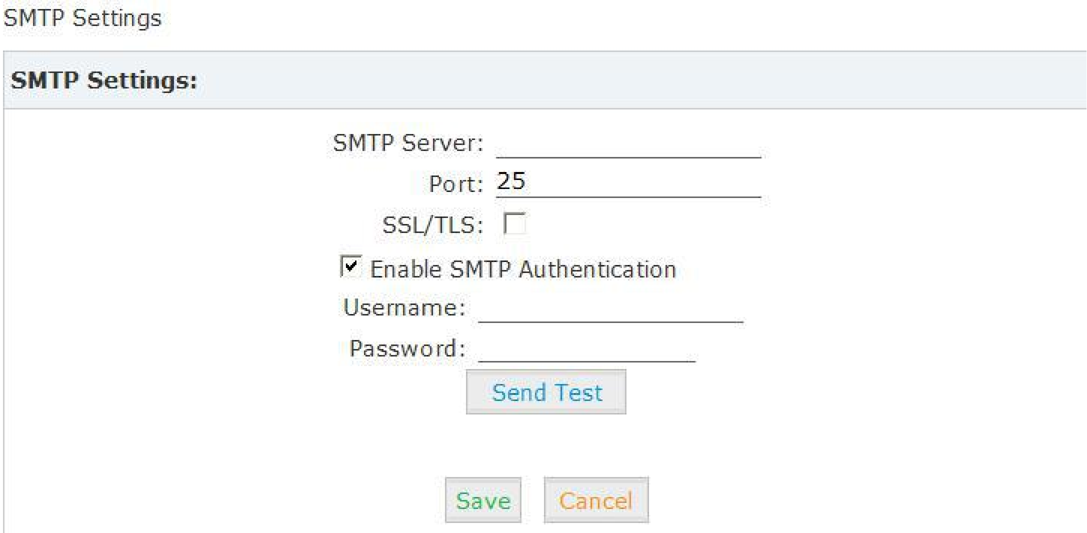
Reference:

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.

3.3 SMTP Settings

To allow email messages to be sent to users with attached voicemail and faxmail messages, the SMTP settings need to be configured.

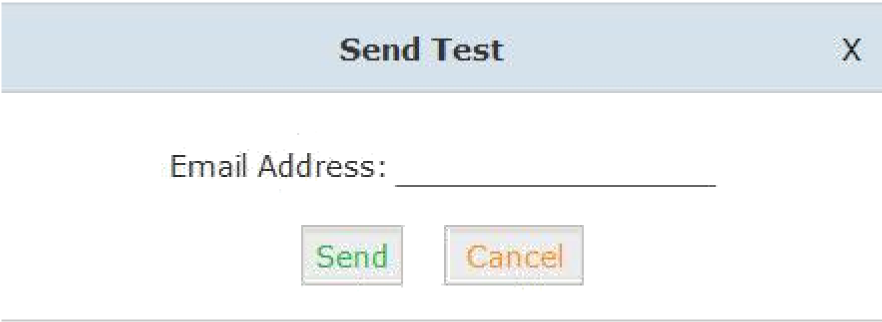
Click【Advance】->【SMTP Settings】:



Reference

Item	Explanation
SMTP Server	You must set SMTP Server address or domain connected to the LAVoice IP PBX, which is used for sending the voice message to Email.
Port	Port number for SMTP server. Default is 25, and it will be changed to 465 when you enable SSL/TLS.
SSL/TSL	Enable SSL/TLS.
Enable SMTP Authentication	If your SMTP server needs authentication, please enable this option, and configure the following.
Username	Input username of your Email.
Password	Input password of your Email.

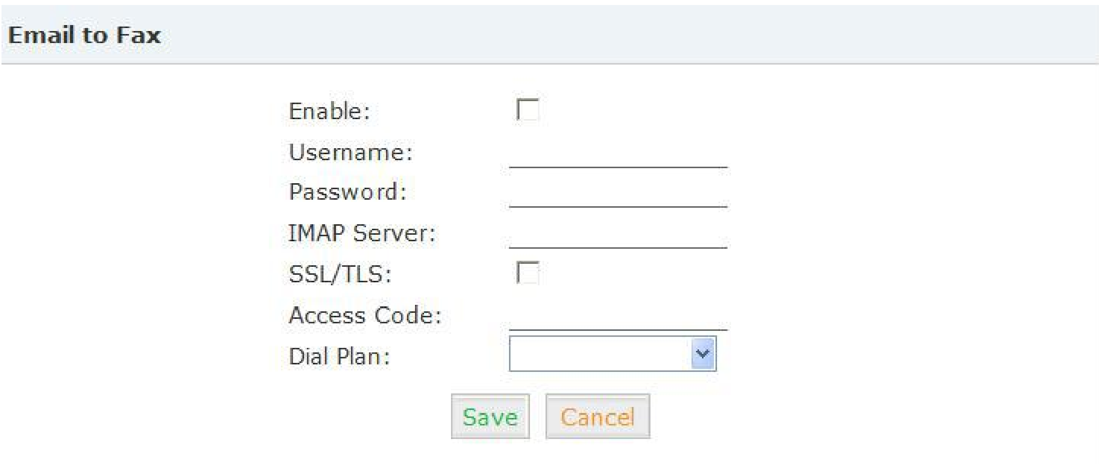
Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.



Specify the email address and click【Send】-to send the test email. Verify that email was successfully sent or not. If no email was received, please modify the SMTP settings and retry.

3.4 Email to Fax

Users can send fax by Email. Please configure as below.
Click【Advanced】->【Email to Fax】



Check "Enable", input username, password and IMAP Server(server format: imap.XX.com), select the DialPlan, then "Save" and "Activate".

Practical Case:

To Send a fax to telephone number 85337096: In DialPlan 1, there is prefix "9" before the telephone number; you need input the 【Access Code】: 985337096 and make this the subject when sending Email. Then the fax will be sent by Email as attachment.

If you need dial the extension when sending fax, e.g.: fax number: 85337096 ext.800, you need use the 【Access Code】: 985337096-800 as subject.

3.5 Music Settings

Management of Music on Hold, Music on Ringback, Music on Queue.
【Music Settings】:

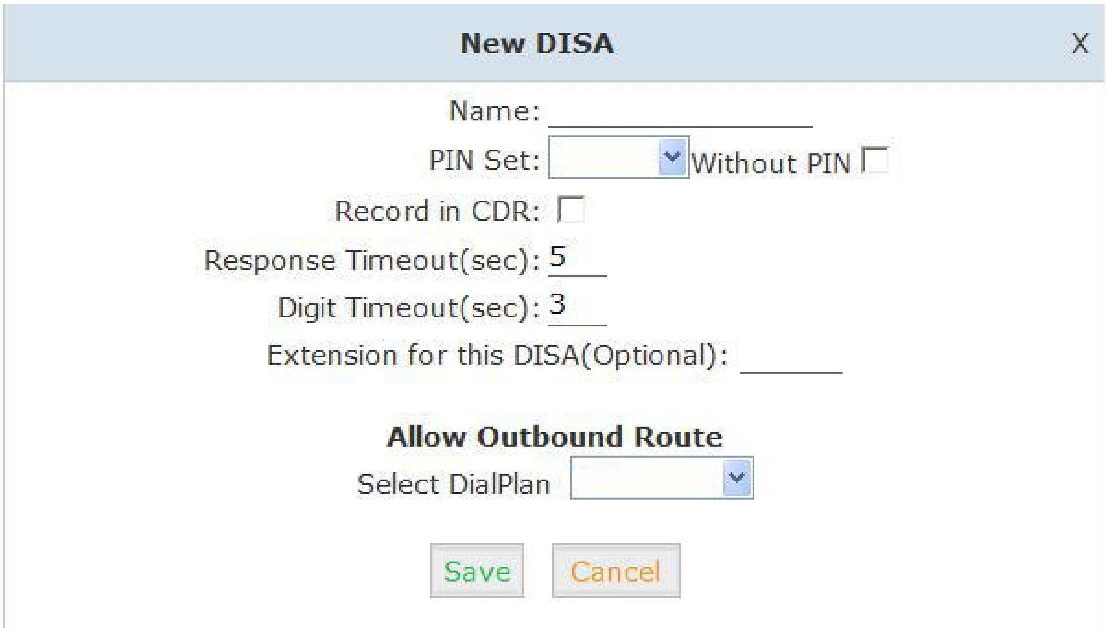
Select the different music file for different Music.
【Music Management】

Item	Explanation
Select Music Directory	Select which Music Directory you wish to load.
File	Display music name under the music file, you can delete it.
Select Music Directory	Select the file where you want to save your uploaded music.
Please choose file to upload	Select the music you want to upload. Note: music file must be WAV(16bit/8000Hz/Single), GSM, ulaw or alaw, and less than 15MB.

3.6 DISA

This feature allows an authorized user to call into the PBX and then place an outbound call using another trunk. For example, an employee working out of the office who needs to make an international call using trunks connected to the PBX. By calling the DISA number, after PIN authentication, the caller hears dial tone and can dial the call.

Please configure as below.
Click【Advance】->【DISA】--【New DISA】



Reference

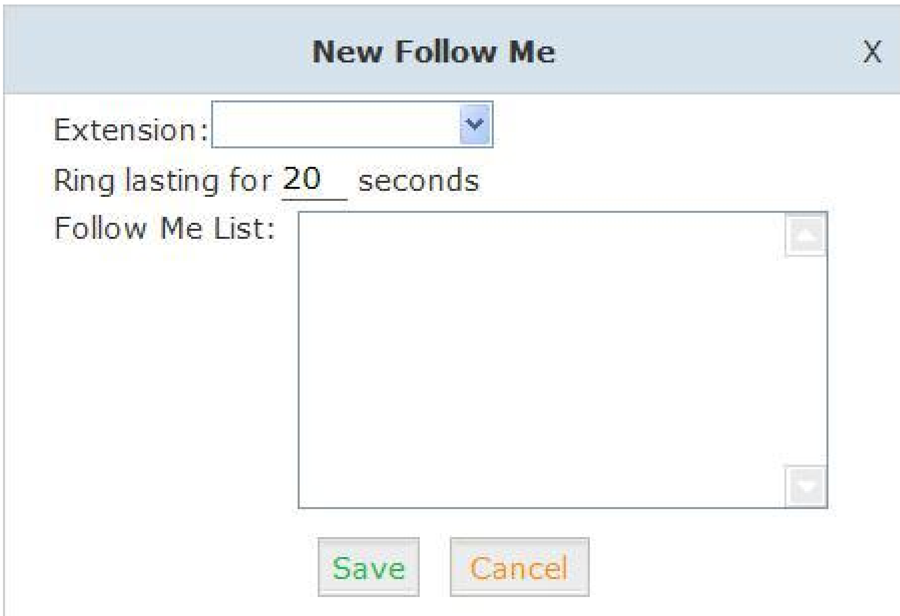
Item	Explanation
Name	Define a name for DISA.
PIN Set	User will be prompted to input this number when PIN Authentication is needed.
Record in CDR	Check to record.
Response Timeout(sec)	The maximum time for waiting before hanging up if the dialed number is

	incomplete or invalid. Default is 10 seconds
Digit Timeout(sec)	The maximum interval time between digits when typing extension number. Default is 5 seconds.
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 DialPlan	Select the DialPlan for this DISA.

3.7 Follow Me

This feature allows callers to automatically be forwarded to one or more internal extensions and/or one or more external phone numbers when the call is not answered at the primary extension.

Please configure as below:
Click【Advanced】->【Follow Me】->【New Follow Me】:



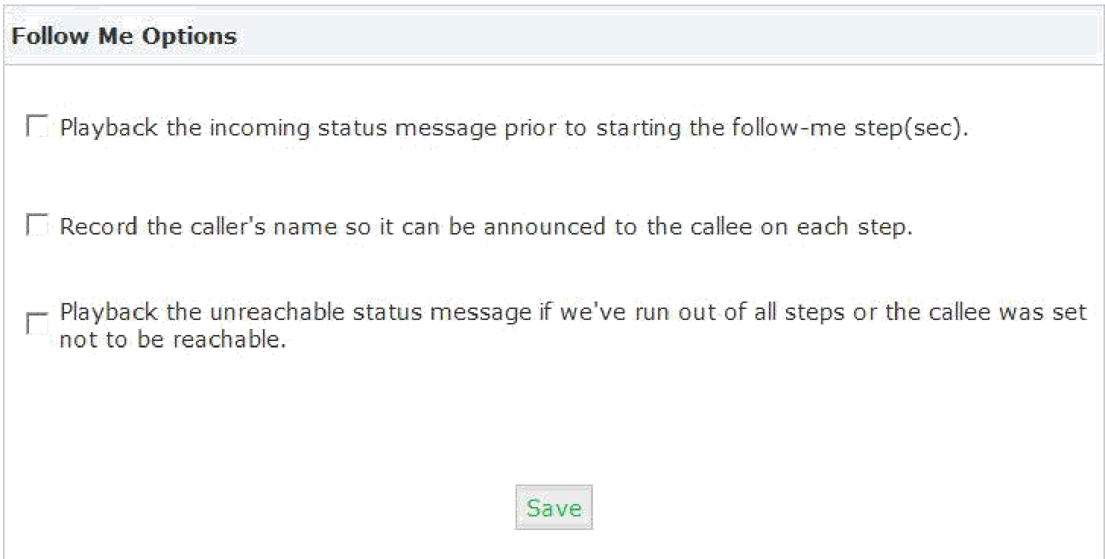
Select an extension, set the ring duration, and add the numbers in the Follow Me List;【Save】 and 【Activate】.

