智科通信

CTMC (Cloud Telephony Management Center)

User Manual

V1.0

Contents

Chapter 1 BRIEF INTRODUCTION

CTMC (Cloud Telephony Management Center) is a network node management center which has been independently developed by ZYCOO, and can be utilized by VoIP service providers and enterprises users to manage multiple CooVox CTNs (Cloud Telephony Nodes).

CTMC provides a multitude of features for CTN's including auto-provision, software/firmware upgrade management, status/performance monitoring, warning log diagnosis etc. CMTC is a powerful solution that delivers the features and functionality required to manage and maintain a highly dispersed telephony environment through the use of a single centralized management system.

CTMC comprises of the following major features:

- Centralized monitoring/configuration and upgrading of CTN
- Cost-savings for calls between each CTN via Numbering Plan
- Monitor system information, configuration and service status
- View and backup of system log, operation log and call log of each CTN
- Manage configuration for individual or multiple devices based on user groups
- Flexible upgrading control strategy allowing for convenient software and firmware upgrades
- Adopting B/S managing mode, multi-language GUI, humanized management process, and easy operation
- Based on TR069 protocol, allowing nodes to pass through private networks
- Manage multi-service and user groups based on template
- Based on Linux which ensures the device is secure and reliable

Chapter 2 SYSTEM INSTALLATION

2.1 Obtain CTMC System Image

Download the latest CTMC firmware (CTMC_Install_Package.iso) on ZYCOO website: http://www.zycoo.com/files/upload/CTMC_Install_Package.iso, or request ZYCOO to provide you with an installation disk.

2.2 Install CTMC

Install CTMC system by CD or USB drive. During the installation, you will be required to configure language, input keyboard, time zone, password to root user. The default IP is DHCP.

Notice: Before using CTMC, you need to upgrade your current CooVox IP PBX to CTN. Download the latest CTN firmware (CTN_Upgrade_Package.zip) on ZYCOO website: http://www.zycoo.com/files/upload/CTN_Upgrade.Package.zip, unzip and use respective firmware to upgrade U20, U50 or U100 to the node mode.

2.3 Free Trial & License Purchase

A lifetime free trial can be available for Three (3) CTNs on CTMC; more than Three (3), you will need to purchase formal license.
Purchase license for:
* Expansion number for CTNs: 10 CTNs/ 20 CTNs/ 30 CTNs

Once purchased, the license gets the lifetime effectiveness.

2.4 Hardware Requirement for CTMC Installation

1. PC Requirements

CPU: Intel(R) Atom(TM) CPU or higher
RAM: 512MB or higher
HDD: 10GB free space at least
Graphic Card: VGA compatible or higher
CD Driver: CD-ROM/DVD-ROM
Others: Audio Card, Network, Modem and so on

2. System Requirements

Linux OS installed by formal installation process
Default IP is DHCP

3. Bandwidth Requirements

CTMC operating bandwidth is the sum of bandwidth of all managed nodes.
Respective bandwidth as below:

Model	Amount	Bandwidth
CooVox-U20	1	100 Kbps
CooVox-U50	1	100 Kbps
CooVox-U100	1	100 Kbps

For example, there are 10pcs U20 managed by CTMC, the bandwidth of CTMC is: $10 \times$ 100Kbps $\times$ 2 > 2Mbps (uplink & downlink), which is the minimum bandwidth in the case of full concurrent calls is 2Mbps.

4. Port Requirements

Port	Functionality
8505	Port for CTMC background monitoring nodes
8506	Port CTMC download
9998	Port for Web access to CTMC
9898	Port for nodes configuration on CTMC
1194	Port for VPN

Note: If CTMC is in public network, 8505 / 8506 / 1194 must stay open.

Chapter 3 QUICK START GUIDE

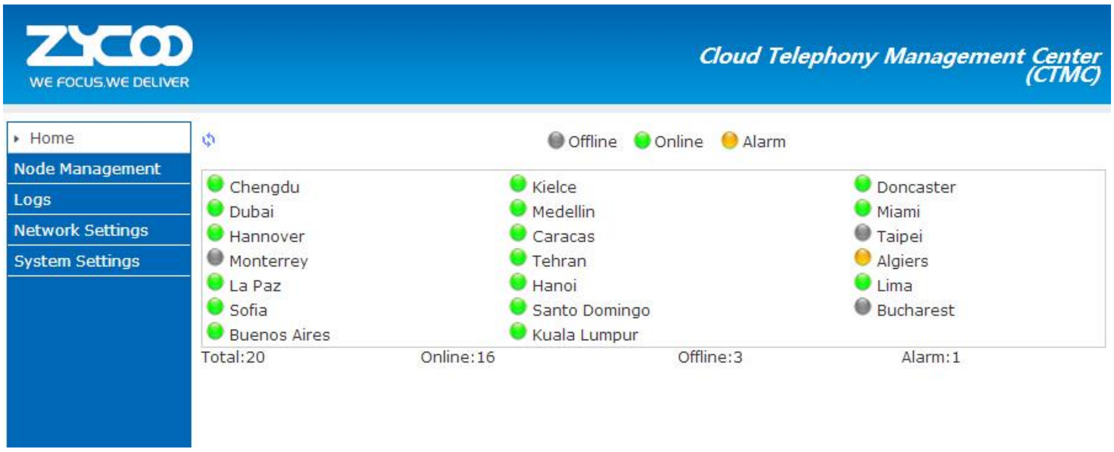
3.1 Quick START Guide

This Quick Start Guide includes a few simple examples that will introduce you to how CTMC manages the CTN. For detailed information, please refer to the respective chapters.

3.2 Detailed Steps

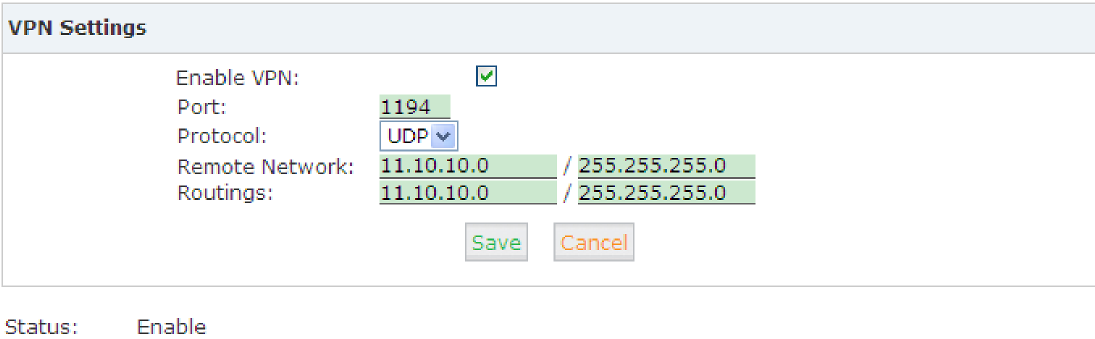
CTMC is a simple and easy-used Web GUI, which can be easily accessed by IP address on a Linux server. To access the CMTC GUI for the first time, go to the default address http://192.168.1.100:9998. The default username is admin, and the default password is admin.

Once logged in you will see the following page:



Step 1: VPN settings on server

VPN settings is a basic and necessary application in CTMS for network construction.
This step can be skipped; please just use the default VPN settings.
To configure the VPN server, select 【Network Settings】 -> 【VPN Settings】:



Reference:

Item	Explanation
Enable	Enable/Disable VPN
Port	1194 by default
Protocol	UDP/TCP
Remote Network	Default
Routings	Default

Note: Port number 9998, 8505, 9898, 1194 must be open if CTMC is in public network.

Step 2: Check Node List

To check the Node list, select【Node Management】->【Node List】:

Node List

	Name	Model	Last Update Time	Version	Status	Options
1	Chengdu	CooVox-U50	2014-06-08 11:57:51	1.0.5	Online	Detail Status Edit Delete
2	Kielce	CooVox-U20	2014-06-09 11:11:42	1.0.5	Online	Detail Status Edit Delete
3	Doncaster	CooVox-U20	2014-06-09 11:11:47	1.0.4	Online	Detail Status Edit Delete
4	Dubai	CooVox-U50	2014-06-09 18:38:11	1.0.5	Online	Detail Status Edit Delete
5	Medellin	CooVox-U50	2014-06-09 11:11:48	1.0.5	Online	Detail Status Edit Delete
6	Miami	CooVox-U50	2014-06-09 11:11:42	1.0.5	Online	Detail Status Edit Delete
7	Hannover	CooVox-U100	2014-06-09 11:11:51	1.0.5	Online	Detail Status Edit Delete
8	Caracas	CooVox-U50	2014-06-08 18:06:25	1.0.5	Online	Detail Status Edit Delete
9	Taipei	CooVox-U20	2014-06-09 11:11:51	1.0.5	Offline	Detail Status Edit Delete
10	Monterrey	CooVox-U20	2014-06-07 21:49:01	1.0.5	Offline	Detail Status Edit Delete
11	Tehran	CooVox-U50	2014-06-05 13:49:54	1.0.5	Online	Detail Status Edit Delete
12	Algiers	CooVox-U20	2014-06-09 11:00:19	1.0.5	Alarm	Detail Status Edit Delete
13	La Paz	CooVox-U20	2014-06-09 11:11:45	1.0.5	Online	Detail Status Edit Delete
14	Hanoi	CooVox-U20	2014-06-09 11:11:49	1.0.5	Online	Detail Status Edit Delete
15	Lima	CooVox-U20	2014-06-08 21:15:11	1.0.5	Online	Detail Status Edit Delete
16	Sofia	CooVox-U50	2014-06-05 09:51:36	1.0.5	Online	Detail Status Edit Delete
17	Santo Domingo	CooVox-U20	2014-06-07 18:02:17	1.0.4	Online	Detail Status Edit Delete
18	Bucharest	CooVox-U100	2014-06-07 18:02:17	1.0.5	Offline	Detail Status Edit Delete
19	Buenos Aires	CooVox-U20	2014-06-07 18:02:17	1.0.5	Online	Detail Status Edit Delete
20	Kuala Lumpur	CooVox-U20	2014-06-07 18:02:17	1.0.5	Online	Detail Status Edit Delete

Total:20 20 per page pages: << 1 >>

In this list there are 20 nodes registered to CTMC.

Reference:

Item	Explanation
Name	Name of node
Model	Model of node
Last Update Time	Last update time
Version	Version of node
Status	Connection status for node to CTMC (online/offline)
Options	Detail: Details of node Status: Monitor status of node, such as memory usage, port operation status Edit: Edit configuration of node, such as DialRule, configure special modules Delete: Delete node

Step 3: Configure node

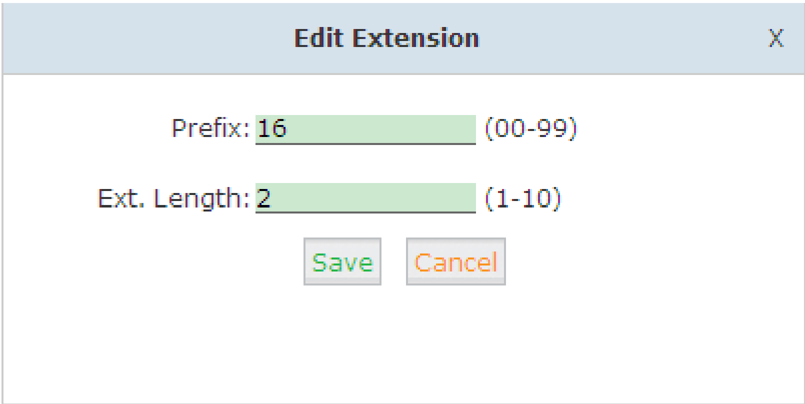
We will use "Chengdu" as an example:

First we need to configure the Numbering Plan of the node, select【Node Management】-> 【Numbering Plan】:

Numbering Plan

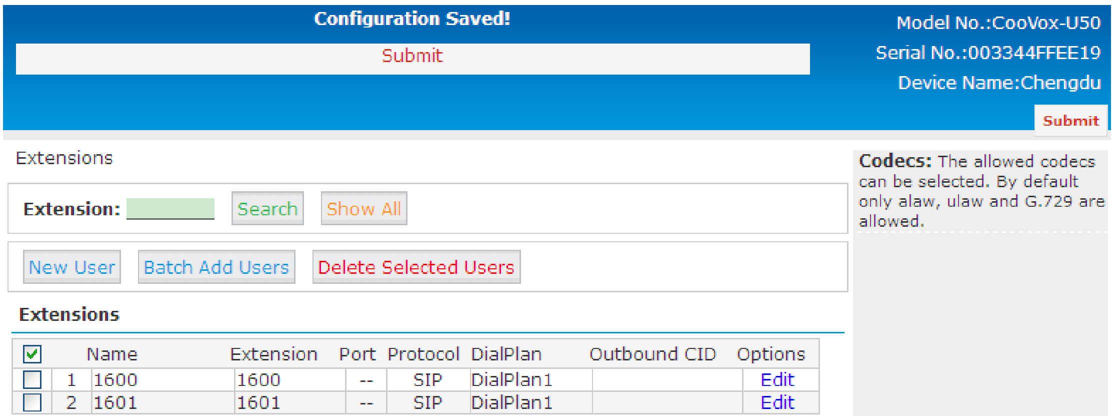
	Name	Numbering Plan. Prefix	Ext. Length	
1	Chengdu			Edit
2	Kielce	51	1	Edit
3	Doncaster	64	1	Edit
4	Dubai	32	3	Edit
5	Medellin	16	2	Edit
6	Miami	13	3	Edit

Select "Edit":



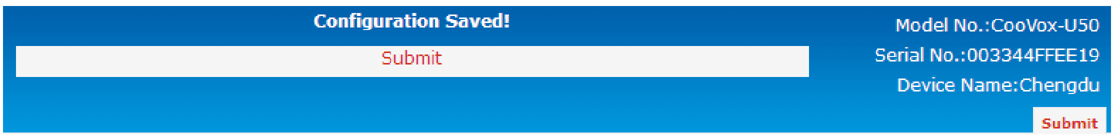
We need to add a prefix in front of the extension number and also define the length of the following extension number. Click "Save" when completed.

Next we need to configure detailed settings, please click【Node Management】->【Node List】, then click "Edit":



For detail information, please refer to Chapter 8.

Finally, click "Submit" to save.



Step 4: Push Configure File

After configuration has completed, you are required to push the configuration file to the CTN: Service Reload or Restart, System Reboot or Upgrade, Obtain System Log or Call Log, all of these configuration files can be pushed here.

Click【Node Management】->【Operation】:

Operation

	Name	Version	Push Conf.	Status	Operation	Firmware	Result	☐
1	Chengdu	1.0.5	Yes	Online	Service Reload			☐
2	Kielce	1.0.5	No	Online	Service Reload			☐
3	Doncaster	1.0.4	No	Online	Service Reload			☐
4	Dubai	1.0.5	No	Online	Service Reload			☐
5	Medellin	1.0.5	No	Online	Service Reload			☐
6	Miami	1.0.5	No	Online	Service Reload			☐
7	Hannover	1.0.5	No	Online	Service Reload			☐
8	Caracas	1.0.5	No	Online	Service Reload			☐
9	Taipei	1.0.5	No	Offline	Service Reload			☐
10	Monterrey	1.0.5	No	Offline	Service Reload			☐

Apply

For example, set the Push Conf option to "Yes" and the Command option to "Service Reload", and select the check box at the end of the line as shown below. Finally select "Apply", and the CTN will reboot and become active.

Finally the Result option will show "success".

Operation

	Name	Version	Push Conf.	Status	Operation	Firmware	Result	☐
1	Chengdu	1.0.5	Yes	Online	Service Reload		success	☑
2	Kielce	1.0.5	No	Online	Service Reload			☐
3	Doncaster	1.0.4	No	Online	Service Reload			☐
4	Dubai	1.0.5	No	Online	Service Reload			☐
5	Medellin	1.0.5	No	Online	Service Reload			☐
6	Miami	1.0.5	No	Online	Service Reload			☐
7	Hannover	1.0.5	No	Online	Service Reload			☐
8	Caracas	1.0.5	No	Online	Service Reload			☐
9	Taipei	1.0.5	No	Offline	Service Reload			☐
10	Monterrey	1.0.5	No	Offline	Service Reload			☐

Apply

Chapter 4 NODE MANAGEMENT

This chapter is about how to manage nodes, including Edit Configuration, Status Check, Command Order, Numbering Plan, Package List and so on.

4.1 Node List

From this list of nodes, you can check the status, detail configurations of each node, edit or delete nodes.

To check the node list, click【Node Management】->【Node List】:

Node List

	Name	Model	Last Update Time	Version	Status	Options
1	Chengdu	CooVox-U50	2014-06-08 11:57:51	1.0.5	Online	Detail Status Edit Delete
2	Kielce	CooVox-U20	2014-06-09 11:11:42	1.0.5	Online	Detail Status Edit Delete
3	Doncaster	CooVox-U20	2014-06-09 11:11:47	1.0.4	Online	Detail Status Edit Delete
4	Dubai	CooVox-U50	2014-06-09 18:38:11	1.0.5	Online	Detail Status Edit Delete
5	Medellin	CooVox-U50	2014-06-09 11:11:48	1.0.5	Online	Detail Status Edit Delete
6	Miami	CooVox-U50	2014-06-09 11:11:42	1.0.5	Online	Detail Status Edit Delete
7	Hannover	CooVox-U100	2014-06-09 11:11:51	1.0.5	Online	Detail Status Edit Delete
8	Caracas	CooVox-U50	2014-06-08 18:06:25	1.0.5	Online	Detail Status Edit Delete
9	Taipei	CooVox-U20	2014-06-09 11:11:51	1.0.5	Offline	Detail Status Edit Delete
10	Monterrey	CooVox-U20	2014-06-07 21:49:01	1.0.5	Offline	Detail Status Edit Delete
11	Tehran	CooVox-U50	2014-06-05 13:49:54	1.0.5	Online	Detail Status Edit Delete
12	Algiers	CooVox-U20	2014-06-09 11:00:19	1.0.5	Alarm	Detail Status Edit Delete
13	La Paz	CooVox-U20	2014-06-09 11:11:45	1.0.5	Online	Detail Status Edit Delete
14	Hanoi	CooVox-U20	2014-06-09 11:11:49	1.0.5	Online	Detail Status Edit Delete
15	Lima	CooVox-U20	2014-06-08 21:15:11	1.0.5	Online	Detail Status Edit Delete
16	Sofia	CooVox-U50	2014-06-05 09:51:36	1.0.5	Online	Detail Status Edit Delete
17	Santo Domingo	CooVox-U20	2014-06-07 18:02:17	1.0.4	Online	Detail Status Edit Delete
18	Bucharest	CooVox-U100	2014-06-07 18:02:17	1.0.5	Offline	Detail Status Edit Delete
19	Buenos Aires	CooVox-U20	2014-06-07 18:02:17	1.0.5	Online	Detail Status Edit Delete
20	Kuala Lumpur	CooVox-U20	2014-06-07 18:02:17	1.0.5	Online	Detail Status Edit Delete

Total:20 20 per page pages: << 1 >>

Reference:

Item	Explanation
Name	Name of node
Model	Model of node
Last Update Time	Last update time
Version	Version of node
Status	Connection status for node to CTMC (online/offline)
Options	Detail: Details of node

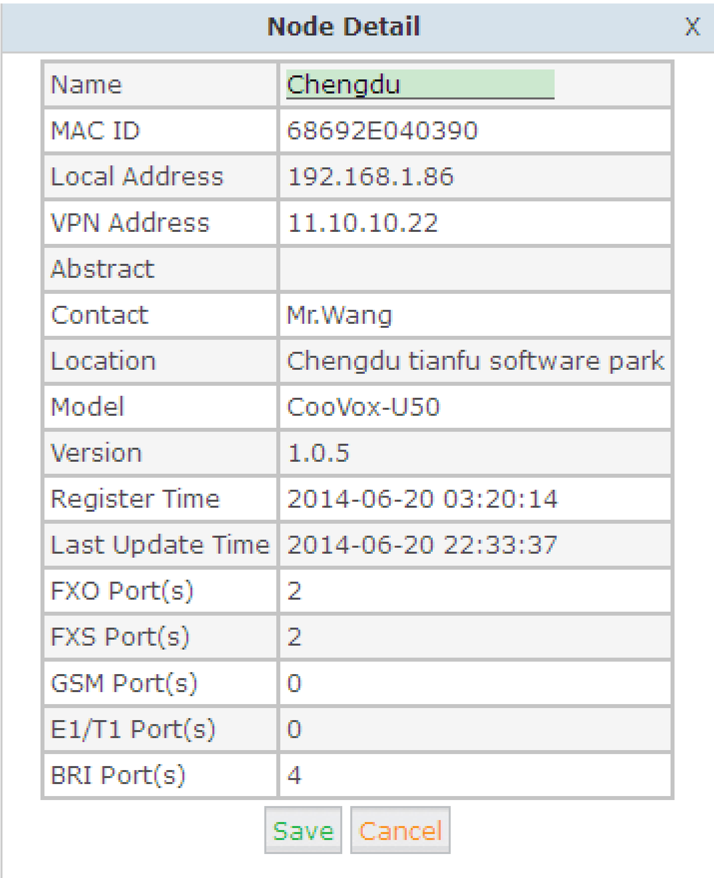
	Status： Monitor status of node, such as memory usage, port operation status Edit： Edit configuration of nodes, such as DialRule, configure special modules Delete： Delete node

Reference:

Item	Explanation
Online	This node can be monitored, managed and operated by CTMC when the node is online. It cannot be deleted .
Offline	This node cannot be monitored, managed or operated by CTMC when the node is offiline. It can be deleted.
Alarm	When VPN connection fails, this node cannot be monitored, managed and operated remotely. It can be deleted.

