S50 IP-PBX

User Manual

XonTel Technology

Table of Contents

1. Introduction

S50 IP-PBX—an IP-PBX for Small Businesses/Home Office

S50 is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations. S50 also offers a hybrid solution (a combination of VoIP applications using legacy telecom equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

Applications

1.1 Features

• Auto-provision	• Follow me
• Blind Transfer	• Interactive Voice Response (IVR)
• BLF Support	• Intercom / Zone Intercom
• Blacklist	• Music On Hold
• Call Detail Records(CDR)	• Music On Transfer
• Call Forward	• Paging / Zone Paging
• Call Parking	• PIN Users
• Call Recording	• Queue
• Call Pickup	• QOS
• Call Routing	• Ring Group
• Call Transfer	• Route by Caller ID
• Call Waiting	• Skype Integration (Skype Connect)
• Caller ID	• Speed dial
• Conference	• Three-way Calling
• Define Office Time	• Do Not Disturb(DND)
• Direct Inward System Access(DISA)	• Voicemail
• Distinctive Ringtone	• VLAN
• Firewalls	

1.2 Hardware Specifications

1.2.1 Exterior Appearance

Front Side

Figure1-1 S50 Front Panel Picture

No.	Identifying
①	Green LED：Indicates the power connections normal.
②	Green LED：Indicates the server system is in working order
③	Green LED：Indicates the system is ready.
④	Green LED：Indicates the internet interface is in use
⑤	Red LED：stands for FXO port Orange LED: Indicates presence of a BRI port. Green LED：stands for FXS port LED Dual - Red blink：FXO port isn't connecting PSTN line. LED Dual - Red and Green blink：FXO port receive an incoming call. LED Dual - Red and Green fast blink：FXO port is in talking. LED Dual - Green and Red blink：FXS port is ringing. LED Dual - Green and Red fast blink：FXS port is in talking.

2) Back Side

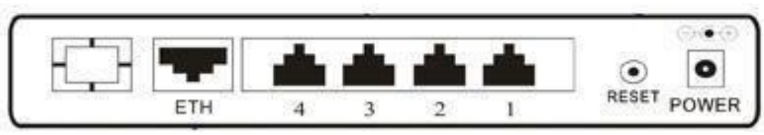


Figure1-2 S50Back Side Picture

2. System set up

2.1 Connection Drawing

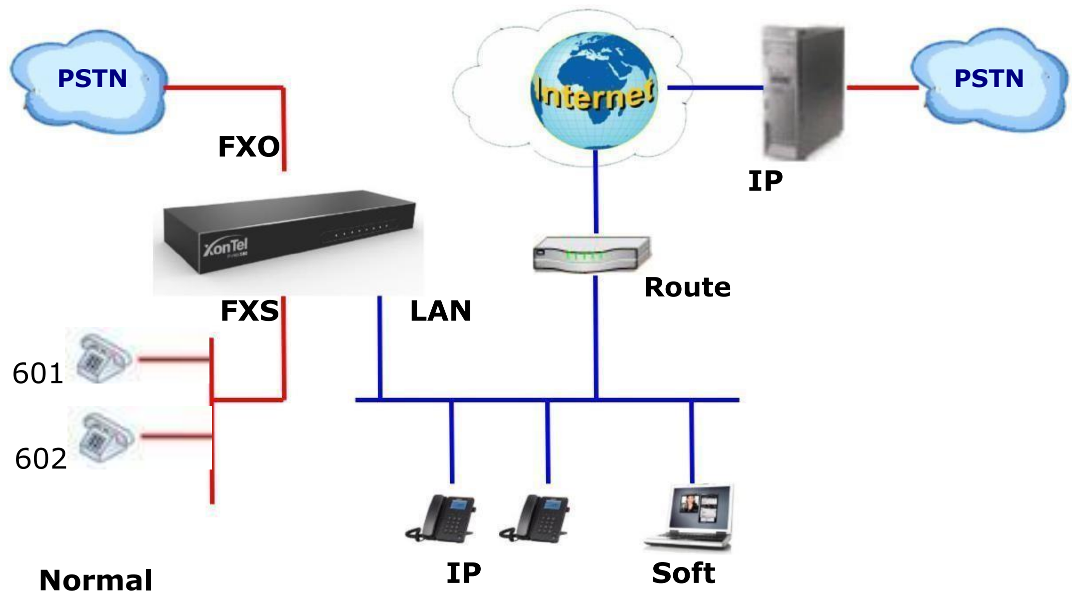


Figure 2-1

2.2 Connecting Ethernet Line

S50 provides two 10/100MEthernet ports with RJ45 interface and LED indicator. Plug Ethernet line into S50's Ethernet port, and then connect the other end of the Ethernet line with a hub, switch, router, LAN or WAN. Once connected, check the status of the LED indicator. A yellow LED indicates the port is in the connection process, and a green LED indicates the port is properly connected.

2.3 Supplying Power

S50 utilizes the high-performance switch power supply, which supplies the required power for the unit.
AC Input: 100~240V
DC Output: 12V,1A

Please follow the steps below to connect the S50 unit to a power outlet:

1. Connect the small end of the power cable to the power input port on the S50 back panel, and plug the other end of the cable into a 100VAC power outlet.
2. Check the Power LED on the front panel. A solid green LED indicates that power is being supplied correctly.

3. Managing S50

3.1 Administrator Login

From your web browser, input the IP address of the S50 server.
If this is the first time you are configuring S50, please use the default settings below:
IP Address: http://192.168.5.150

Note: S50 supports multiple administrators in hierarchical mode (Administrator, User, and CDR)

·**Administrator**
have all authority .
Username: admin , Password:xontel

·**User**
have basic authority; without the advanced authority to create VoIP trunks, reset, update, backup and restore S50.
Username: user, Password:xontel

·**CDR**
only have the authority to check the call recordings. Username: cdr, Password:xontel

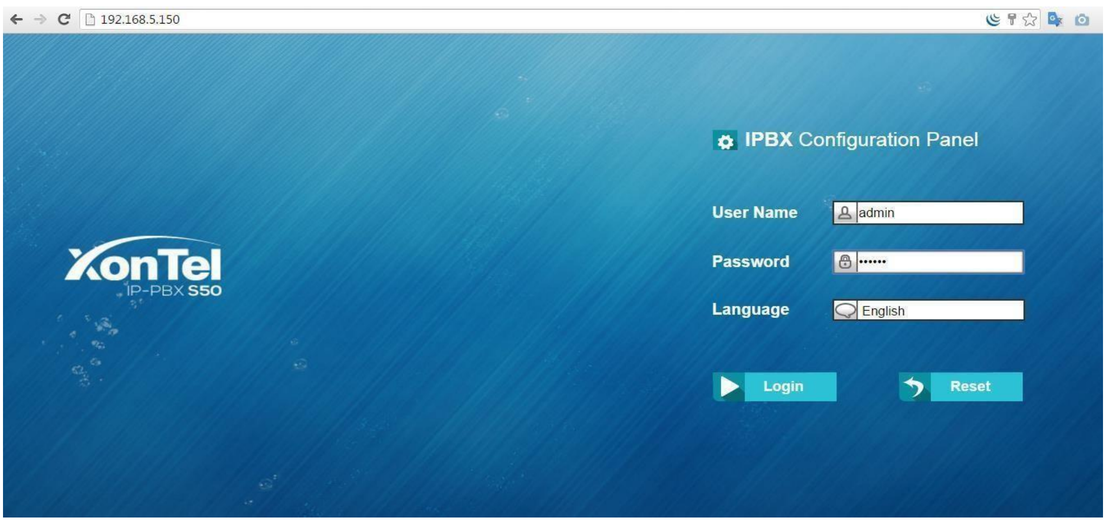


Figure 3-1

3.2 Status Monitor

3.2.1 Line Status

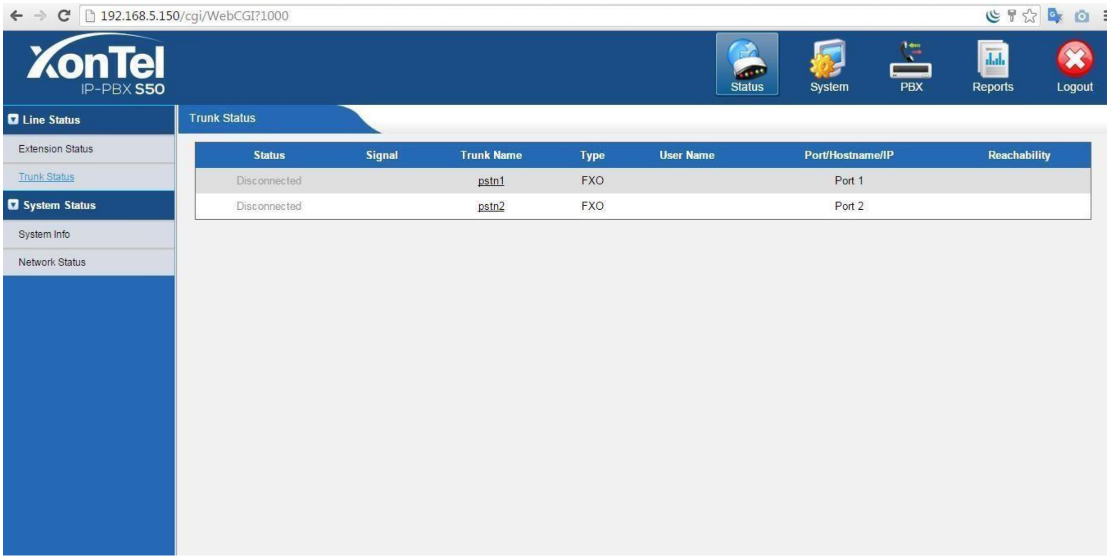


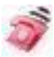
Figure 3.2.1

S50 Status Description:

Extensions:

1)  :Extension is unregistered

2)  :Extension is idle3）

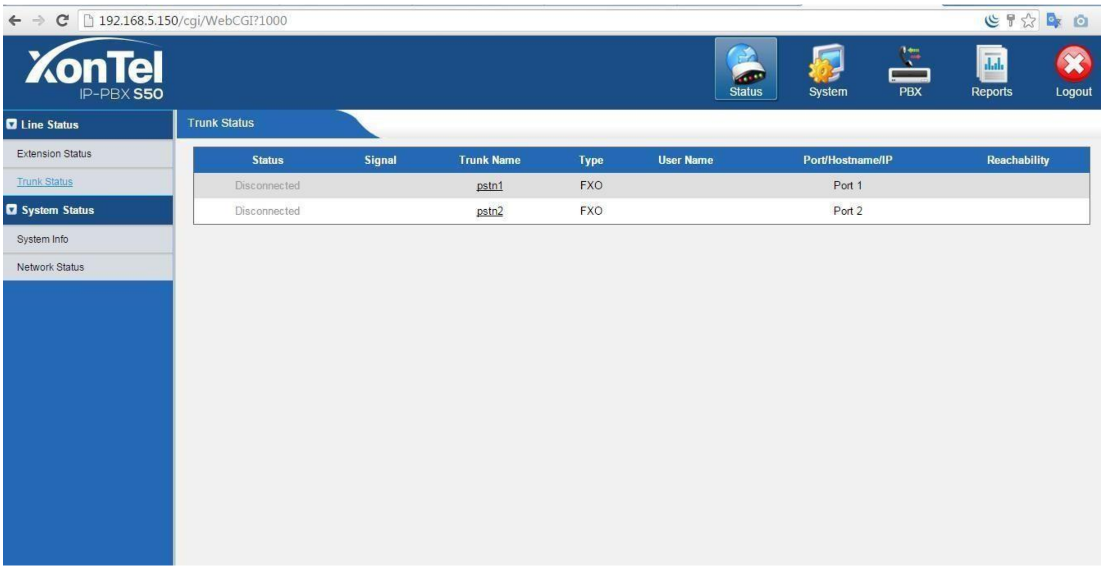
 :Extension is ringing 4）

 :Extension is busy in a call

5）  : Extension is on hold

Trunks:

VOIP Trunk:



Status

Unregistered: Trunk registration failed.

Registered: Succeed registration, trunk is ready for use. Request Send: Registering.

Waiting: Waiting for authentication.

FXO Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

Disconnected: The port hasn’t connected the PSTN line.

More detail message, please refer to the LED identifying of front panel.

BRI Trunk:

Status

Ok: The ports connect correctly.

Error: The port is error.

Disconnected: The port hasn’t connected the PSTN line.

Service Provider:

Status

Ok: Succeed registration, trunk is ready for use.

Failed: Trunk registration failed.

3.3 Basic

3.3.1 Extension

Extension has two types: Analog extensions (FXS) and VOIP extensions (SIP extension).

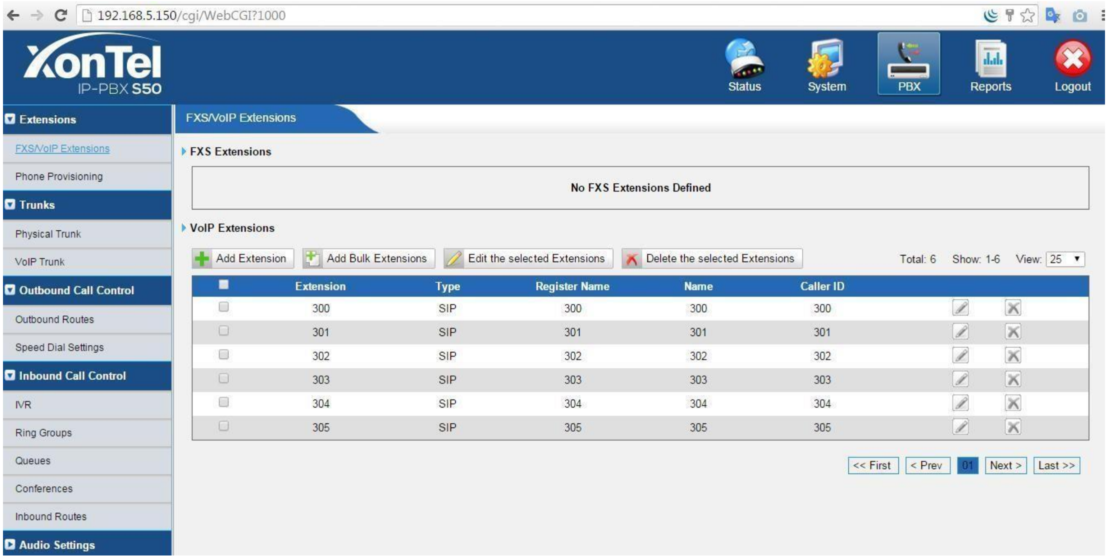


Figure 3.3.1

3.3.1.1 Analog Extensions (FXS)

Edit Analog Extensions

On the administration page of FXS extensions, click 'Edit' on the extension that you want to edit, and modify the following information on the popup window.

1) General

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Port

The extension correspond port.

·Name

A character-based name for this user, i.e. 'Mostafa , Ahmed ' .

·Caller ID

The Caller ID (CID) string will be used when this user calls another internal user.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #
Voicemail Password for this extension, i.e. '1234'.

3) Mail Setting
·Enable Send Voicemail
Once enabled, the voicemail will be sent to the below email address as an attachment.

·Send Voicemail to Email Address
This option defines whether or not voicemails/Fax is sent to the Email address as an attachment.
Note: Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

4) Flash
·Hook Flash Detection
Sets the amount of time, in milliseconds that must pass since the last hook-flash event received by S50 before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then S50 will ignore the event. The default value of Flash is 1000 ms, and it can be configured in 1ms increments.

5) Group
·Pickup Group
If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default *4).
Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.
S50 supports max 20 pickup groups.

6) Follow me (Call Forwarding)
This function sets inbound call forwarding on an extension. An administrator can configure Follow Me for this extension.

7) Other Options
·Call Waiting
Check this option if the extension should have Call Waiting capability.If this option is checked, the 'When busy' follow me options will not be available.

· DND
Don't Disturb.

·User Web Interface

Check this option to allow the user to login to the S50 User Web interface, which can be used to access voicemail and extension recordings. Users may login to the S50 User Web interface by using their extension number and voicemail PIN
as the login and password respectively.

·Ring Out

Check this option if you want to custom the ring time. Tone will stop over the time defined.

8) Volume Settings

Rxgain: The Volume sent to FXS extension.
Txgain: The Volume sent out by the FXS extension.

9) Mobility Extension

S50 allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call S50 with this mobility phone, you will hear a dial tone. S50 will recognize your call as a call from an extension. You can dial the number of other extensions (Your caller ID will be the number of your extension) and use all outbound route that your extension can use of S50.

·Mobility Extension Number

Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number field.E.g. if you want to set "15960XXXXXX" as mobile extension, and the dial pattern of the outbound route is "9"; you should set "915960XXXXXX" here.

Note: If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled.

10) Spy Settings

S50 allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

·spy modes

There are 4 spy modes available for choice:
General spy: you have the permission to use the following 3 modes.

Normal spy: you can only hear the call, but can't talk
Whisper spy: you can hear the call, and can talk with the monitored extension
Barge spy: you can hear the call and talk with them both

Note: for example, if 500 want to monitor extension 501, we need to enable the 'allow being spied ' for 501, and choose the spy mode for extension 500.

Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call.