List Format: Extension Number, Ring Duration

E.g.: 806,30
808,20

806 rings, after 30 seconds, the call is going to 808

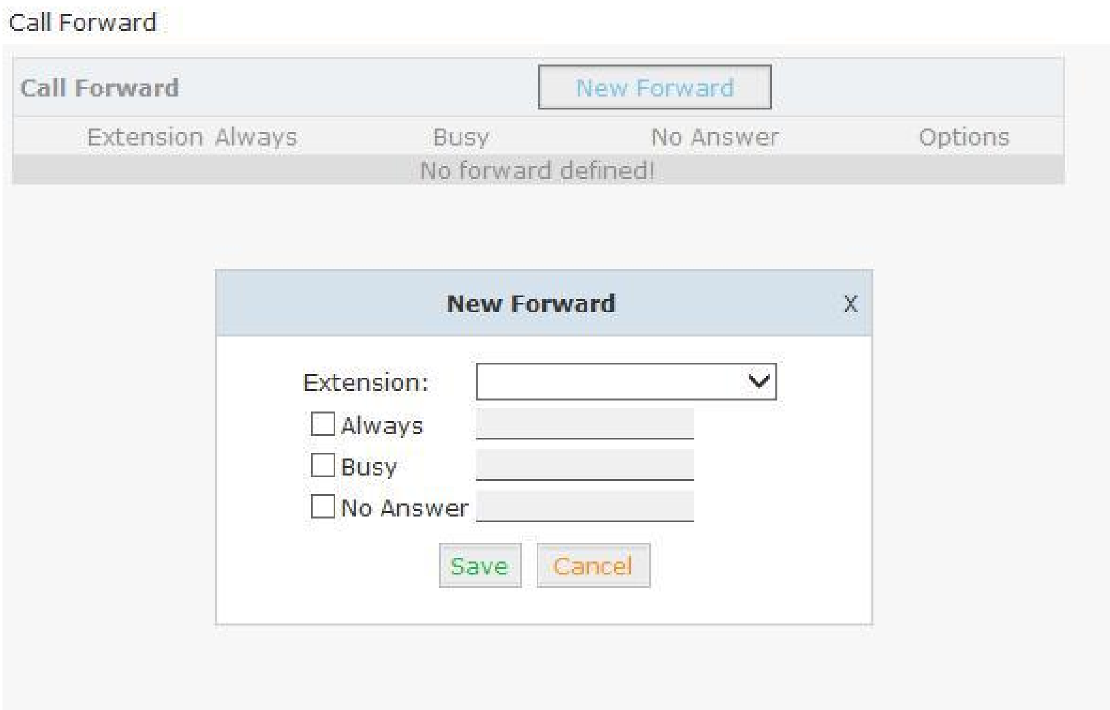
【Follow Me Options】



3.8 Call Forward

The administrator can config the Call Forward on this page:

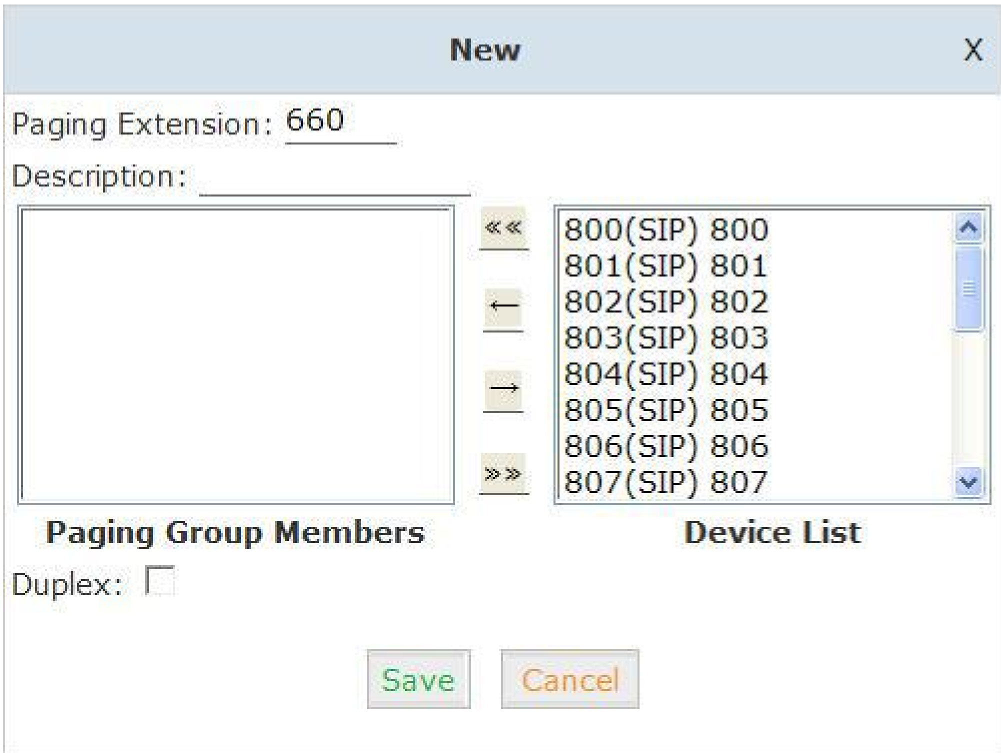
Click【Advanced】->【Call Forward】:



3.9 Paging and Intercom

This feature allows setting up a Paging group so when the Paging extension is dialed, the listed extensions allow the caller to speak through the speaker phone. The extensions in the Paging group must use phones that support this feature. If the Duplex option is selected, and

the listed extensions use phones that support Duplex, then all the phones in the paging group will be able to have two-way conversations.
Click【Advanced】->【Paging and Intercom】->【New Paging Group】:



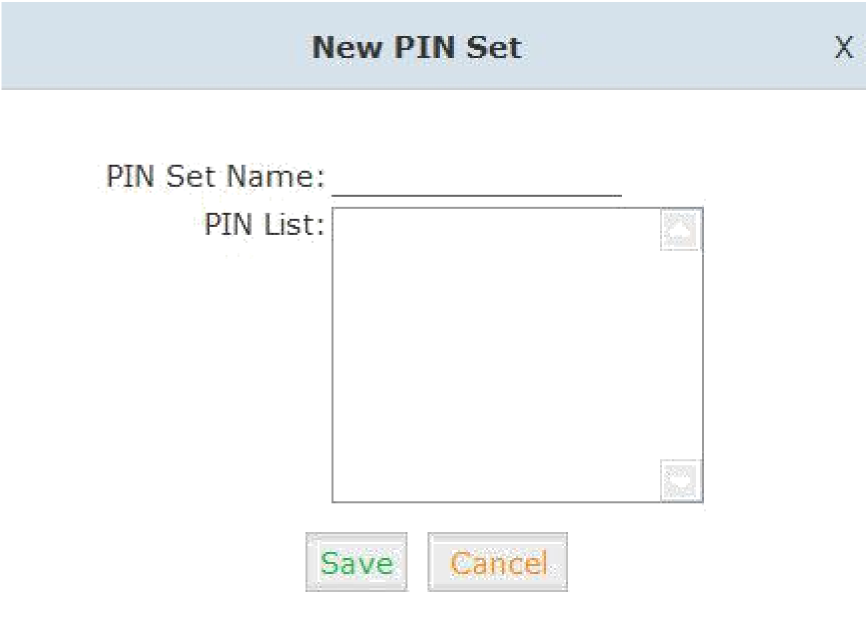
Reference:

Item	Explanation
Paging Extension	Define an extension for this Paging Group.
Description	Define a name for this Paging Group.
Paging Group Members	Selected devices in this Paging Group.
Device List	Select device(s) here to Paging Group.
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 look like an "instant conference".

3.10 PIN Sets

This feature allows an administrator to specify a list of PIN codes in a PIN Set. An Outbound Route can be specified that a valid PIN code from a selected PIN Set must be used in order to have access to a give Outbound route (e.g. for long distance or international calling).

Please configure as below.
Click【Advanced】->【PIN Sets】->【New PIN Set】:

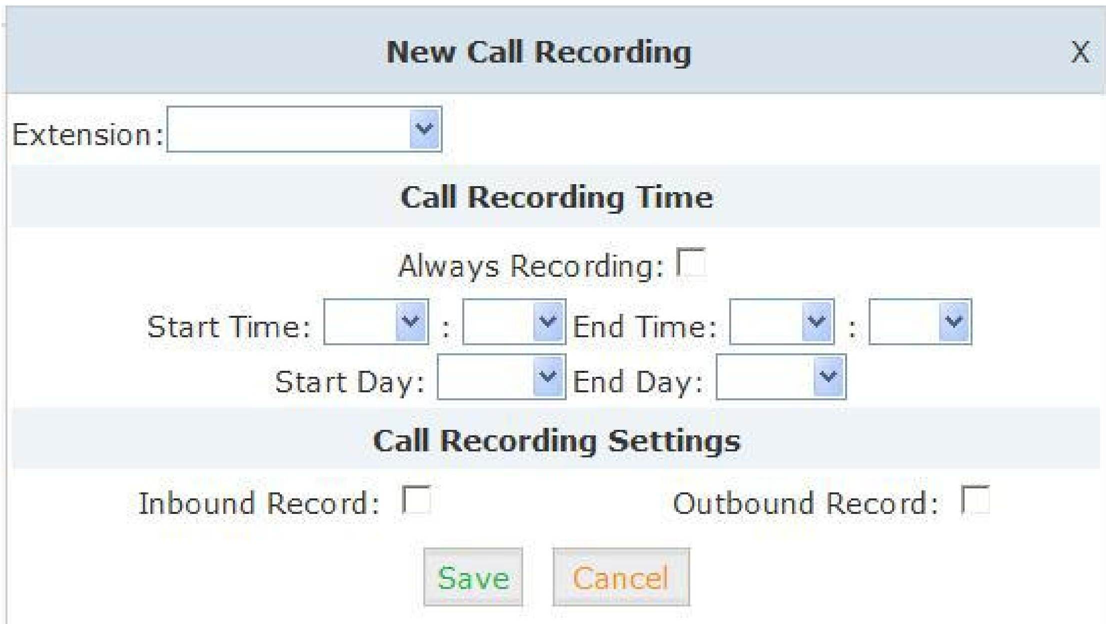


1. PIN Set Name Define the name for this PIN Set.
2. PIN List Define PIN codes in this list.

3.11 Call Recording

This feature allows an administrator to enable Call Recording to record incoming and/or outgoing calls related to the specified extension.

Please configure as below:
Click【Advanced】->【Call Recording】->【New Call Recording】:



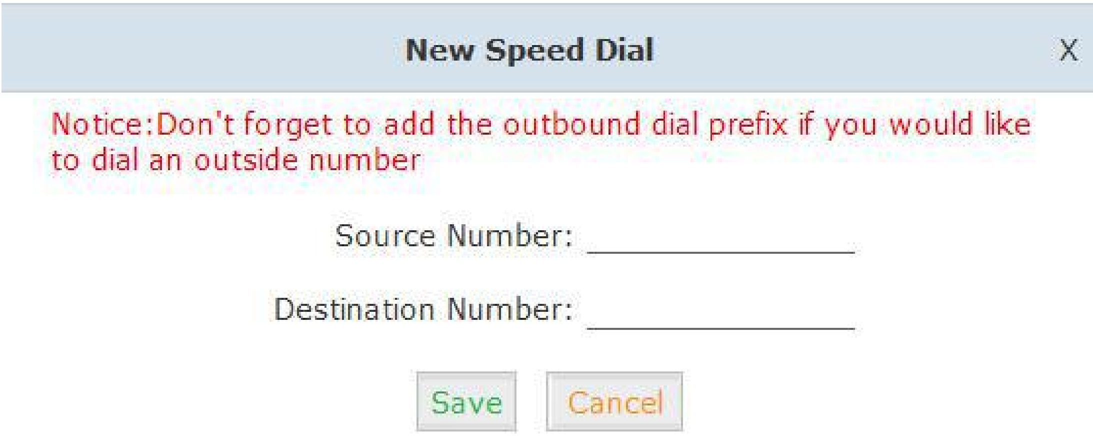
Reference:

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.

3.12 Speed Dial

This feature allows setting up system wide speed dial numbers that translate a feature code (*99) plus a two-digit code (00-99) into an external phone number.

Please configure as below.
Click【Advanced】->【Speed Dial】->【New Speed Dial】:

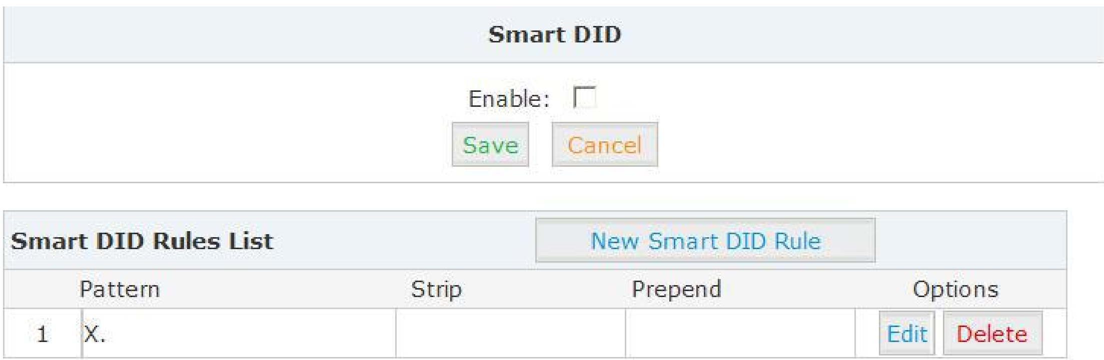


E.g.: prefix is *99 , speed number is 00, destination telephone number is 85337096. When dial *9900, the call is going to 85337096 automatically.

3.13 Smart DID

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

Click【Advanced】->【Smart DID】:



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

Click【New Smart DID Rule】to display the following diagram:

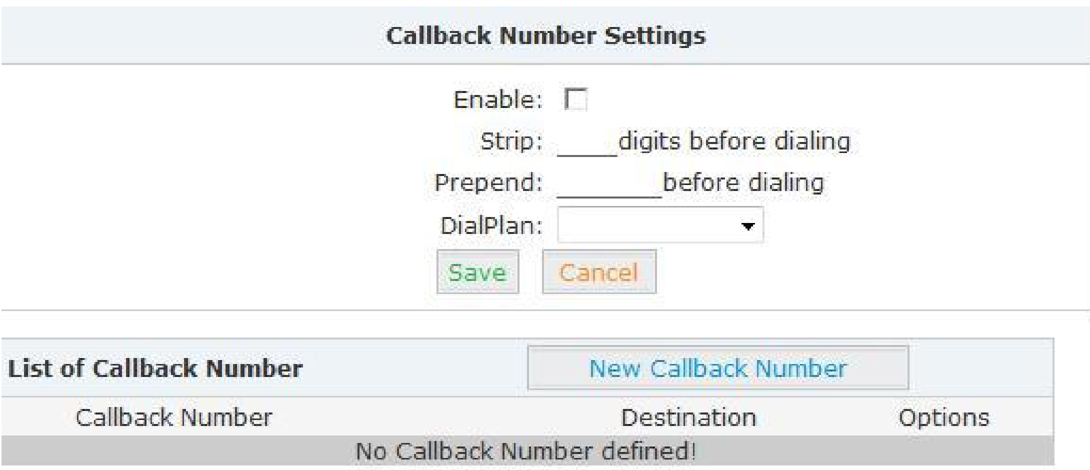
Input the pattern and define how many digits need to be stripped or prepend, then click “Save”--“Activate”.

3.14 Callback

This feature allows an external caller to place an inbound call to the IP PBX. The inbound call will be disconnected and subsequently the PBX will place an outbound call back to this number and forwarded to defined destination after the call is connected.

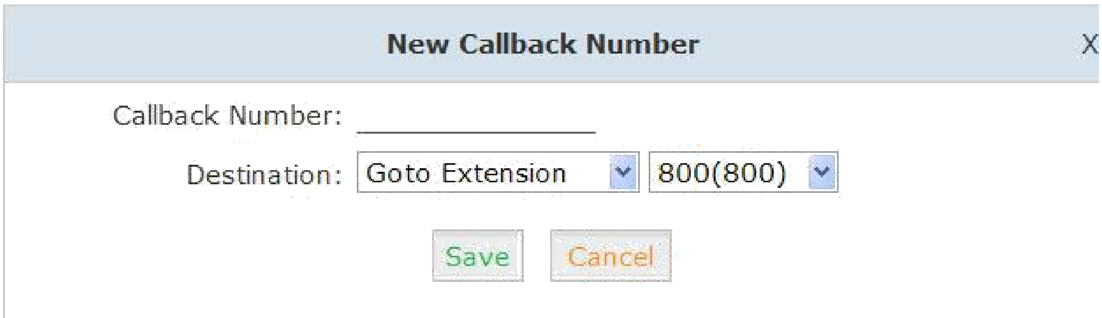
Please configure as below.

Click【Advanced】->【Callback】:



Enable this function; select DialPlan, and define the callback rule (strip digits or prepend prefix).

Click【New Callback Number】to add callback number.