4.1.1 Node Detail

To check details of a node, click【Node List】->【Options】->【Detail】:


Node Detail X

Name	Chengdu
MAC ID	68692E040390
Local Address	192.168.1.86
VPN Address	11.10.10.22
Abstract	
Contact	Mr.Wang
Location	Chengdu tianfu software park
Model	CooVox-U50
Version	1.0.5
Register Time	2014-06-20 03:20:14
Last Update Time	2014-06-20 22:33:37
FXO Port(s)	2
FXS Port(s)	2
GSM Port(s)	0
E1/T1 Port(s)	0
BRI Port(s)	4

Save Cancel

Reference:

Item	Explanation
Name	Name of node. It can be edited when node is online.
MAC ID	MAC of node, the sole identifier of device
Local Address	Local Address of node
VPN Address	VPN address of node
Abstract	Abstract of node
Contact	Contacts of node
Location	Location of node
Model	Model of node
Version	Version of node
Register Time	The first time node is registered to CTMC
Last Update Time	Last update time
Version	Version of node
Status	Connection status of node with CTMC (online/offline)
FXO Port(s)	Number of FXO on nodes
FXS Port(s)	Number of FXS on nodes
GSM Port(s)	Number of GSM on nodes
E1 Port(s)	Number of E1 on nodes
BRI Port(s)	Number of BRI on nodes

4.1.2 Status of Nodes

To check node status, memory usage, port operation status, click【Node List】->【Options】->【Status】:

Node List → Status

Basic Status：

MAC ID	Disk Size	Time Zone	Host Name	Run Time	Reload Time	Version
68692E040390	/dev/root 3.0G 188.6M 2.6G 7% /	Asia/Chongqing	CooVox	System runtime: 5 days, 2 hours, 21 minutes, 3 seconds	Last reload: 2 minutes, 4 seconds	1.0.5
Bytes sent on ETH0	Bytes received on ETH0	Bytes sent on ETH1	Bytes received on ETH1	Register Status	Register Count	
220728860	895081908	0	0	Online	46	

FXO：

	Connect Status	call status
32	Disconnected	idle
33	Disconnected	idle

E1:

	Model	Protocol	Connect Status	Protocol Status
1	E1	CPE	Disconnected	Down

Reference:

Item	Explanation
MAC ID	MAC address of node, the sole identifier of device
Storage	Storage usage status
Time Zone	Time zone of node
Host Name	Host name of node
Run Time	Run time of node
Reload Time	Asterisk reload time of node
Version	Version of node
Bytes sent on ETH0	Bytes sent on ETH0 of node
Bytes received on ETH0	Bytes received on ETH0 of node
Bytes sent on ETH1	Bytes sent on ETH1 of node
Bytes received on ETH1	Bytes received on ETH1 of node
Registration Status	Registration status of node on CTMC (Online/Offline/Alarm)
Registration Count	Reconnection count of offline node

Check FXO instant status:

FXO：

	Connection Status	call status
1	Disconnected	idle
2	Disconnected	idle

Reference:

Item	Explanation
Port	Port of FXO
Connection Status	Connected Disconnected
Call Status	Idle Busy

Check GSM instant status:

GSM:

	Connect Status	call status
1	23	idle
2	NO SIM CARD	

Reference:

Item	Explanation
Port	Port of GSM

Connection Status	Signal strength of SIM card No SIM card
Call Status	Idle Busy

Check E1/T1 instant status:

E1:

	Model	Protocol	Connect Status	Protocol Status
1	E1	CPE	Disconnected	

Reference:

Item	Explanation
Port	Port of E1/T1
Mode	E1, T1
Protocol	NET, CPE, MFC
Physical Port	Connected Disconnected
Protocol Status	UP, DOWN

Check BRI instant status:

BRI:

	Connect Status	BRI Model
1	Not Settings	TE
2	Not Settings	TE
3	Not Settings	TE
4	Not Settings	TE

Reference:

Item	Explanation
Port	Port of BRI
Connection Status	Connected Disconnected
BRI Model	TE NT

4.1.3 Node Configurations

【Node Management】->【Edit】

For detail configuration of node, please refer to Chapter 8.

4.1.4 Delete Nodes

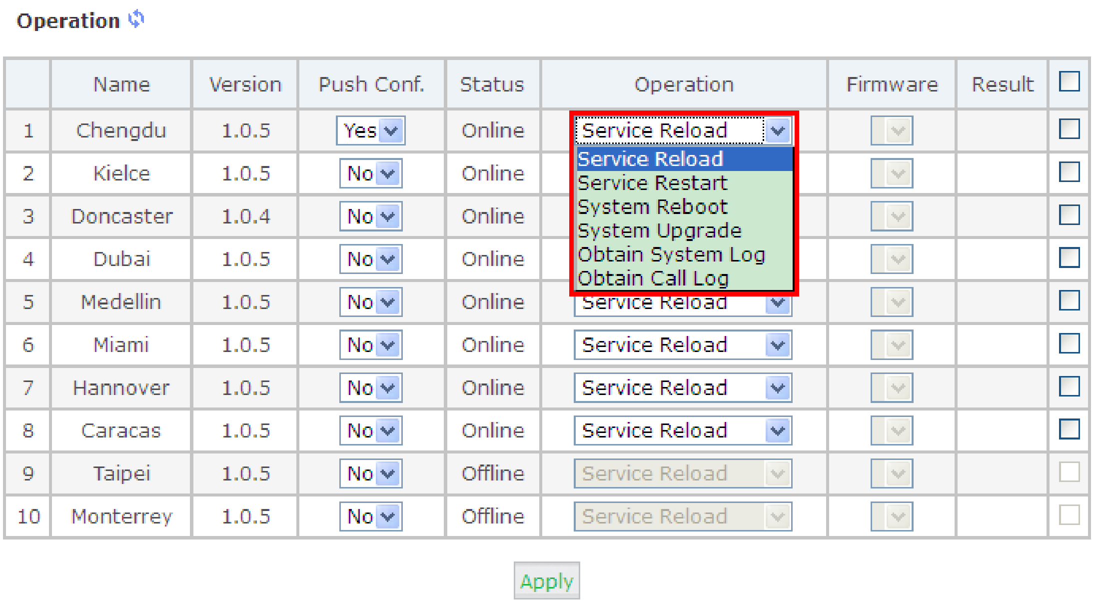
【Node Management】->【Delete】

When the node is offline or alarm, it's able to be deleted.

4.2 Operation

Every node is under control by CTMC; configure file need to be pushed via remote operation, including service reload & restart, system reboot & upgrade, obtain system log & call log.

Operation menu for nodes:


Operation

	Name	Version	Push Conf.	Status	Operation	Firmware	Result	
1	Chengdu	1.0.5	Yes	Online	Service Reload			
2	Kielce	1.0.5	No	Online				
3	Doncaster	1.0.4	No	Online				
4	Dubai	1.0.5	No	Online				
5	Medellin	1.0.5	No	Online	Service Reload			
6	Miami	1.0.5	No	Online	Service Reload			
7	Hannover	1.0.5	No	Online	Service Reload			
8	Caracas	1.0.5	No	Online	Service Reload			
9	Taipei	1.0.5	No	Offline	Service Reload			
10	Monterrey	1.0.5	No	Offline	Service Reload			

Reference:

Item	Explanation
Name	Name of node
Version	Version of node
Push Conf.	Push Configure file or not

Status	Connection status of node with CTMC (online/offline)
Operation	Service Reload Service Restart System Reboot System Upgrade Obtain System Log Obtain Call Log
Firmware	Firmware of node
Result	Results for APPLY (Query/Executing/Success/Fail)
□	Check item
APPLY	Click to apply

4.3 Numbering Plan

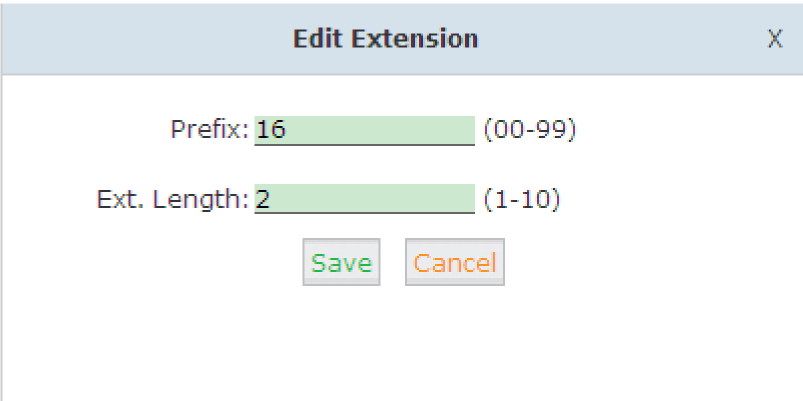
Numbering Plan is the most featured functionality for enterprises to achieve cost-savings. Each node will be assigned a numbering plan from CTMC centrally.

To check node's Numbering Plan, click【Node Management】->【Numbering Plan】:

Numbering Plan

	Name	Numbering Plan.		
		Prefix	Ext. Length	
1	Chengdu			Edit
2	Kielce	51	1	Edit
3	Doncaster	64	1	Edit
4	Dubai	32	3	Edit
5	Medellin	16	2	Edit
6	Miami	13	3	Edit

We will take node "Chengdu" as an example:

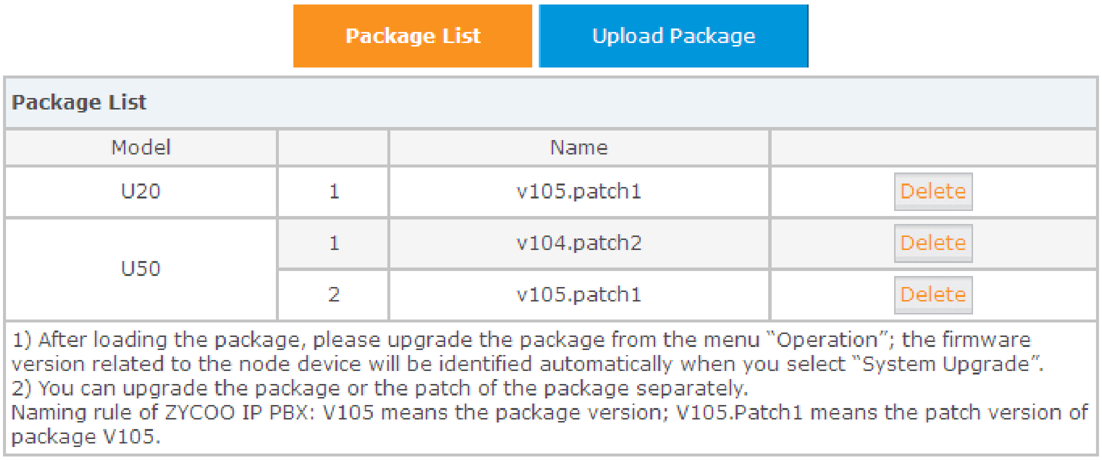


The device has been configured with a Prefix of 16 and following with 2 digits-extension, to confirm this click "Save". After creating the extensions, please dial 6500 to test.

4.4 Package List

All the package will be listed here. If you want to upgrade the node, please upload the relative firmware or patch package here; after uploaded the package, you need to push this configuration file from “Operation” .
Of cos, it’s allowed to be deleted if necessary.

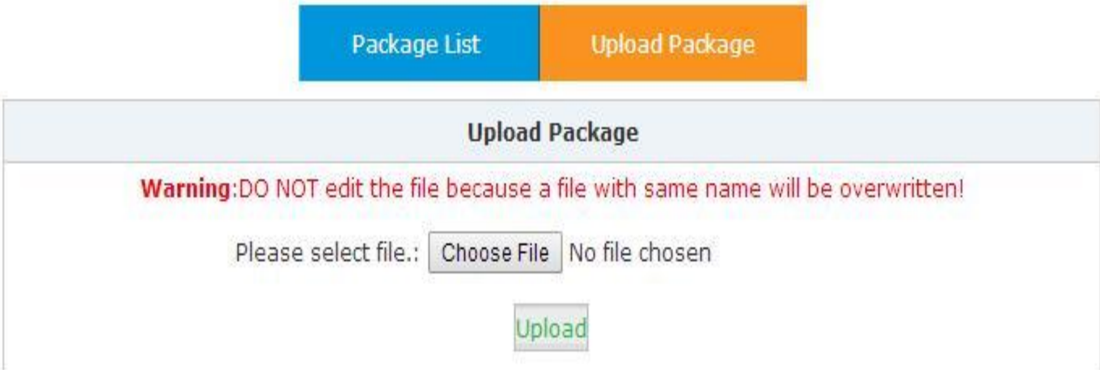
To check the package of nodes, click【Node Management】->【Package List】:



Package List			
Model		Name	
U20	1	v105.patch1	Delete
U50	1	v104.patch2	Delete
	2	v105.patch1	Delete

1) After loading the package, please upgrade the package from the menu "Operation"; the firmware version related to the node device will be identified automatically when you select "System Upgrade".
2) You can upgrade the package or the patch of the package separately.
Naming rule of ZYCOO IP PBX: V105 means the package version; V105.Patch1 means the patch version of package V105.

To upload a package or update a file, click “Choose File”:



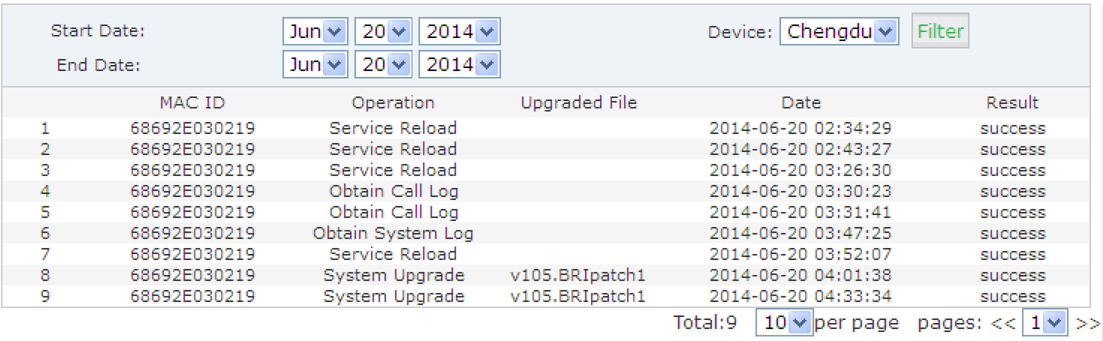
Chapter 5 LOGS

As the central management system, all the logs of each node are monitored by CTMC. This is much helpful for enterprises to manage and monitor the branch offices' operations remotely. System logs, operation logs and call logs are included.

5.1 Operation Logs

Check the operation history by operation logs.

Click【Logs】->【Operation Logs】, select the node you want to check and start date & end date, then click【Filter】:



Start Date: Jun 20 2014 Device: Chengdu Filter
End Date: Jun 20 2014

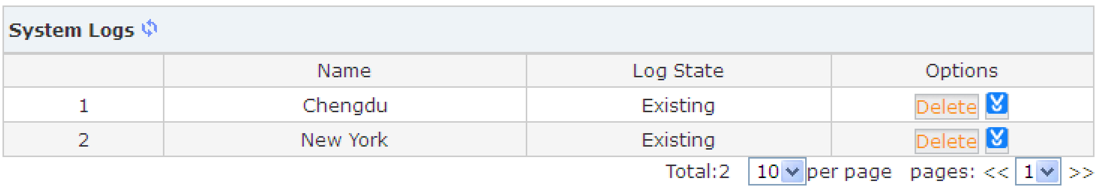
	MAC ID	Operation	Upgraded File	Date	Result
1	68692E030219	Service Reload		2014-06-20 02:34:29	success
2	68692E030219	Service Reload		2014-06-20 02:43:27	success
3	68692E030219	Service Reload		2014-06-20 03:26:30	success
4	68692E030219	Obtain Call Log		2014-06-20 03:30:23	success
5	68692E030219	Obtain Call Log		2014-06-20 03:31:41	success
6	68692E030219	Obtain System Log		2014-06-20 03:47:25	success
7	68692E030219	Service Reload		2014-06-20 03:52:07	success
8	68692E030219	System Upgrade	v105.BRIpatch1	2014-06-20 04:01:38	success
9	68692E030219	System Upgrade	v105.BRIpatch1	2014-06-20 04:33:34	success

Total:9 10 per page pages: << 1 >>

5.2 System Logs

Check the system logs of each node. It's allowed to be downloaded to the local to analyze and debug.

To check system logs of each node, click【Logs】->【System Logs】:



System Logs

	Name	Log State	Options
1	Chengdu	Existing	Delete
2	New York	Existing	Delete

Total:2 10 per page pages: << 1 >>

Reference:

Item	Explanation
Name	Name of node
Log Status	Not Existing: No logs, get system log on【Node Management】->【Operation】 Existing: Node system log obtained

Options	Download：Download to browse or backup Delete：Delete uploaded logs

5.3 Call Logs

Check the call logs of nodes.

Call logs allow you to run a report of calls on your node. As the management center, CTMC can check all the call logs of each node by filtering the device.

Select the start date & end date, and node device, then click【Filter】:

	Call Start	Caller ID	Destination ID	Duration(second)	Disposition
1	2014-05-29 15:40:38	"9600" <9600>	9400	2	ANSWERED
2	2014-06-12 11:24:49	"86002" <86002>	9401	32	ANSWERED
3	2014-06-12 11:25:26	"9401" <9401>	86002	8199	ANSWERED
4	2014-06-12 15:04:26	"9401" <9401>	86002	697	ANSWERED
5	2014-06-12 15:16:10	"9401" <9401>	86002	39	ANSWERED
6	2014-06-17 15:54:17	"9400" <9400>	8700	2	ANSWERED
7	2014-06-17 15:56:36	"8700" <8700>	9400	75	ANSWERED
8	2014-06-19 14:49:37	"9300" <9300>	9400	7	ANSWERED
9	2014-06-19 14:49:59	"9300" <9300>	9400	2386	ANSWERED

Total:9 10 per page pages: << 1 >>

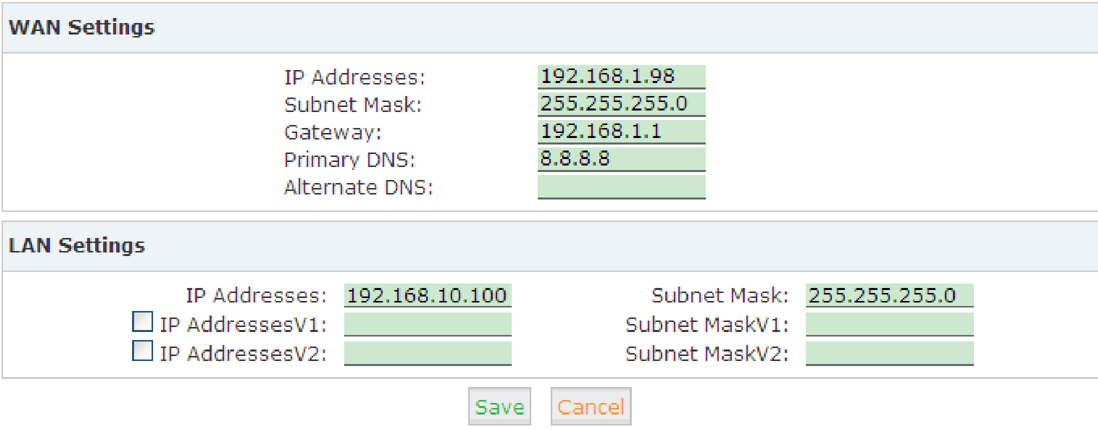
Chapter 6 NETWORK SETTINGS

This chapter is about how to make local network settings of CTMC.

6.1 Network Settings

Local network settings, including WAN settings & LAN settings. It's necessary to configure this to ensure the CTMC is connected to the public network.