If 500 choose 'normal spy', it should dial'*90501' to start monitoring; if 500 choose 'whisper spy', it should dial '*91501' to start monitoring; if 500 choose 'barge spy', it should dial '*92501' to start monitor;
if 500 choose 'general spy', it can dial '*90501','*91501' or '*92501' to start monitor.

11) Call Duration Settings

Setup the max cull duration for every call of this extension, but it's only valid for outbound calls. And if enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

Note: This setting will not be valid for internal calls.

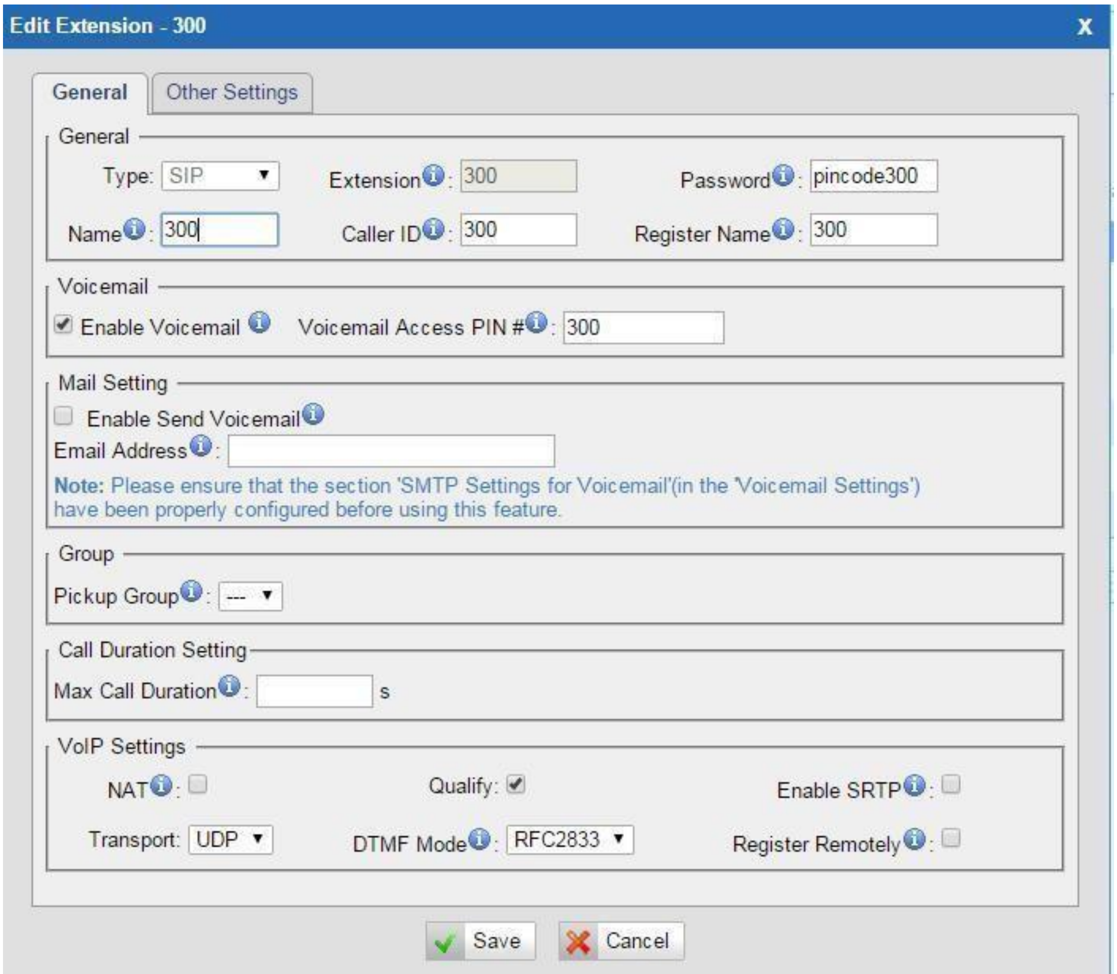


Figure 3.3.1.1

3.3.1.2 VOIP Extension

A VOIP extension is an SIP Account that allows an IP Phone or an IP Soft-Phone client to register onS50.

1. Add VOIP Extension

Go to Extensions → VOIP Extensions → Create New Extension

1) General

·Type

Extension type: SIP

SIP – The extension sends and receives calls using the VoIP protocol SIP.

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Password

The password for this extension, Ex: ‘12t3f6’

·Name

A character-based name for this user, EX: ‘Mostafa , Ahmed’

·Caller ID

The Caller ID will be used when this user calls another internal extension.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #

The voicemail Password for this extension, i.e. ‘1234’.

3) Mail Setting

This option defines whether or not voicemails or faxes are sent to an Email Address as attachment.

·Enable Send Voicemail

Once enabled, the voicemail will be sent to email as an attachment.

·Email Address

Email address used to receive the voicemail or Fax.

Note: Please ensure that the section 'SMTP Settings For Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

4) Group

·Pickup Group

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default is *4).

Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

5) Follow me (Call Forwarding)

Call forwarding for an extension can be configured here. The administrator can configure Follow Me option for this extension. If you want to transfer the call to an outbound number, please follow the dial pattern of outbound route filled in the outbound number.

For example: transferring to your mobile phone number 123456789, the dial pattern of outbound route is '9.', you should fill in 9123456789 here.

6）Other Options

.Call Waiting

Check this option if the extension should have Call Waiting capability.If this option is checked, the 'When busy' follow me options will not be available. The call waiting function of IP phone has higher priority than S50's call waiting function.

.DND

Don't Disturb. When DND is enabled for an extension, the extension will be not available.

.User Web Interface

Check this option to allow the user to login to the S50 User Web interface, which can be used to check voicemail and extension recordings. Users may login to S50 User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

.Ring Out

Check this option if you want to customize the ring time. Ring tone will stop over the time defined.

7) Spy Settings

S50 allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

·spy modes

There are 4 spy modes available for choice:

General spy: you have the permission to use the following 3 modes. Normal spy: you can only hear the call, but can't talk
Whisper spy: you can hear the call, and can talk with the monitored extension
Barge spy: you can hear the call and talk with them both

Note: for example, if 500 want to monitor extension 501, we need to enable the 'allow being spied ' for 501, and choose the spy mode for extension 500.
Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call.
If 500 choose 'normal spy', it should dial'*90501' to start monitoring; If 500 choose 'whisper spy', it should dial '*91501' to start monitoring;
If 500 choose 'barge spy', it should dial '*92501' to start monitor;
If 500 choose 'general spy', it can dial '*90501','*91501' or '*92501' to start monitor.

8) VoIP Settings

·NAT

This setting should be used when the system is using a public IP address to communicate with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.

·Qualify

Send check alive packets to IP phones

·SIP Transport

This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.

·DTMF Mode – RFC2833, Info, Inband, Auto.

·Remote Register

Allow to register remote extensions.
This option is used to enhance the system security, it's disabled by default.

9) IP Restriction

·Enable IP Restriction

Check this option to enhance the VoIP security for S50.If this option is enabled, only the permitted IP/Subnet mask will be able to register this extension number. In this way, the VoIP security will be enhanced.

·Permitted 'IP address/Subnet mask'

The input format should be 'IP address'+'/'+'Subnet mask'.
e.g."192.168.5.100/255.255.255.255" means only the device whose IP address is 192.168.5.100 is allowed to register this extension number.
e.g."192.168.5.0/255.255.255.0" means only the device whose IP address is 192.168.5.XXX is allowed to register this extension number.

10) Mobility Extension

S50 allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call S50 with this mobility phone, you will hear a dial tone. S50 will recognize your call as a call from an extension. You can dial other extension numbers (Your caller ID will be the number of your extension) and dial out using all outbound route that your extension can use on S50.

·Mobility Extension Number

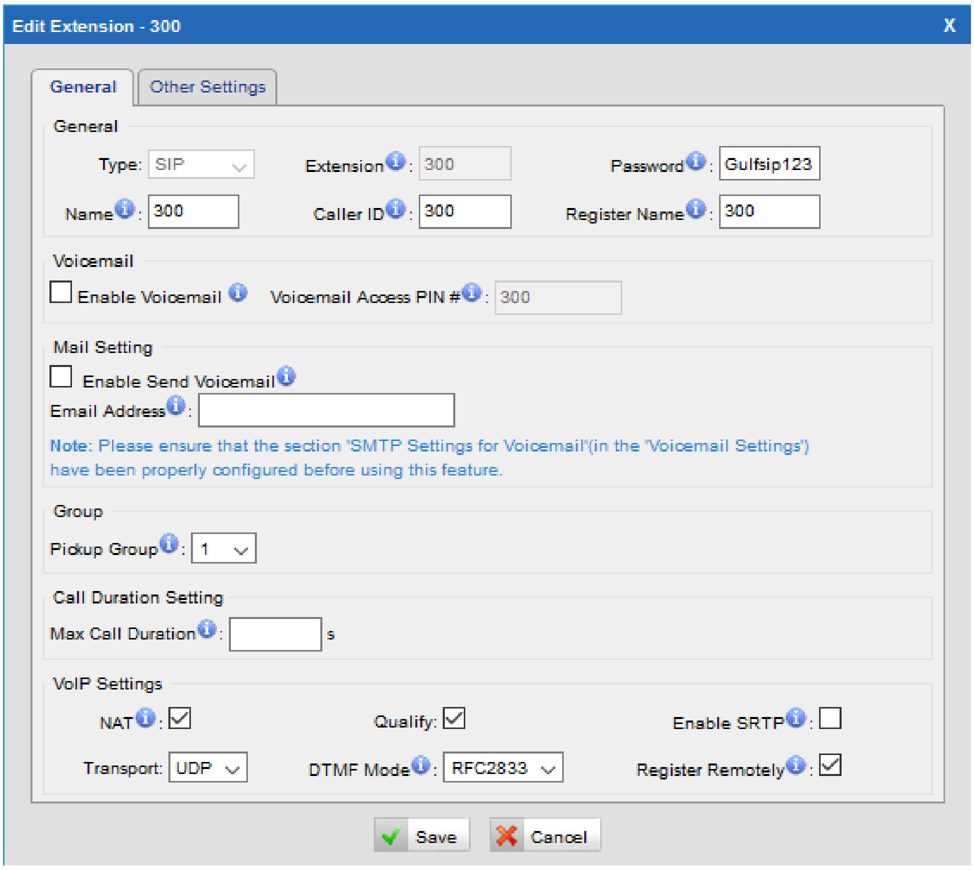
Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number field.E.g. if you want to set "15960XXXXXX" as mobile extension, and the dial pattern of the outbound route is "9"; you should set "915960XXXXXX" here.

Note: If callback is enabled on the inbound route, the mobility extension function of this inbound route will be disabled.

11) Call Duration Settings

Setup the max cull duration for every call of this extension, but it's only valid for outbound calls. And if enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

Note: This setting will not be valid for internal calls.


Edit Extension - 300
X
General
Other Settings
General
Type: SIP
Extension: 300
Password: Gulfsip123
Name: 300
Caller ID: 300
Register Name: 300
Voicemail
Enable Voicemail
Voicemail Access PIN #: 300
Mail Setting
Enable Send Voicemail
Email Address:
Note: Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.
Group
Pickup Group: 1
Call Duration Setting
Max Call Duration: s
VoIP Settings
NAT:
Qualify:
Enable SRTP:
Transport: UDP
DTMF Mode: RFC2833
Register Remotely:
Save
Cancel

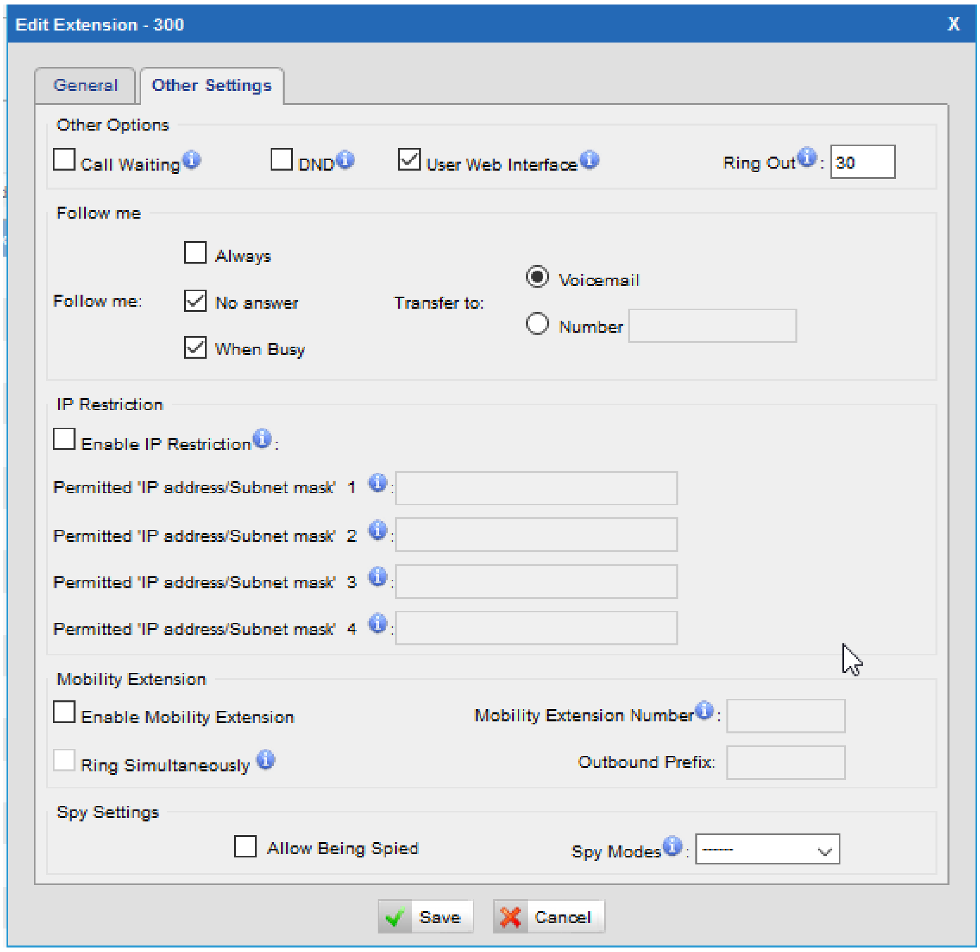

Edit Extension - 300
X
General
Other Settings
Other Options
Call Waiting
DND
User Web Interface
Ring Out: 30
Follow me
Always
Follow me:
No answer
When Busy
Transfer to:
Voicemail
Number
IP Restriction
Enable IP Restriction:
Permitted 'IP address/Subnet mask' 1:
Permitted 'IP address/Subnet mask' 2:
Permitted 'IP address/Subnet mask' 3:
Permitted 'IP address/Subnet mask' 4:
Mobility Extension
Enable Mobility Extension
Mobility Extension Number:
Ring Simultaneously
Outbound Prefix:
Spy Settings
Allow Being Spied
Spy Modes: ------
Save
Cancel

Figure 3.3.1.2

2. Add Multiple Extensions

Go to Extensions → VOIP Extensions → Add Bulk Extensions.
1) Choose the number of extensions that you would like to create.
2) Choose the type of extension that you would like to create.
3) Fill in the starting extension number.

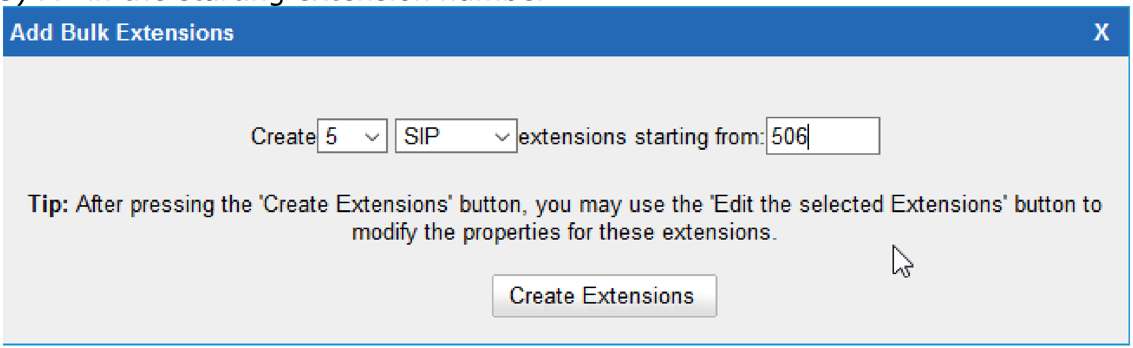


Figure 3.3.1.3

3. Edit VOIP Extension

Click 'Edit' on VOIP Extension administration page or click 'Modify Selected Extensions' to edit extension.

3.3.2 Trunk

There are four types of trunks: Analog trunks (FXO),BRI Trunk, VoIP Trunk and Service Provider trunks.