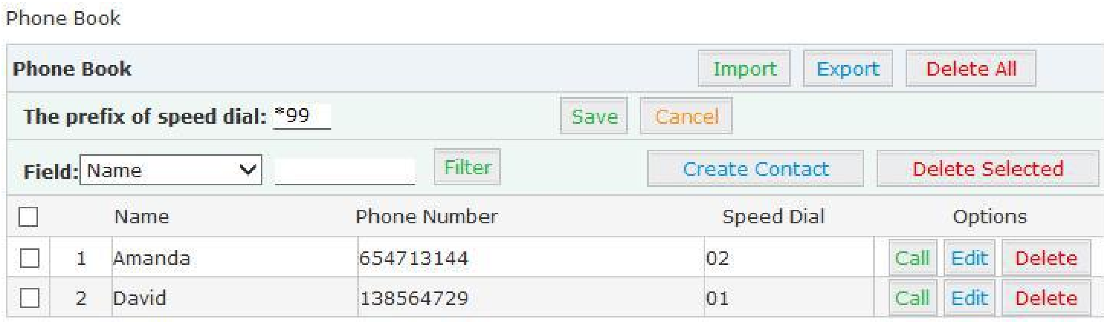


Input callback number and define the destination.

3.15 Phone Book

When incoming call Caller ID matches the number in the phone book, the name of matched number will be displayed. Please configure as below.

Click【Advanced】->【Phone Book】:



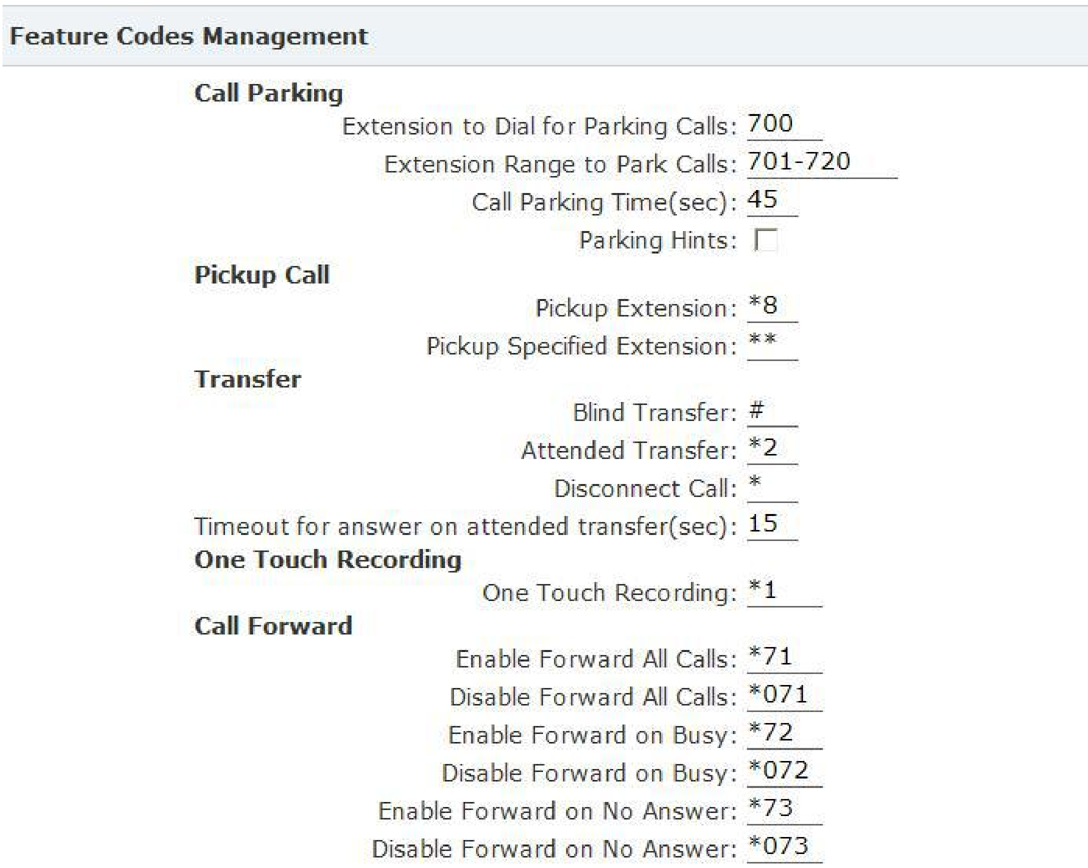
Reference:

Item	Explanation
Import & Export	Import & Export a list, make sure it's UTF-8 coded if it's not in English
Delete All	Delete all the contacts from the phone book
The prefix of speed dial	Set the prefix of speed dial
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

Click【Create Contact】to see the following diagram:

3.16 Feature Codes

Click【Advanced】->【Feature Codes】to see the following diagram, and you can define the code for each feature.



Reference:

Item	Explanation
Extension to Dial for Parking Calls	Define an extension for parking calls.
Extension Range to Park Calls	Define the extension range for parking calls. (e.g.: 701-720)
Call Parking Time(sec)	Define the time for parking calls. LAVoice IP PBX will return the call to the extension after this time limit has expired.
Pickup Extension	This feature code will pick up a call given that the callers extension and the ringing extension are in the same pickup group and call group.

Pickup Specified Extension	This feature code allows a caller to Pickup a call ringing on the specified extension. Default: Dial**+extension number to pickup the specified extension.
Blind Transfer	To Allow unattended or blind transfer while on a call based on the following steps: 1. While on a call with caller “A”, the user dials the blind transfer key sequence (in this case “#”). The system places the original call with “A” on hold, says "Transfer" then gives a dial tone. 2. dial the transferee extension or phone number you wish to transfer the call to “B” and hangup the phone. 3. The original caller “A” is transferred immediately to the transferee “B” and “B” sees the callerid of “A”.
Attended Transfer	To Allow attended or supervised transfer while on a call based on the following steps: 1. While on a call with caller “A”, the user dials the supervised transfer key sequence (in this case “*2”). The system places the original call with “A” on hold, says "Transfer" then gives a dial tone. 2. dial the transferee extension or phone number you wish to transfer the call to “B” and wait for “B” to answer the phone and talk to “B” to introduce the call. 1. If “B” does not wish to take the call, “B” can hang up the call and you are returned to your call with “A”. 2. If “B” wishes to accept the call, you hang up the phone and caller “A” is transferred to the transferee “B”. 3. If the call goes to voicemail or you wish to abort the transfer, simply press the “disconnect call” key sequence (in this case “*”) and the transfer will be aborted and you will be back on the call with the original caller “A”.
Disconnect Call	Disconnect the current transfer call (for Attended transfer).
Timeout for answer on attended transfer（sec）	Set the timeout value
One Touch Recording	Configure the function key for One Touch Recording
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable “Do Not Disturb”
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklist number.
Voicemail	Configure the function keys for entering voicemail and check extension voicemail.
Invite Participant	In conference, the administrator can invite people into the conference by dialing “0”. After pressing “0”, you will get dialtone, and you can dial to invite people. After the call is connected, please press ** to direct the people into the conference, or *# to hang up the current call and return to the conference.
Create Conference	During the call, you can dial *0 to forward to the conference with the callee.
Return to conference with participant	In conference, the administrator can dial “0” to invite people into the

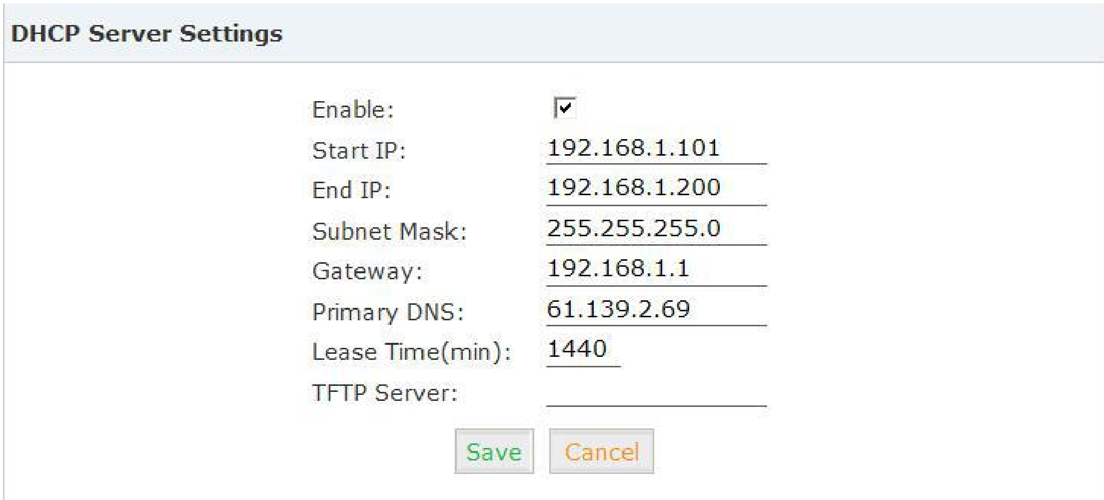
	conference. After pressing "0", you will get dialtone, and you can dial to invite the participant; when the call is connected, dial "**" to return to the conference with invited participant.
Return to conference without participant	In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dialtone, and you can dial to invite the participant. When the call is connected, you can dial "*#" to hang up and return the conference yourself.
Pause Queue Member Extension	Pause the agent, and the agent cannot receive the call.
Unpause Queue Member Extension	Unpause the agent, and the agent can receive the call.
Others	Function key for Intercom/ Paging/ Directory

3.17 IP Phone Provisioning

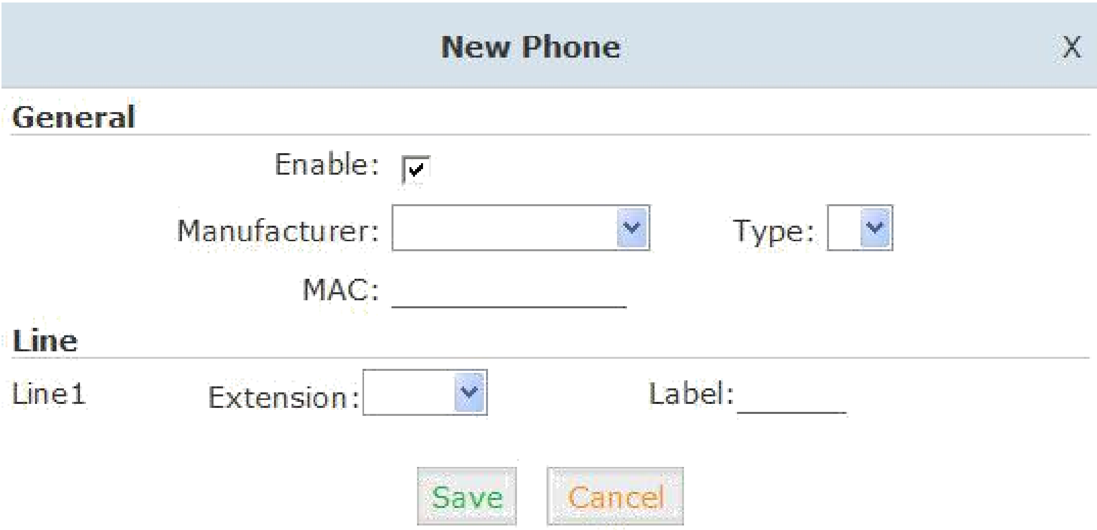
When many IP Phones are needed, please record the MAC, extension number, and username of each phone according to the format (please take reference of the auto provision script file model for details) , then import the format file, once the phone is connected to the local network, it will get the extension 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 following diagram:



Then Click【Advanced】->【Phone Provisioning】->【New Phone】:



Enable Phone Provisioning in【Basic】, select the IP Phone manufacture, input MAC of the phone, and select the extension for provisioning.

Notice

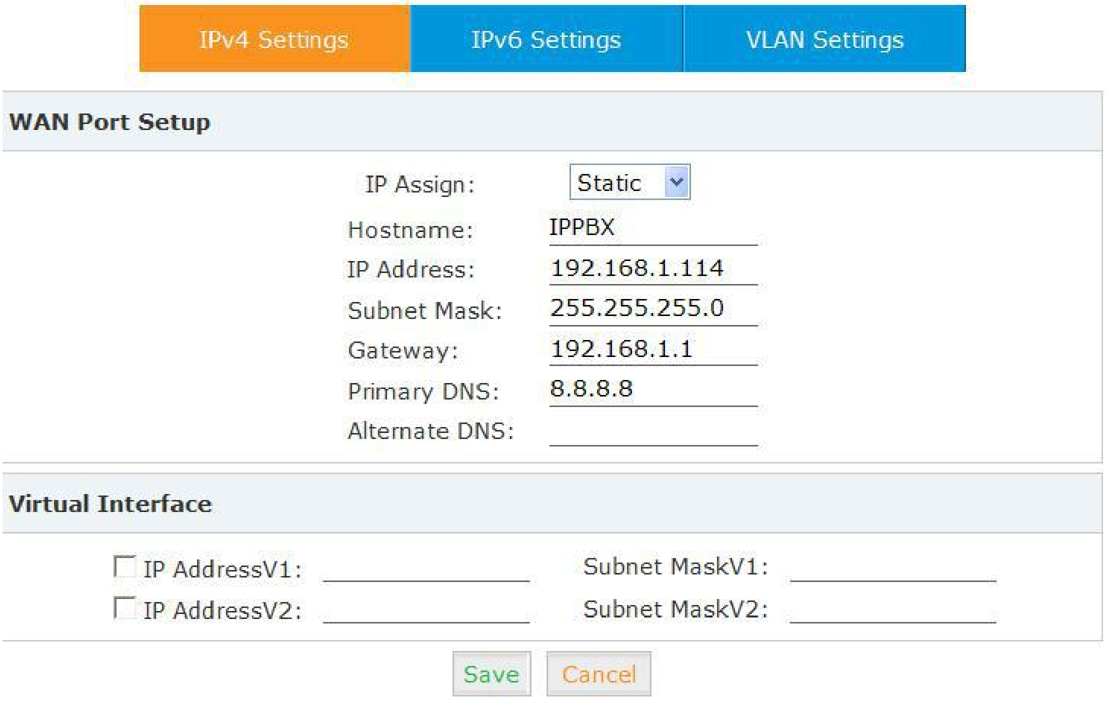
LAVoice IP PBX supports IP Phones from Fanvil/ Yealink/ Grandstream now.

Chapter 4 Network Settings

4.1 Network

You can configure the WAN Port, and define the Virtual Interface.