To configure network settings of CTMC, click【Network Settings】->【Network Settings】:



WAN Settings:

Item	Explanation
IP Address	Set a static IP address
Subnet Mask	Default
Gateway	Default
Primary DNS	Default
Alternative DNS	Default

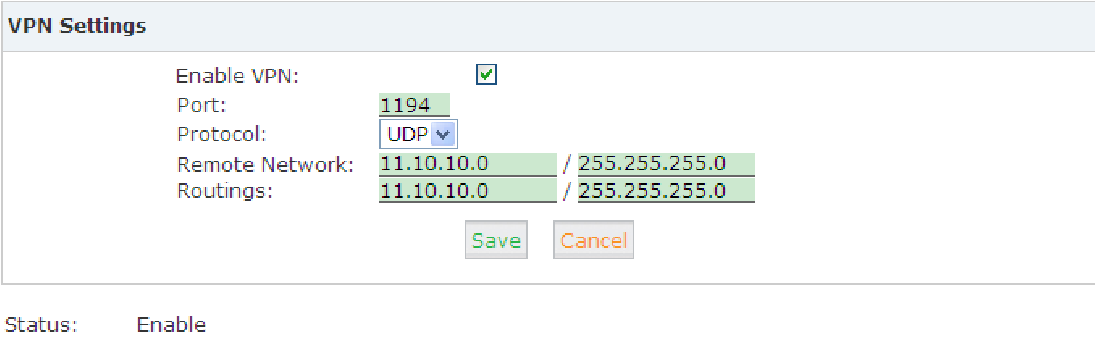
LAN Settings:

Item	Explanation
IP Address	Set a static IP address
Subnet Mask	Default

6.2 VPN Settings

The interactive communication between CTMC and its CTNs must be passed through VPN. The default is OpenVPN. Here is OpenVPN server settings of CTMC, including port, protocol, IP segment and so on.

To configure the VPN server, click【Network Settings】->【VPN Settings】:



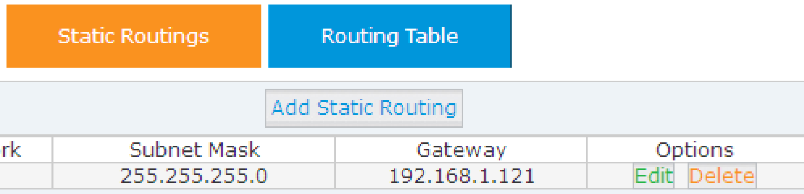
Reference:

Item	Explanation
Enable VPN	Enable/Disable VPN
Port	Default 1194
Protocol	UDP/TCP
Remote Network	Default
Routings	Default

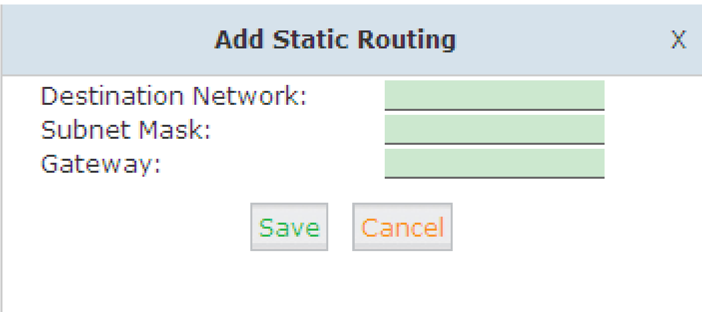
6.3 Static Routings

Local static routings management. There is default static routing; if you need more static routings, please add by the following instructions:

To configure static address routings, click【Network Settings】->【Static Routings】->【Edit】:



Click【Add Static Routing】

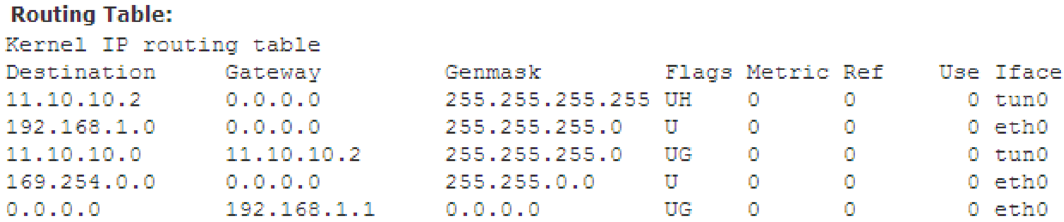


Reference:

Item	Explanation
Remote Network	Network (IP segment) of remote network
Subnet Mask	Subnet mask of remote network

Gateway	Gateway of remote network

Click【Network Settings】->【Static Routings】->【Routing Table】to check status of current routing:



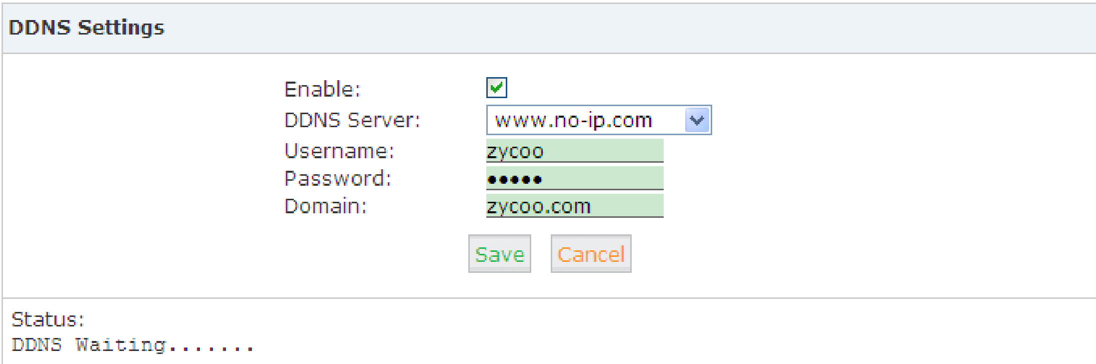
Routing Table:

```
Kernel IP routing table
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
11.10.10.2      0.0.0.0         255.255.255.255 UH    0      0        0 tun0
192.168.1.0     0.0.0.0         255.255.255.0   U     0      0        0 eth0
11.10.10.0      11.10.10.2      255.255.255.0   UG    0      0        0 tun0
169.254.0.0     0.0.0.0         255.255.0.0     U     0      0        0 eth0
0.0.0.0         192.168.1.1     0.0.0.0         UG    0      0        0 eth0
```

6.4 DDNS Settings

DDNS allows dynamic addresses to be mapped and tracked to a host name.

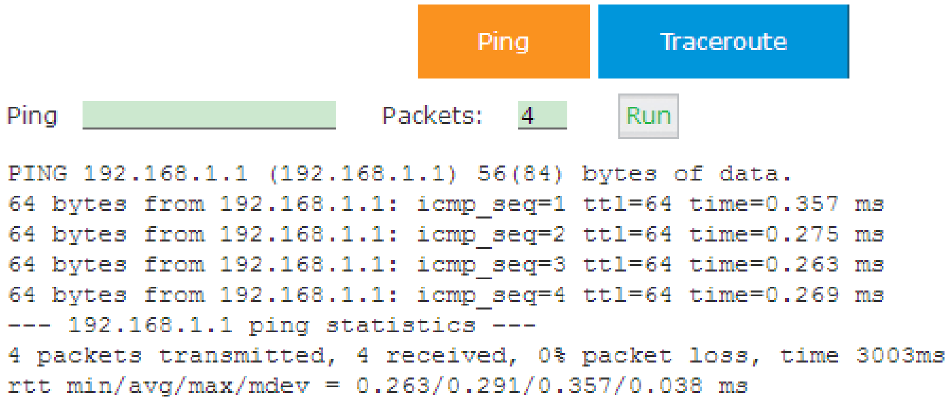
To configure DDNS, Click【Network Settings】->【DDNS Settings】:



The device supports Dyndns.org/No-ip.com/zoneedit.com for now.

6.5 Troubleshooting

Troubleshooting section allows you to confirm the status of the network by performing simple diagnostics including, ping to other network devices or Traceroute command to trace network routings, click【Network Settings】->【Trouble Shooting】:



Note: Port 9998, 8505, 8506, 1194 must be open when CTMC is in the public network.

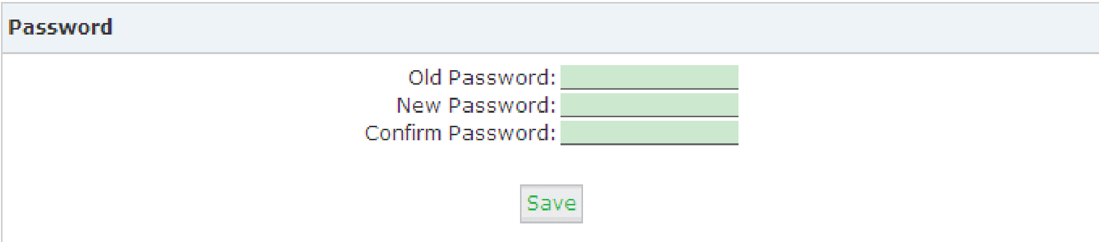
Chapter 7 SYSTEM SETTINGS

This chapter is about how to make system settings of CTMC.

7.1 Change Password

Change password of CTMC.

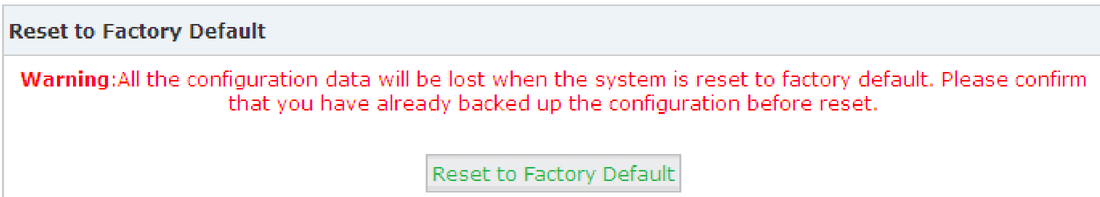
To change CTMC's login password, click【System Settings】->【Password】



7.2 Reset & Reboot

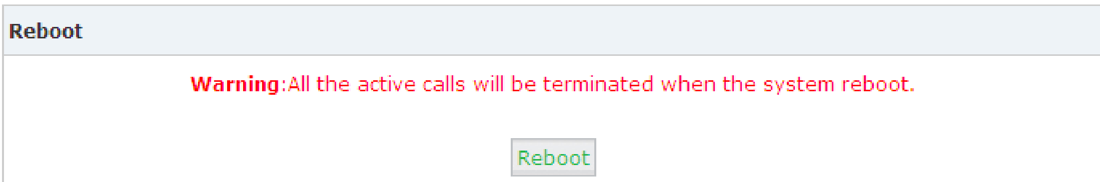
Manage CTMC, to reset to factory default, or reboot.

To reset to factory default, click【System Settings】-> 【Reset & Reboot】->【Reset to factory Default】:



Note: All the configuration data will be lost when the system is reset to factory default. Please confirm that you have already backed up the configuration before reset.

To reboot CTMC, click【System Settings】->【Reset & Reboot】->【Reboot】:

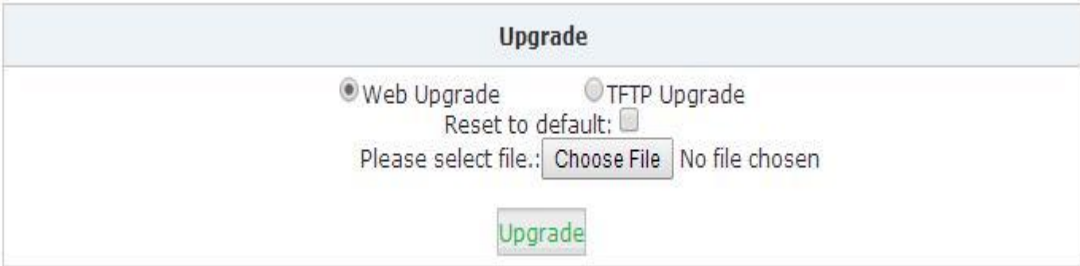


Note: All the active calls will be terminated when the system reboots.

7.3 Upgrade

Upgrade CTMC to latest firmware, to make the system working under the best condition. Before upgrading, you need to download the latest firmware of CTMC from ZYCOO official website: www.zycoo.com.

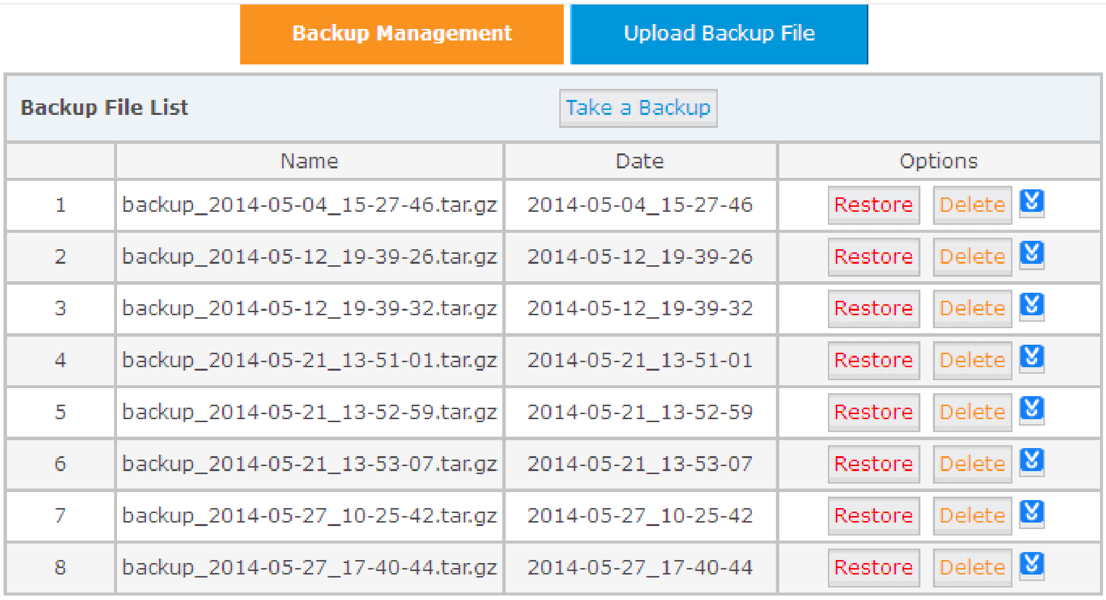
To upgrade CMTC firmware, click【System Settings】->【Upgrade】



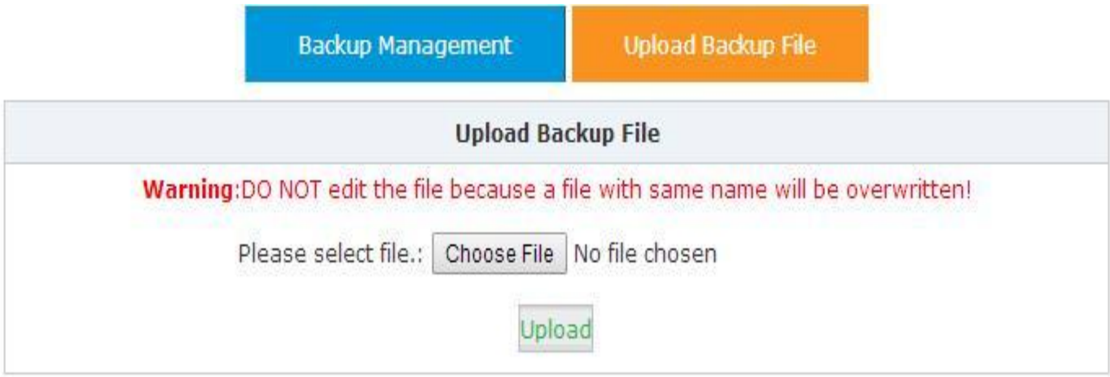
7.4 Backup

Backup is important for any enterprise in case you need restore the system. The backup includes the settings of CTMC and function configuration of its nodes. Here you need backup the CTMC system manually, or upload to the local for restoration purpose.

To backup the configuration and logs files of CTMC or nodes, click【System Settings】->【Backup】



	Name	Date	Options
1	backup_2014-05-04_15-27-46.tar.gz	2014-05-04_15-27-46	Restore Delete
2	backup_2014-05-12_19-39-26.tar.gz	2014-05-12_19-39-26	Restore Delete
3	backup_2014-05-12_19-39-32.tar.gz	2014-05-12_19-39-32	Restore Delete
4	backup_2014-05-21_13-51-01.tar.gz	2014-05-21_13-51-01	Restore Delete
5	backup_2014-05-21_13-52-59.tar.gz	2014-05-21_13-52-59	Restore Delete
6	backup_2014-05-21_13-53-07.tar.gz	2014-05-21_13-53-07	Restore Delete
7	backup_2014-05-27_10-25-42.tar.gz	2014-05-27_10-25-42	Restore Delete
8	backup_2014-05-27_17-40-44.tar.gz	2014-05-27_17-40-44	Restore Delete

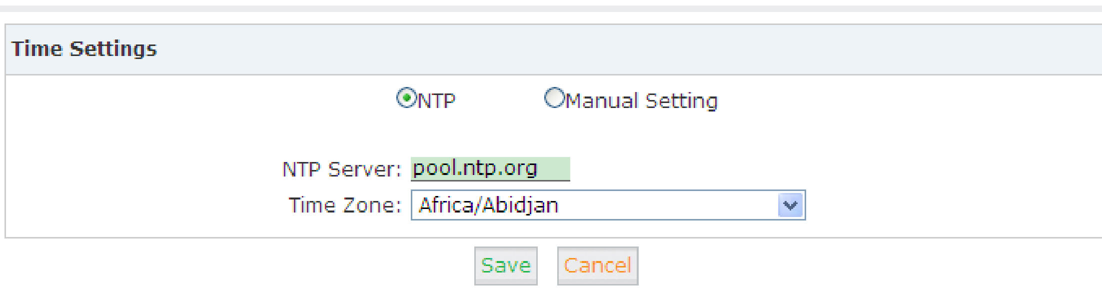


7.5 Time Settings

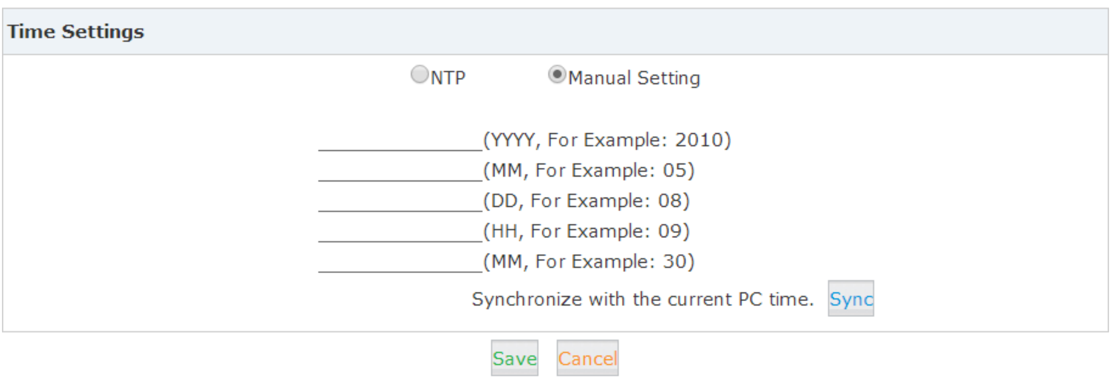
Time settings of CTMC. Time can be synchronized via setting the NTP server, or set manually.