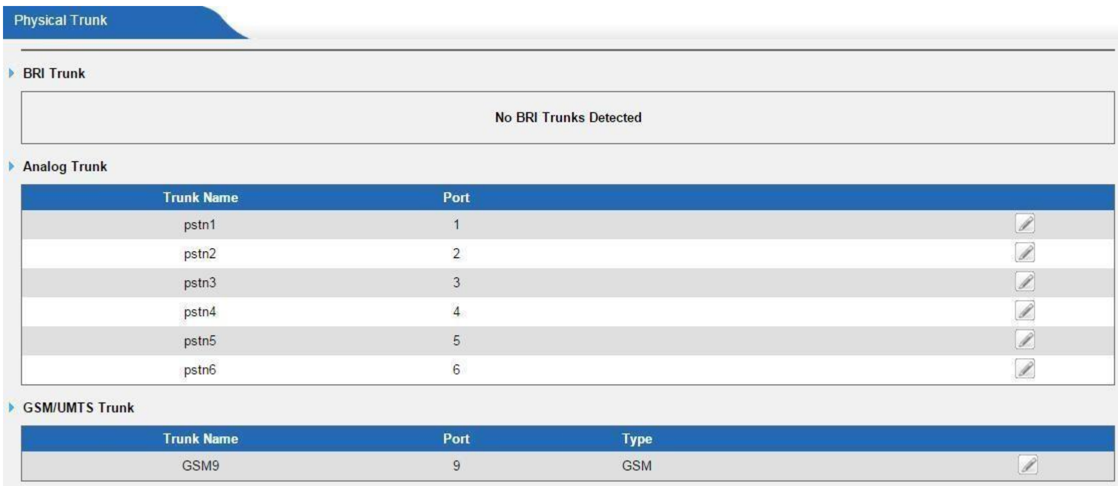


Figure 3.3.2

3.3.2.1 Analog Trunk (FXO)

1. Edit Analog Trunk (FXO)

On the Trunk administration page, click 'Edit' on the selected trunk and modify its properties in the popup window:

1） General

·Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'pstn5'

·Volume Setting

Used to modify the volume level of this trunk. Normally, this setting does not need to be changed.

2） Busy Detection

·Busy Detection

Busy Detection is used to detect far end hang-up or for detecting a busy signal. Select "Yes" to turn this feature on.

·Busy Count

If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before disconnecting the call. The default is 4, but better results can be achieved if set to 6 or even 8. Remember, the higher the number, the more time will be required to release a channel. A higher setting lowers the probability that you will encounter random hang-ups.

·Busy Interval

The busy detection interval

·Busy Pattern

If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many Countries, it is 500msec on, 500msec off. Without Busy Pattern specified, S50 will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then S50 will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnect.

·Frequency Detection

Used for Frequency Detection (Enable detecting the busy signal frequency or not).

·Busy Frequency

If the Frequency Detection is enabled, you must specify the local frequency.

·Polarity Detection

Configure if the call needs to be hung up when a polarity signal arrived

3） Advanced Options

·Caller ID Start

This option allows you to define the start of a Caller ID signal:
Ring: Start when a ring is received (Caller ID Signaling: Bell_USA, DTMF). Polarity: Start when a polarity reversal is started (Caller ID Signaling: V23_UK,V23_JP,DTMF).
Before Ring: Start before a ring is received (Caller ID Signaling: DTMF).

·Caller ID Signaling

This option defines the type of Caller ID signaling to use. It can be set to one of the following:
Bell: bell202 as used in the United
States v23_UK: suitable in the UK
v23_Japan: suitable in Japan
v23-Japan pure: suitable in Japan
DTMF: suitable in Denmark, Sweden, and Holland

.Caller ID Detection

For fxo trunks, this option forces S50 to clarify Caller ID incoming calls.

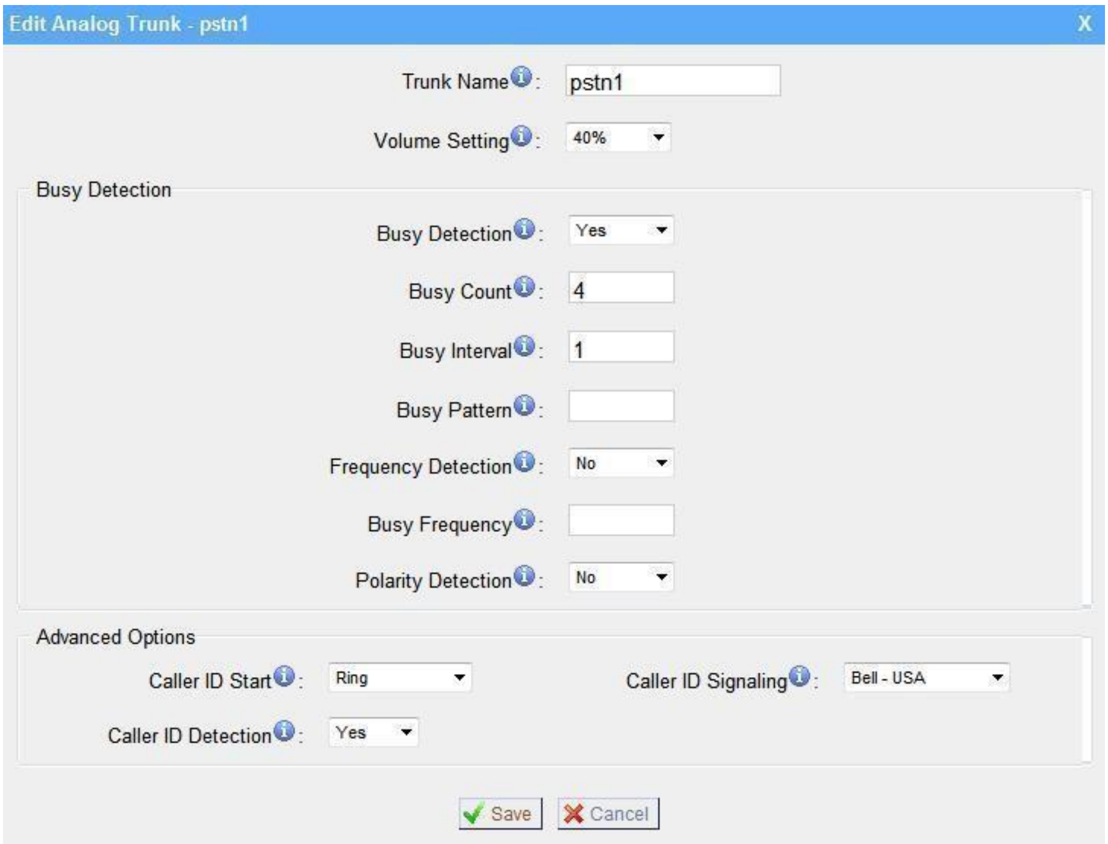


Figure 3.3.2.1

3.3.2.2 BRI Trunk

1. Edit BRI Trunk

1）General

·Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'BriTrunk1'

·Signaling

Signaling method.
BRI-CPE: ISDN BRI in TE mode and Point to Point. BRI-CPE-PTMP: ISDN BRI in TE mode and Point to multi Point. BRI-NET: ISDN BRI in NET mode and Point to Point. BRI-NET-PTMP: ISDN BRI in NET mode and Point to multi Point.

·Switch Type

National: National ISDN type2 (common in the US)
ni1: National ISDN type 1
dms100: Nortel DMS100
4ess: AT&T 4ESS
5ess: Lucent 5ESS
euroisdn: EuroISDN
qsig: D-channel signaling protocol at Q reference point for PBX networking .

·PRI Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Reset interval

Sets the time in seconds between restart of unused channels. Some PBXs don't like channel restarts. So set the interval to a very long interval e.g. 100000000 or 'never' to disable *entirely*. If you are in Israel, the following is important: As Bezeq in Israel doesn't like the B-Channel resets happening on the lines, it is best to set the reset interval to 'never' when installing a box in Israel. Our past experience also shows that this parameter may also cause issues on local switches in the UK and China.

·PRI Local Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Over Lap Dial
Whether S50 can dial this switch using overlap digits. If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then S50 will wait after the last digit it receives.

·PRI Indication
Tells how Device should indicate Busy() and Congestion() to the switch/user. Accepted values are:
inband: Device plays indication tones without answering; not available on all PRI/BRI subscription lines .
outofband: Device disconnects with busy/congestion information code so the switch will play the indication tones to the caller. Busy() will now do same as setting PRI_CAUSE=17 and Hangup().

·Enable Facility
To enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility).

·NSF
Used with AT&T PRIs. If outbound calls are being rejected due to "Mandatory information element missing" and the missing IE is 0x20, then you need this setting.

·Echo Cancellation
Echo cancel Obviously this disables or enables echo cancellation, it is recommended to not turn this off.

·Hide CallerID
If you want others to see your CID, please disable this option.

·Codec
You can choose alaw or ulaw codes.

2) CallerID Prefix
·International Prefix
When there are international calls coming in via this BRI trunk, the International Prefix you have set here will be added before the CID. So you can know this is an international call before you answer it.

·National Prefix
When there are national calls coming in via this BRI trunk, the National Prefix

you have set here will be added before the CID. So you can know this is a national call before you answer it.

·Local Prefix
When there are Local calls coming in via this BRI trunk, the Local Prefix you have set here will be added before the CID. So you can know this is a local call before you answer it.

·Private Prefix
When there are Private calls coming in via this BRI trunk, the Private Prefix you have set here will be added before the CID. So you can know this is a Private call before you answer it.

·Unknown Prefix
When there are calls with unknown number coming via this BRI trunk, the Unknown Prefix you set here will be shown as the caller ID.

3) DOD Setting
·Global DOD
Global direct outward dialing number.

·DOD
Direct Outward Dialing Number.

·Associated Extension
The extension make call out via BRI Trunk will display the associated DOD.

Figure 3.3.2.2

3.3.2.3 VOIP Trunk

. Add SIP Trunk

Input correct SIP information (provide by VOIP provider). Inaccurate information will prevent the trunk from registering.

1) General setting

·Type

SIP– Identifies whether the trunk sends and receives calls using the VoIP protocol SIP

·Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'Xontel'.

·Hostname/IP

Service provider’s hostname or IP address.5060 is the standard port number used by SIP protocol. Don’t change this part if it is not required.

.Domain

VoIP provider’s server domain name.

·Username

Username of SIP account. Used for SIP trunk registration.

.Authorization name

Used for SIP authentication. Leave this blank if not required.

·Password

Password of SIP account.

.From User

All outgoing calls from this SIP Trunk will use the From User (In this case the account name for SIP Registration) in From Header of the SIP Invite.

.Online number

Define the online number that expected by ‛Skype Connect ‛and some other SIP service providers. Leave this field blank if it's no required.

·Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

·Caller ID

Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the ‛extension’ screen will override the caller ID set in the ‛VOIP trunk’ screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

·Outbound Proxy Server

A proxy that receives requests from a client, even though it may not be the server resolved by the Request-URI.

·Codecs

Define the codec for this sip trunk and its priority
Note: codec can only display when edit it（it is edited after creating the trunk).

·Transport
This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

·Enable SRTP
Define if SRTP is enabled for this trunk

·Qualify
Send check alive packets to the sip provider.

·DTMF mode
Set default mode for sending DTMF of this trunk. Default setting: rfc2833

2)DOD Setting
·DOD
Direct Outward Dialing Number.

·**Associated Extension**
The extension make call out via SIP Trunk will display the associated DOD.

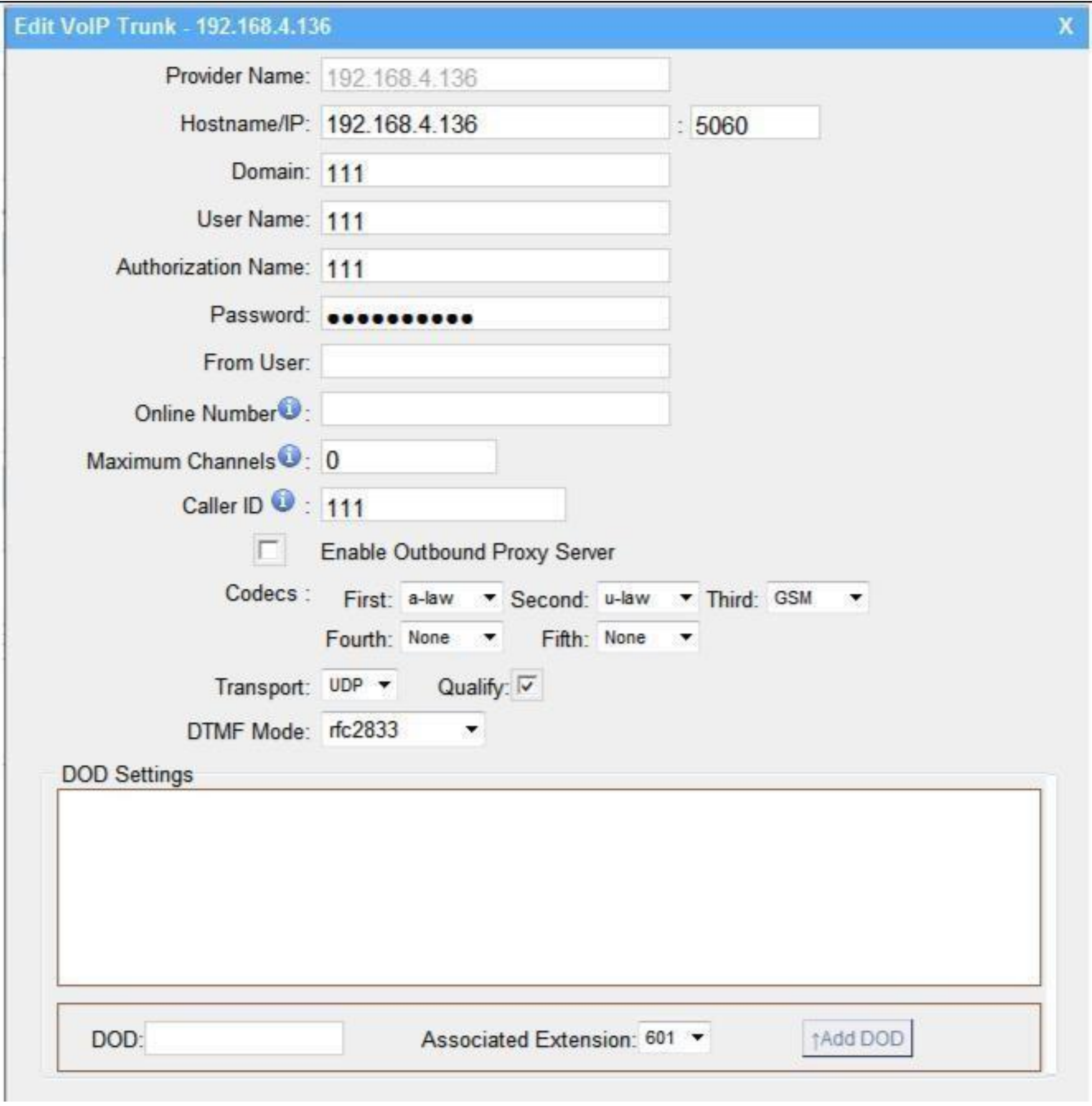


Figure 3.3.2.3

3.3.2.4 Service Provider

1. Add Service Provider

To Create the Service provider definition you need to complete the following VoIP fields.

1) General

·Type

SIP

SIP – Identifies whether the trunk sends and receives calls by using the VoIP protocol SIP.

·Provider Name

A unique label would help to you identify this trunk. Ex: 'Provider2'.

·Hostname/IP

Service provider's hostname or IP address.

Note: 5060 is the standard port number used by SIP protocol, Don't change this part if it is not required.

·Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

.

·Codecs

Define the codec for this sip trunk and its priority

Note: codec can only display when edit it after creating the trunk.

·Transport

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

·Qualify

Send check alive packets to the sip provider.

·DTMF mode

Set default mode for sending DTMF of this trunk. Default setting: rfc2833

2) DOD Setting

·DOD

Direct Outward Dialing Number.

·Associated Extension

The extension make call out via BRI Trunk will display the associated DOD.