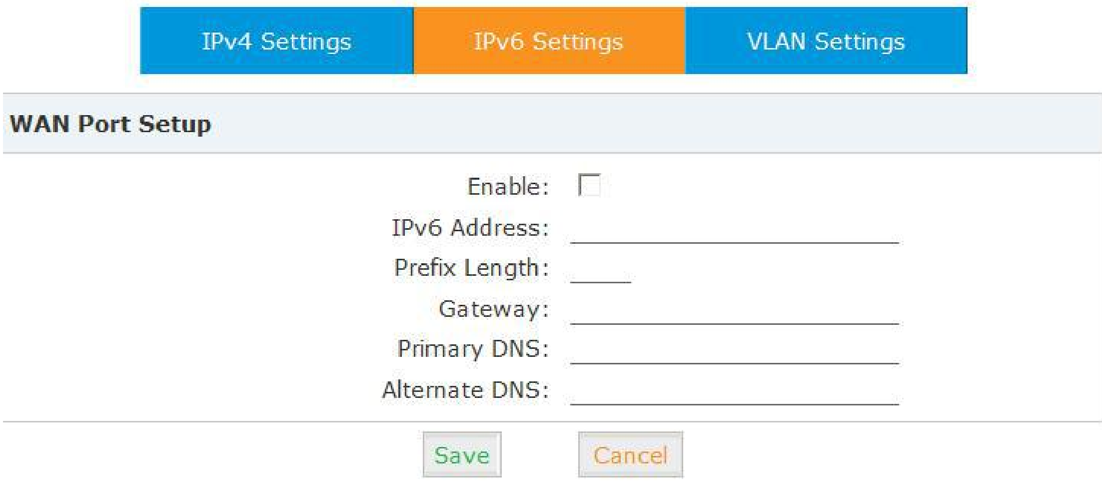
Click【Network Settings】->【Network】->【IPv4 Settings】:



Reference

Item	Explanation
IP Assign	Static/ DHCP/PPPoE supported.
Virtual Interface	Define the virtual interface for WAN Port.

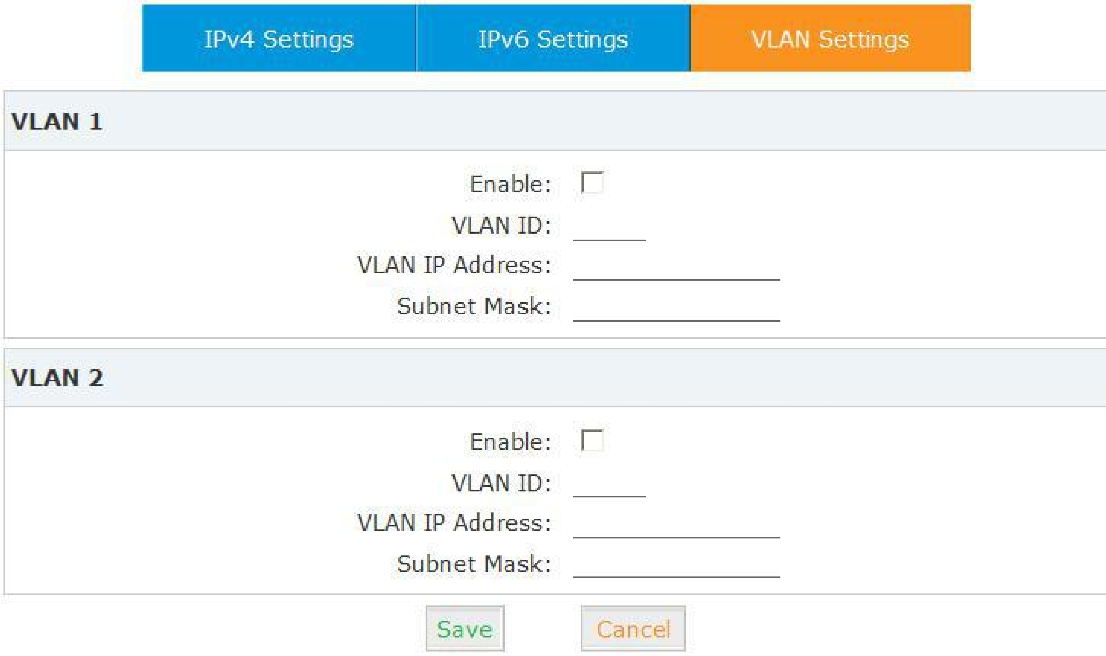
Click【Network Settings】->【Network】->【IPv6 Settings】



IPv6 Reference:

Item	Explanation
Enable	Enable IPv6, define the IPv6 address, gateway, and DNS.

Click【Network Settings】->【Network】->【VLAN Settings】:

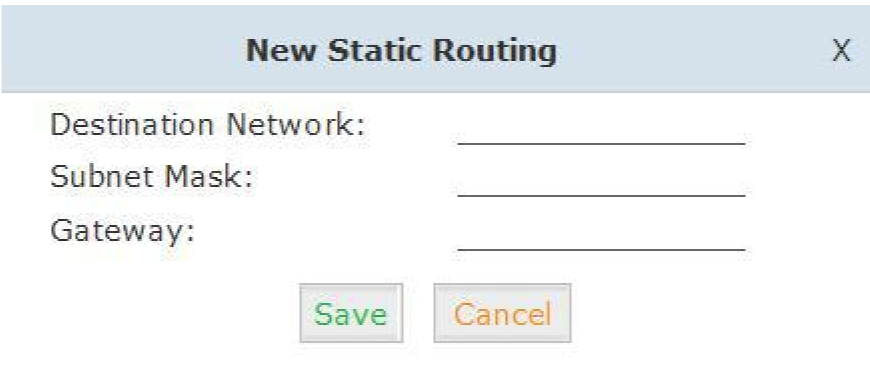


VLAN Reference:

Item	Explanation
Enable	Enable VLAN, define the VLAN address and VLAN ID.

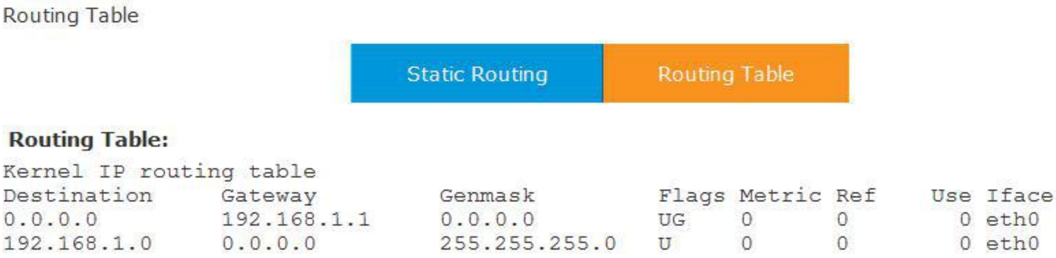
4.2 Static Routing

Click【Network Settings】->【Static Routing】:



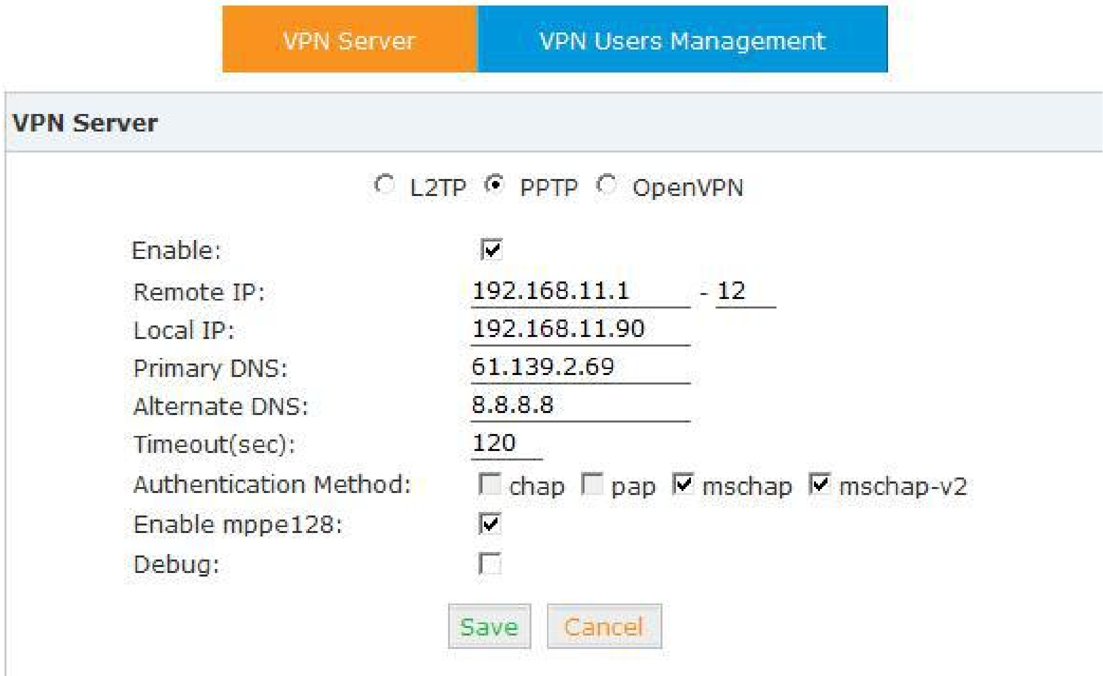
Item	Explanation
Destination	Set destination network for static routing.
Subnet Mask	Set subnet mask of the destination network.
Gateway	Define the gateway accessing the destination network.

Click【Network Settings】->【Static Routing】->【Routing Table】, the current routing information will be displayed as below:



4.3 VPN Server

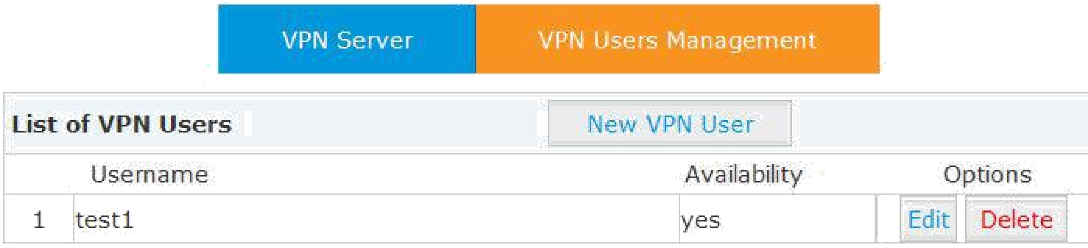
IP PBX supports three kinds of VPN servers: L2TP/PPTP/OpenVPN.
Click【Network Settings】->【VPN Server】:



Reference:

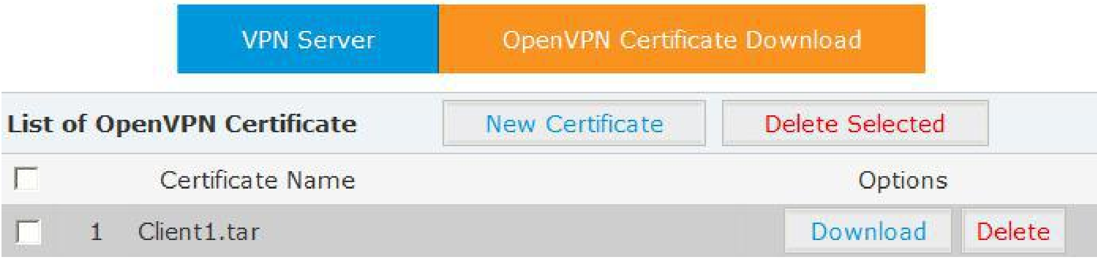
Item	Explanation
VPN Server Mode	Three kinds of VPN servers L2TP/PPTP/OpenVPN supported（Only one mode can be enabled simultaneously)
Enable	Enable/Disable VPN Server

When the mode is L2TP or PPTP VPN server, click【Network Settings】->【VPN Server】->【VPN Users Management】:



This page is used for management of VPN username and password.

When the mode is OpenVPN server, click【Network Settings】->【VPN Server】->【OpenVPN Certificate Download】：



This page is used for management of OpenVPN certificate file.

4.4 VPN Client

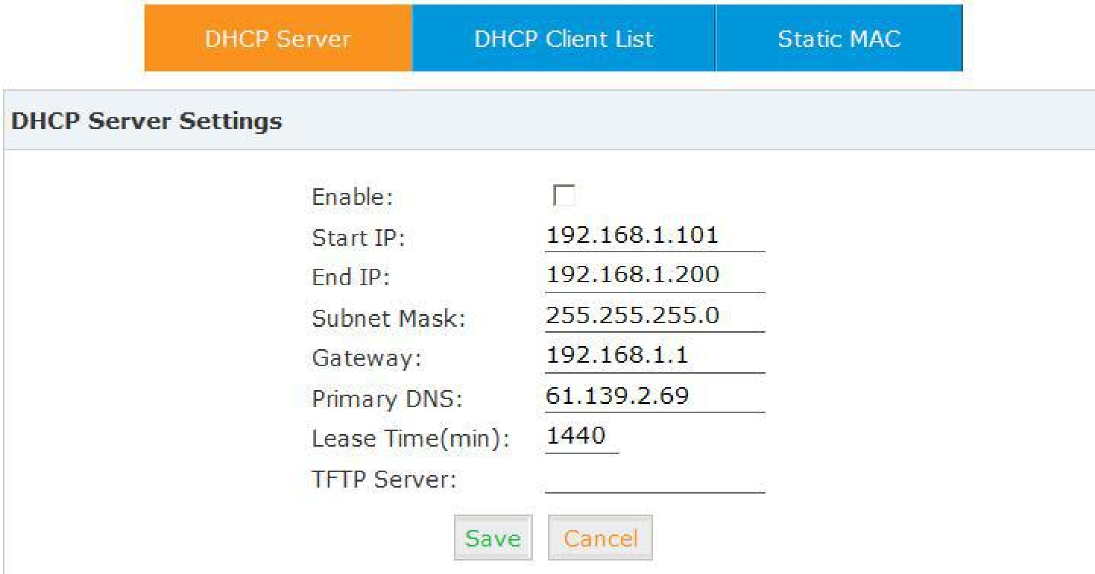
IP PBX supports four kinds of VPN Clients: L2TP /PPTP /OpenVPN /N2N
Click【Network Settings】->【VPN Client】：

Reference:

Item	Explanation
VPN Client	Four kinds of VPN Clients supported: L2TP/PPTP/OpenVPN/N2N（Only one mode can be enabled simultaneously）
Enable	Enable/Disable VPN Client

4.5 DHCP Server

Click【Network Settings】->【DHCP Server】：



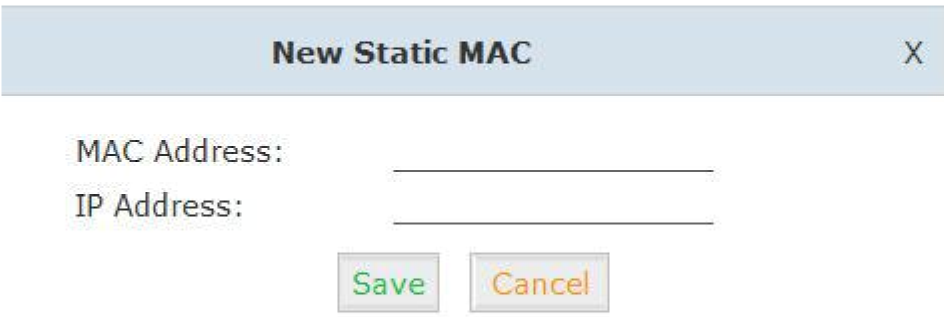
Click【Network Settings】->【DHCP Server】->【DHCP Client List】:

Mac Address	IP Address	Host Name	Expires in
6c:3e:6d:e0:f2:00	192.168.1.101	iPhone	expired
00:03:58:45:87:9a	192.168.1.102		expired
0c:74:c2:47:71:6d	192.168.1.103	hnteki-iPhone	expired
20:c9:d0:85:3b:fb	192.168.1.104		expired
08:ed:b9:e7:c5:7f	192.168.1.105	DPVYE1J0WCAAC7I	expired
78:e4:00:8e:c3:99	192.168.1.106	LBSZLACHCIC	22:10:25
68:a3:c4:ef:5d:8b	192.168.1.107	HBWang	1 days 00:00:00
0c:72:2c:5a:39:41	192.168.1.108	MW150R	00:00:57

This page is used to display DHCP Client address and related information.