To set the time zone & time of CTMC, click 【System Settings】 -> 【Time Settings】

Synchronize via NTP Server:



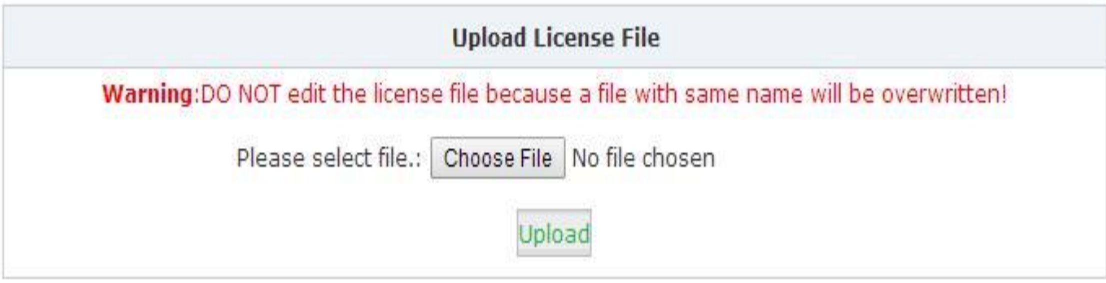
Manual Setting:



7.6 License Upload

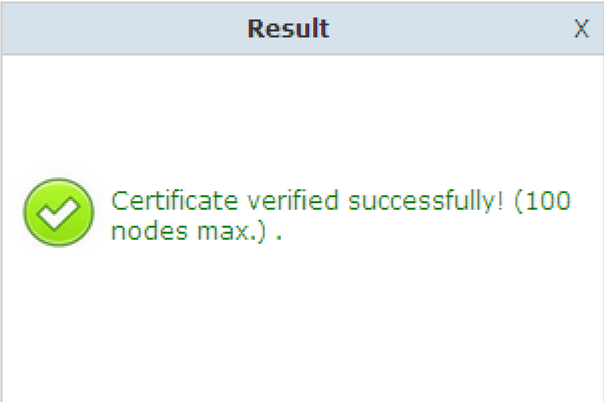
With expanding of the enterprise business, you need to increase the number of nodes on CTMC then you will need to buy the license and upload a new license, click【System Settings】

->【License Upload】

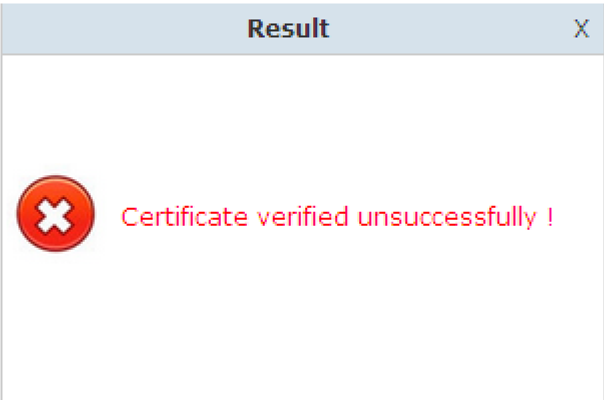


Note: License is provided by ZYCOO or ZYCOO Distributor.

If license upload was successful then you will receive the following notification:



If license upload failed then you will receive the following notification:

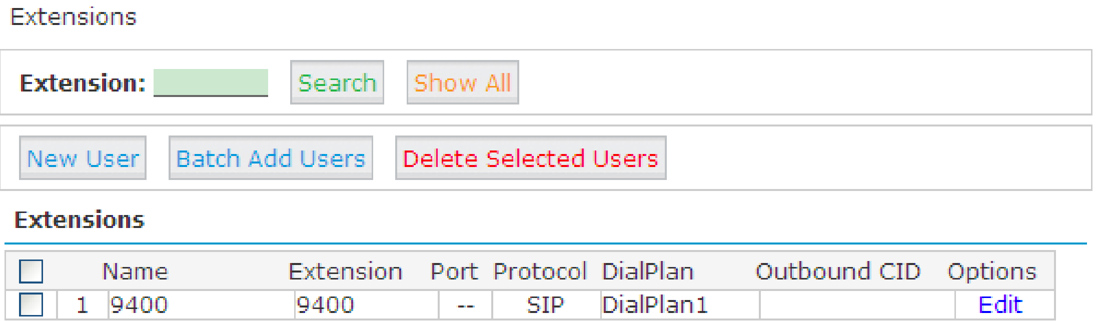


Chapter 8 NODES CONFIGURATION

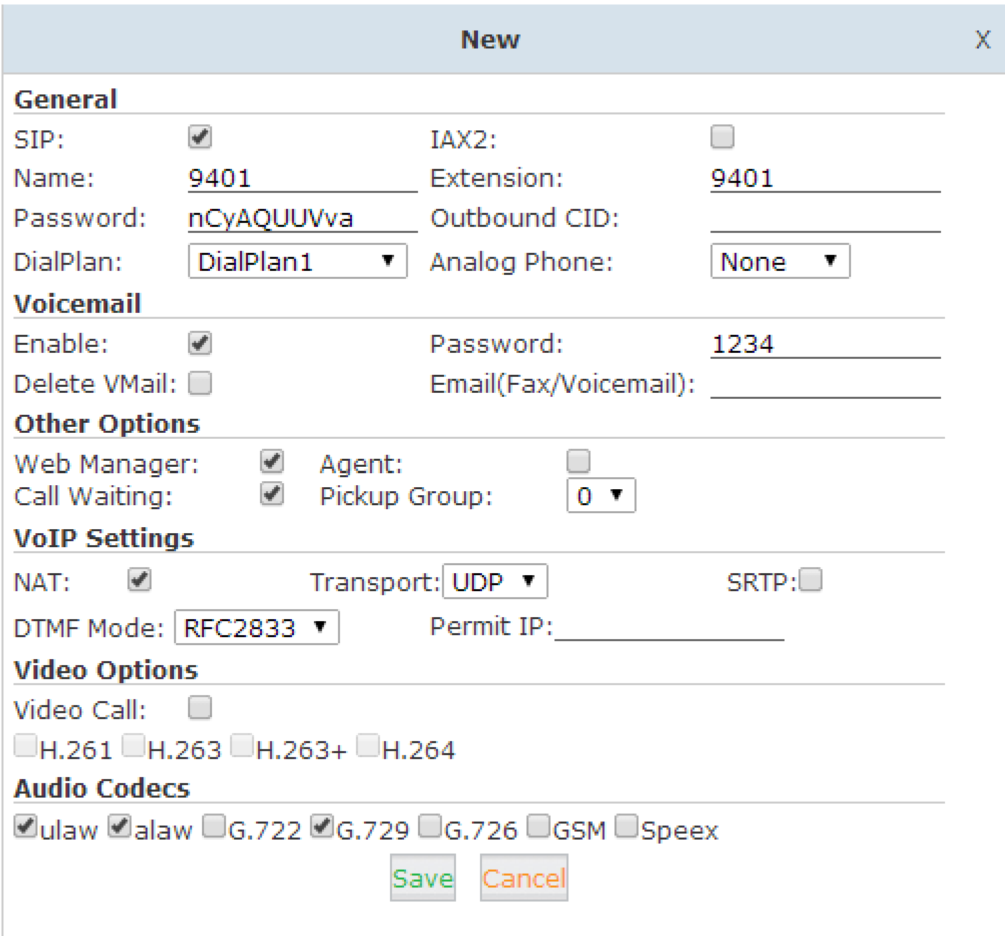
8.1 Extensions

This device 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:

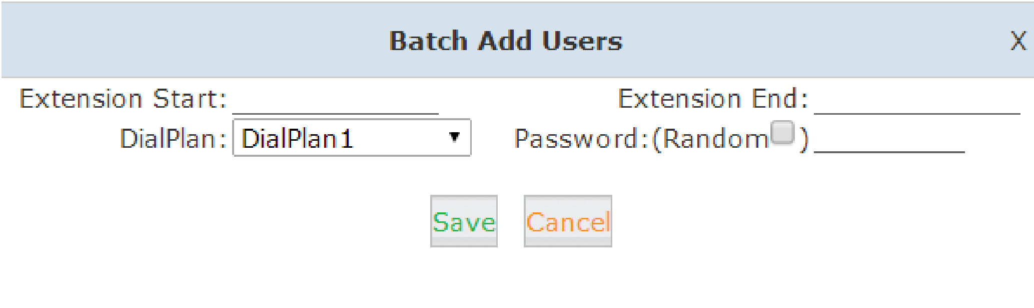


Extension Settings Reference:

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
Pickup Group	Select the Pickup Group which the extension user belongs to.
NAT	Check this option if extension user or the phone is located outside the NAT(Network Address Translation) available gateway.
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..
Permit IP	Set device ip address or subnet permitted to register to 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 permitted to register to this extension.
Video Call	Check to enable video calling for this extension. And select the video codecs you need to use.
Audio Codecs	Select what audio codecs you need to use.

Batch Add Users

It's available to batch add users, please click【Batch Add Users】to see the following window:



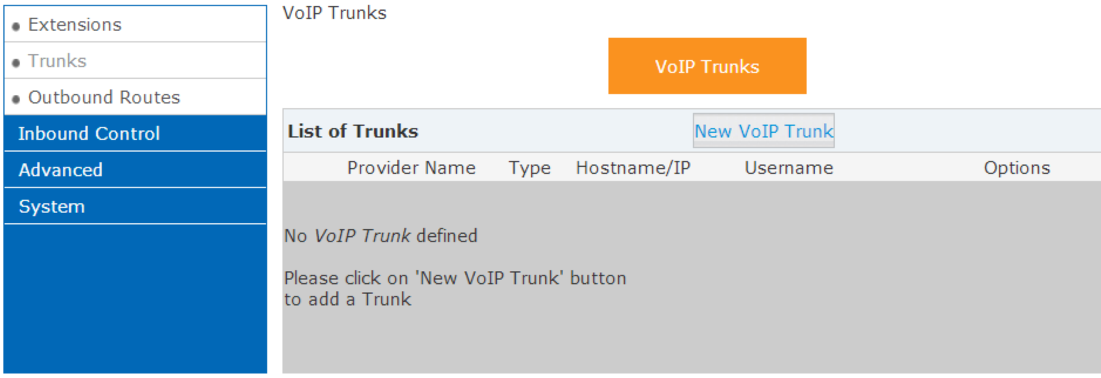
Input the extension number to start and end to define the extension range, select the dialplan for the extension; password can be random for each extension or defined to the same

Notice:

1. The quantity of extension numbers for each node is determined by the node device model. E.g.: Node Chengdu is CooVox-U50, CooVox-U50 supports 100 extensions max., then 100 extensions can be added to the node maximumly.
2. The admin can set the extension rule for each node to distinguish the node. E.g.: Node Chengdu is 8xx, Node NewYork is 7xx, Node Doncaster is 6xx...

8.2 Trunks

If you wish to configure an outbound trunk to connect to the PSTN (Public Switch Telephone Network) or VoIP provider then you need to configure it here: 【Basic】 -> 【Trunks】

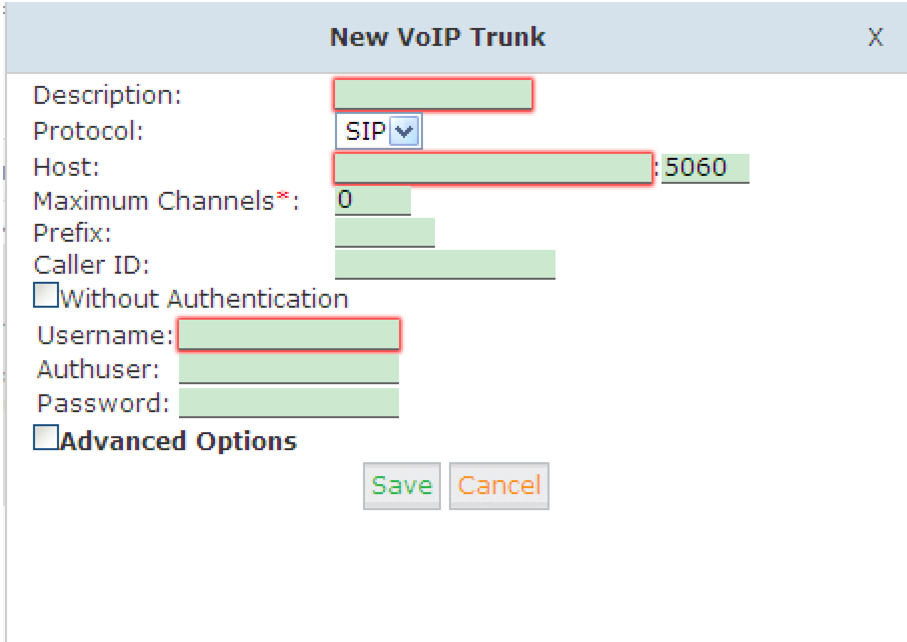


The device supports a choice of two types of trunk, customized VoIP/SIP trunk or FXO/GSM/BRI/PRI trunk. VoIP trunk can be configured here, but FXO/GSM/BRI/PRI trunk have to be configured from each CTN separately for different module settings.

The instructions below detail how to configure VoIP trunk type:

VoIP Trunks

Click【VoIP Trunk】->【New VoIP Trunk】:



VoIP Trunks Reference:

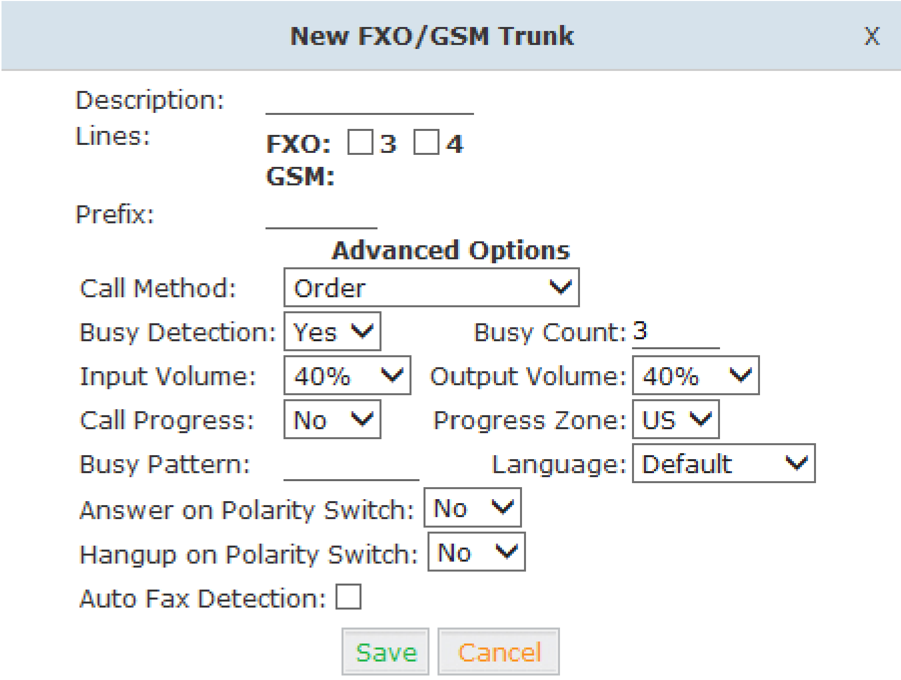
Item	Explanation
Description	Description of SIP trunk.
Protocol	Select protocol for outbound route, SIP or IAX2.
Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; “0” = no limitation).
Prefix	The prefix will be added 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 Authentication	If your trunk is static IP based and does not require a registration string when connecting the CooVox IP PBX, select 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 can be viewed in the list of VoIP Trunks when the trunk has been added successfully.

FXO/GSM Trunk

The following page is for CooVox-U20 only; FXO /GSM trunk settings on CooVox-U50/U100 will be similar as this.

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

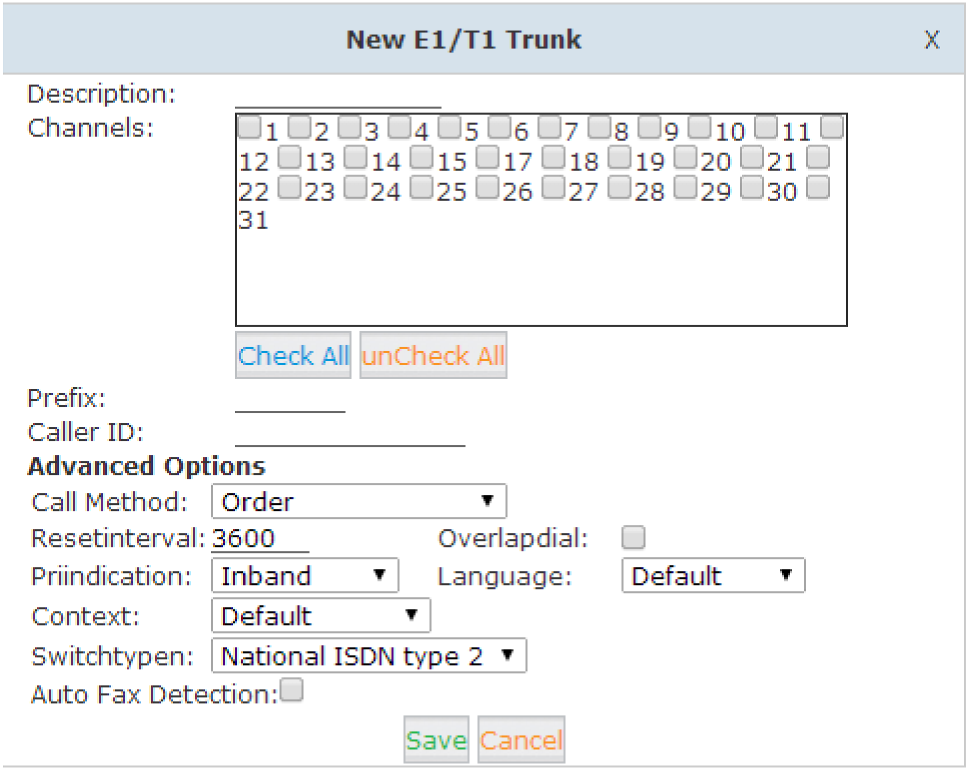


FXO/GSM Trunk Reference:

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.

E1/T1 Trunk

Click【E1/T1Trunks】->【New E1/T1 Trunk】:



E1/T1 Trunk Reference:

Item	Explanation
Description	Define description for the trunk
Channels	Individual channel of the trunk
Prefix	The prefix will be added to the dialed number automatically when this trunk is used.
Caller ID	Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'VoIP Trunks' screen will override the caller ID set in the 'Extensions' screen. Please note that aren't all service providers support this feature. Contact your service provider for more information.
Advanced Options	
Call Method	Call Method: It's used for how to use analog ports for this trunk. Order -- Order to select the non-busy analog channel. Reverse Order -- Reverse order to select the non-busy analog channel . Order Cycle -- Use round-robin search, starting at the next channel of the one that worked last time. Reverse Circulation -- Use round-robin search, starting at the previous channel of the one that worked last time.
Resetinterval	sets the time in seconds between restart of unused channels,defaults to 3600 minimum 60 seconds
Overlapdial	Whether Asterisk can dial this switch using overlap digits. Default: no.
Priindication	Tells how Asterisk should indicate Busy and Congestion to the switch/user. Default: inband.

Language	Define voice prompt language for the trunk.
Context	Select the dialplan for this trunk
Switchtypen	Sets the type of PRI switch being used. Default: National ISDN type 2.; Switchtype: Only used for PRI ; national: National ISDN 2 (default) ; dms100: Nortel DMS100 ; 4ess: AT&T 4ESS ; 5ess: Lucent 5ESS ; euroisdn: EuroISDN (common in Europe) ; ni1: Old National ISDN 1 ; qsig: Q.SIG
Auto Fax Detection	Detect the fax automatically

BRI Trunk

Click【BRI Trunks】->【New BRI Trunk】

BRI Trunk Reference:

Item	Explanation
Description	Define description for the trunk
Lines	Individual channel of the trunk
Prefix	The prefix will be added to the dialed number automatically when this trunk is used.
Caller ID	Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'VoIP Trunks' screen will override the caller ID set in the 'Extensions' screen. Please note that aren't all service providers support this feature. Contact your service provider for more information.
Advanced Options	
Echo Cancel	Echo cancellation
Overlapdial	Overlap dialing mode (sending overlap digits)
Method	Set the method to use for channel selection: Standard - Use the first free channel starting from the lowest number. Standard is default value! ReverseOrder – (standard_dec) Use the first free channel starting from the highest number. RoundRobin - Use the round robin algorithm to select a channel. Use this if you want to balance your load.

Context	Select the dialplan of this trunk
Language	Define voice prompt language for the trunk.

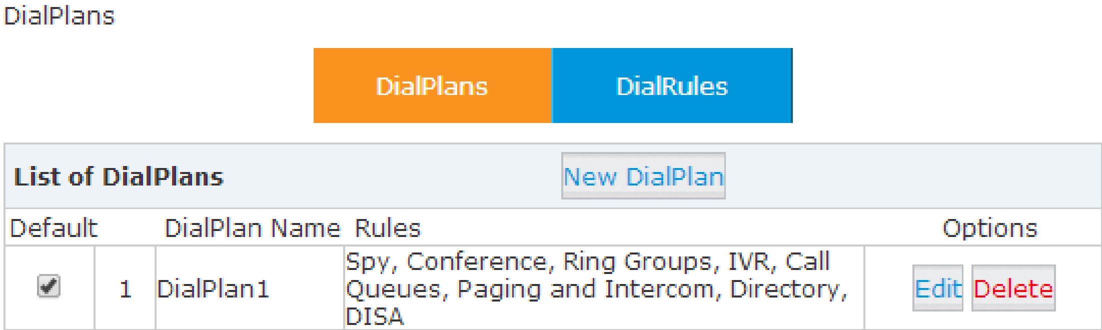
Select one or more of the available channels to be used for the trunk group.

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

8.3 Outbound Routes

Outbound Routes are used to define and control how outbound calls are made and controlled. If you do not allow an extension user to place external calls, please ignore this section.