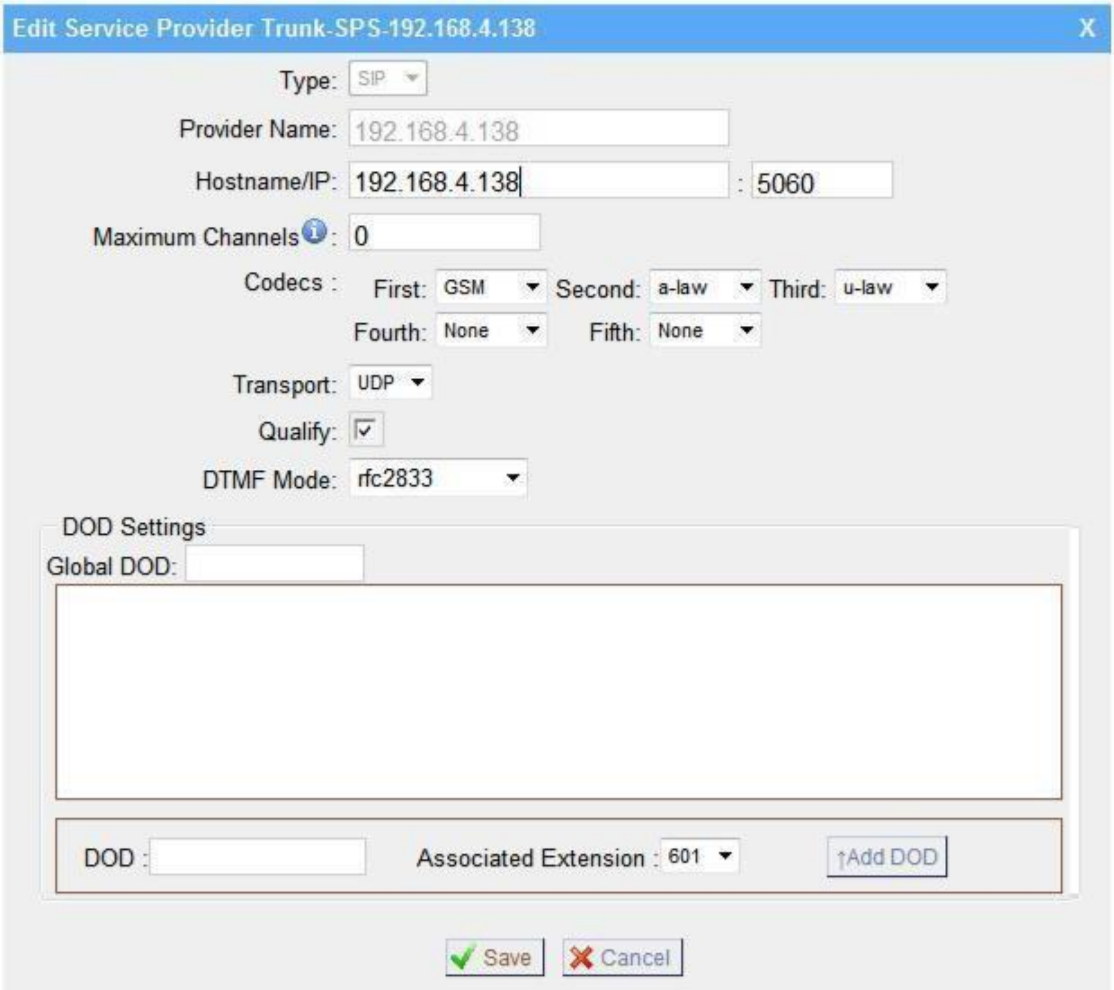


Figure 3.3.2.4

3.3.3 Outbound Routes

Outbound routing defines how outgoing calls are processed through the trunks.

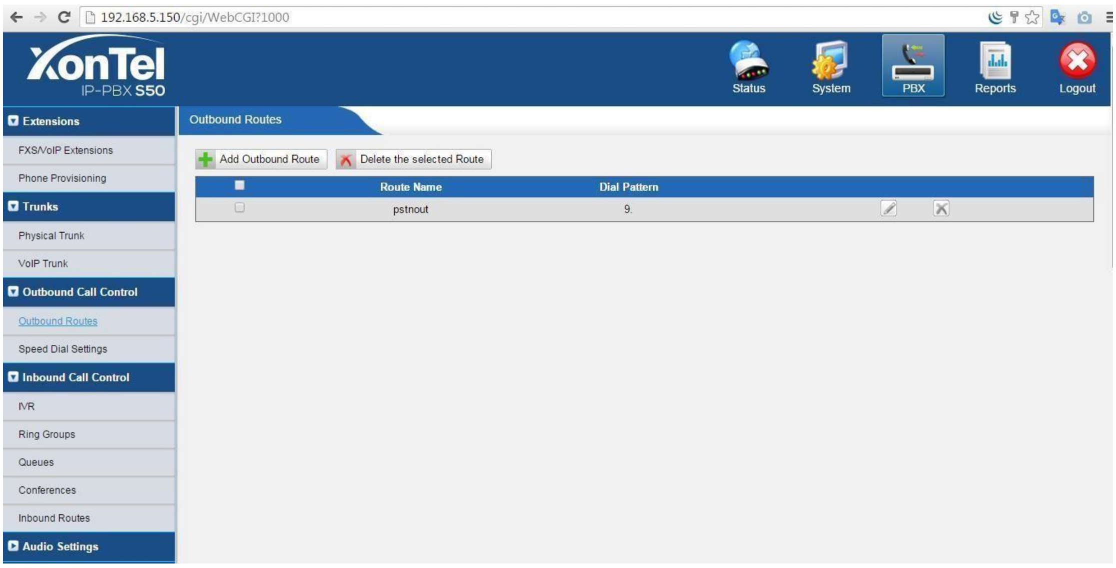


Figure 3.3.3

3.3.3.1 New Outbound Route

Click ‘New Outbound Route’ and fill in the corresponding information in the popup window.

1）General

·Route Name

Name of this Outbound Route. Ex: 'Local' or 'Long Distance' etc.

·Dial Pattern

Outbound calls that match this dial pattern will use this outbound route. There are a number of dial pattern characters that have special meanings:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9) The ‘**.**’Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The ‘**!**’ will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX**will match any 7 digits phone number.

Example 2: **1NXXNXXXXXX**will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Strip digits from front

Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.

·Prepend these digits before dialing

These digits will be prepended to the phone number before the call is placed. For example, if a trunk requires 10 digit dialing, but users are more comfortable with 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit phone numbers before calls are placed. When using analog trunks, a 'w' character may also be prepended to provide a slight delay before dialing.

·Password

The route password can be used to protect this route from being accessed without a password.

Memory Hunt

Round robin with memory, remembers which trunk was used last time, and then use the next available trunk to call out.

·Member Extensions

Defines the extensions that will be permitted to use this outbound route.

·Member Trunks

Defines the trunks that can be used for this outbound route.

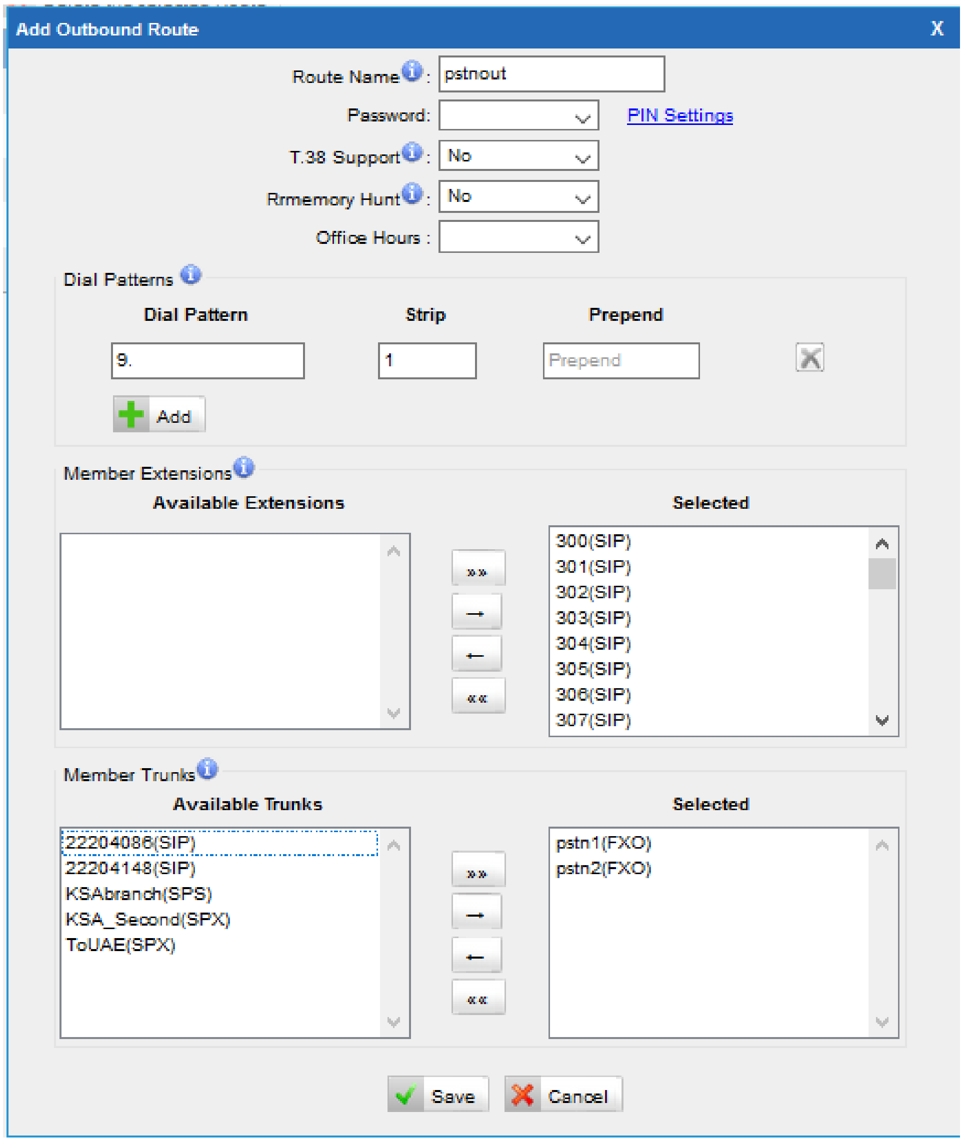


Figure 3.3.1

3.3.4 Phone Provisioning

The Auto Provision sub menu provides users a method to Auto Provision IP Phone after the Express Setup process.

Note: Auto Provision functions fully test with these models:

XonTel (S10P , S11P , S12P . S16P , S17P , S18P , S19P , S20P , S21P , S22P ,

S23P)
Snom (300,320,360,370)
Polycom (IP 6000,IP 7000,IP 32X,IP33X,IP430,IP450,IP550,IP560,VVX1500)
Cisco (IP7940,IP7960)
Aastra(480i,480i CT,6757i,6757i CT, 6737i)
Grandstream(GXP1450,GXP2100,GXP2110,GXP2120)
Escene(ES220,ES320,ES330,ES410,ES620)
Fanvil(C56,C58,C60,C62)

News:
When provisioning XonTel and Yealink IP phone, S50 is not needed to be set as the only DHCP server any more.

☐	4	00a859cf93e8	Gulfsip	192.168.1.13	--
☐	5	00156516aefc	Yealink	192.168.1.16	--
☐	6	00156532b5f8	Yealink	192.168.1.17	--
☐	7	d4676123ff02	Xontel	192.168.1.19	--
☐	8	d467619ac61c	Xontel	192.168.1.190	--
☐	9	d467619ac5e5	Xontel	192.168.1.191	--

Figure 3.3.4.2

3.3.4.3 Phone book

You can add your contacts here and provision them to your IP phone.

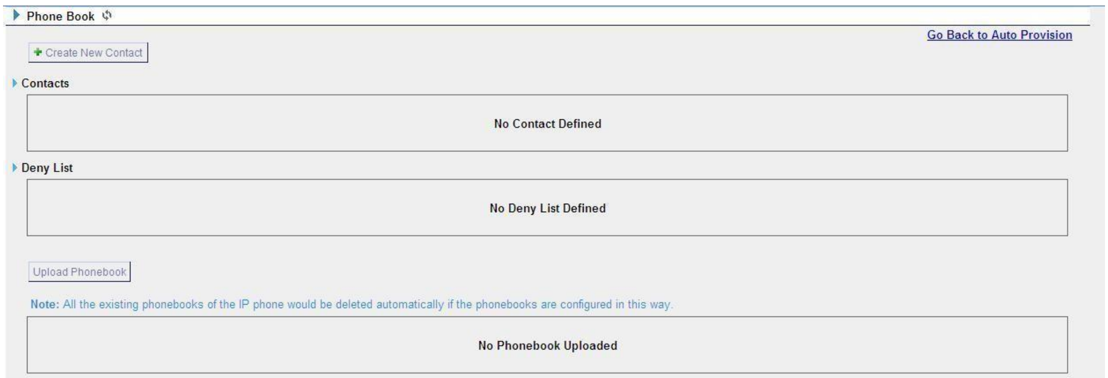


Figure 3.3.4.2

1) Add Contact

·Type

There are three types: None, VIP and Deny list (Blacklist).

·Group

There are 5 groups: None, Friends, Family, Work, Colleagues list.

·Nick Name

You can set a nick name for this number.

·Favorite
Only works with snom phone.

·Organization
Input the organization of this contact. Only works with snom phone.

·Title
Input the title of this contact. Only works with snom phone.

·Email
Input the email of this contact. Only works with snom phone.

·Birthday
Input the birthday of this contact. Only works with snom phone.

·First Name
Input the first name of this contact. Only works with snom phone.

·Family Name
Input the family of this contact. Only works with snom phone.

·Office Number
Input the office number here.

·Mobile Number
Input the mobile number here.

·Home Number
Input the home number here.

·Sub Number
Add sub number of this contact. Only works with snom phone.

·Note
Take some note of this contact. Only works with snom phone.

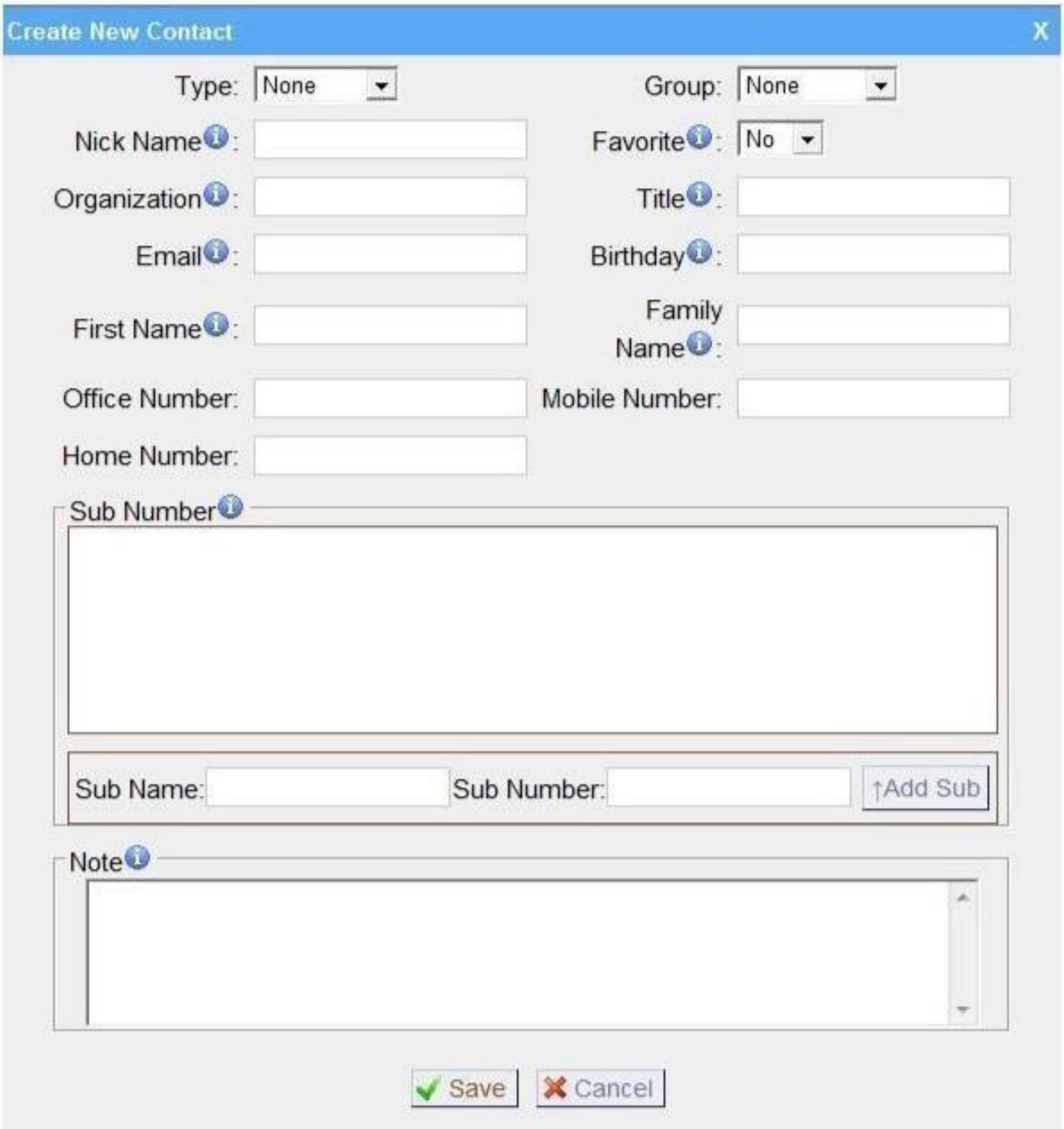


Figure 3.3.4.2.1

2) Upload Phonebook

You can upload a phonebook before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones. The format of phonebook should be *.xml.

Note: All the existing phonebooks of the IP phone will be replaced automatically if the phonebooks are configured in this way.

3.3.4.4 configure phone

Let's take provisioning yealink as an example

Create New Phone have two modes,

Create New phone in webpage and Upload the IP Phone's configure file.

1. Add new phone via webpage

Click 'Add Phone' and fill in the corresponding information in the popup window. 1) General

· Enabled

Choose yes or no to enable or disable this extension.

· NewConfig

If the firmware version number is greater than or equal to 70,please enable this option.

· MAC address

Input the MAC address of IP phone

·Name

Put the name of this Phone here.

·Manufacturer

You can choose the Manufacturer of IP phone

·Phone Type

Choose the model of your phone. Only for snom phone

·Call Waiting

This call feature allows your phone to accept other incoming calls to an extension already in an active call.