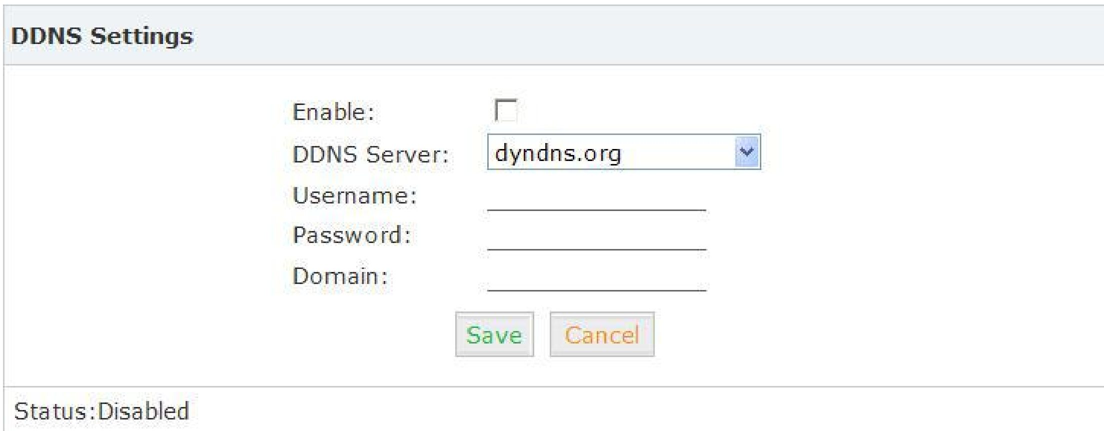
When DHCP Server distributes address, the Client's MAC address is associated with the IP address, and then the device will get the same IP address every time.

Click【Network Settings】->【DHCP Server】->【Static MAC】->【New Static MAC】:



4.6 DDNS Settings

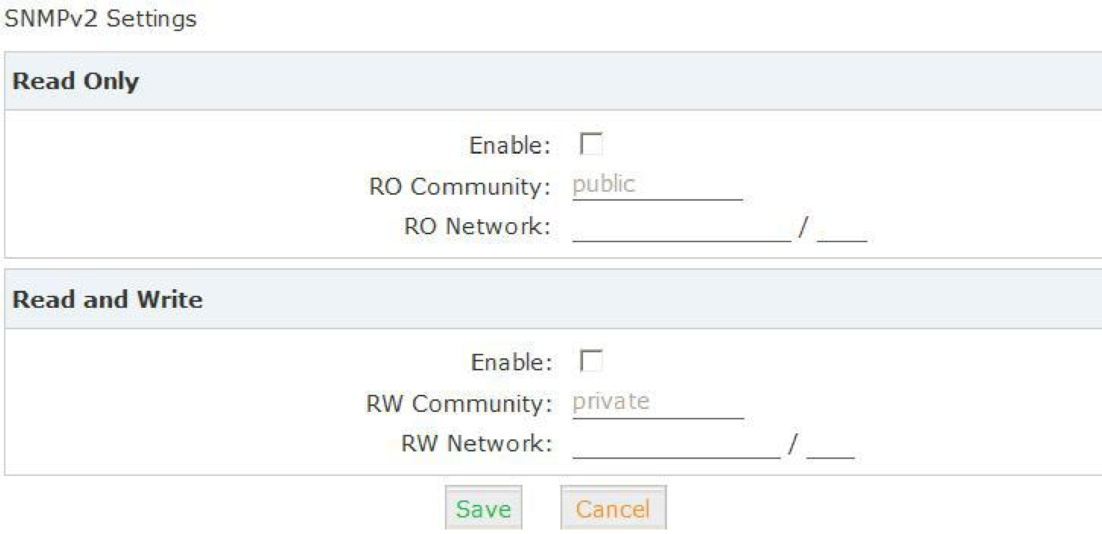
After setting DDNS (Dynamic Domain Network Server), LAVoice IP PBX settings will be visited remotely. Click【Network Settings】->【DDNS Settings】:



LAVoice supports DDNS provided by Dyndns.org / No-ip.com / zoneedit.com.

4.7 SNMPv2 Settings

SNMP(Simple Network Management Protocol): Used for remote management.
Click【Network Settings】->【SNMPv2 Settings】:



Reference

Item	Explanation
Enable	Enable “Read Only” of SNMP
RO Community	Define the name of RO Community of SNMP
RO Network	Define network of RO

4.8 Trouble Shooting

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

Click【Network Settings】->【TroubleShooting】:

Troubleshooting

Ping | Traceroute

Ping 192.168.1.1 Packets: 4 Run Stop

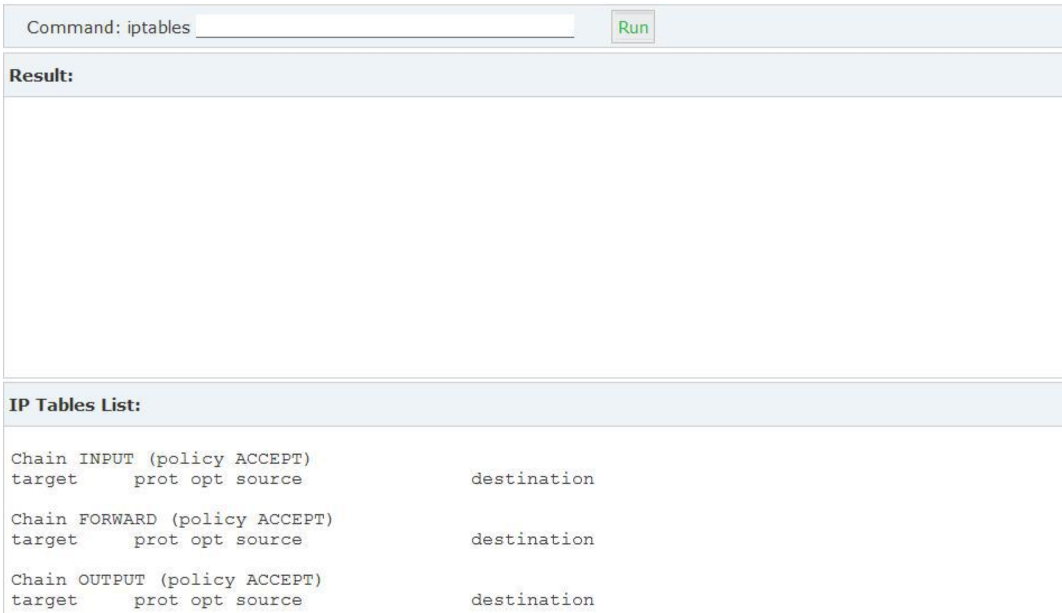
```
PING 192.168.1.1 (192.168.1.1): 56 data bytes
64 bytes from 192.168.1.1: seq=0 ttl=64 time=1.677 ms
64 bytes from 192.168.1.1: seq=1 ttl=64 time=0.964 ms
64 bytes from 192.168.1.1: seq=2 ttl=64 time=1.057 ms
64 bytes from 192.168.1.1: seq=3 ttl=64 time=0.950 ms

--- 192.168.1.1 ping statistics ---
4 packets transmitted, 4 packets received, 0% packet loss
round-trip min/avg/max = 0.950/1.162/1.677 ms
```

Chapter 5 Security

5.1 Firewall

Click【Security】->【Firewall】



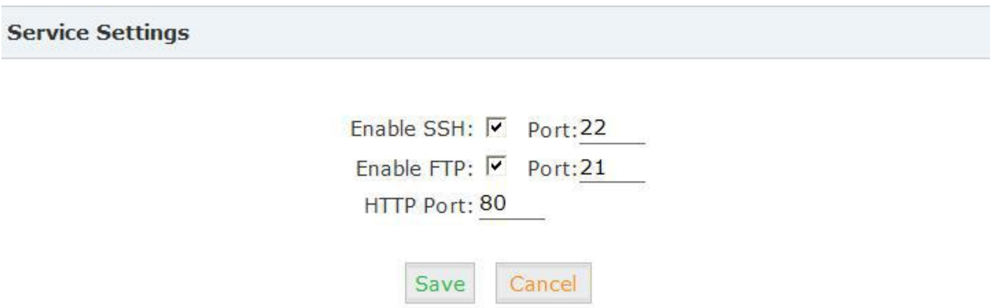
Iptables Command:

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 80 port	iptables -A INPUT -s 192.168.0.3 -p tcp --dport 80-j DROP

5.2 Service

【Service】: settings of SSH/FTP and HTTP Port.

Click【Security】->【Service】:



Enable SSH to login background management system through SSH.

Enable FTP to allow uploading files to system through FTP.

5.3 SIP Allowed Address

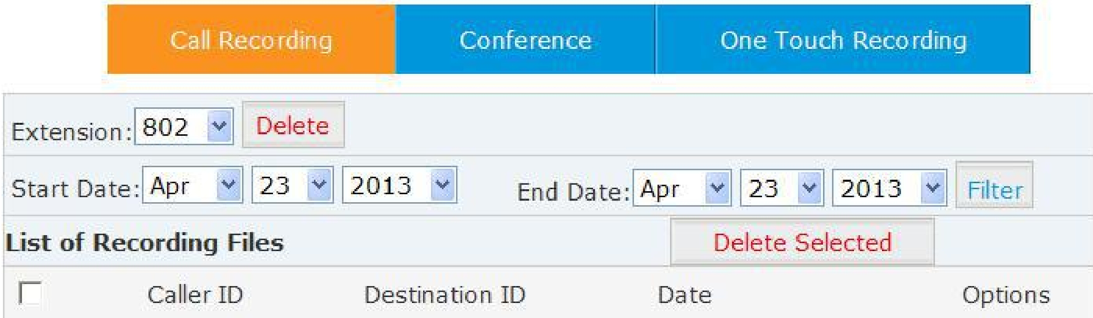
Define an allowed address, from which every SIP request will never be filtered or refused.

Click【Security】->【SIP Allowed Address】:

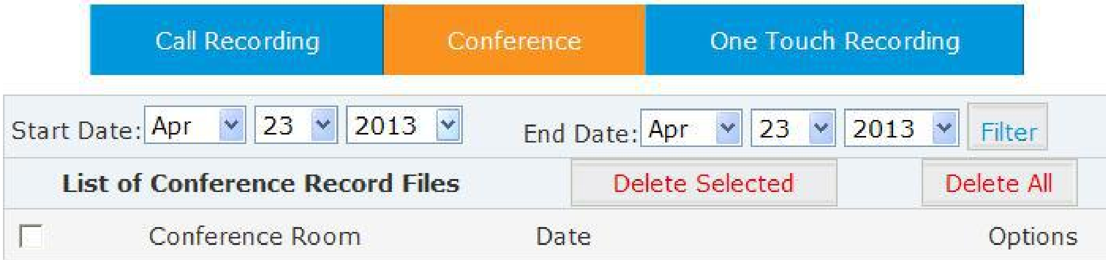
Chapter 6 Report

6.1 Record List

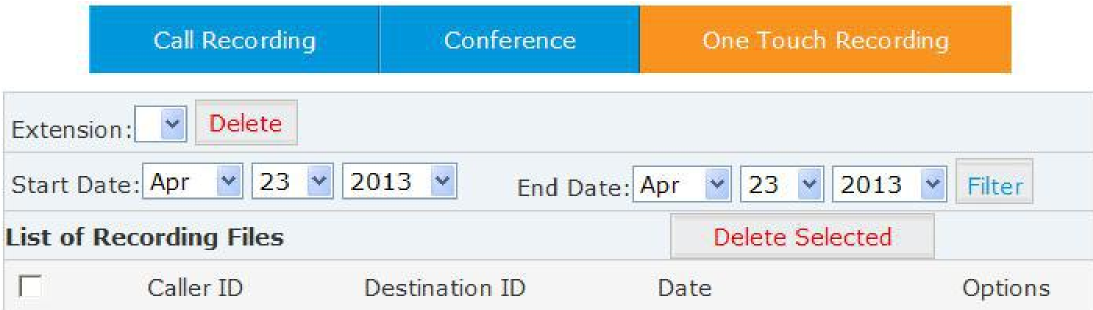
Check recordings of specified extension or conference here, or delete the recording file.
【Record List】:



【Conference】:



【One Touch Recording】



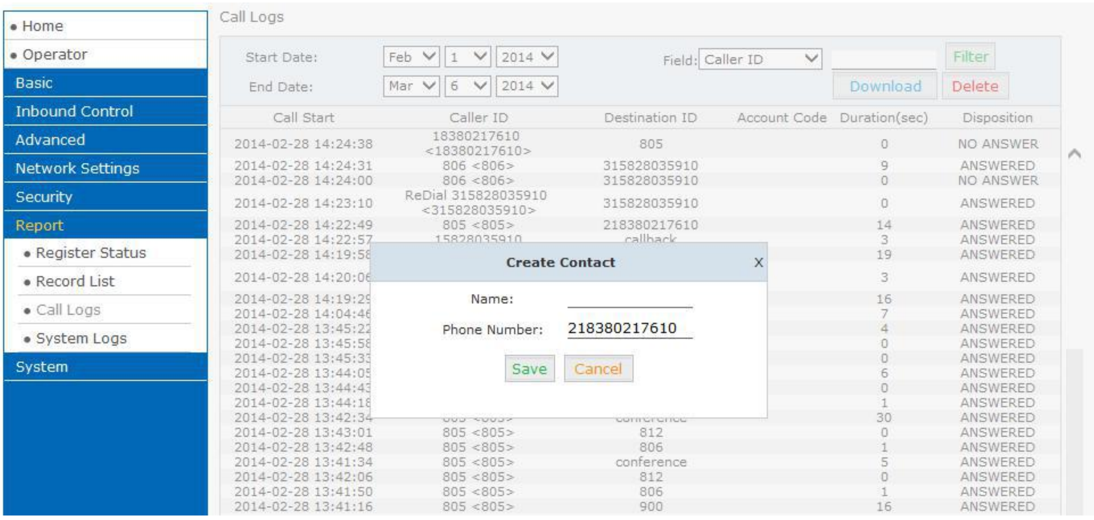
6.2 Call Logs

Check call logs by caller ID or callee ID.
Click【Report】->【Call Logs】:

Notice

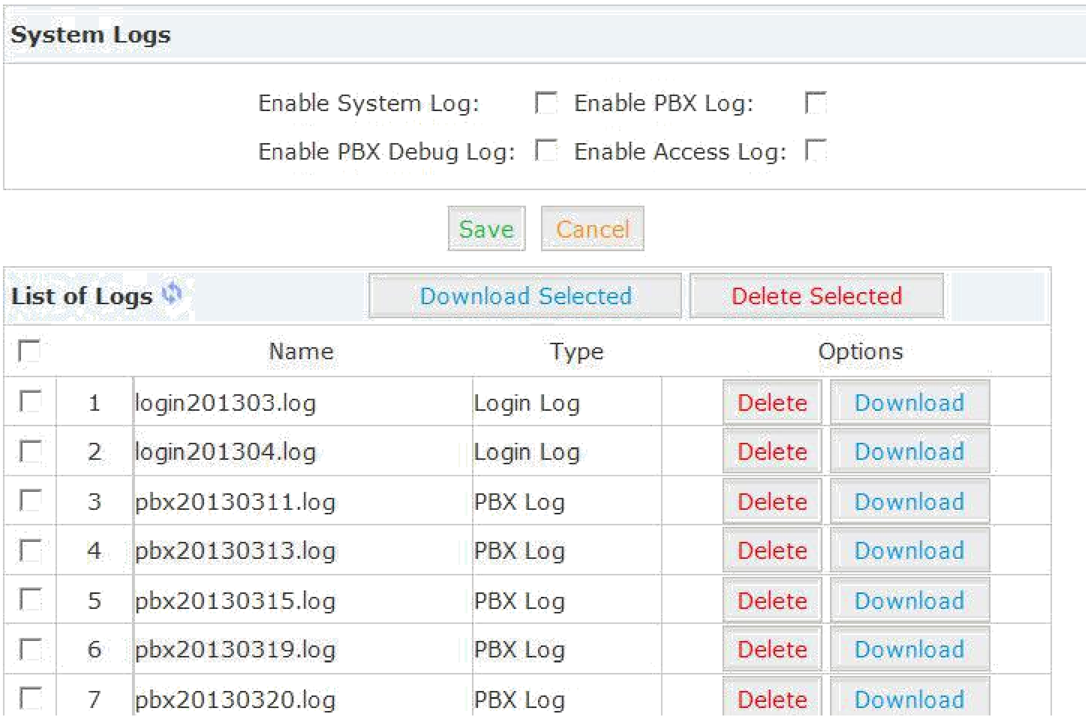
Duration in the call logs is not real charged duration. If you need billing, PSTN must support polarity reversal function, and meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for the LAVoice IP PBX.