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

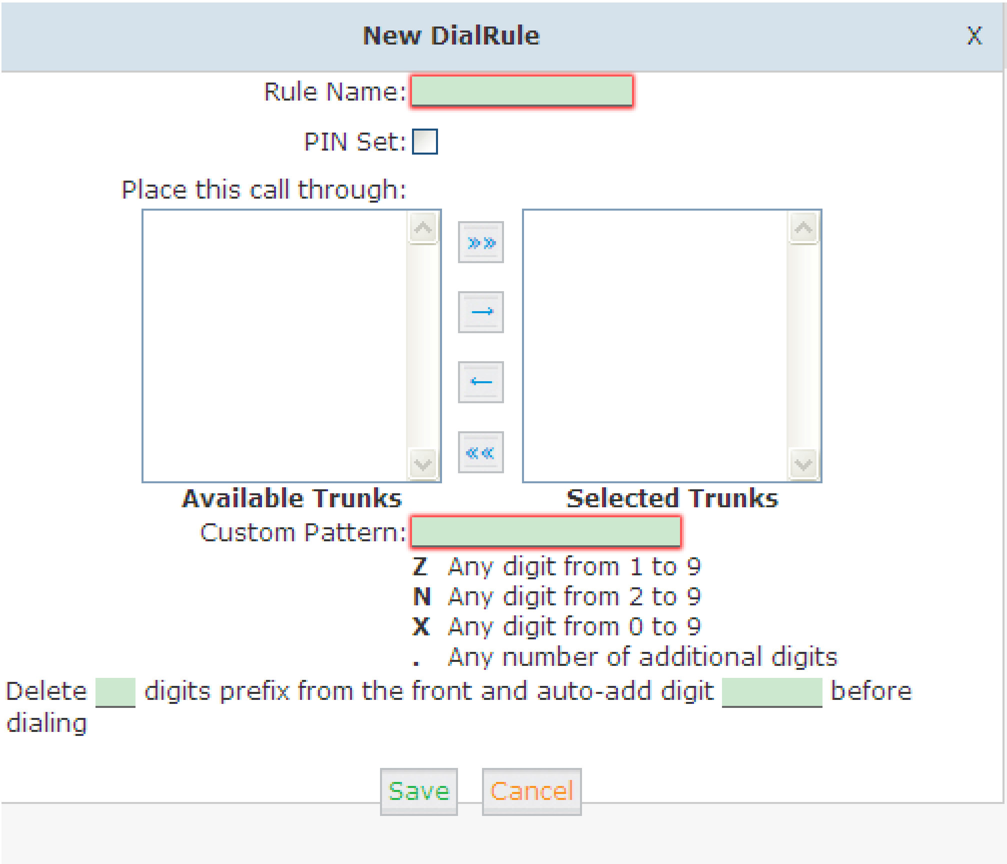


You can configure a basic match pattern of outbound routes and create different dial plans on this page. Dialplans are assigned to extensions and determine what types of calls an extension can make. For example, create "InternalDialPlan" to include all Internal Calling Rules but do not select any outbound dial rules. Select "InternalDialPlan" for all extension users that do not need the ability to make external calls.

Select【DialPlans】->【New DialPlan】:

DialRules

Dialrules defines patterns that will be used by the system to determine how to route a call. These are particularly useful if you have multiple trunks and you want to control how these trunks should be used.



Reference:

Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule (security feature).
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 (security feature)
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.

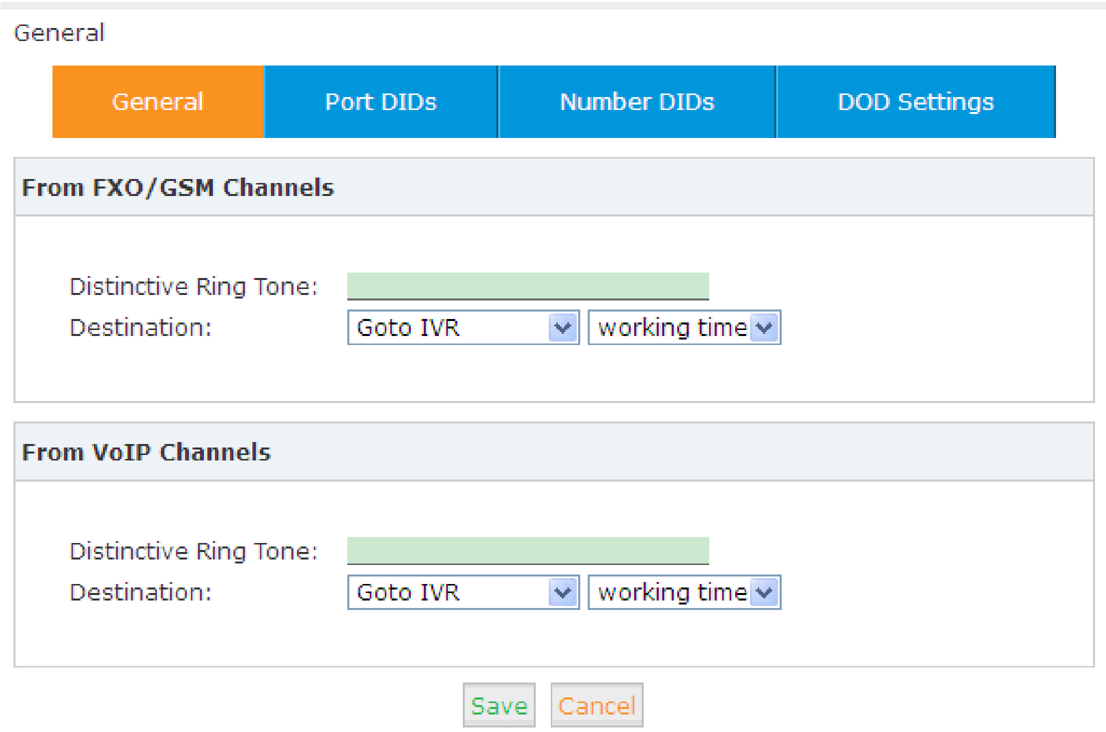
8.4 Inbound Control

8.4.1 Inbound Routes

Inbound Routes are used by the system to determine how external call should be routed, e.g. to an extension, IVR etc.

Select【Inbound Control】->【Inbound Routes】

（This page differs when the device model is different）



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.

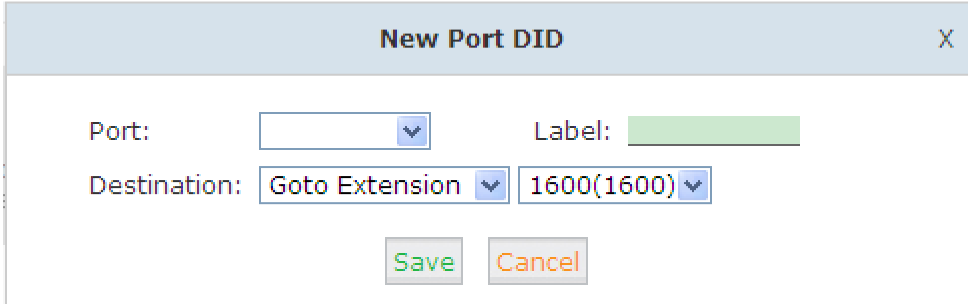
Note: The phone must support this feature as well.

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:

Select【Port DIDs】->【New Port DIDs】:

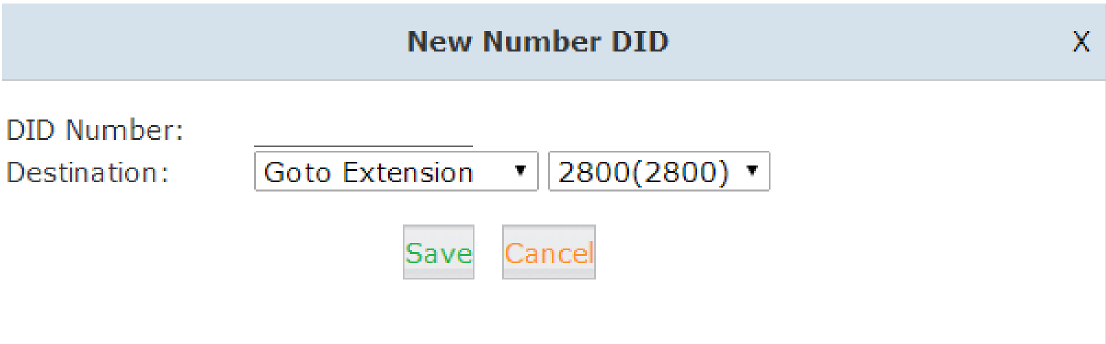


1. Port Select the trunk group port
2. Label Set a label for this port. Incoming calls from this port will display the specified label.
3. Destination Incoming calls will be answered by the specified destination (extension user, call queue, conference bridge, or IVR)

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:

Select【 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)

8.4.2 IVR

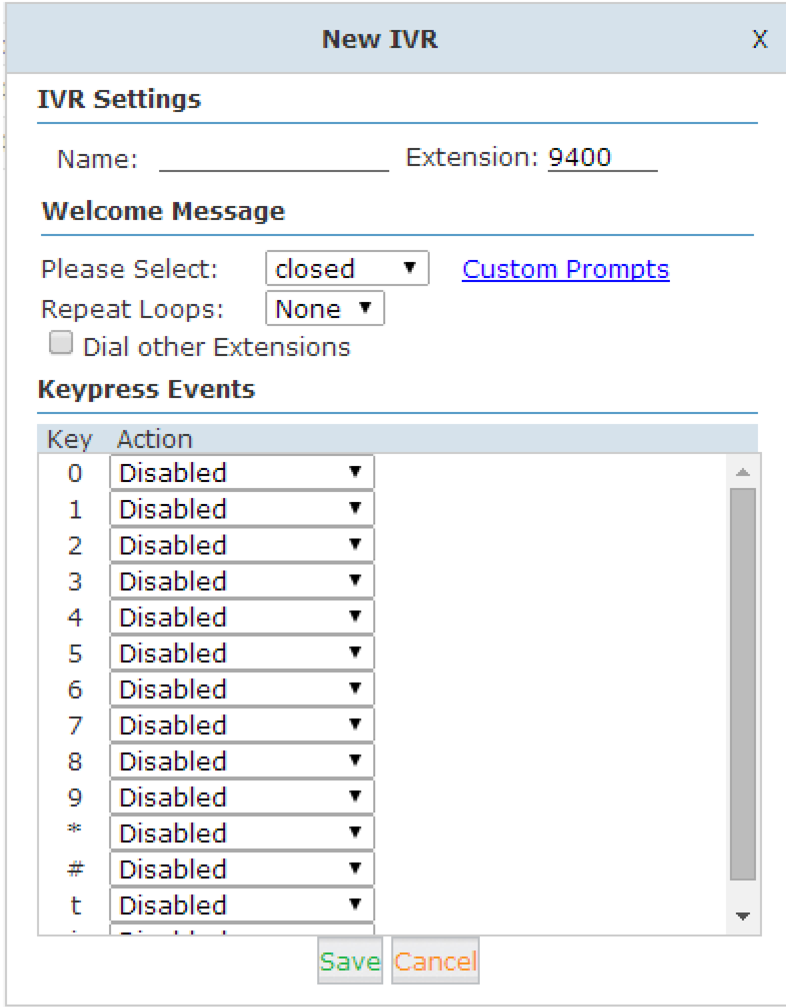
IVR (Interactive Voice Response) or Automated Attendant will allow callers to select from a specific set of options by pressing the selected digit on their telephone dial pad.

Select【Inbound Control】->【IVR】:

IVR

List of IVRs			New IVR	
	Extension	Name	Dial other Extensions	Options
1	9401	working time	Yes	Edit Delete
2	9402	closed time	No	Edit Delete

Select【New IVR】to create a new IVR:



IVR Reference:

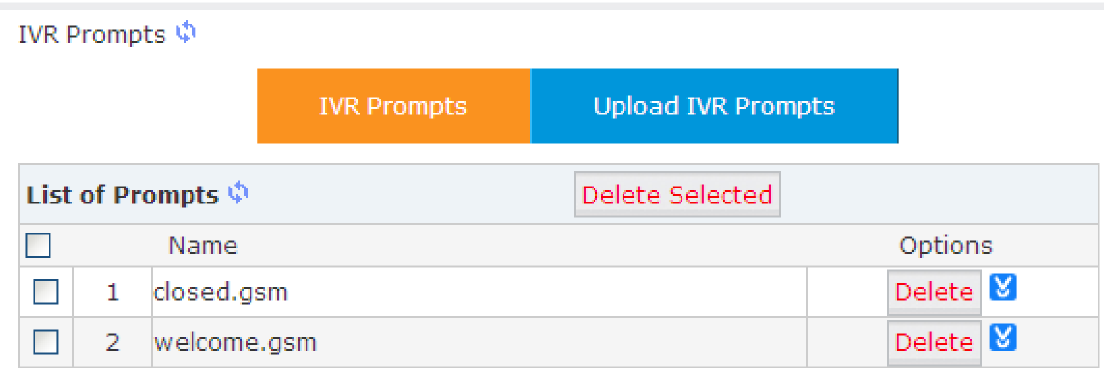
Item	Explanation
Name	Enter a descriptive name for the IVR
Extension	Enter a unique extension or IVR number. This number is used to access the IVR from an internal extension
Custom	Click "Custom" to choose a DialPlan for IVR
Please Select	Select the IVR prompt that will provide the caller with instructions 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 the listed options
Keypress Event	Select the available options beside the designated digit

8.4.3 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】



Upload IVR prompt

【Upload IVR prompt】

Notice:

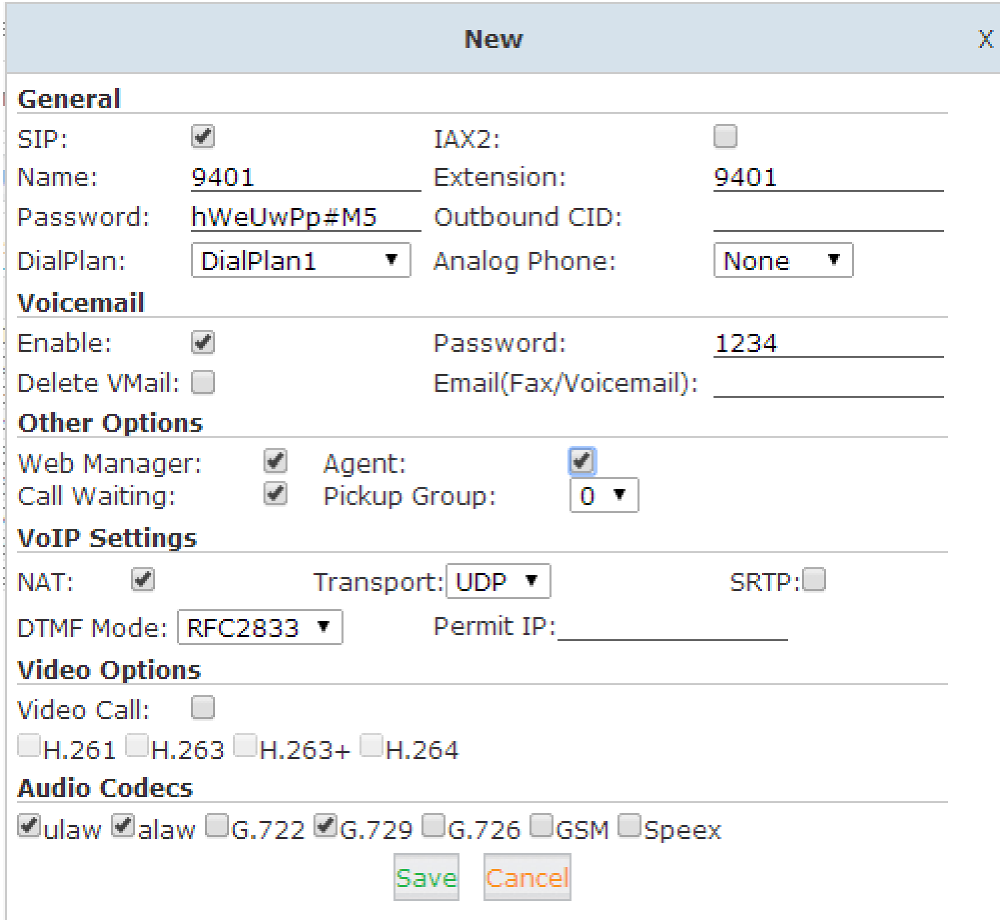
The device supports custom audio file with wav,gsm,ulaw,alaw format.
Recordings must be smaller than 15MB.

8.4.4 Call Queue

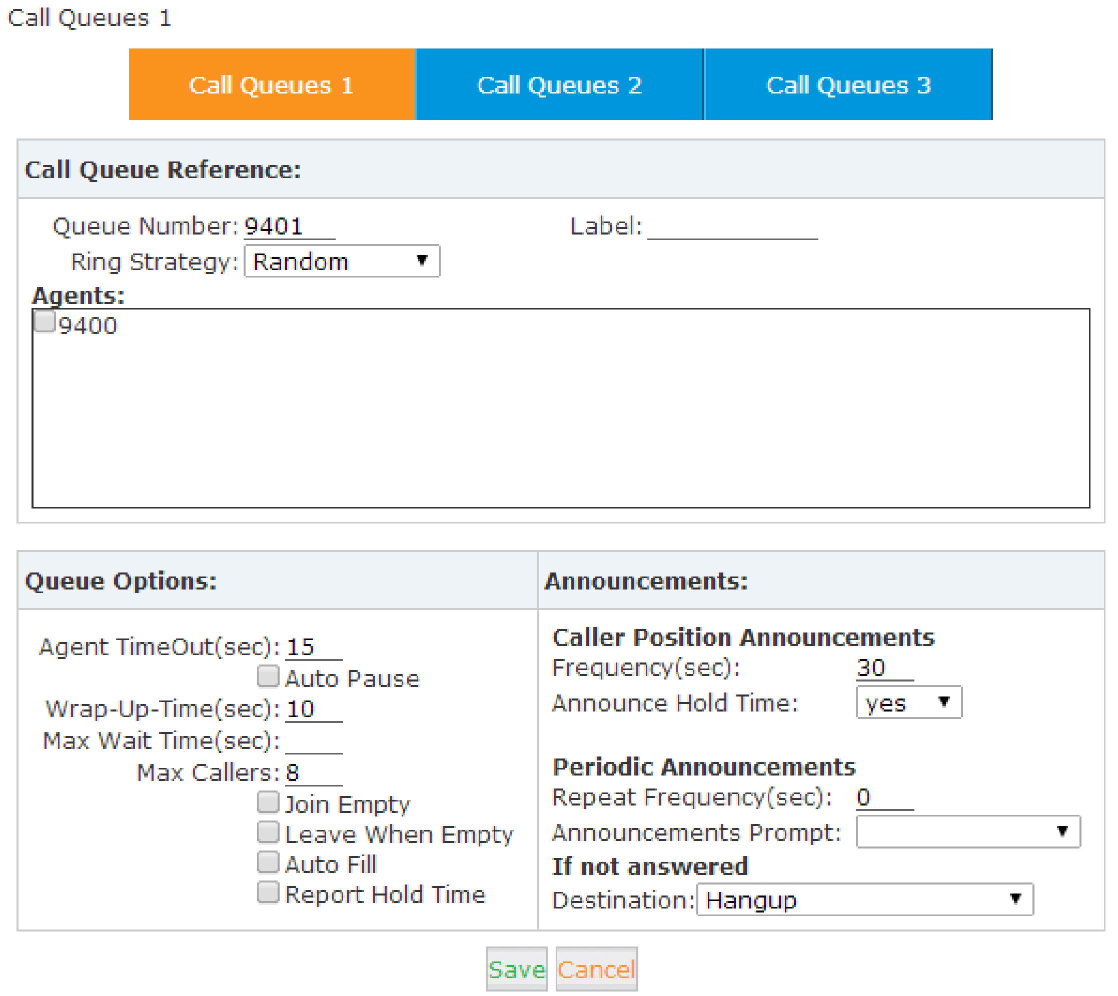
Create Agent

To allow a user to be considered an agent in a Call Center queue, please select the “Agent” option for the specific user extension.
Select【Basic】->【Extension】->【Edit】the extension you want to configure:

Step1: Select【Agent】and【Save】



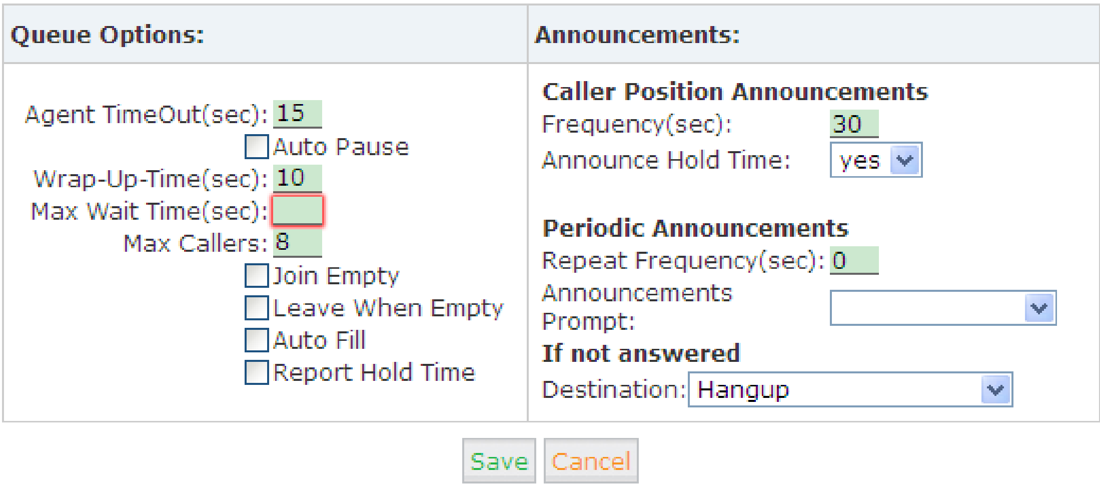
Step2: Select【Inbound Control】->【Call Queues】



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 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.