·Key as Send

Configure the key as send, you choose # ,* or disable it

·Auto redial

Enable the auto redial for IP Phone

·Auto answer

Configure if auto answer is allowed for IP phone

·Phone book

Enable the feature of phone book of IP phone

·Line

Extension: Selected the extension number for IP Phone. Label: It is shown on the LCD for users to identify the account.

Line Active: You can choose on/off to enable/disable the account respectively.

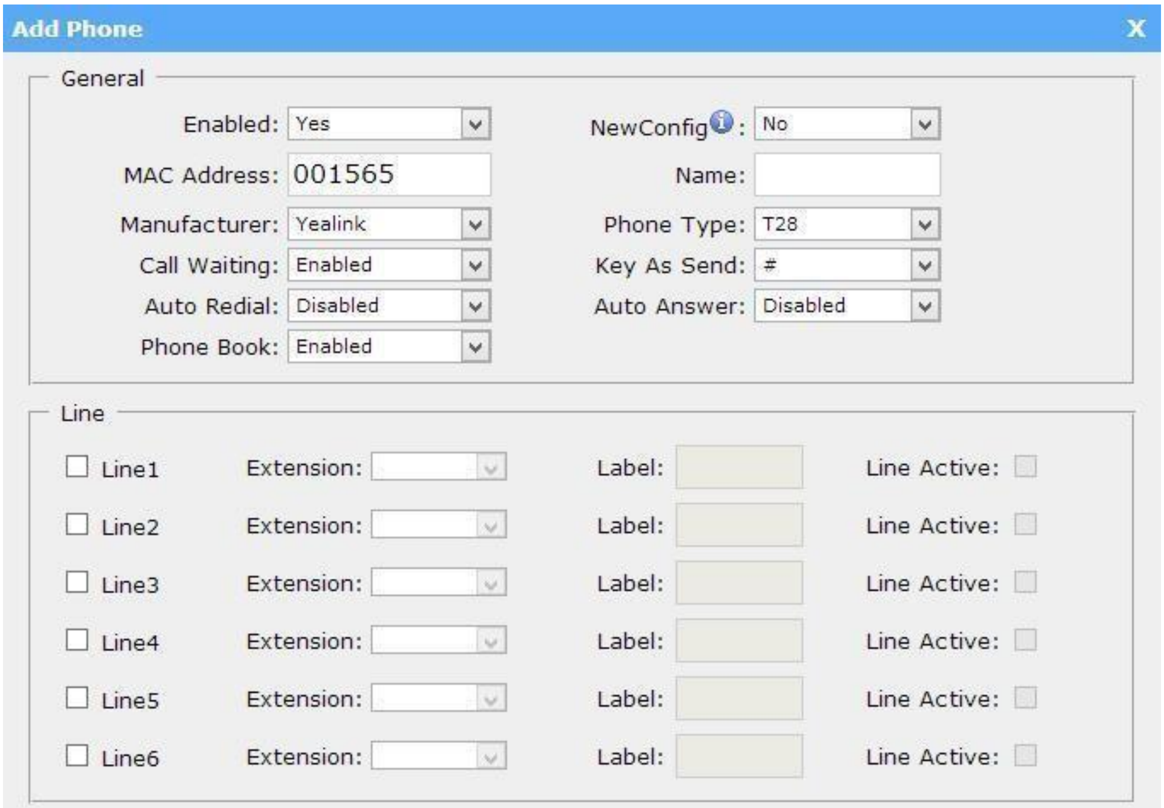


Figure 3.3.4.4.1

2) Audio codec

In this section, we can design the allowed codec for IP phone

Figure 3.3.4.4.2

3) Line keys settings

Configure the DSS keys/Function Keys

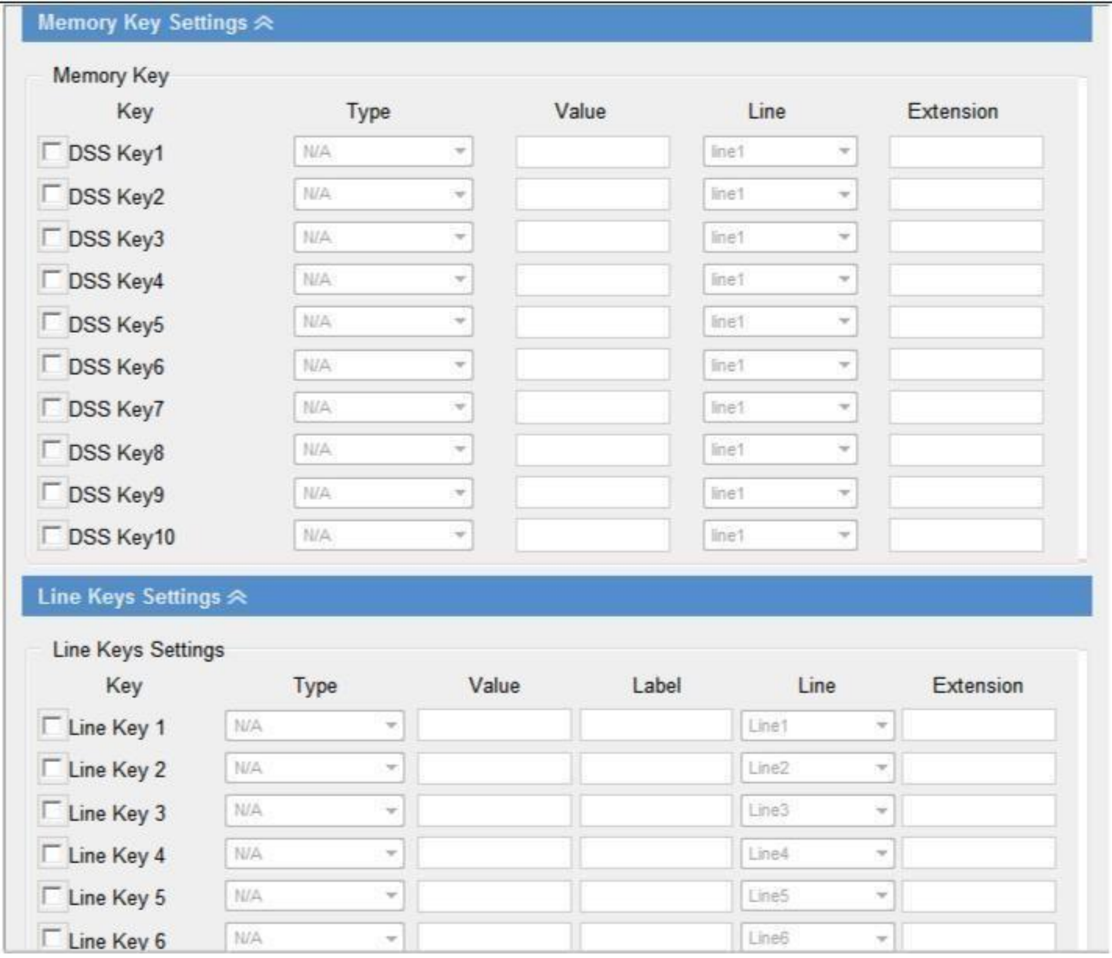


Figure 3.3.4.4.3

3.3.4.5 Not configured phone

In this section, S50 will scan all the supported IP phones and display here, we can click the 'MAC address' of IP phone and input the corresponding information in the popup window, like the picture shows below

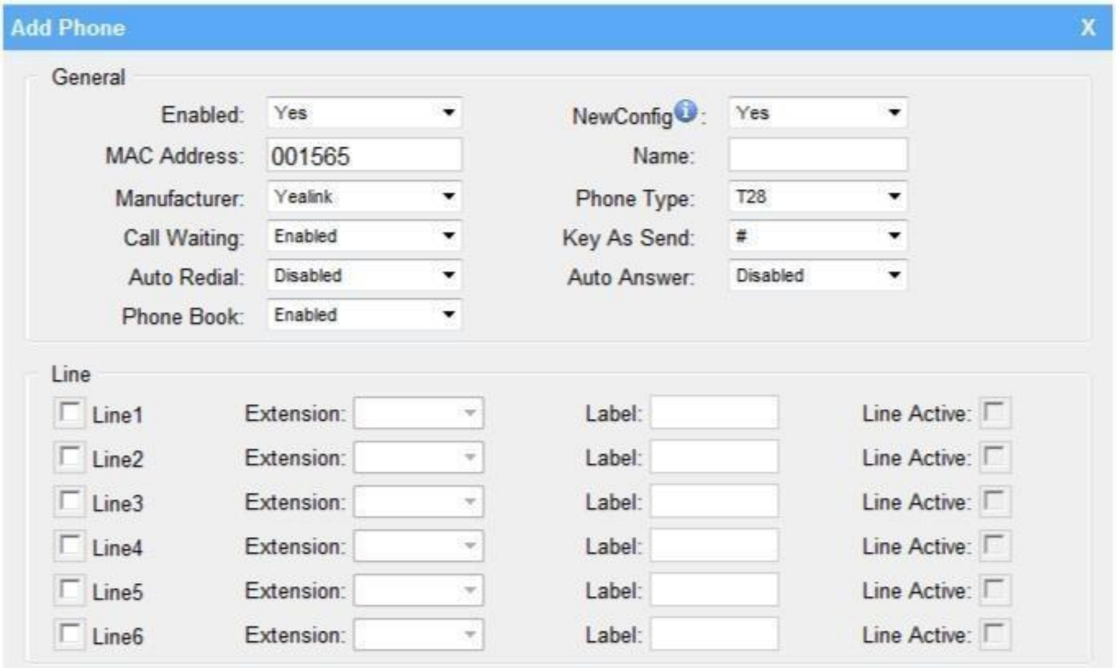


Figure 3.3.4.5

3.3.4.6 Upload a file

Click ‘Upload a file’ and choose the configure file of IP phonein the popup window.

Note: the file format must be .cfg

Please edit the configuration files in advance before uploading.

3.4 Inbound Call Control

3.4.1 IVR

When there’s an inbound call aims at Auto Attendant, S50 will play an IVR recording and route the caller to the requesteddestination (for example, ‘Welcome to XX company, for sales press 1, for technical support press 2, for operator press 0, etc’). The system will transfer the call to corresponding extension according to DTMFdigits inputted by the user.

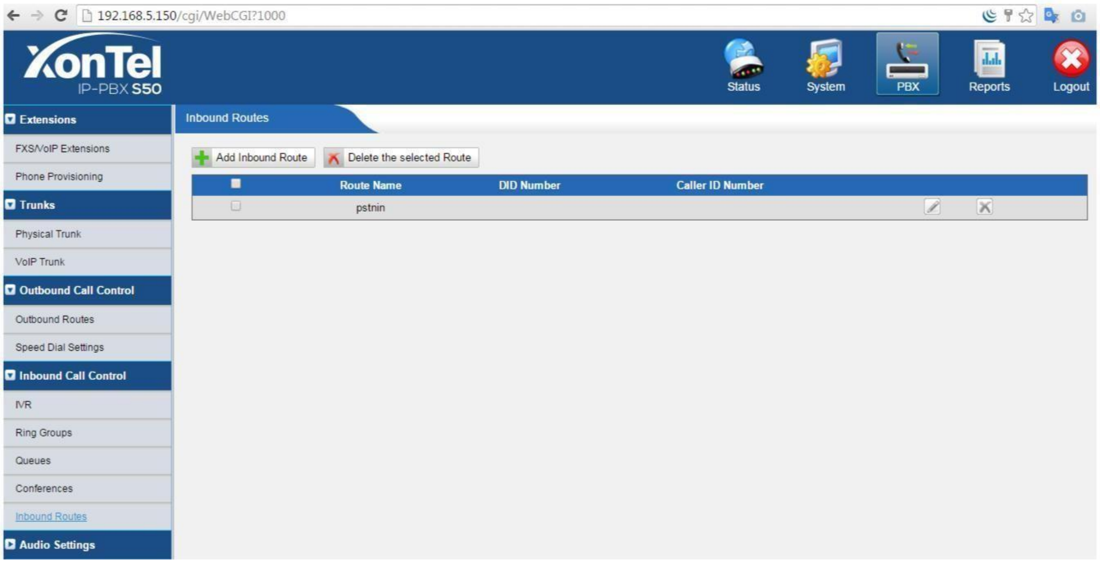


Figure 3.4.1

3.4.1.1 Create IVR

Click 'Create New IVR'.

1) General

·Number

S50 treats IVR as an extension; you can dial this extension number to reach the IVR.

·Name

A name for the IVR

·Prompt

The prompt recording that will be played when this IVR is reached.

·Repeat Count

The number of times that the selected IVR prompt will be played.

·Key Timeout

Wait for the user to enter a new extension for a specified number of seconds.

·Allow Dialing of Other Extensions

Allow the caller to dial other extensions other than the ones explicitly defined.

2) Key Press Events

A list of actions that can be performed depending on the digit dialed by the user.

·Time Out

Defines the timeout action. A timeout occurs after the IVR prompt has finished playing for the number of times specified by the 'Repeat Count' field.

·Invalid

Defines the invalid action. The invalid action is triggered if the user enters a DTMF digit that is not defined for this IVR.

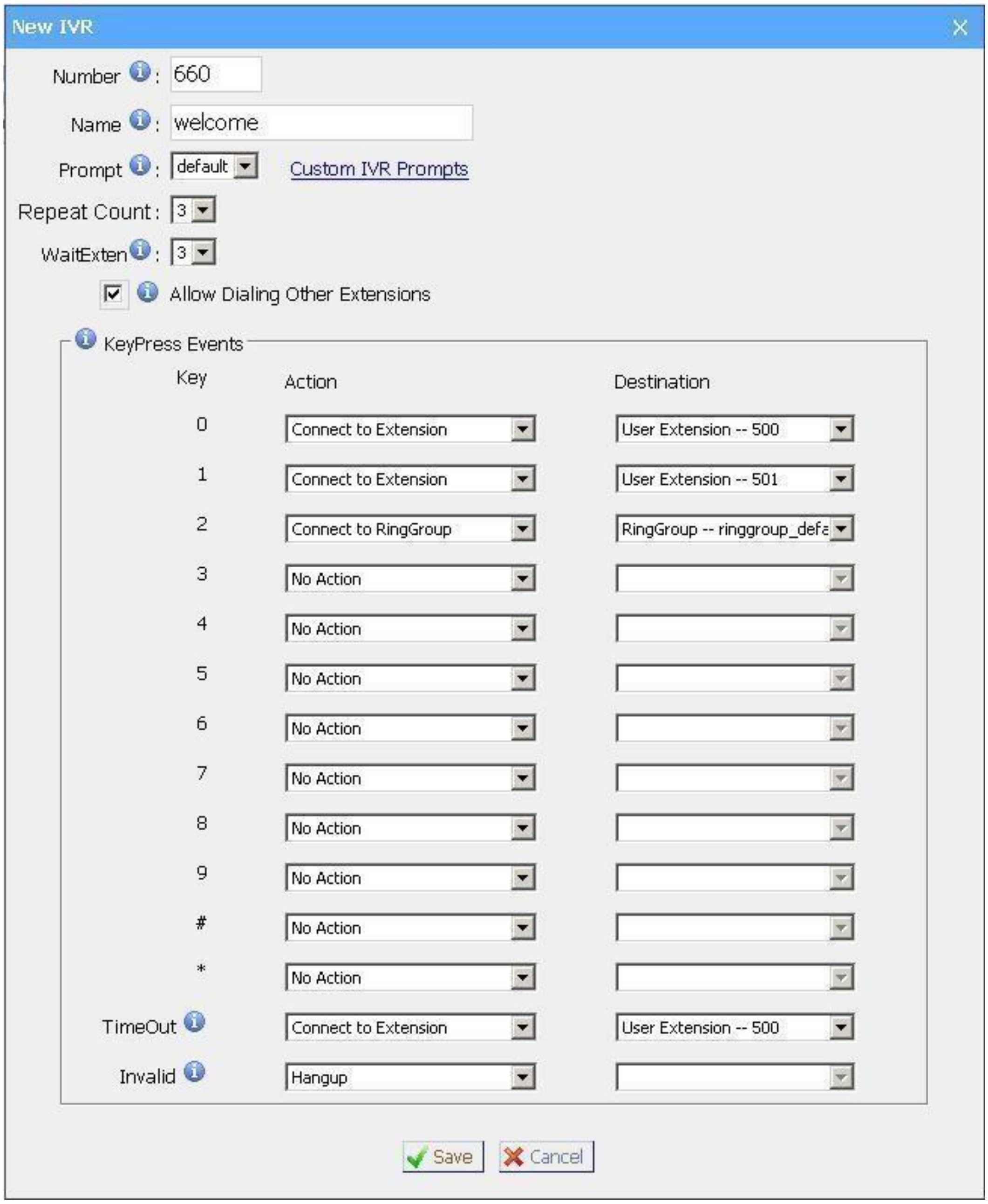


Figure 3.4.1

3.4.2 Queues

Call Queues give users (i.e. call centers) an efficient means to have their calls answered in the order they were received to deliver top tier customer service.

Call queues allow calls to be sequenced to one or more agents.

Note: Dial 'Queue number + '*'' to log in or 'Queue number + '**'' to log out the queue. For example, if the queue number is '680', then agent can dial '680*' to log in or '680**' to log out.

1） General

·Queue Name

A name for the Queue.

·Queue Number

Use this number to dial into the queue, or transfer callers to this number to put them into the queue.