The number in the call logs can be added in the phone book directly:



6.3 System Logs

Click【Report】->【System Logs】, you can download/ delete the system logs.

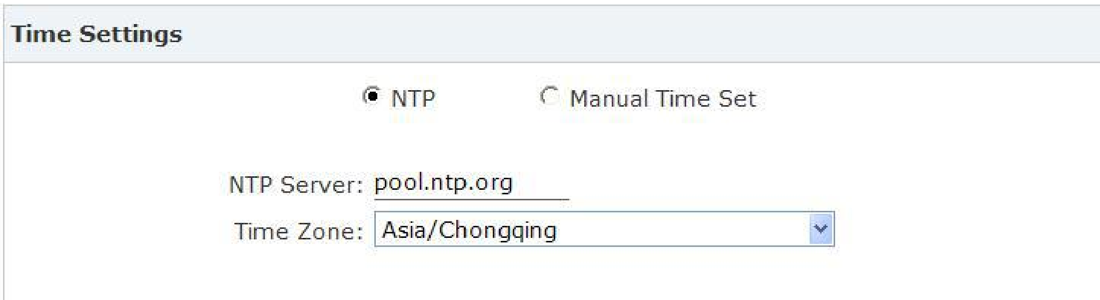

System Logs
Enable System Log:
Enable PBX Log:
Enable PBX Debug Log:
Enable Access Log:
Save
Cancel
List of Logs
Download Selected
Delete Selected
Name
Type
Options
1 login201303.log Login Log Delete Download
2 login201304.log Login Log Delete Download
3 pbx20130311.log PBX Log Delete Download
4 pbx20130313.log PBX Log Delete Download
5 pbx20130315.log PBX Log Delete Download
6 pbx20130319.log PBX Log Delete Download
7 pbx20130320.log PBX Log Delete Download

Chapter 7 System

7.1 Time Settings

Time settings for LAVoice system. The system supports either NTP or Manual Time Set.

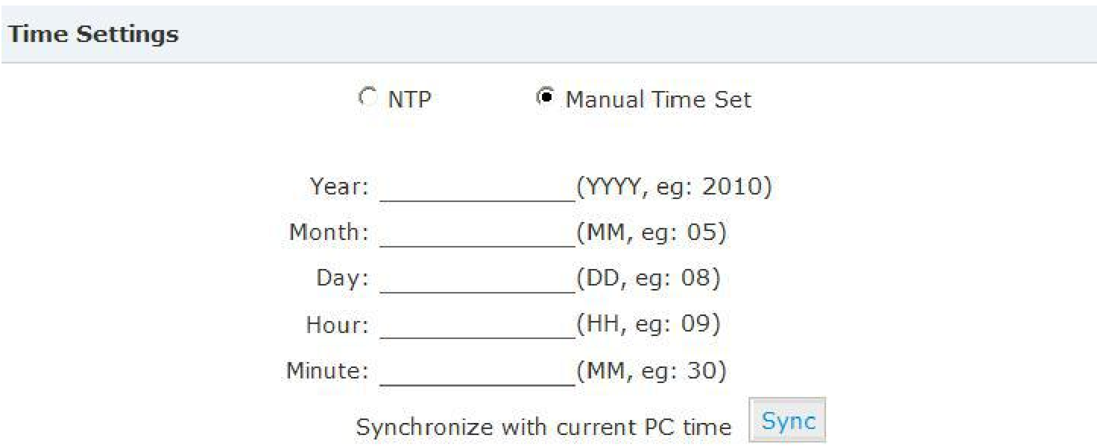
【NTP】:



Reference:

Item	Explanation
NTP Server	Define the NTP Server. You can input the IP address or domain of this server, whether it's local or remote. Default server is pool.ntp.org. Be aware that the LAVoice IP PBX needs to be able to connect to an NTP server to properly function.
Time Zone	Select your time zone so that the system will set time based on the time zone.

【Manual Time Set】:

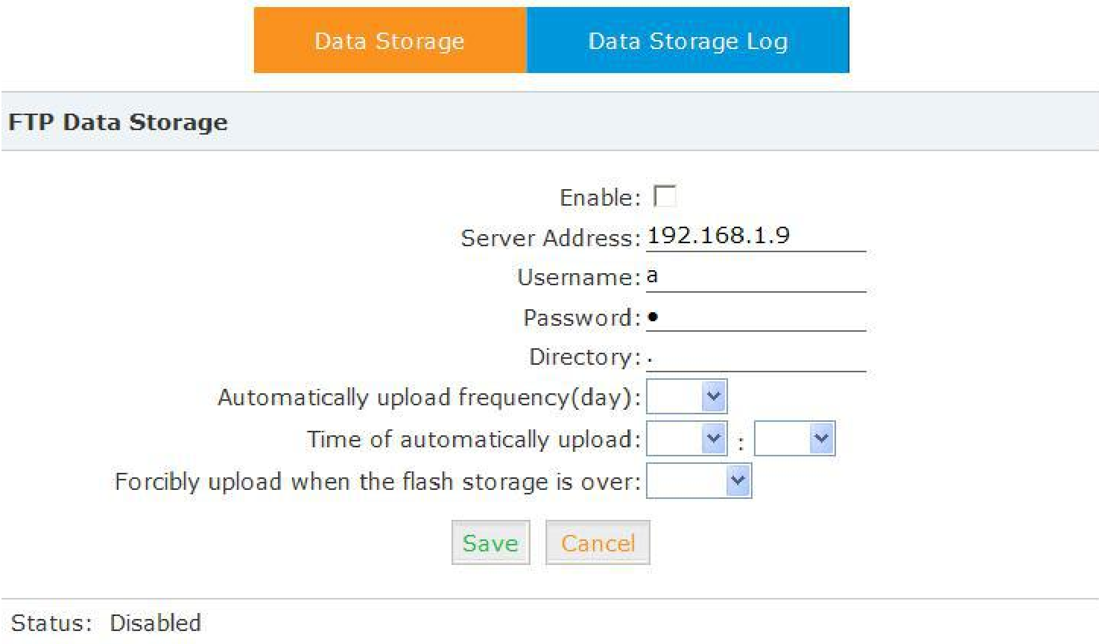


After entering Year/ Month/ Day/ Hour/ Minute, then save and activate.

Or, you can click【Sync】to synchronize with current PC time.

7.2 Data Storage

When you need mass storage of recording files, voicemails, call logs, etc, you can upload these files to FTP server through FTP Data Storage based on the specified time frequency
Click【System】->【Data Storage】:



Reference

Item	Explanation
Enable	Enable FTP Data Storage.
Server Address	Set FTP server address (IP address or domain).
Username	Username for login FTP.
Password	Password for login FTP.
Directory	Define a directory used for storage on FTP server.
Automatically upload frequency (day)	Define frequency by days to upload the data.
Time of automatically upload	Define the time to upload the data.
Forcibly upload when the flash storage is over	Forcibly upload data when flash storage is over the percentage value.

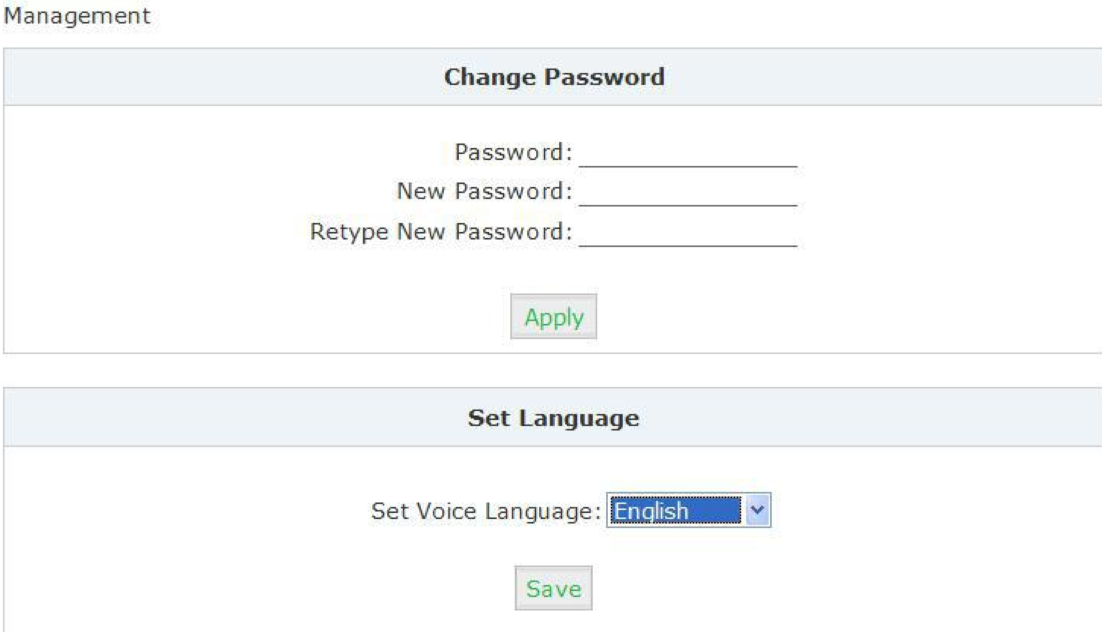
Check from【Data Storage Log】:

Click【Refresh】to refresh data storage log.
Click【clear】to clear data storage log.

7.3 Management

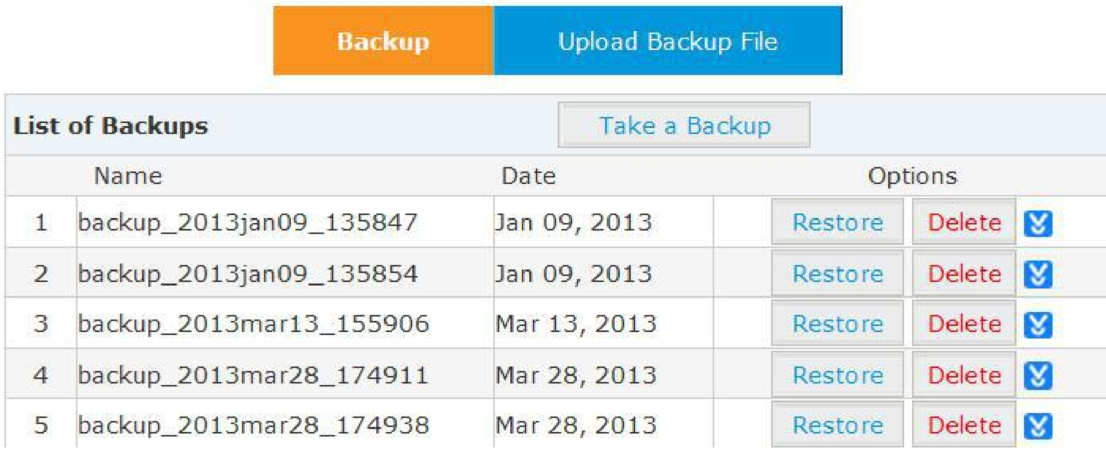
【Management】is used for modify password of LAVoice system, and the settings of system voice.

Click 【System】->【Management】:



7.4 Backup

Click【System】->【Backup】

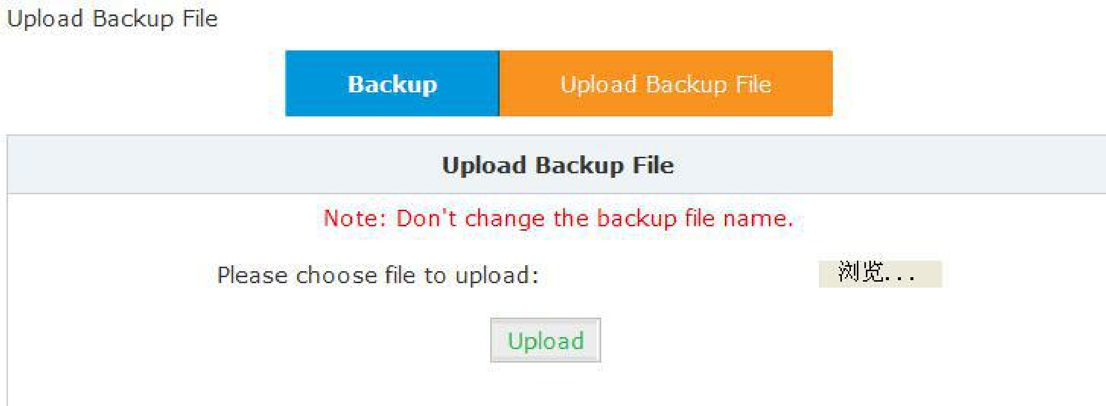


Reference:

Item	Explanation
Take a Backup	Take a backup of the current system configuration.
Restore	Restore system to the specified backup configuration.
Delete	Delete specified backup file.

Click the download button “ ” to download the specified backup file and manage locally.