Queue Options & Announcements:



Reference:

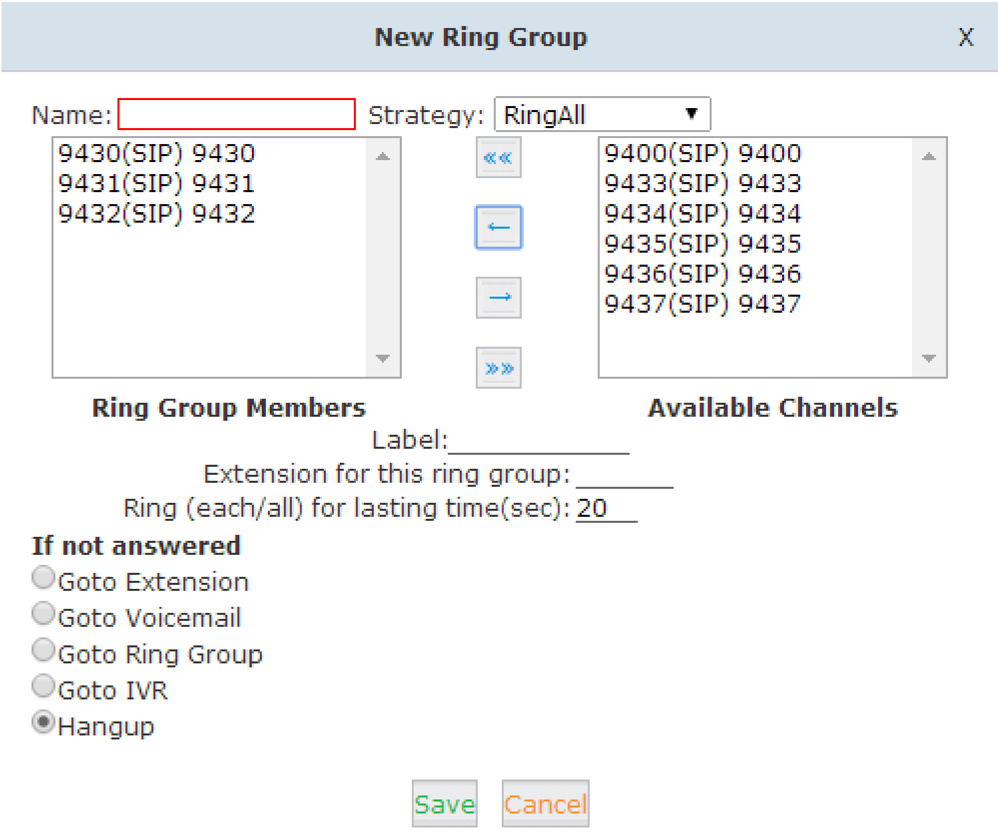
Item	Explanation
Agent TimeOut(sec)	Specify the number of seconds to ring an agent's extension before sending the call to the next Agent (based on Ring Strategy).
Auto Pause	If an Agent's extension rings and the Agent fails to answer the call, automatically pause that agent so they stop receiving calls 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	Announce the hold time. Announce (yes), do not announce (no) or

Time	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.

8.4.5 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 select【Inbound Control】-> 【Ring Groups】->【New Ring Group】:



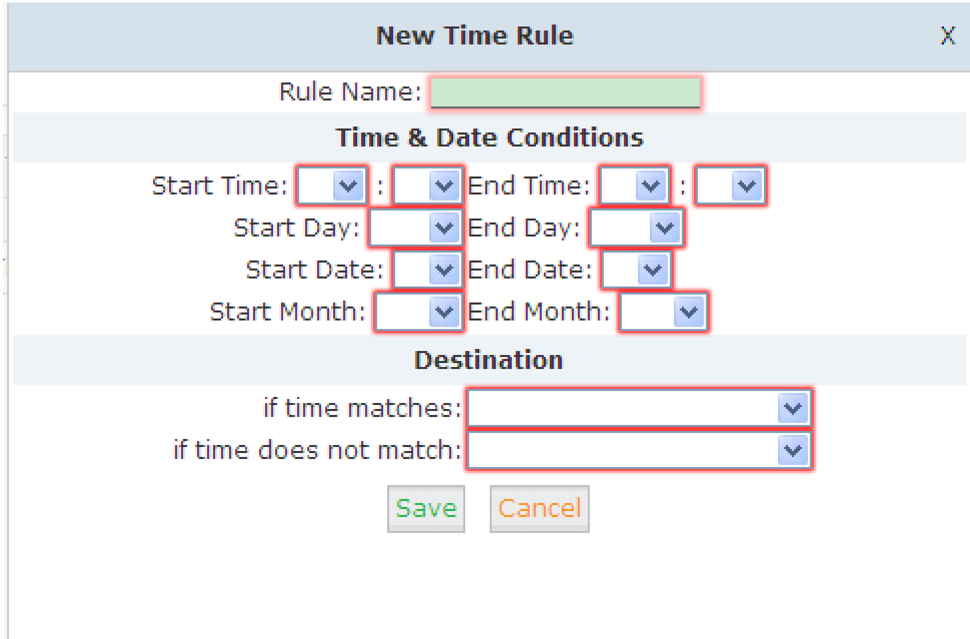
Item	Explanation
Name	Define a name for the Ring Group
Strategy	Select "Ring All" or "Ring in order"
Ring Group Members	Select the Ring Group Member from "the Available Channels", click  to add.

If not answered	You can choose to forward the call to extension, voicemail, ring group, IVR or hang up if not answered.

8.4.6 Time Based Rules

Create a Time Rule. For example, BusinessHours.
Select the start & end time, start & end days of the week, specific start & end dates and start & end 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 not used for the call.

Please configure a new time based rules 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

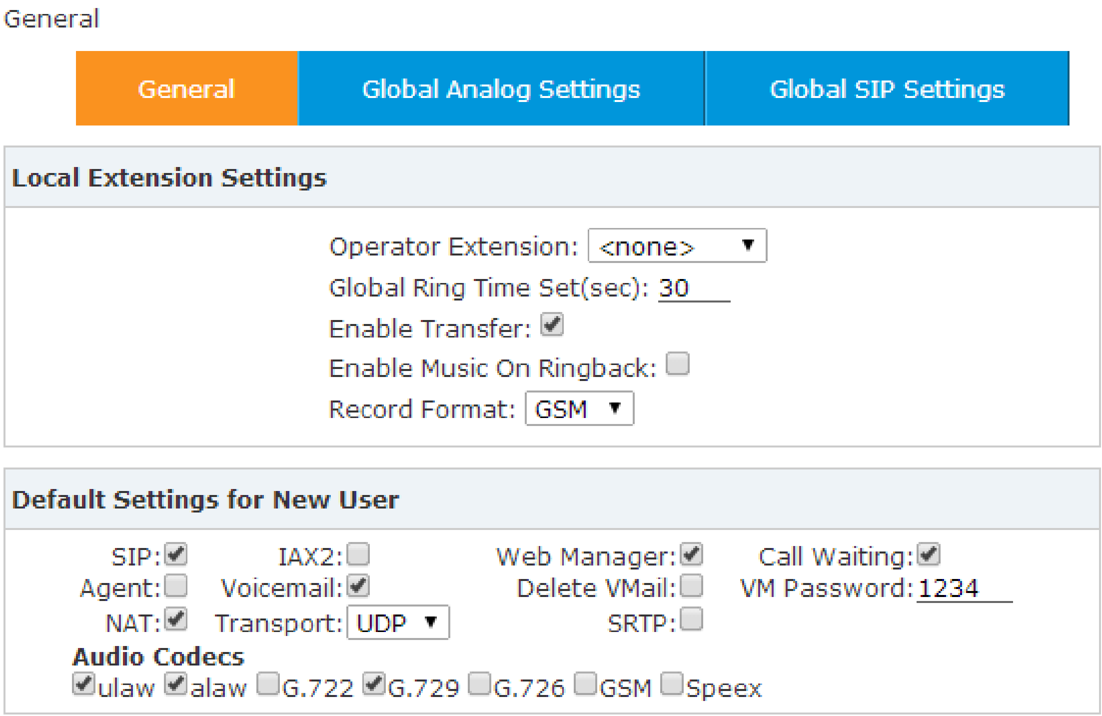
8.5 Advanced

8.5.1 Options

General

Default settings for local extensions and new extensions.

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

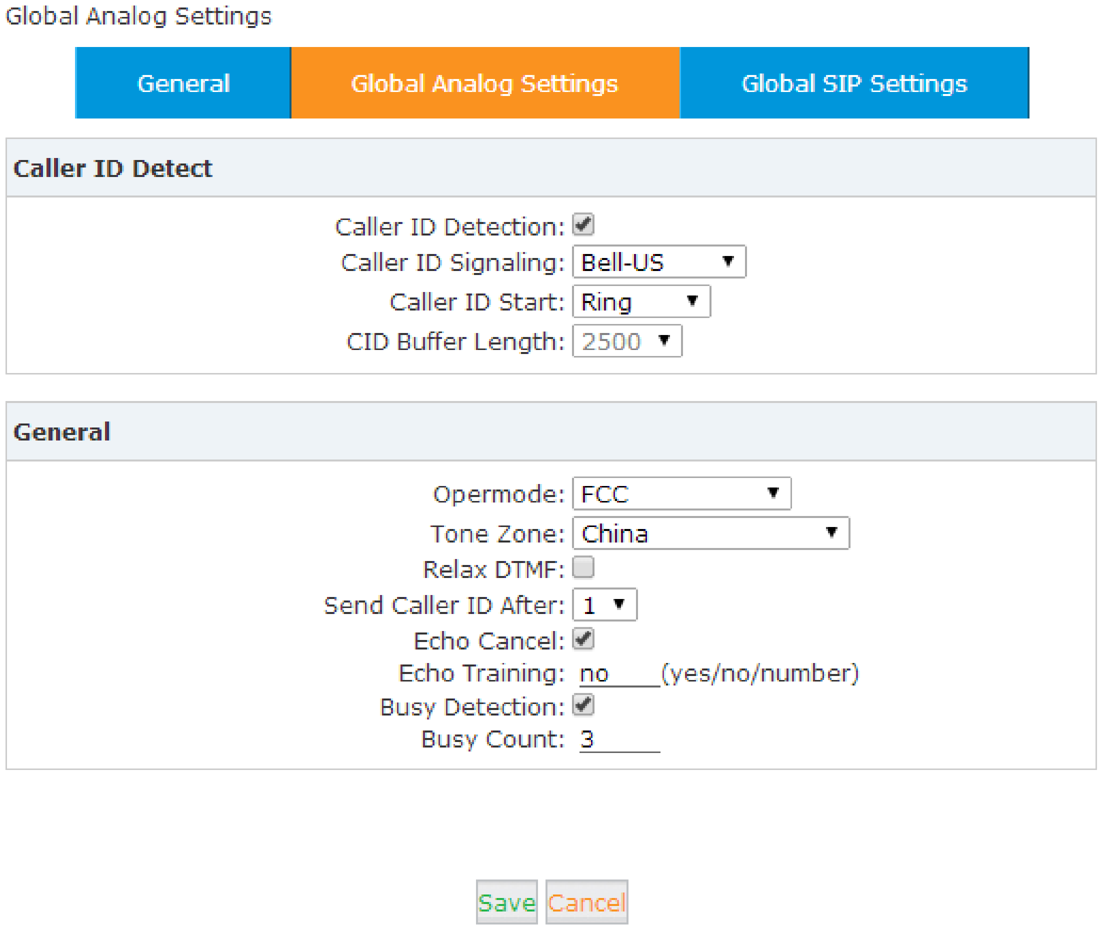


Reference:

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

Global Analog Settings

Select 【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.
Send Caller ID After	Some countries (UK) have ring tones with different ring tones (ring-ring), which means the caller ID needs to be set later on, and not just after the first ring
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

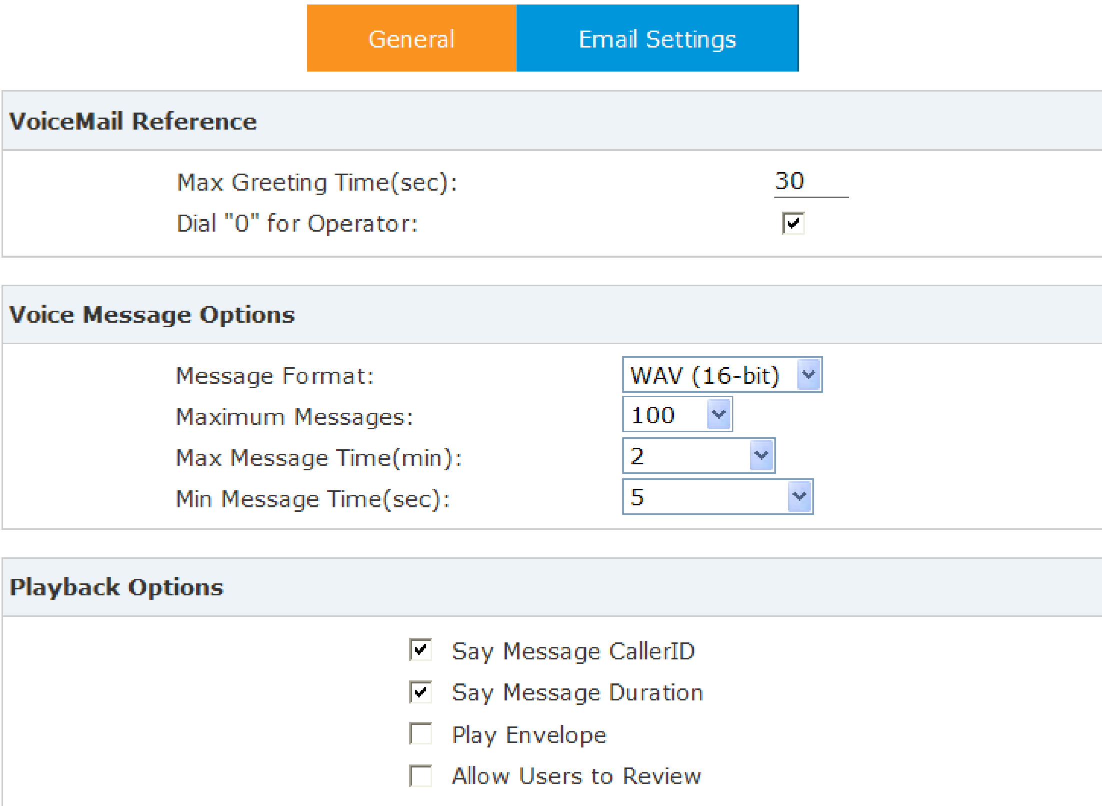
【Global SIP Settings】 is designed for advanced administrators.

Please contact our technical support department before modifying anything in this section.

8.5.2 Voicemail

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

General

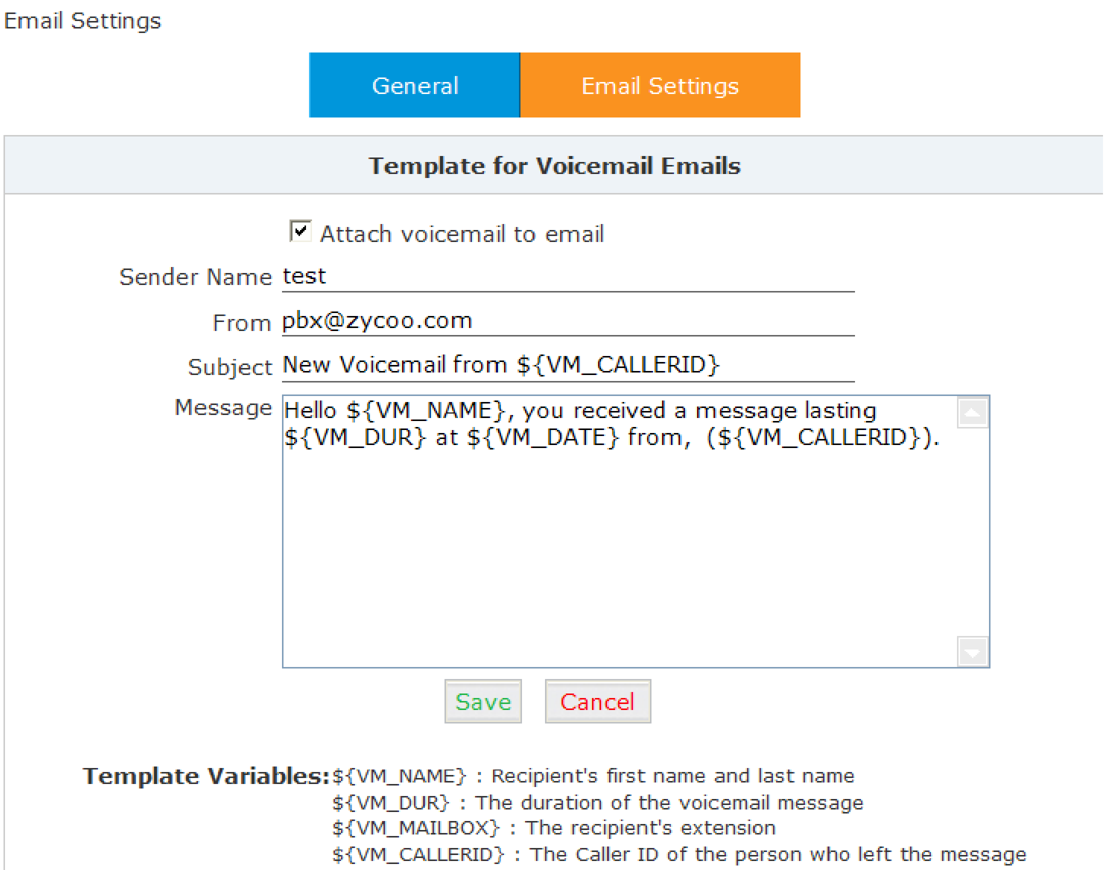


Reference:

Item	Explanation
MaxGreeting Time(sec)	Maximum recording length for voicemail greetings
Dial "0" for Operator	Select this option to allow callers to 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.

Select【Advance】->【Voicemail】->【Email Settings】:



Reference:

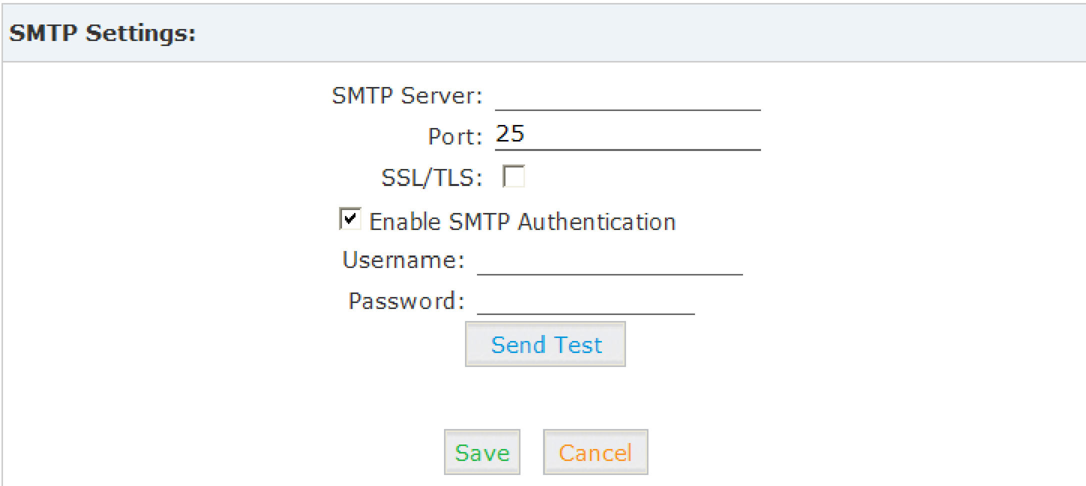
Item	Explanation
Attach voicemail to Email	The voicemail will be sent as attachment to the user's Email.
Sender Name	The sender's name will be displayed when you receive the Email.
From	Mailbox to send email
Subject	Subject of the Email.
Message	Input the Email template.