·Queue Password

You can require agents to enter a password before they can login to this queue.

·Queue Agent Timeout

The number of seconds an agent's phone can ring before we consider it a timeout.

·Queue Max Wait Time

The maximum number of seconds a caller can wait in a queue before being pulled out. (0 for unlimited).

·Queue Ring Strategy

This option sets the Ringing Strategy for this Queue.The options are

RingAll: Ring All available Agents simultaneously until one answers.

LeastRecent: Ring the Agent which was least recently called.

FewestCalls: Ring the Agent with the fewest completed calls. Random: Ring a Random Agent.

RRmemory: Round Robin with Memory, Remembers where it left off in the last ring pass".

2） Agents

This selection shows all users. Select a user here makes them a agent of the current queue.

3) Caller Position Announcements

·Announce Position

Announce position of caller in the queue

·Announce Hold Time

Enabling this option causes S50 to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

·Frequency
How often to announce queue position and estimated hold time.
Note: '0 seconds' means disable the announcement
4) Periodic Announcements
·Prompt
Select a prompt file to play periodically.

·Frequency
How often to announce a prompt to the caller.

5) Events
If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.

6) Failover-Destination
Defines the failover action. A failover occurs after the user reach the Queue max wait time.

7) Others
·Music On Hold
Select the 'Music on Hold' Class for this Queue.

·Leave When Empty
This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.
Yes: Callers are forced out of a queue when no agents are logged
in. No: Callers will remain in a queue with no agents.

·Join Empty
This option controls whether callers can join a call queue that has no agents. There are two options,
Yes: Callers can join a call queue with no agents or only unavailable
agents No: Callers cannot join a queue with no agents
The default option is No.

·Agent Announcement
Announcement played to the Agent prior to bridging in the caller.

·Join Announcement
Announcement played to callers once prior to joining the queue.

·Retry

The number of seconds we wait before trying all the phones again.

·Wrap-up time

How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 30.

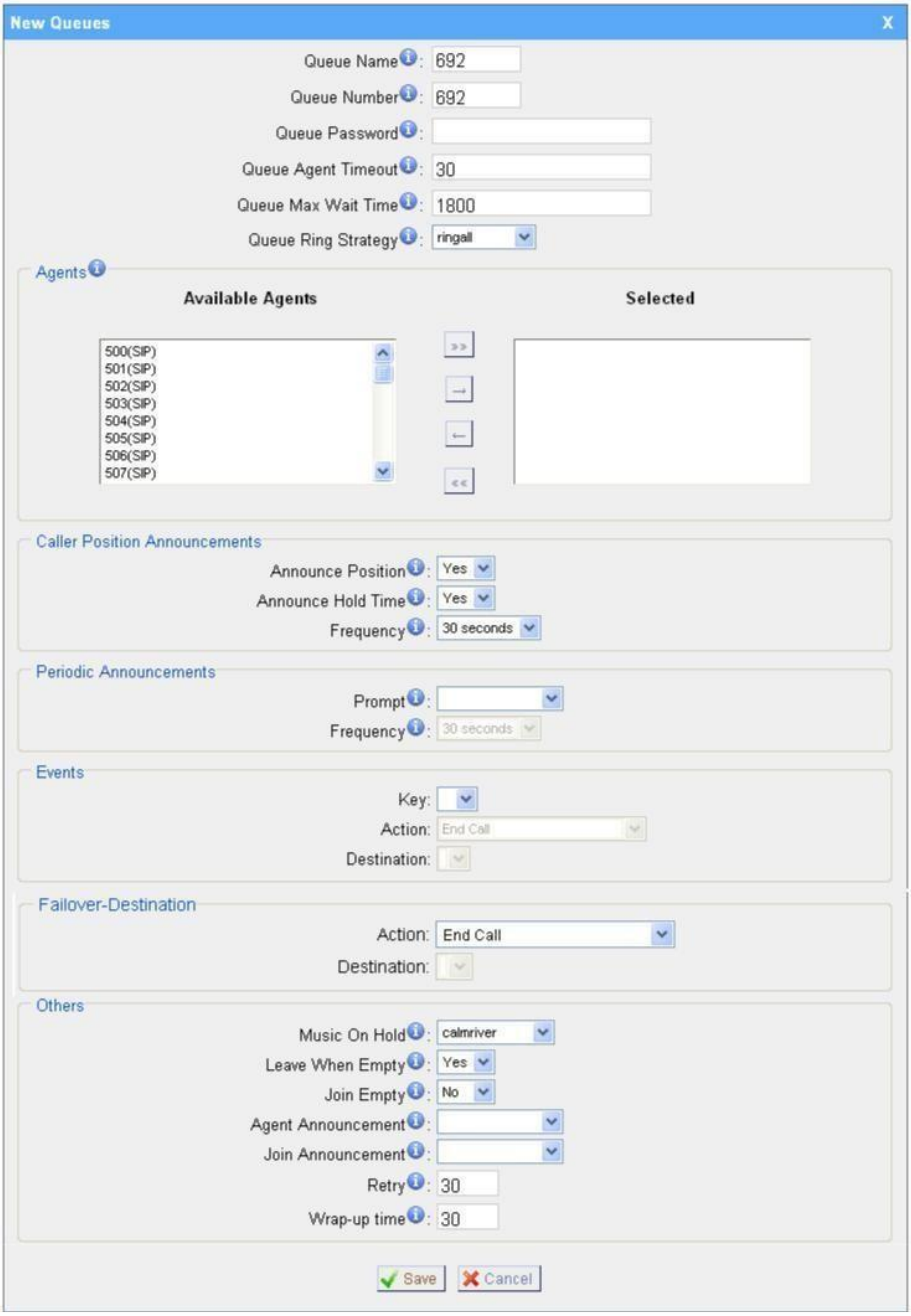

New Queues
X
Queue Name: 692
Queue Number: 692
Queue Password:
Queue Agent Timeout: 30
Queue Max Wait Time: 1800
Queue Ring Strategy: ringall
Agents
Available Agents
Selected
500(SIP)
501(SIP)
502(SIP)
503(SIP)
504(SIP)
505(SIP)
506(SIP)
507(SIP)
Caller Position Announcements
Announce Position: Yes
Announce Hold Time: Yes
Frequency: 30 seconds
Periodic Announcements
Prompt:
Frequency: 30 seconds
Events
Key:
Action: End Call
Destination:
Failover-Destination
Action: End Call
Destination:
Others
Music On Hold: calmriver
Leave When Empty: Yes
Join Empty: No
Agent Announcement:
Join Announcement:
Retry: 30
Wrap-up time: 30
Save
Cancel

Figure 3.4.2

.

3.4.3 Custom Prompts

1. Record new Prompt

The administrator can use this screen to record custom prompts by doing the following:

1） Click 'Record New Custom Prompt'

2) Input the desired file name on the popup window and choose an extension to call for recording (such as 500).

3) Click 'Record'. The selected extension will ring and you can pick up the phone to start recording.

Figure 3.4.3.1

2. Upload Prompt

The administrator can also upload prompts by doing the following:
1)） Click 'Upload Prompt'.

2) Click 'Browse' to choose the desired prompt.

3) Click 'Upload 'to upload the selected prompt.

Note: The file size must not be larger than 1.8MB, and the file must be WAV format:

GSM 6.10 8kHz, Mono, 1Kb/s;

Alaw/Ulaw 8kHz, Mono, 1Kb/s;

PCM 8kHz, Mono, 16Kb/s.

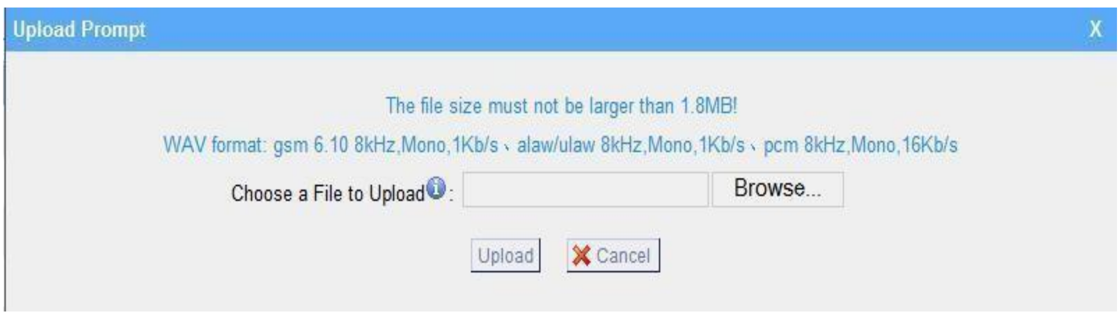


Figure 3.4.3.2

Note: the maximum amount of custom prompt is 16.

3.4.4 Ring Groups

Ring groups can be configured to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

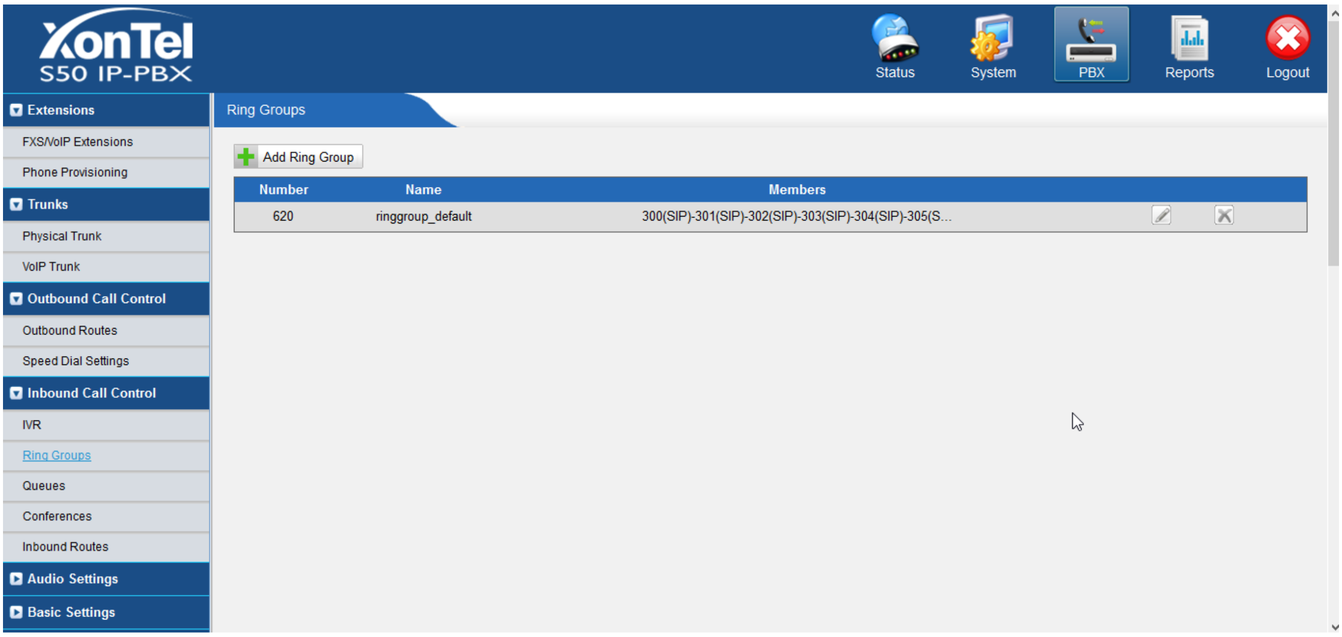


Figure 3.4.4

3.4.4.1 Create Ring Group

Click 'New Ring Group' to enter into the Manage Ring Groups page

1） General

·Ring Group Name

This option defines a name for this group, i.e. 'Sales'. 'Ring Group Name' is a label to help you identify this group in the group list.

·Ring Group Number

This option defines the numbered extension that can be dialed to reach this group.

·Strategy

This option sets the Ringing Strategy for this Group. The options are as follows:

1. Ring All Simultaneously: Ring all available Extensions simultaneously.
2. Ring Sequentially: Ring each extension in the group one at a time.

·Seconds to ring each member

1. If the strategy is 'Ring All Simultaneously', it means set the number of seconds to ring this group before routing the call according to the 'Destination if No Answer' settings.
2. If the strategy is 'Ring Sequentially', it means set the number of seconds to ring a single extension before moving onto the next one.

2) Ring Group Members

An extension can be made a member of this ring group by moving it into the 'Selected' box.

3) Destination If No Answer

When all members on this group fail to answer the call, system will handle the call according to the selected destination.

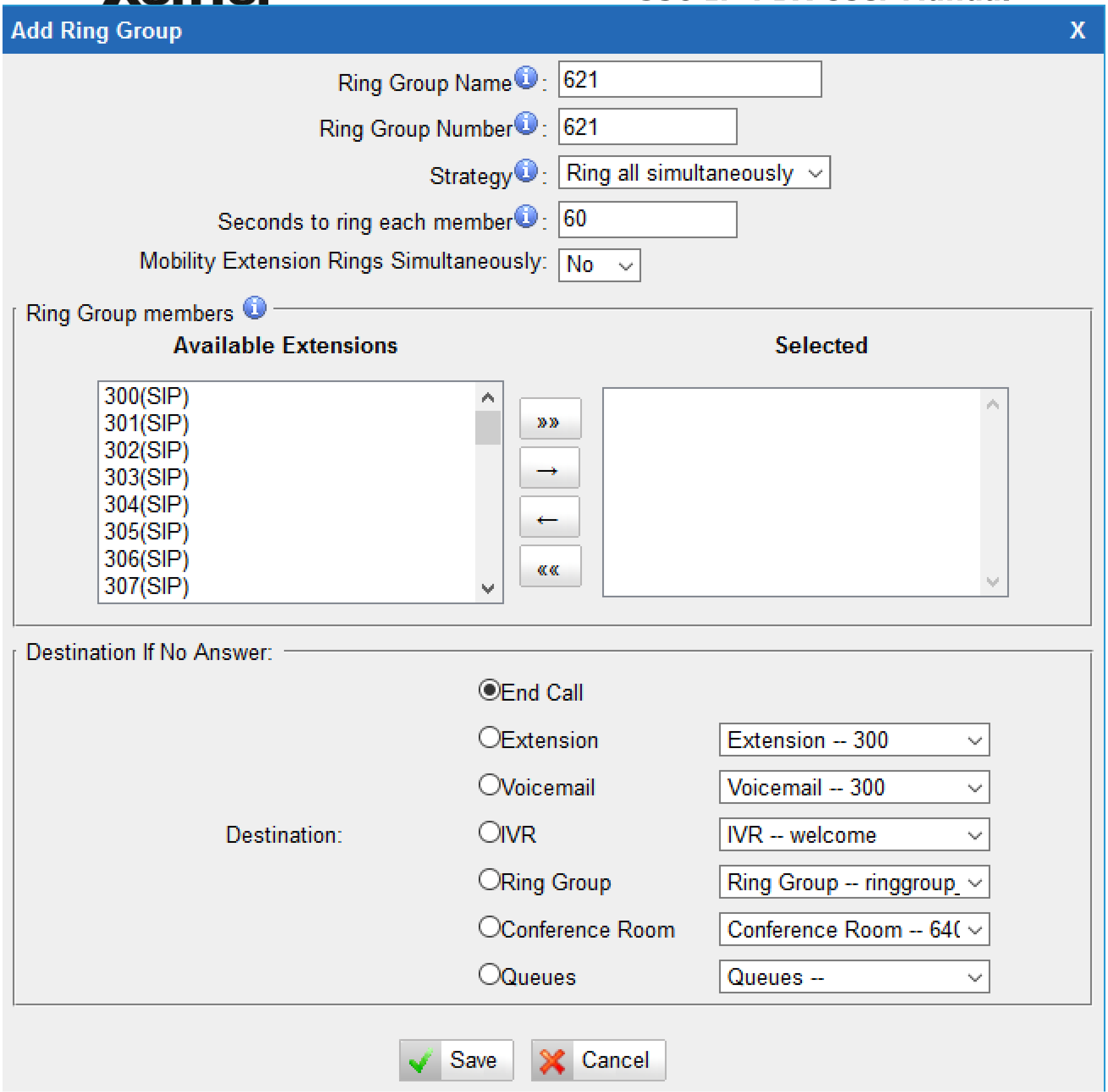


Figure 3.4.4.1

Note: the maximum amount of ring group is 16.Each group includes 20 members at most.

3.4.5 Inbound Routes

Inbound routing processes incoming call traffic to destination extensions during office hours or outside office hours.