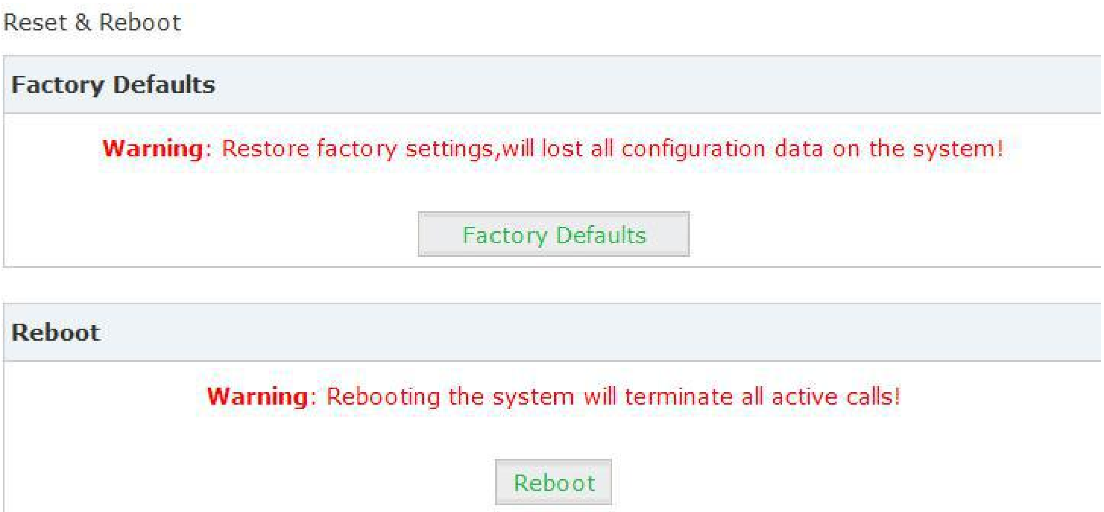
Click【Upload Backup File】to upload the backup file here.



Click【browse】to select the local backup file, and click【Upload】to upload the backup file to system.

7.5 Reset & Reboot

If you need reset the system to factory defaults or reset, please click【System】->【Reset & Reboot】: Restoring factory settings will make configuration data in the system lost.



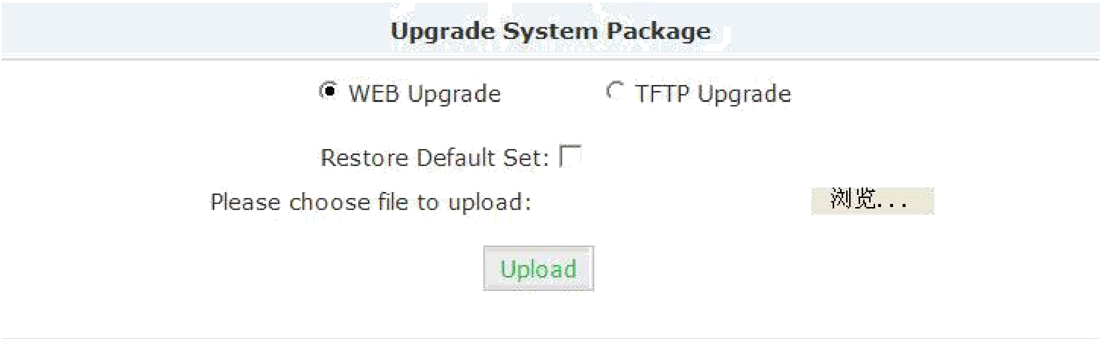
Click【Factory Defaults】to reset the system to factory defaults.

Click【Reboot】to reboot the system.

7.6 Upgrade

7.6.1 WEB Upgrade

Click【System】->【Upgrade】->【WEB Upgrade】:

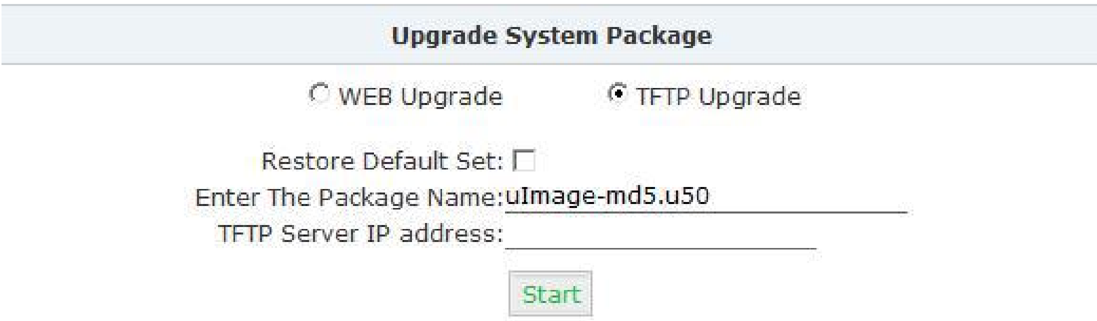


Click【Browse】to select the firmware file, then click【Upload】to upload the selected firmware to system and finish the upgrading automatically.

If check【Restore Default Set】, the system will clear all the configuration and reset to factory default.

7.6.2 TFTP Upgrade

Click【System】->【Upgrade】->【TFTP Upgrade】:



Reference:

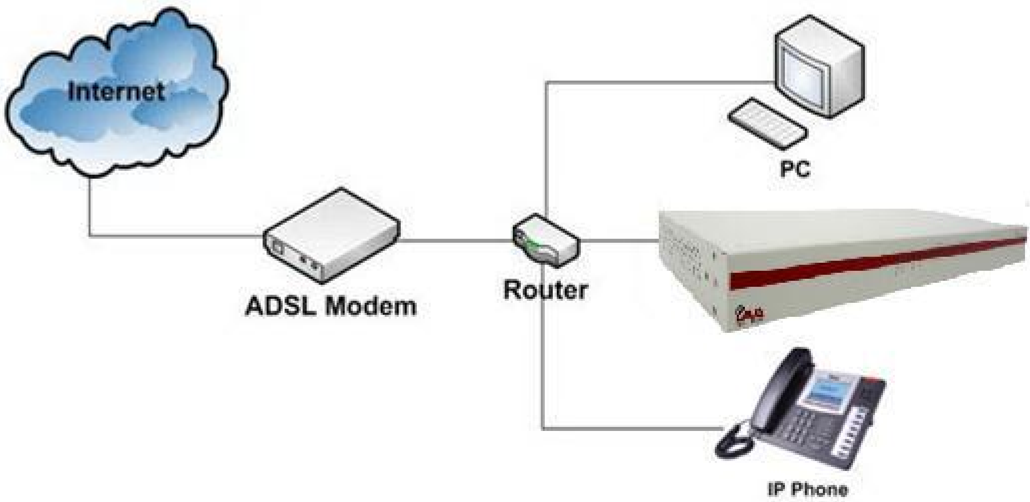
Item	Explanation
Restore Default Set	System will restore to factory defaults after checking this option.
Enter The Package Name	Enter the package name for upgrading.
TFTP Server IP address	Enter your TFTP server IP address.

Chapter 8 Operating Instructions

(Take LVX-100S as example)

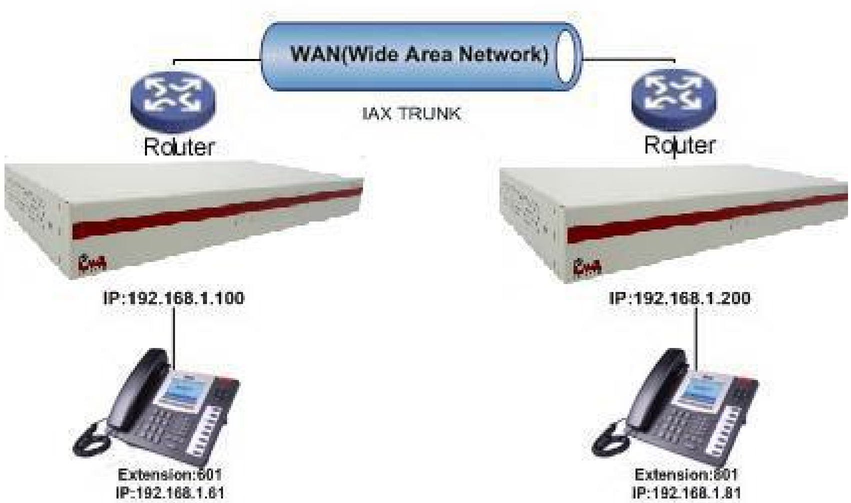
8.1 How to connect LVX-100S in the Network

If your office accesses the public network through router, you can put the LAVoice IP PBX behind the router. You should connect the WAN port of the IP PBX to the LAN port of the router.



8.2 How to combine two sets LAVoice IP PBX in the same network

We start to combine two IP PBXs in the same network and then try to expand to different network. Combine two IP PBXs in the same LAN from the structure as below:



Register 100S-B IP to a trunk of 100S-A, and register 100S-A IP to a trunk of 100S-B, without authentication for each registration.

Configuration Rule:

1. IP Phone registers on LVX-100S-A with extension number 601.
2. Another IP Phone registers on LVX-100S-B with extension number 801.
3. LVX-100S-A WAN IP: 192.168.1.100.
4. LVX-100S-B WAN IP: 192.168.1.200.
5. Extension format of LVX-100S-A: 6XX.
6. Extension format of LVX-100S-B: 8XX.
7. All extensions on 100S-A can call extensions on 100S-B by 8XX format.
8. All extensions on 100S-B can call extensions on 100S-A by 6XX format.

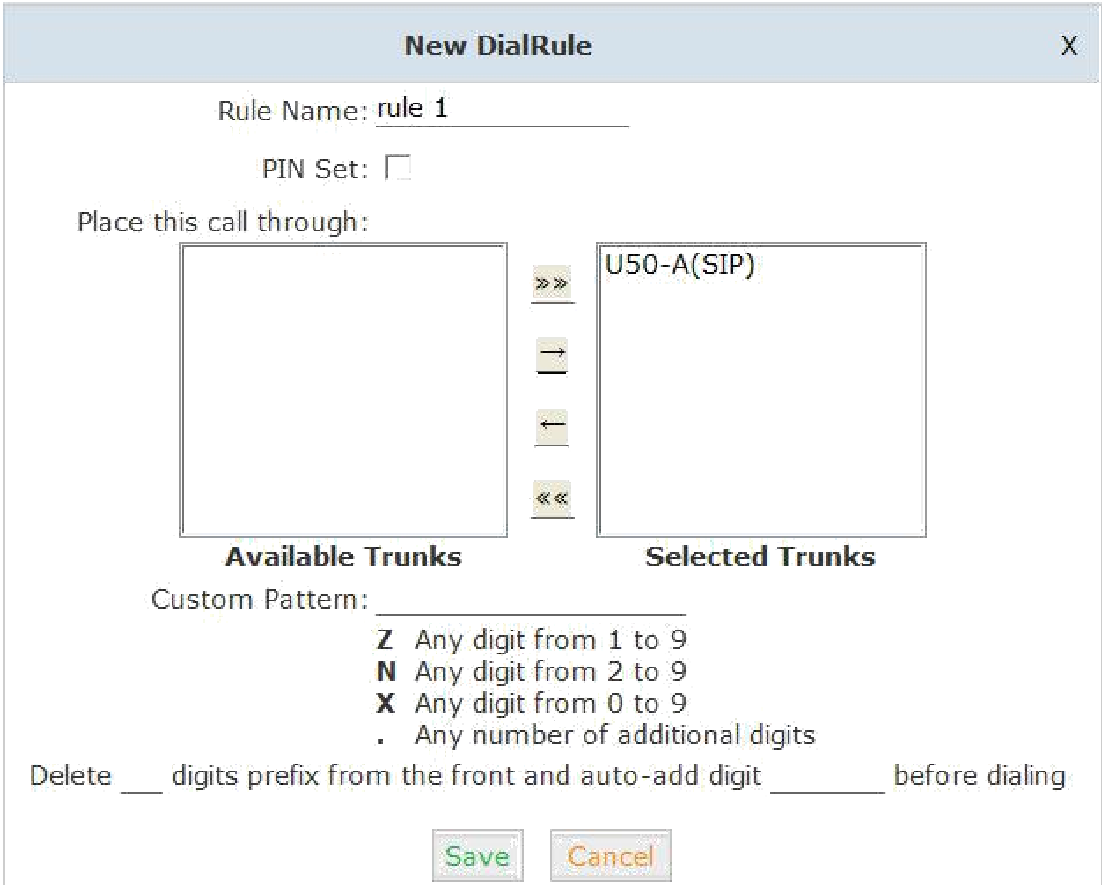
Step1: Register 100S-B IP to a trunk of 100S-A

LVX-100S-A: Click【Basic】->【Trunks】->【New VoIP Trunk】:

Step2: Register 100S-A IP to a trunk of 100S-B as the same way of step 1.

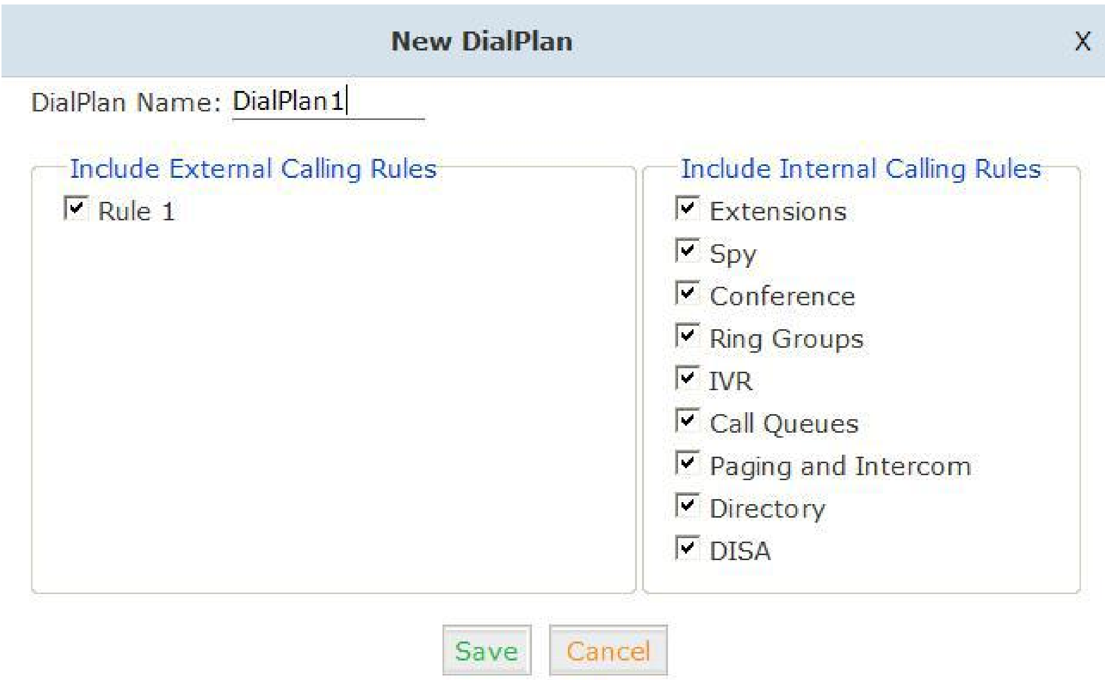
Step 3: Create DialRule on 100S-A, and add the DailRule to the DialPlan

Click【Outbound Routes】->【DialRules】->【New Dial Rule】:



Select the created line 192.168.1.200 to 【Selected Trunks】, custom pattern is XXX, save and activate.

Click【DialPlans】->【New Dial Plan】:



Check the created calling rule, save and activate.

Step4: Create dialrule on LVX-100S-B, add the created dialrule to the dialplan as the same way of Step 3.

Step 5: Activate the current configuration and test:
1. Register IP Phone to 100S-A as extension 601.
2. Register another IP Phone to 100S-B as extension 801.
3. Make a call from 801 to 601, 601 rings and the call is connected.
4. Make a call from 601 to 801, 801 rings, and the call is connected.

8.3 How to connect two sets LAVoice IP PBXs in different network?

E.g.: two sets LVX-100S in the internet.
Normally, the two sets LVX-100S are located in different place; but they are in the internet, and have public IP address.