8.5.3 SMTP Settings

In order to allow email messages to be sent to users with attached voicemail and faxmail messages, the SMTP settings must to be configured.

Select【Advance】->【SMTP Settings】:

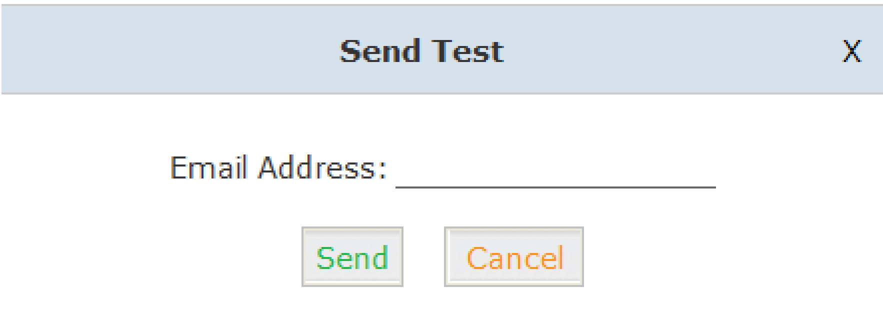
SMTP Settings



Reference:

Item	Explanation
SMTP Server	You must set SMTP Server address or domain connected to the CooVox 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/TLS	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.

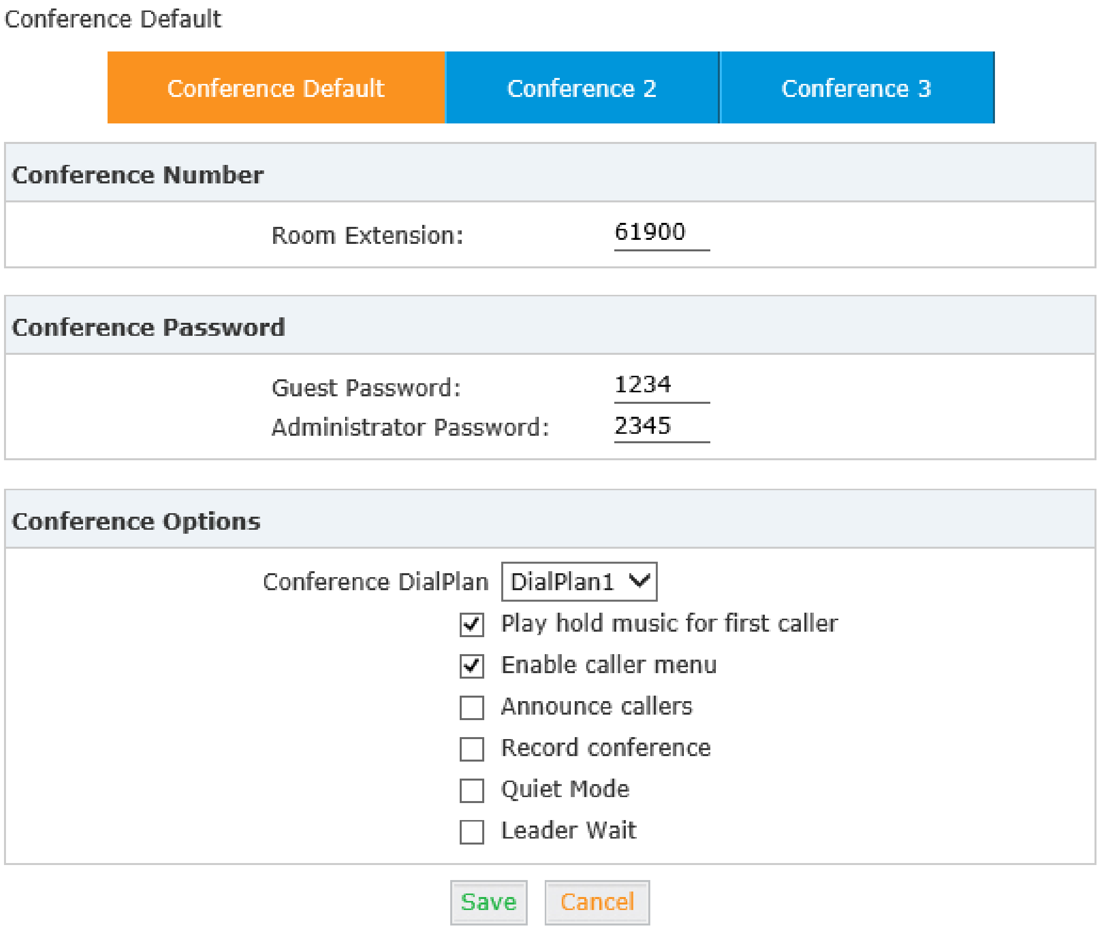
Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.



Specify the email address and click【Send】to send a test email, to verify that the email was successfully sent. If no email was received, please modify the SMTP settings and retry.

8.5.4 Conference

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】:

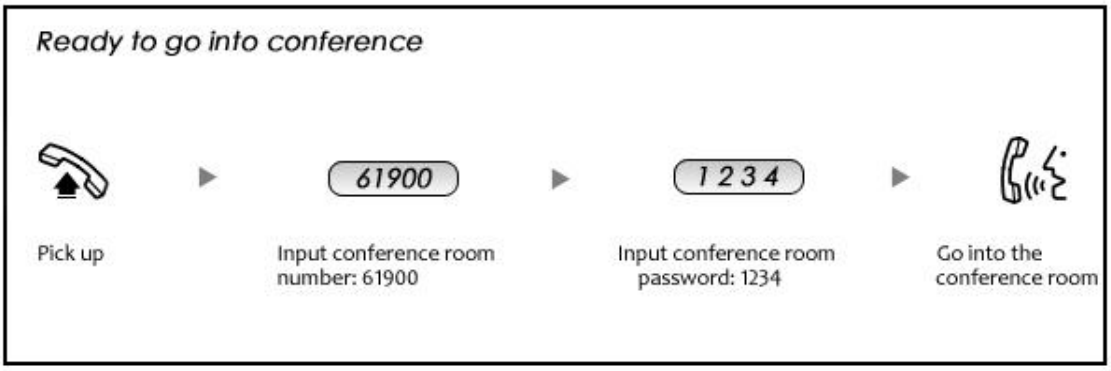


Reference:

Item	Explanation
Conference Number	The number that internal callers use to access the conference room, the default number is“NP+900”; each node will be set different NP for conference number to achieve the free conference calls in the whole CTMS.
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

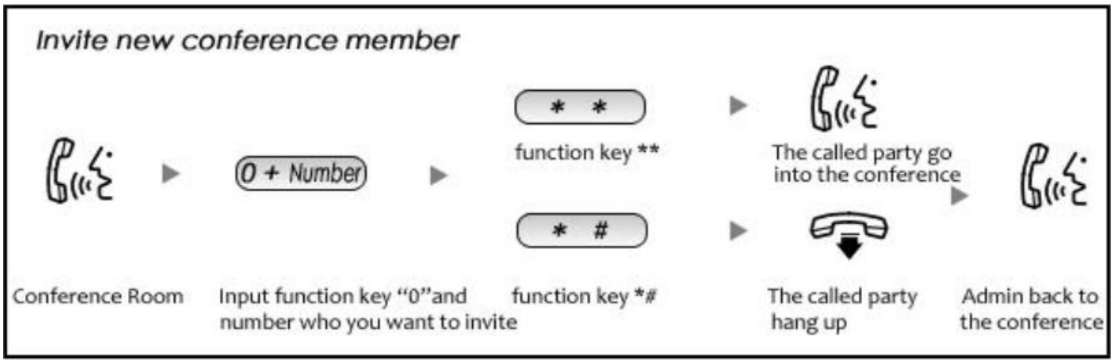
	Bridge menu by pressing "*" on the dialpad.
Announce callers	Check this option to announce to all Bridge participants that new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If check this option, all the participants in the conference can hear only, 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 guests (extension user or external number) into the conference. (Default password for admin is 2345)

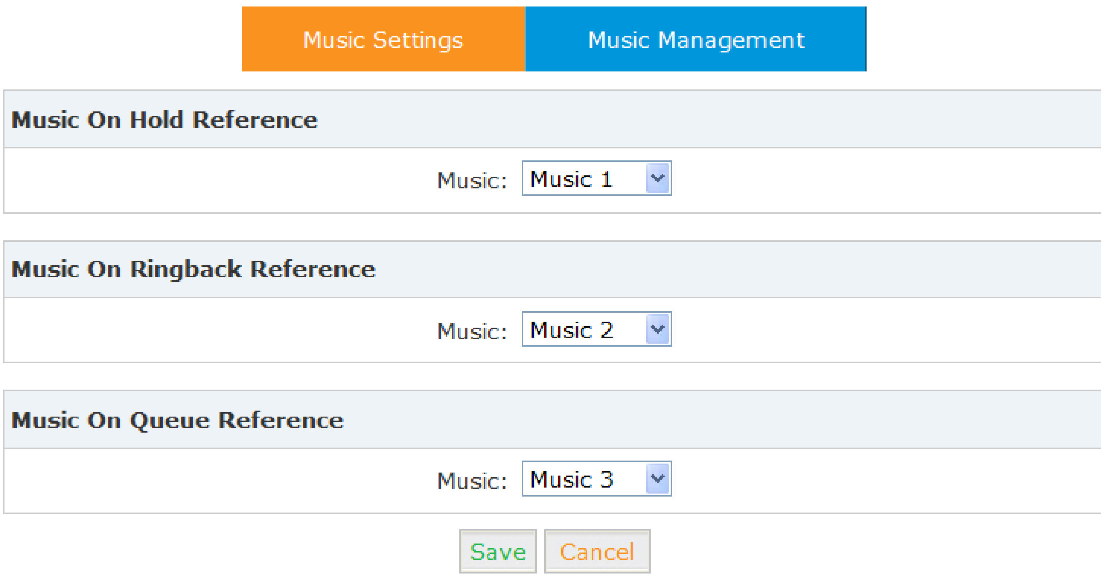
As an administrator, to invite a new guest to the conference, refer to the diagram as below:



8.5.5 Music Settings

Management of Music on Hold, Music on Ringback, Music on Queue.

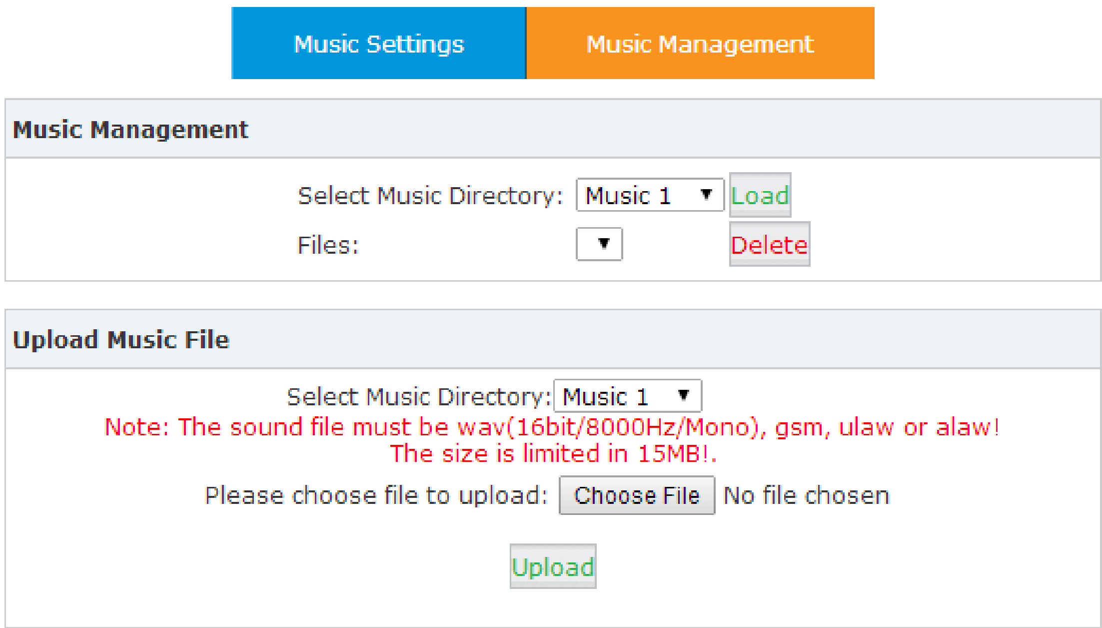
【Music Settings】:



Select the different music file for different Music.

【Music Management】

Music Management



Reference:

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,

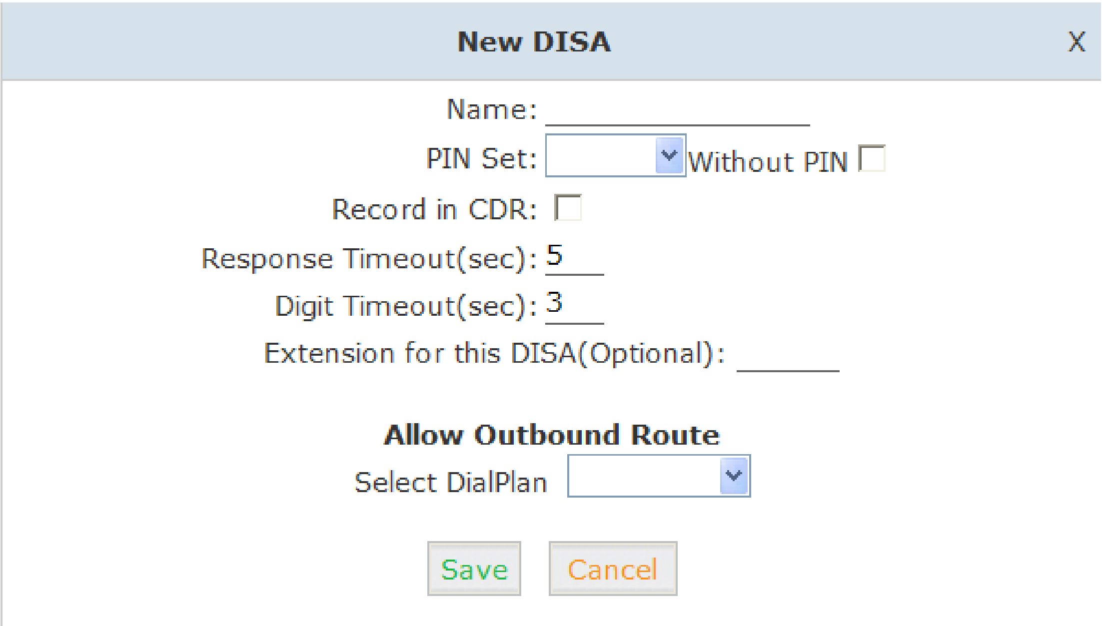
	ulaw or alaw, and less than 15MB.

8.5.6 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, and after entering the user PIN as authentication, the caller will hear a dial tone and can make a call as if they were an extension on the PBX. Especially in the CTMS, DISA is a powerful feature to save long distance call cost for companies.

Please configure as below.

Select【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	Select 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.

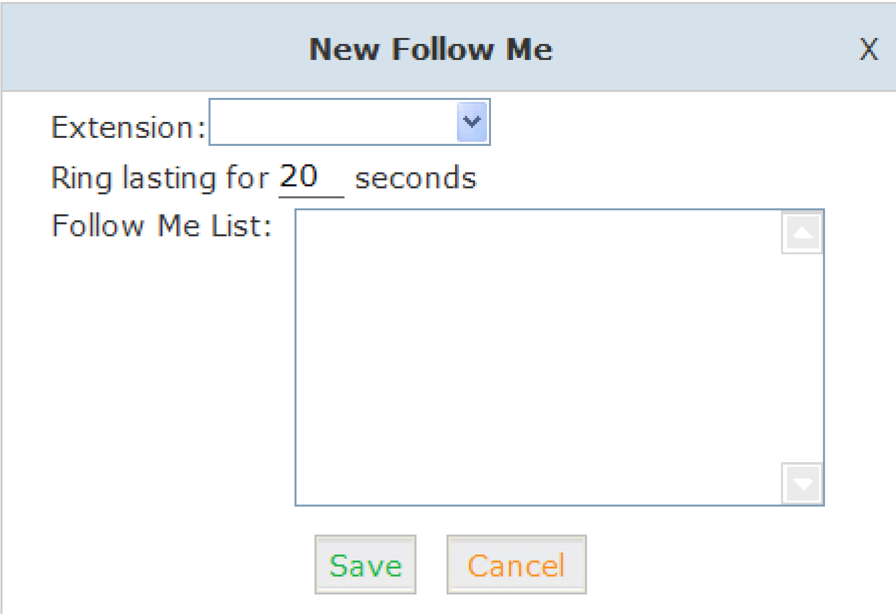
Select DialPlan	Select the DialPlan for this DISA.

8.5.7 Follow Me

This feature allows callers to define a list of numbers (internal and external) where they can be reached and have their calls automatically forwarded when the call is not answered at their primary extension. It is suitable for employees who are out of office.

Please configure as below:

Select【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】.

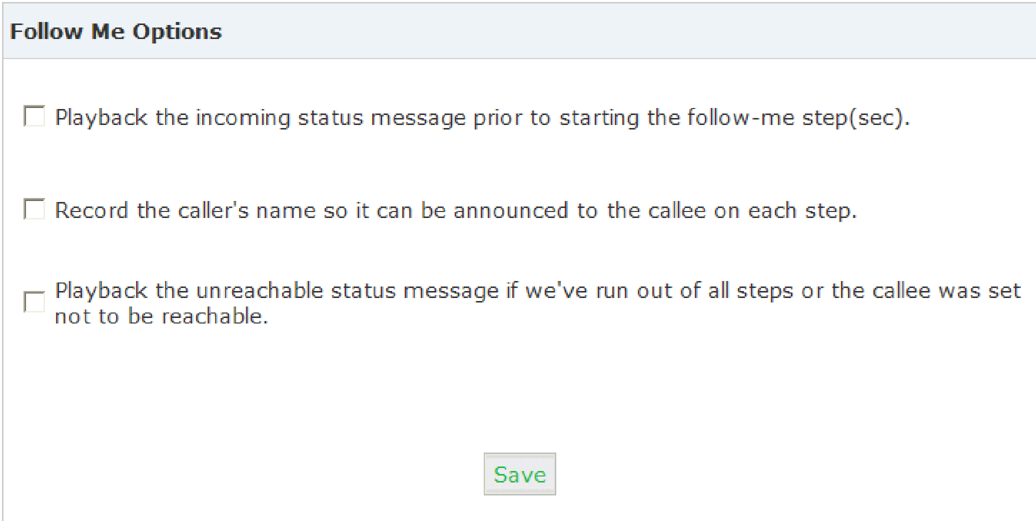
List Format: Extension Number, Ring Duration

E.g.: 9431,30

9433,20

9431 rings, after 30 seconds, the call is going to 9433

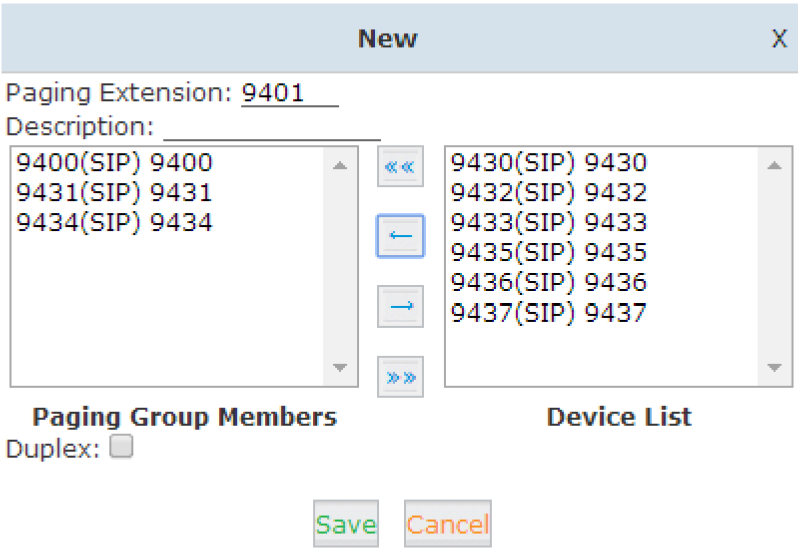
【Follow Me Options】



8.5.8 Paging & Intercom

This feature allows setting up a Paging group so that when the Paging extension is dialed, the listed extensions allows the caller to speak through the external speaker phone of the extension. 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.