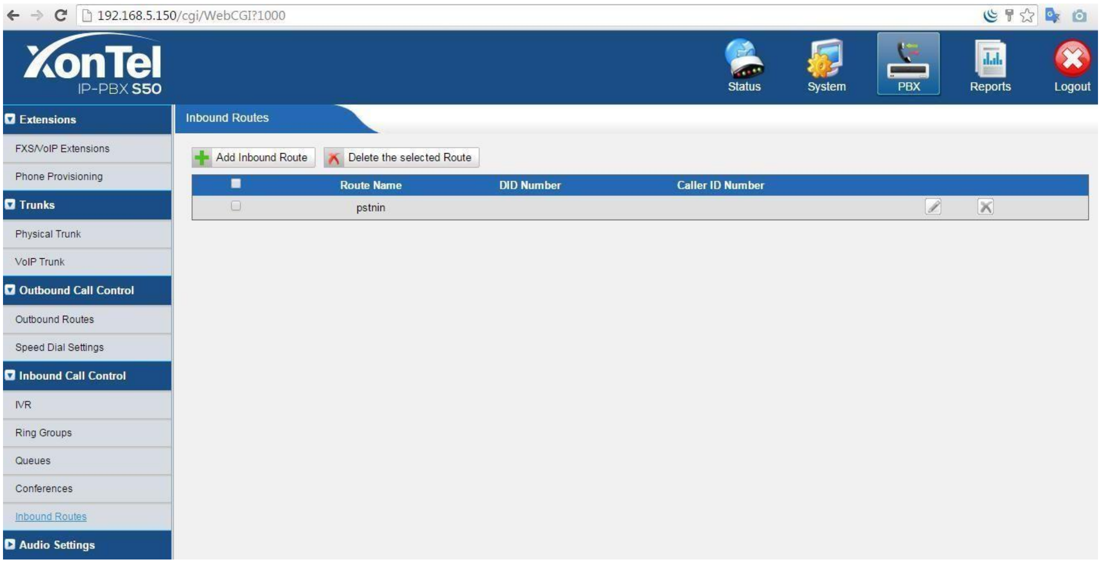


Figure 3.4.5

3.4.5.1 Create Inbound Route

Click 'New Inbound Route' to enter to the Manage Inbound Routes page. When an incoming call arrives, the system will first check 'fax detection', then 'Holidays', at last 'Business Days'.

1) General

·Route Name

A name for this inbound route. Ex: 'pstnin' etc.

·DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only service provider, E1 trunks, BRI trunks or SIP trunks need to be configured with this setting.

You can also use pattern matching to match a range of numbers.

The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9) The '**.**'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.
Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Extension

Define the extension for DID number. This field is only valid when you use BRI, SIP, SPS or SPX trunk for this inbound router. You can only input number and '-'in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

·Caller ID Number

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no DID info.
You can also use a pattern match (e.g. 2[345]X) to match a range of numbers. The following patterns may be used:
X: Any Digit from 0-9
Z: Any Digit from 1-9
N: Any Digit from 2-9
[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9) The '**.**'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.
The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.
Example 2: **1NXXNXXXXX**will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Distinctive Ringtone

S50 support mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to '**Family**', the ring tone will be played if the phone receives the incoming call.

Note: If you want to use feature Distinctive ringtone, please confirm you're your IP phone support this feature firstly. Currently distinctive ringtone can be compatible with XonTel and Snom phone.

2) Member Trunks

This area allows you to select which trunks will be member trunks for this route. To make a trunk a member of this route, please move it to the 'Selected' box.

4) Business Days
Define where the calls will be routed during Business Days.

·Office Days
Select one defined business days office days.

·Office Hours Destination
Configure where to route the incoming calls during office hours.

·End Calls
Route the incoming calls to end calls, System will auto hang-up the call.

·Extension
Route the incoming calls to a specific extension.

·Voicemail
Route the incoming calls to extension's voicemail.

·IVR
Route the incoming calls to a specific IVR.

·Ring Group
Route the incoming calls to a specific Ring Group.

·Conference Room
Route the incoming calls to a specific Conference Room.

·DISA
Route the incoming calls to a specific DISA.

·Queues
Route the incoming calls to a specific Queue.

·Outbound Routes
Route the incoming calls to a specific outbound route.

This function is mainly used for the connection of two branches.
For example: Company A locates headquarters in the USA with a branch B in China. A and B both have S50 phone systems.
Now if staff of A would like to make a call to a telephone or mobile phone in China from the extension of A but via the FXS line of B, that can be done by this configuration.

·Non-office Hours Destination

Configure where to route the incoming calls during non-office hours.

5) During Holidays

Define where the calls will be routed during Holidays.

·Holiday

Select the defined Holiday to use. When a time is defined in both Business Days and Holidays, it will be treated as Holidays.

·Destination

Configure where to route the incoming calls during holidays.

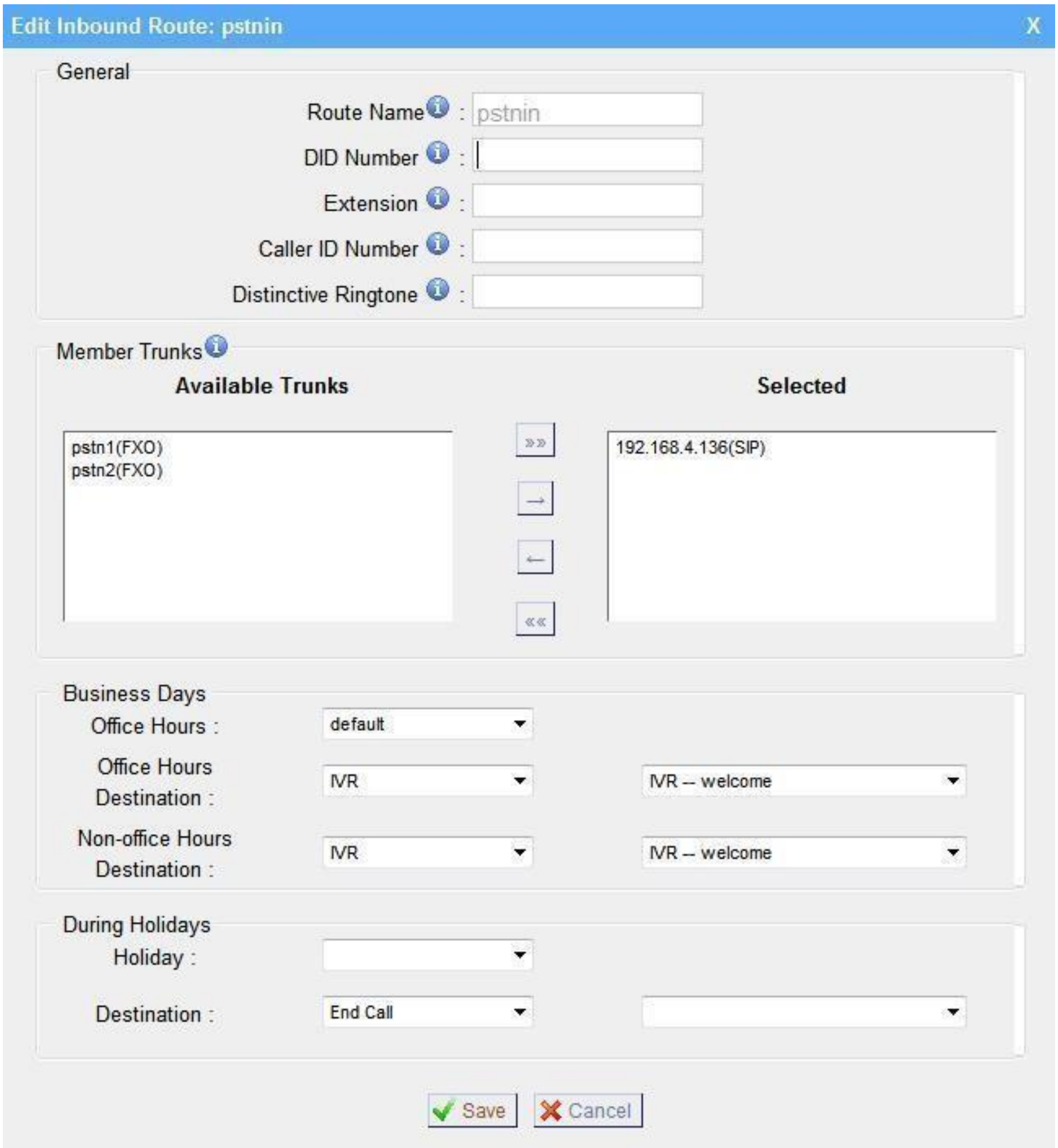


Figure 3.4.5.1

3.4.6 Blacklist

Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is registered in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.

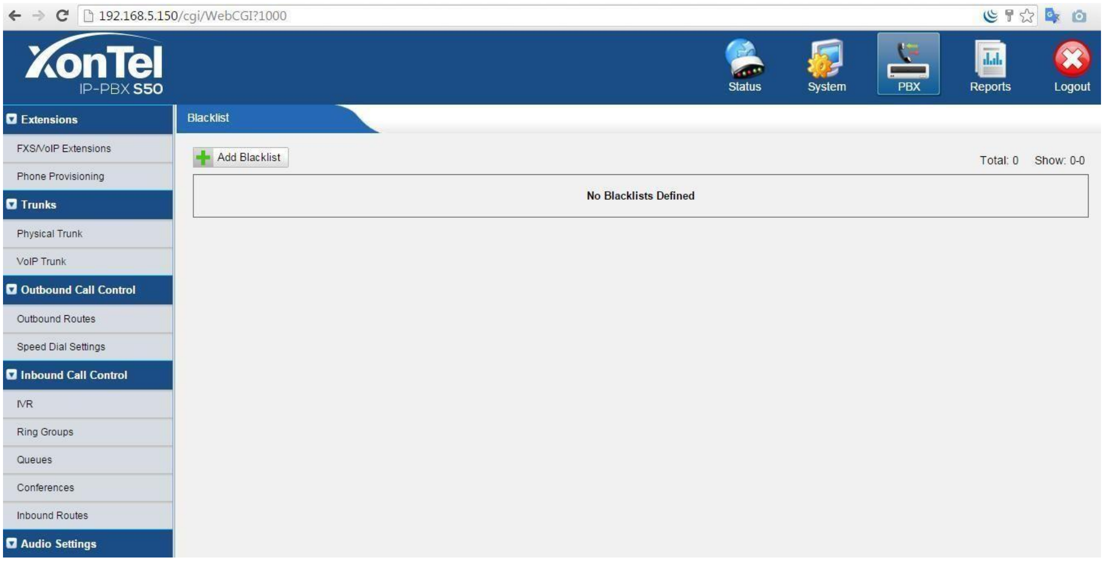


Figure 3.4.6

3.4.6.1 Create Blacklist

Click 'New Blacklist' to create a new number blacklist.

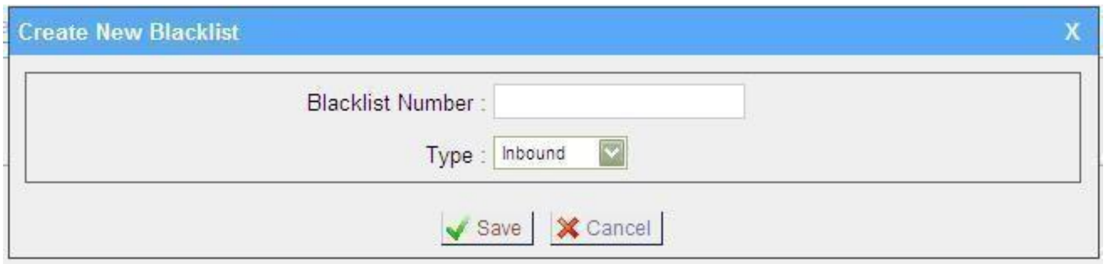


Figure 3.4.6.1

·Blacklist number

Enter the number you would like to block.

·Type

The number blocked for incoming or outgoing calls or both.

3.5 Internal Settings

3.5.1 Options

1) General

·Ring Timeout

Number of seconds to ring a device before handling the call as per the extension's Follow Me settings. Default value is 30s.

.**MAX call duration**

The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. Default value is 6000s.

.**Maximum concurrent calls**

Maximum concurrent calls limits. Default value 0 means no limit

·Music on hold

Used to set hold music for the system.

·Tone Region

Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.
Note: please reboot the system to take it effect.

·HTTP bind port/Web Access Port

Port to use for HTTP sessions. Default: 80
Note: please reboot the system to take it effect.

·FXO Mode

FXO port's operation mode.

· Distinctive Caller ID

When incoming calls are routed from ring group/queue/IVR, the caller ID displays with the name of ring group / queue/IVR , for example 5503302(ringgroup_default)

Note: for displaying (To display) IVR's name, please press the key instead of the extension number directly.

·Enable Follow Me Prompt

When set Follow me to Transfer to number on the extension page (e.g. when 500 is busy, transfer to 501), while 500 is busy, the call will be transferred to 501. If 'Enable Follow Me Prompt' choosing yes, there will be prompt 'please wait while trying to look at the person you are calling 'before transferring the call. Otherwise, the call will be transferred directly without any prompt. Default: Yes.

·Music on hold for Follow Me
Choose to play default music-on-hold, ringing tone or stay mute while 'Follow Me'.

·Invalid Phone Number Prompt
Configure the prompt when the dialed phone number is invalid.

·Busy Line Prompt
Configure the prompt when the dialed phone number is busy.

·Dial Failure Prompt
Configure the prompt when dial failed due to conjunction no-available channel.

2) Extension Preferences
·User Extensions
The default value is 500 to 616

·Ring Group Extensions
The default value is 620 to 629

·Paging Group Extensions
The default value is 630 to 639

·Conference Extensions
The default value is 640 to 659

·IVR Extensions
The default value is 660 to 679

·Queue Extensions
The default value is 680 to 689

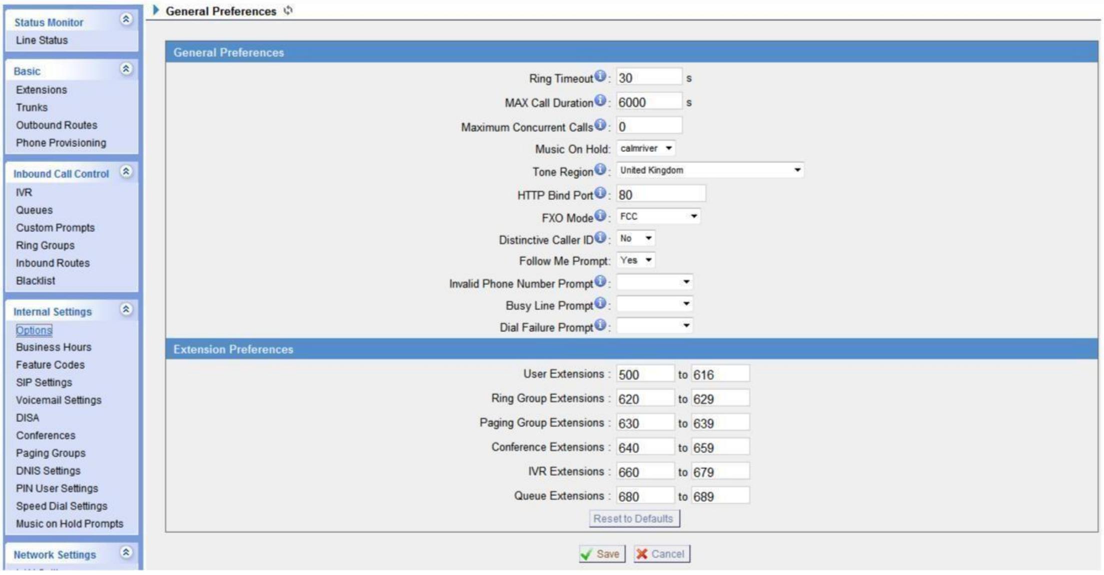


Figure 3.5.1

3.5.2 3.5.2 Business Hours

1) General

·Enable or Disable Business Hours

2) Others

·Enable Office Closed Timing

By dialing *81 (*81 is default) on an extension will force the office time closed for the device whatever the general setting is.

·Enable Office Timing

By dialing *82 (*82 is default) on an extension will force the office time enabled for the device whatever the general setting is.

·Disable Office (closed) timing

By dialing *081 (*081 is default) on an extension will disable the Office Timing or Office Closed Timing.

3) Business Days

You can setup the business hours here.

4) Holidays

You can setup the holidays here.

If a time period is configured as both Holidays and office hours, it will be treated as Holidays.

Figure 3.5.2

3.5.3Feature Codes

1) General

·One Touch Record

A user may initiate or stop call recording by dialing *1during a call. (*1 is default setting)

·Extension for Checking Voicemail

User scan check their Voicemail by dialing *2 on their phone (*2 is default setting).

·Voicemail for Extension

Users can leave a voicemail to other extensions by dialing # on their phone or the incoming call could be forwarded to an extension’s voicemail directly. (# is default setting).

For example, extension 500 want to leave a message for extension 501, users can use 500 dial’#501’ to enter the voicemail of 501.

·Voicemail main menu

User scan go to the main menu by dialing *02 (*02 is default setting).

·Attended Transfer

Users may transfer an incoming call by dialing *3 on their phone (*3 is default setting).

·Attended Transfer Timeout

The time out of transferring a call

·Blind Transfer

Users may blind transfer an incoming call by dialing*03 on their phone (*03 is default setting).

·Call Pickup

Users may pickup an incoming call by dialing *4 on their phone (*4 is default setting)

·Extension Pickup

Users may pickup a specific extension's incoming call by dialing *04+extension number on their phone (*04 is default setting)

·Intercom

Define the feature code that is used to dial an extension in intercom mode. For instance setting this value to *5 would allow you to initiate an intercom call with extension 501 by dialing *5501.

·Normal Spy

In this mode, you can only listen to the extension being spied, for example you can dial *90501 to monitor extension 501

·Whisper Spy

In this mode you can listen/whisper to the extension being spied, for example, dialing *91501 to listen to extension 501, you can also talk with 501 too.