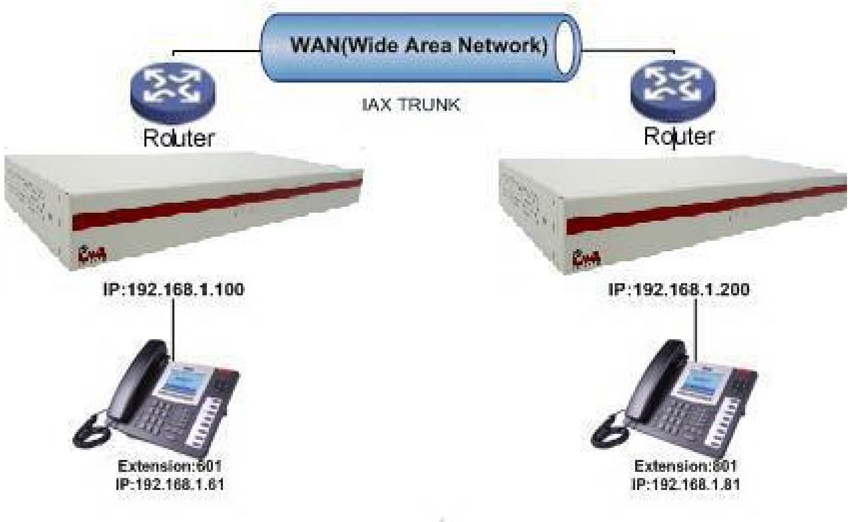
Note: Enable NAT on Router.
For external line configuration, you must use public IP address.
Take the following instructions as example:

Register 100S-B IP to a trunk of 100S-A with authentication.
Configuration Rule:
1. IP Phone registers on 100S-A as extension 601.
2. Another IP Phone registers on 100S-B as extension 801.
3. 100S-A IP:192.168.1.100.
4. 100S-B IP:192.168.1.200.
5. Extension format of 100S-A: 6XX.
6. Extension format of 100S-B: 8XX
7. Create an extension 888 with password 123456 on 100S-B.
8. All extensions on 100S-A can call extensions on 100S-B with format 8XX.
9. All extensions on 100S-B can call extensions on 100S-A with format 6XX.

For detail steps, please take chapter 8.2 as reference.

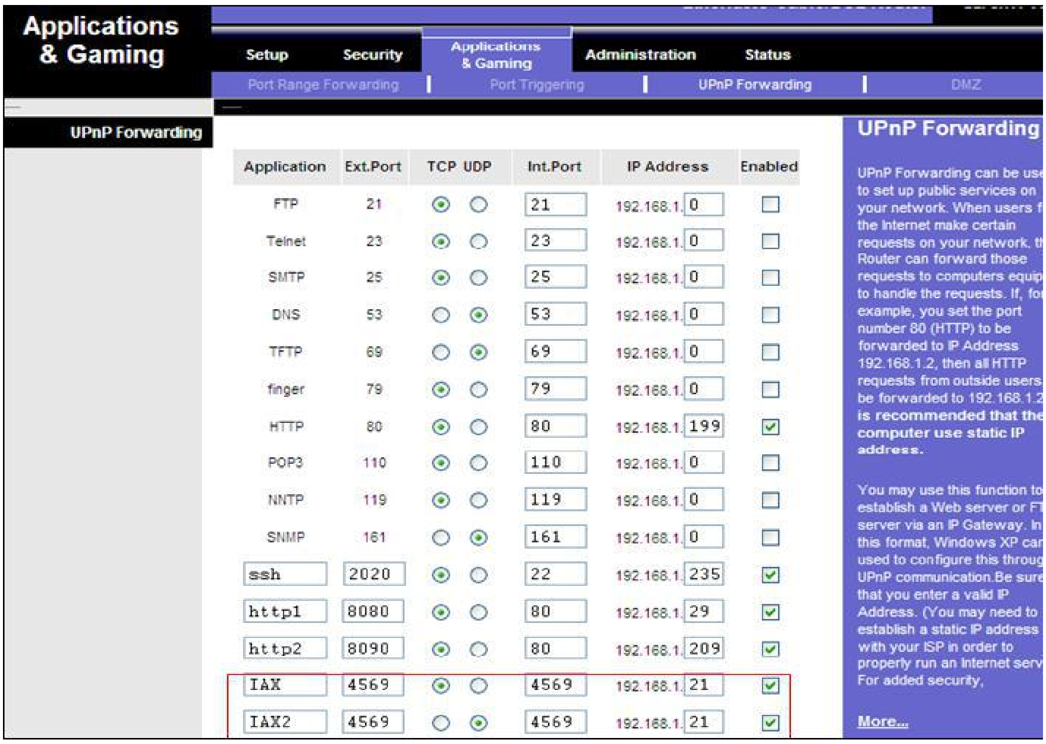
Two sets 100S behind router

Sometimes 100S doesn't have public IP, and you have to configure port mapping for your router.



Step1: Configure the mapping rule of 100S-A on the router. 100S-B is connected behind the router, registers on 100S-A through internet, you need configure the port mapping of IAX2 port(4569) on the router. Then, all data received from WAN port of router(192.168.1.100:4569) will be sent to 100S-A

Now, take the web management panel of Linksys router as example.



Step2: 100S Configuration

Configure the trunk and dialplan on 100S-B, register 100S-B IP to 100S-A, configuration is same as above, but you have to replace the public IP with internal IP:192.168.1.21.

Step3: Configure port mapping rule of 100S-B on the router

Configure port mapping of 100S-B on the router as the same way of step1..

Step4: Connect two sets 100S and make the call

Create extension 601 on 100S-A, extension 801 on 100S-B, and create the correct outbound rule.

Notice

Public IP must be provided by network provider. It could be dynamic IP address, and easy to change; you can resolve this problem by using DDNS.

8.4 How to resolve the problem “one-way” audio problems

If 100S is behind router, to resolve the problem, please set up IP address as below:

Click【Advanced】->【Option】->【Global SIP Settings】:

1. External IP — External IP or domain to replace the device IP
2. External Host — External domain to replace the device IP
3. External Refresh(sec) — Refresh time, default is 10 seconds.
4. Local Network Address — IP address and subnet mask needed to be converted . E.g.: 192.168.1.100/255.255.255.0

8.5 How to use Skype on LVX-100S

8.5.1 Visit the Top-up Page

Visit the top-up page: http://skype.tom.com/products/en/skypeout.html

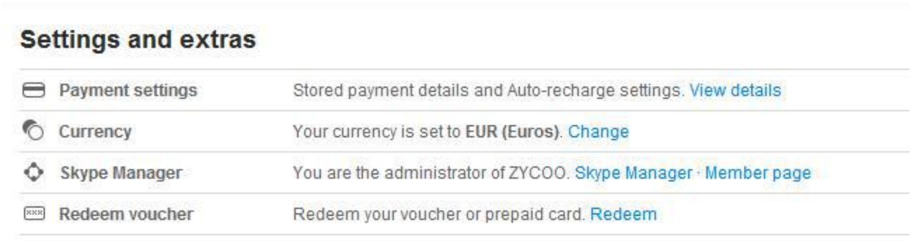
Select subscription, payment method and enter the Skype account to top up credit.

Notice

First top up for business account must be more than €50.

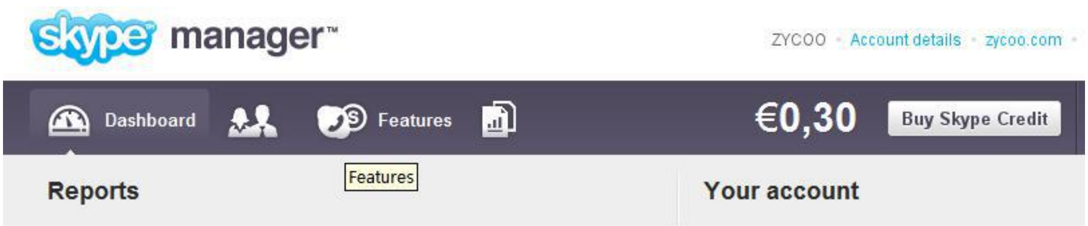
8.5.2 Manage Skype Account

After login, you will find **Skype Manager**, and click it.

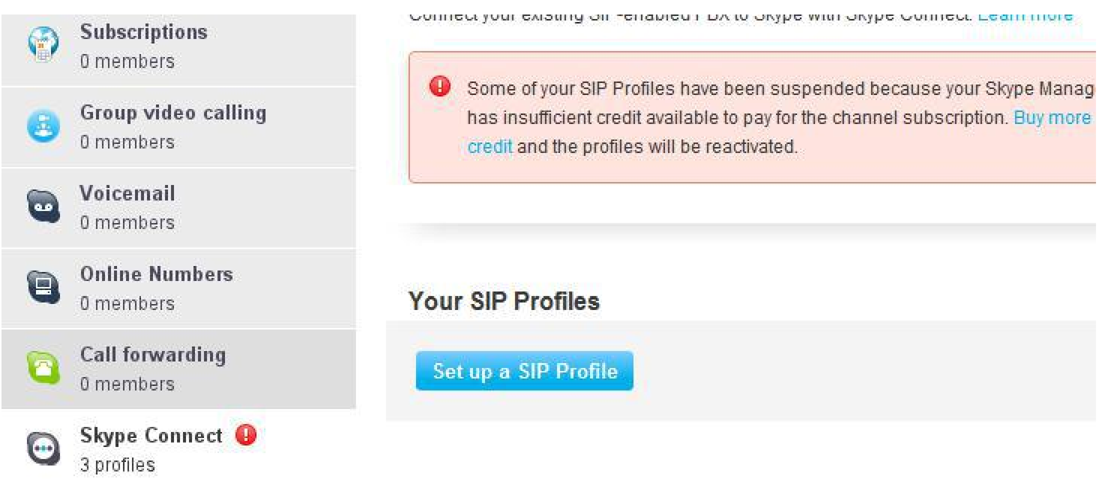


8.5.3 Create a SIP File

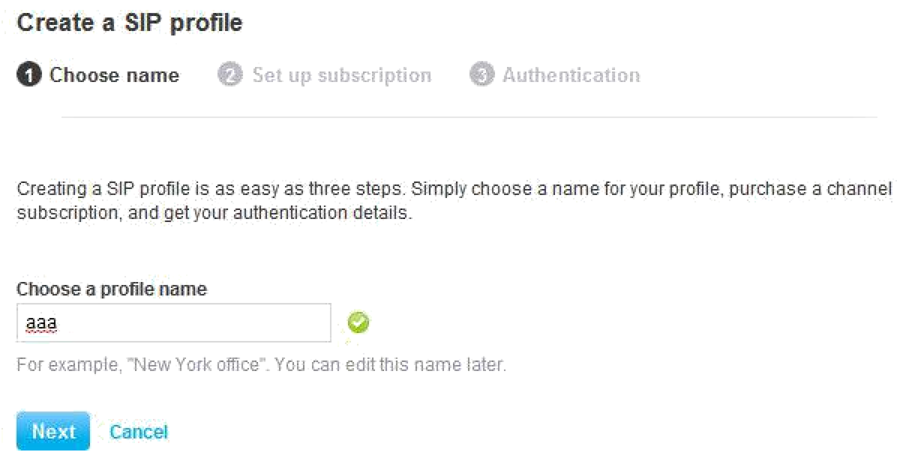
Click **Features:**



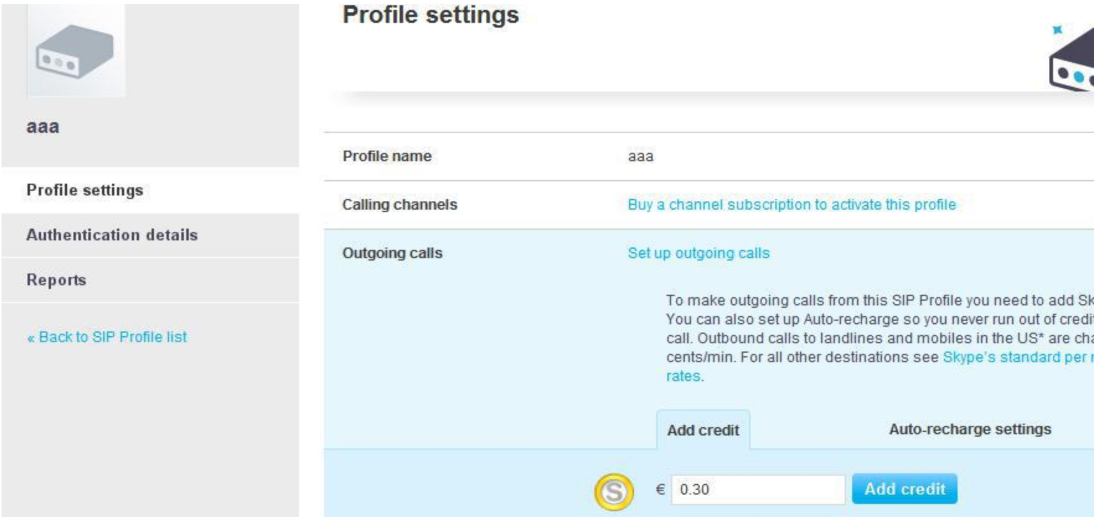
Click **Skype connect:**



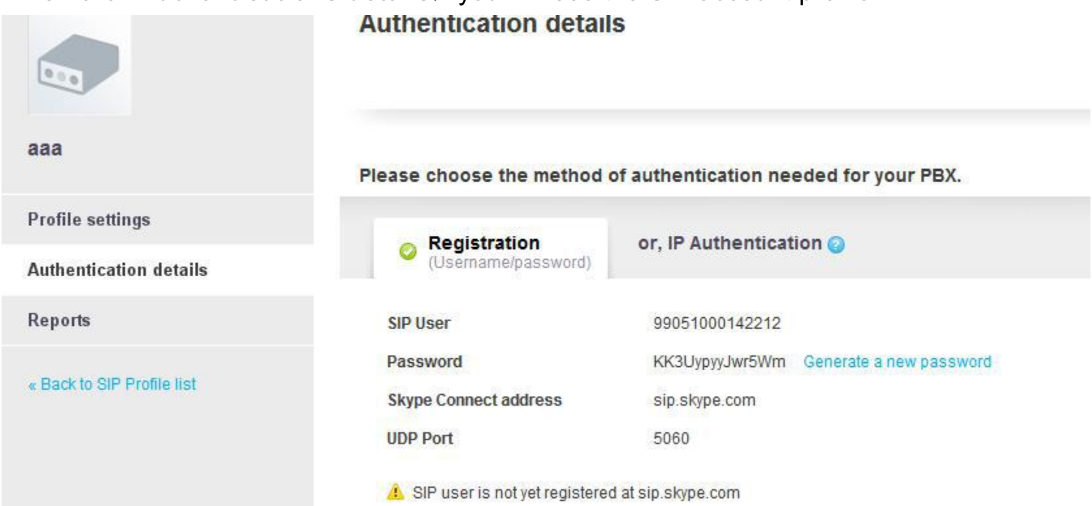
Click **Set up a SIP Profile:**



Create a SIP account, and each account has a channel, you need pay €5 for each channel as monthly rent. Then input the registration profile in the VoIP trunk of 100S and distribute the money to Outgoing calls.



When click **Authentications details**，you will see the SIP account profile:



Select the created line 192.168.1.200 to 【Selected Trunks】, custom pattern is XXX, save and activate.