Select【Advanced】->【Paging and Intercom】->【New Paging Group】:

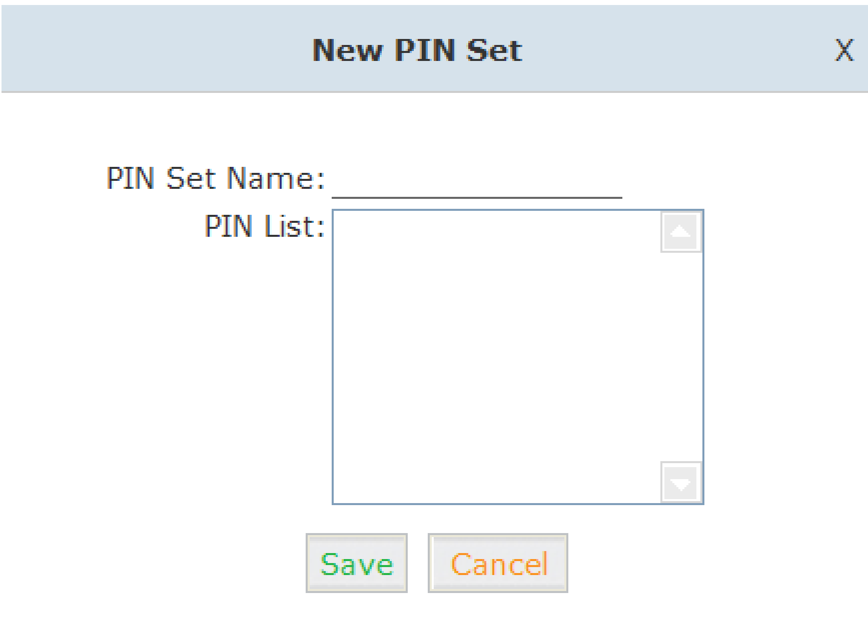


Reference:

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".

8.5.9 PIN Sets

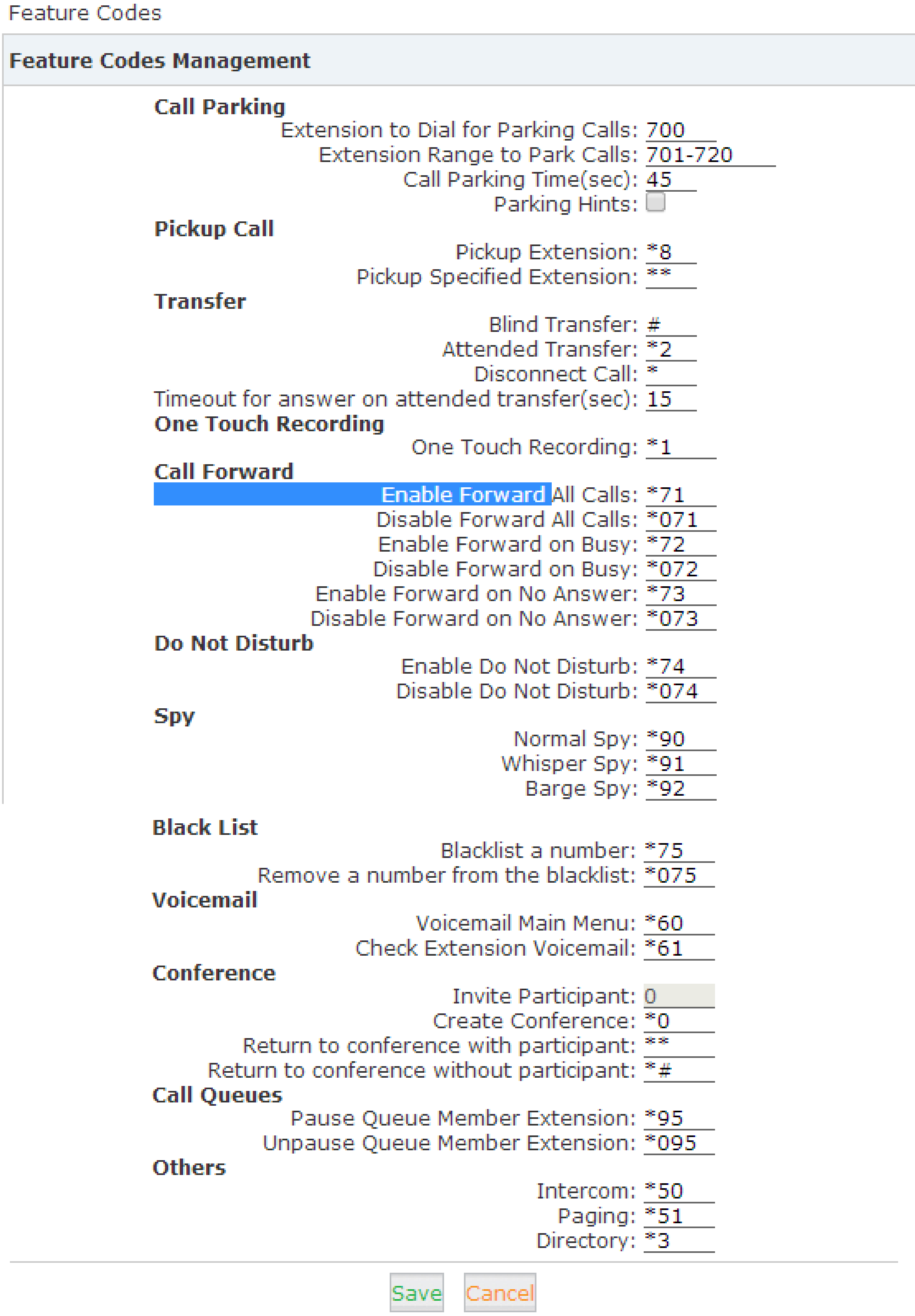
This feature allows an administrator to specify a list of PIN codes in a PIN Set. The PIN can be set on Outbound Routes to ensure only an authorized user is making the call.
Please configure as below.
Select【Advanced】->【PIN Sets】->【New PIN Set】:



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

8.5.10 Feature Codes

Select【Advanced】->【Feature Codes】 and you can view or edit the feature codes listed below.



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. CooVox 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: 3.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. 4.dial the transferee extension or phone number you wish to transfer the call to “B” and hangup the phone. 5.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: 6.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. 7.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

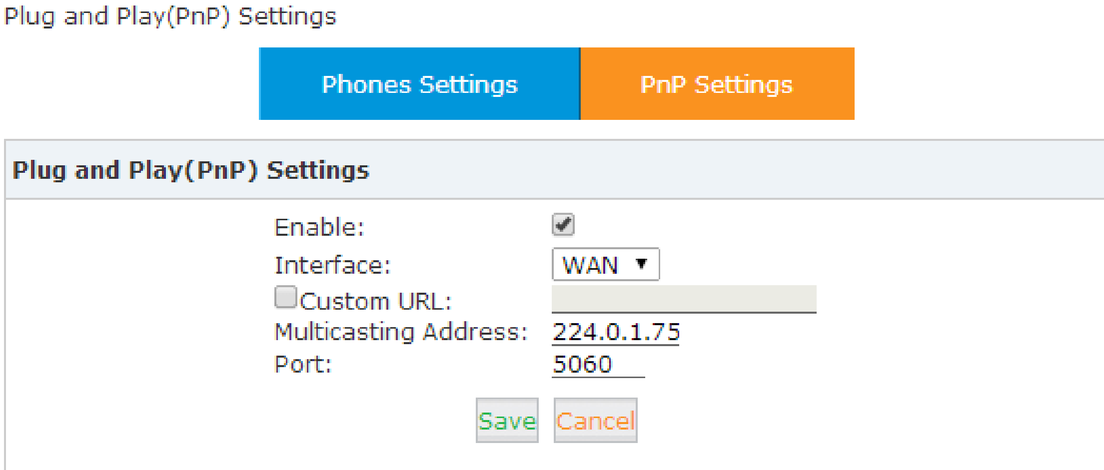
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable “Do Not Disturb”
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklist number.
Voicemail	Configure the function keys for entering voicemail and 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

8.5.11 Phone Provisioning

When deploying large numbers of IP Phones, it is time consuming to have to configure each extension manually. CooVox allows certain IP Phones to be auto-provisioned, then all the phones can be auto-provisioned via CTMC, how it is amazing for enterprise!
To achieve this, please record the MAC, extension number, and username of each phone in the required format (please take reference of the auto provision script file model for details) , then import the formatted 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: DHCP & PnP . Please see details as below:

Method 1: PnP Settings

Select 【Advance】-> 【Phone Provisioning】->【PnP Settings】to enable PnP Settings, the default will be shown as below:

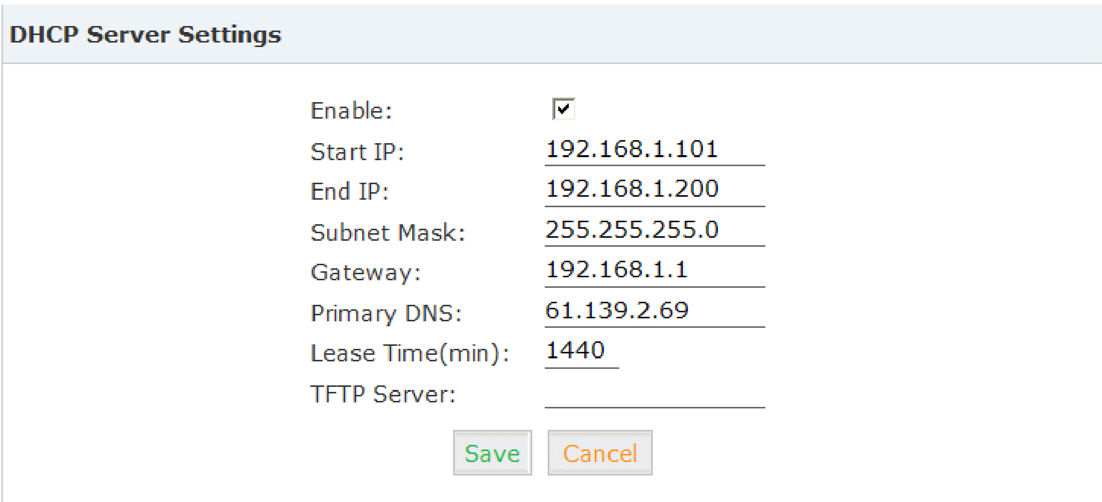


Note: Custom URL is the path for some users to get the phone configuration files specially.

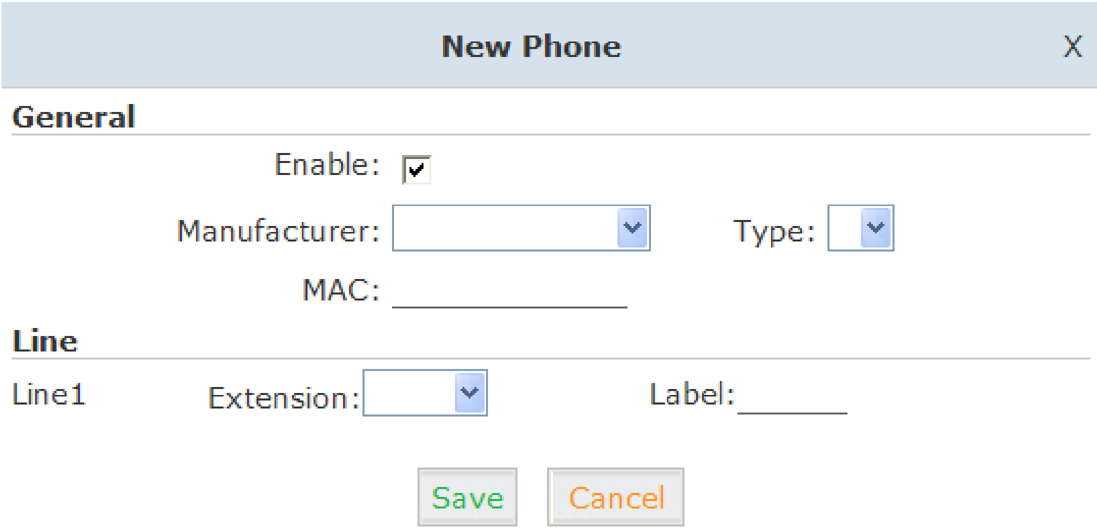
Method 2: Enable DHCP service via CTN

Note: Before using this method, please notice it will cause IP conflict in your network.

Login the CTN, select【System】->【Network Advanced】->【Enable】DHCP Server in the following diagram:



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



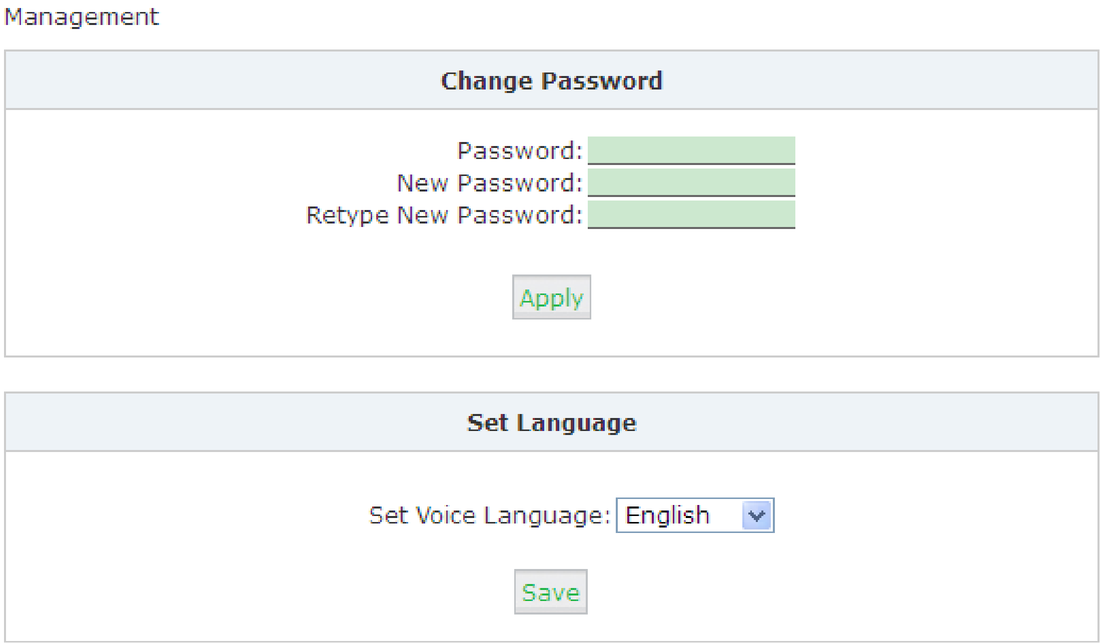
Notice

The device supports IP Phones from ZYCOO/ Yealink/ Grandstream now.

8.6 System

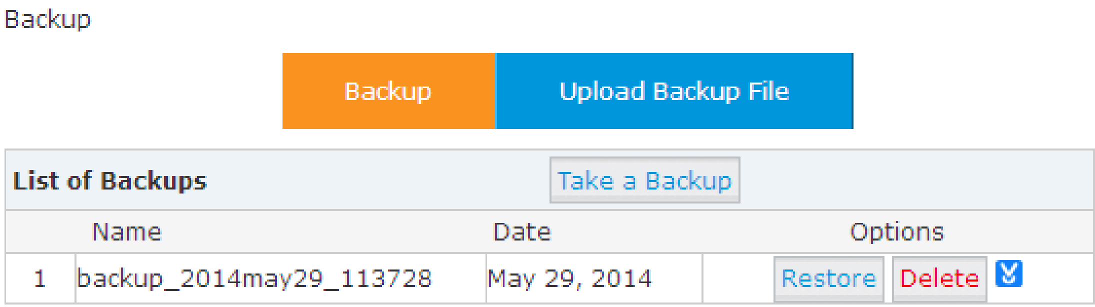
8.6.1 Management

To change the password of a node and select the voice prompt language, please select 【System】->【Management】:



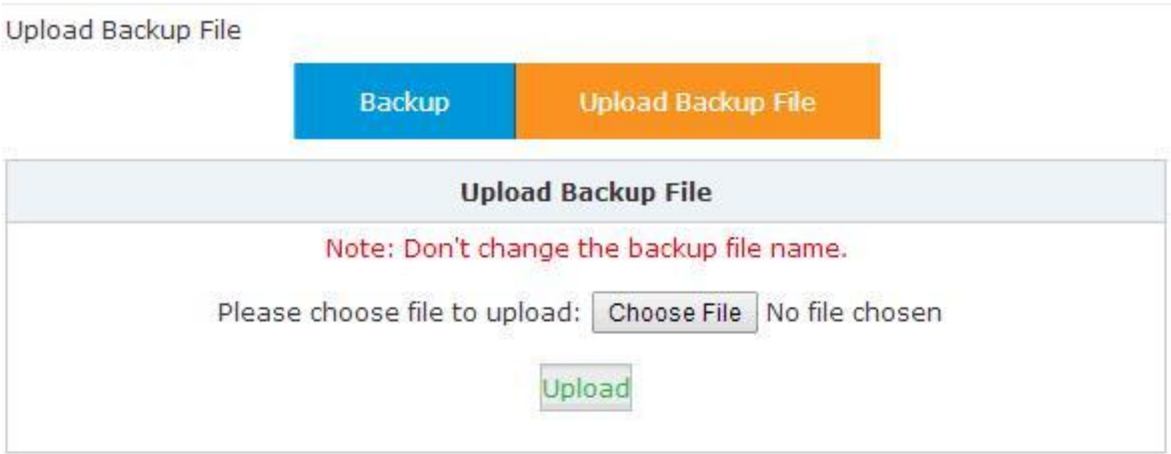
8.6.2 Backup

Backup or upload configuration file of nodes, select【System】->【Backup】->【Backup】:

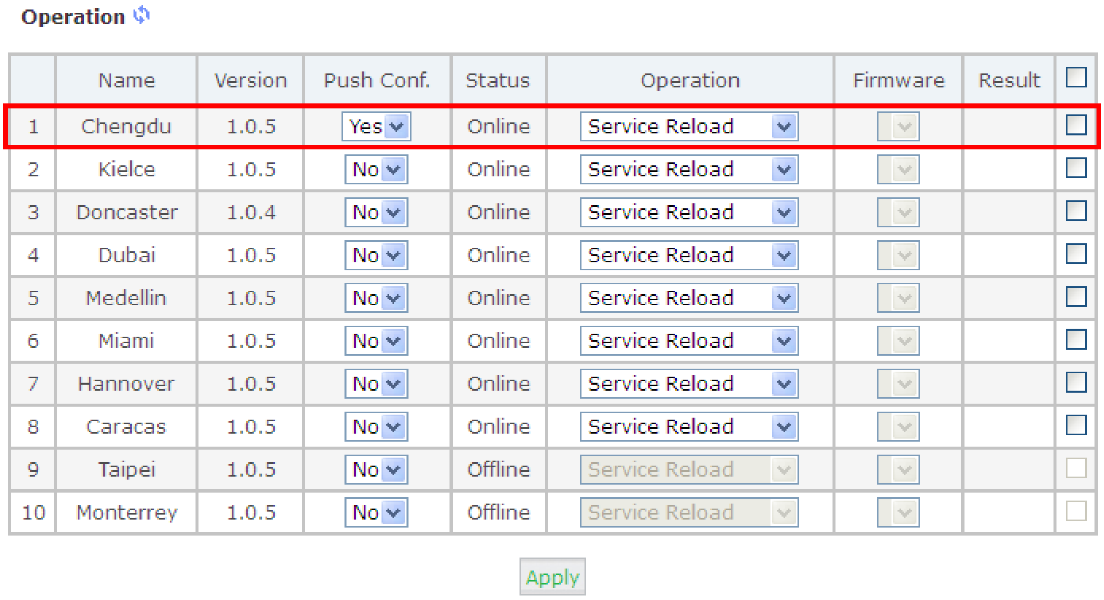


Select "Take a Backup", create the backup file of current system.

To restore the system from a backup file, please select【System】->【Backup】->【Upload Backup File】:



Select the configuration file to restore and select “Upload”. After the operation has completed, go back to【Node Management】->【Operation】, the updated status is now shown.



	Name	Version	Push Conf.	Status	Operation	Firmware	Result	
1	Chengdu	1.0.5	Yes	Online	Service Reload			
2	Kielce	1.0.5	No	Online	Service Reload			
3	Doncaster	1.0.4	No	Online	Service Reload			
4	Dubai	1.0.5	No	Online	Service Reload			
5	Medellin	1.0.5	No	Online	Service Reload			
6	Miami	1.0.5	No	Online	Service Reload			
7	Hannover	1.0.5	No	Online	Service Reload			
8	Caracas	1.0.5	No	Online	Service Reload			
9	Taipei	1.0.5	No	Offline	Service Reload			
10	Monterrey	1.0.5	No	Offline	Service Reload			

Note: a reboot is necessary after being restored

<The End>