·Barge Spy

In this mode, you can barge in both extensions involved the call, for example dialing *92501 to barge in and talk with all the extensions inside

2) Call Park Preferences

·Call Parking

User may park an incoming call on his own telephone by pressing '*6'(*6 is default setting)

·Extension range used to park calls

User may park an incoming call on a designated extension at first and then pick up the call again on any other extension.

·Number of seconds a call can be parked before it is recalled.

Defines the number of seconds that a call can be parked before it is recalled to the station that parked it.

3) Call Forwarding Preferences

·Reset to Defaults

Users may reset all call forward defaults by calling *70 on their phone (*70 is default setting).

Note: When reset to defaults. The call forwarding settings will be configured as follows:
Always forward: Disabled
Busy forward to Voicemail: Enabled
No answer forward to Voicemail: Enabled
Do not disturb: Disabled

·Enable Forward All Calls
Users may enable always forward by calling *71 on their phone (*71 is default setting)

·Disable Forward All Calls
Users may disable always forward by calling *071 on their phone (*071 is default setting)

·Enable Forward When Busy
Users may enable busy forward by dialing *72 on their phone (*72 is default setting)

·Disable Forward When Busy
Users may disable busy forward by calling *072 on their phone (*072 is default setting)

·Enable Forward No Answer
Users may enable no answer forward by calling *73 on their phone (*73 is default setting)

·Disable Forward No Answer
Users may disable no answer forward by calling *073 on their phone (*072 is default setting)

·Forward to number
Users may activate call forwarding by dialing this feature code, followed by the extension or phone number to forward all calls to.

Note: Users may activate Forward to number by dialing *74 + phone number. e.g.: by dialing *74501, all calls will be forwarded to extension 501.

·Forward to Voicemail
Users may forward the call to Voicemail by calling *074 on their phone (*074 is default setting)

·Enable Do Not Disturb

Users may enable do not disturb by calling *75 on their phone (*75 is default setting)

·Disable Do Not Disturb

Users may disable do not disturb by calling *075 on their phone (*075 is default setting)

Figure 3.5.3

3.5.4 SIP Settings

1) General

·UDP Port

Port use for sip registrations, Default is 5060.

·TCP Port

Port use for sip registrations, Default is 5060.

·TLS Port

Port use for sip registrations, Default is 5061.

·RTP Port Start

Beginning of RTP port range

·RTP Port End

End of RTP port range

·DTMF Mode
Set default mode for sending DTMF.Default setting: rfc2833

·Max Registration/Subscription Time
Maximum duration (in seconds) of a SIP registration. Default is 3600 seconds.

·Min Registration/Subscription Time
Minimum duration(in seconds)of a SIP registration.Default is 60seconds.

·Default Incoming/Outgoing Registration Time
Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.

·Register Attempts
The number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default is 8 times.

·Register Timeout
Number of seconds to wait for a response from a SIP Registrar before timed out. Default is 20 seconds.

·Calling Channel Codec Priority
Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected in preference. If not, S50 will follow the priority in your SIP/SPS trunks.

·Video Support
Support for SIP video or no. Default is yes.

·Max Bit Rate
Configure the max bit rate for video stream. The default: 384kb/s

·DNS SRV Look Up
Please enable this option when your SIP trunk contains more than one IP address.

·User Agent
To change the user agent parameter of asterisk, the default is 'S50', you should change it if needed.

2) NAT
Note: Configuration of this section is only required when using remote

extensions.

·Enable STUN

STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

·STUN Address

The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.

·External IP Address

The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.

·External Host

Alternatively you can specify an external host, and the system will perform DNS queries periodically.
This setting is only required when your public IP address is not static. It is recommended that a static public IP address be used with this system. Please contact your ISP for more information.

·External Refresh Interval

If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.

·Local Network Identification

Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall.
Some examples of this are as follows:
'192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks;
'10.0.0.0/255.0.0.0' : Also RFC1918;
'172.16.0.0/12':AnotherRFC1918withCIDRnotation;
'169.254.0.0/255.255.0.0' : Zero conf local network.
Please refer to RFC1918 for more information.

·NAT Mode

Global NAT configuration for the system. The options for this setting are as follows:
Yes = Use NAT. Ignore address information in the SIP/SDP headers and reply to the sender's IP address/port.
No = Use NAT mode only according to RFC3581.
Never = Never attempt NAT mode or RFC3581 support.

Route = Use NAT but do not include rport in headers.

·Allow RTP Reinvite
By default, the system will route media steams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.

3） Codecs
A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.
u-law: A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

a-law: A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

GSM: A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

SPEEX: Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

G.722: G.722 is a wideband speech coding algorithms which supports the bit rate of 64, 56 and 48kbps wideband. It's a broadband voice encoding of G series.

G.726: A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

ADPCM, G.729A, H261, H263, H263p, H264,MPEG4.
Note: If you would like to use G.729, please enter your license.

4） QOS
QoS (Quality of Service) is a major issue in VOIP implementations. The issue

is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic.When the network capacity is insufficient, QoS could provide priority to users by setting the value.

5) Advanced Settings

·From Field

Where to get the caller ID in sip packet.

·To Field

Where to get the DID in sip packet.

·180 Ringing

It is set when the telecom provider needs. Usually it is not needed.

·Remote Party ID

Whether send Remote-Party-ID on SIP header. Default no.

·Allow Guest

Whether allow anonymous registration extension. Default: no.
This option is used to avoid some anonymous calls by hackers.

·Pedantic

Enable pedantic parameter. Default: no.

·Session -timers

Enable sesstion-timer mode, default: yes.

·Sesstion-expires

The max refresh interval.

·Sesstion-minse

The min refresh interval, which mustn't be less than 90 .

·Sesstion-refresher

Choose sesstion-refershher, the default is Uas.

sFigure 3.5.4

3.5.5 Voicemail Settings

1) General Voicemail Settings

a) Message Options

·Max Messages per Folder

Set the maximum number of messages that can be stored in a single voicemail box.

·Max Message Time
Set the maximum length of a single voicemail message.

·Min Message Time
Set the minimum length of a single voicemail message. Messages below this threshold will be automatically deleted.

·Ask Caller to Dial 5
If this option is set, the caller will be prompted to press 5 before leaving a message.

·Operator Breakout from Voicemail
If this option is set, the caller can jump out of the voicemail and go to the destination (IVR) you set by dialing "0".

b) Greeting Settings
·**Busy Prompt**
Greeting played when the extension called is busy.
Skip greeting: Do not play a greeting.
Play busy greeting: play the extension busy greeting.

·Unavailable Prompt
Greeting played when the extension called is Unavailable.
Skip greeting: Do not play a greeting.
Play Unavailable greeting: play the extension Unavailable greeting.

·Leave a Message Prompt
Greeting when ask the caller to dial 5 to leave a message.
Skip greeting: Do not play a greeting.
Play busy greeting: play the extension busy greeting.
Play Unavailable greeting: play the extension Unavailable greeting.

c) Playback Options
·Announce Message Caller ID
If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

·Announce Message Duration
If this option is set, the duration of the message in minutes will be played back before the voicemail message begins playing.

.Announce Message Arrival Time
If this option is set, the arrival time of the message will be played back before the voicemail message begins playing.

.Allow Users to Review Messages
Allow callers to review their recorded message before sending it to voicemail.

2) SMTP Settings for Voicemail
Note: If you want to send voicemail messages as email attachments, please configure this section.

·E-mail Address
The E-mail Address that S50 will use to send voicemail.

·Password
The password for the email address used above.

·SMTP Server
The IP address or hostname of an SMTP server that the S50will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

·Port
SMTP Port: the default value is 25.

·Use SSL/TLS to send secure message to server
If the server of sending email needs to authenticate the sender, you need to select the check box.
Note: Must be selected for Gmail or exchange server.
After filling out the above information, you can click on the 'Test Account Settings' button to check whether the setup is OK.
1) If the test is successful, you can use the email safely.
2) If test failed, please check the above information is correct or network is proper.

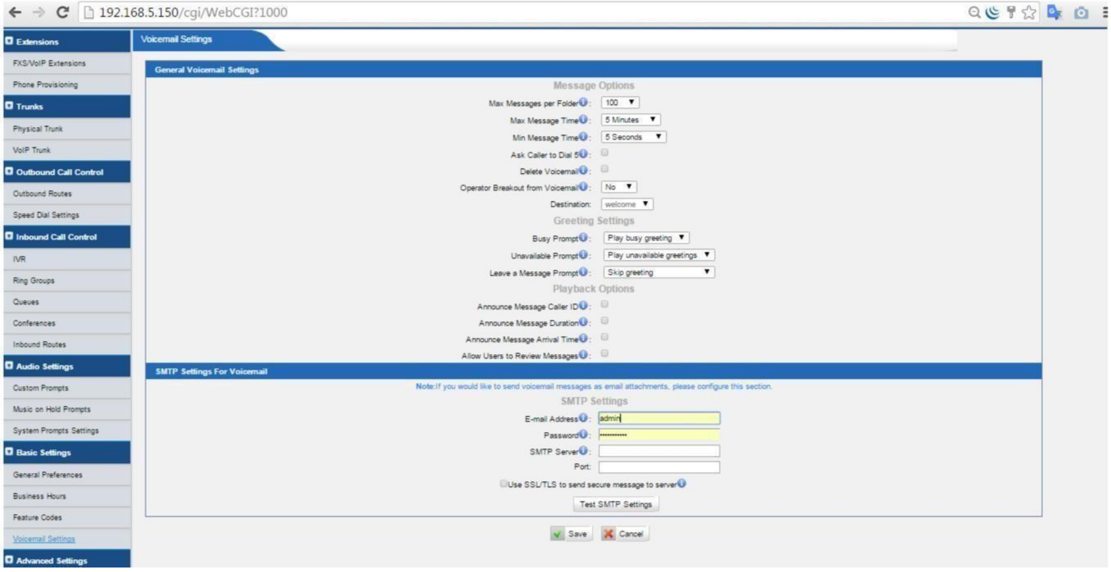


Figure 3.5.5

3.5.6 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an 'internal' system dial tone and make calls as if they were using one of the extensions attached to the telephone switch. To use DISA, a user calls a DISA number,which invokes the DISA application. The DISA application in turn requires the user to enter aPIN number, followed by the pound sign (#). If the PIN number is correct, the user will hear dialtone on which a call may be placed. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security.

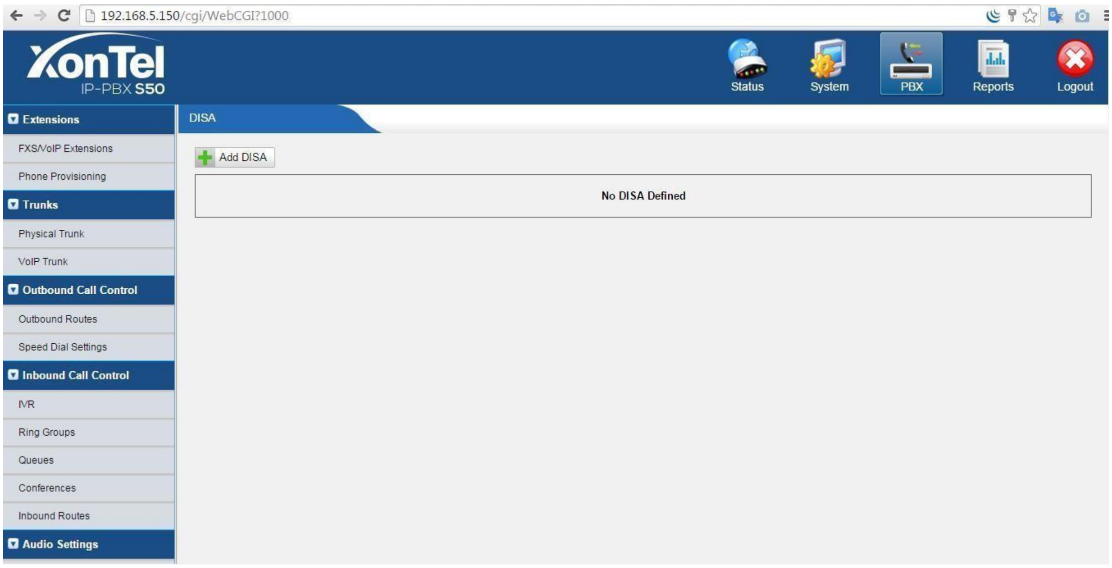


Figure 3.5.6.1

To add a new DISA application, click the New DISA button.

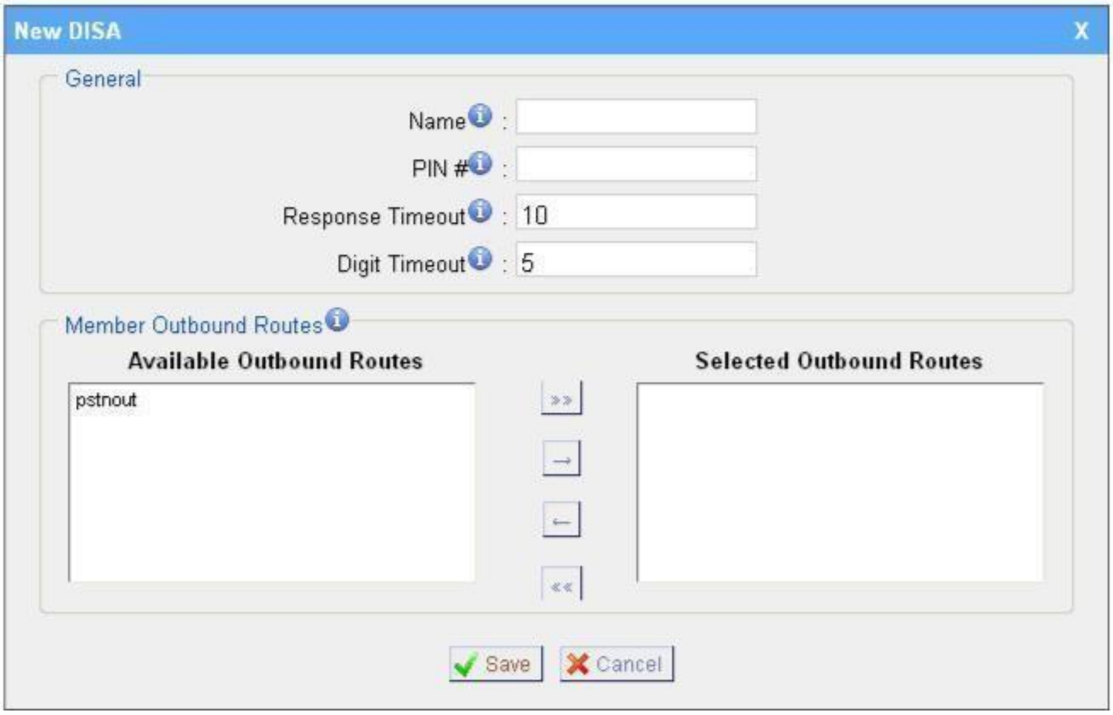


Figure 3.5.6.2

1) General

·DISA Name

Give this DISA application a name to help you identify it.

·PIN#

The password for this DISA.

·Response Timeout

The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. Default is 10 seconds.

·Digit Timeout

The maximum amount of time permitted between each digit when the user is dialing an extension number. Default is 5 seconds.

2)Member Outbound Routes

Used to set the outbound routes that can be accessed from this DISA.

Note: the maximum amount of DISA is 16.

3.5.7 Conferencing

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings.

·Extension

This is the number dialed to reach this Conference Room.

·Admin

Admin can kick a user out and can lock the conference room.

·Pin #

Set a PIN # that must be entered in order to access this conference room (i.e. 1234).

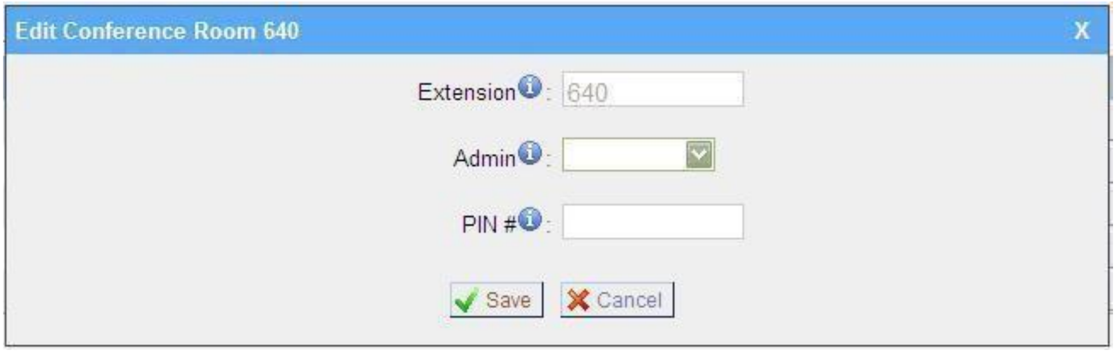


Figure 3.5.7

3.5.8 Paging Groups

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the Other Settings -> Feature Codes screen.
This feature is supported by the following SIP phones:
Yealink's T28,T26,T22,T20,T10T,T9CM. Other SIP devices may also work with this feature but are not officially supported.
Note:A paging group can have a maximum of 20 members.

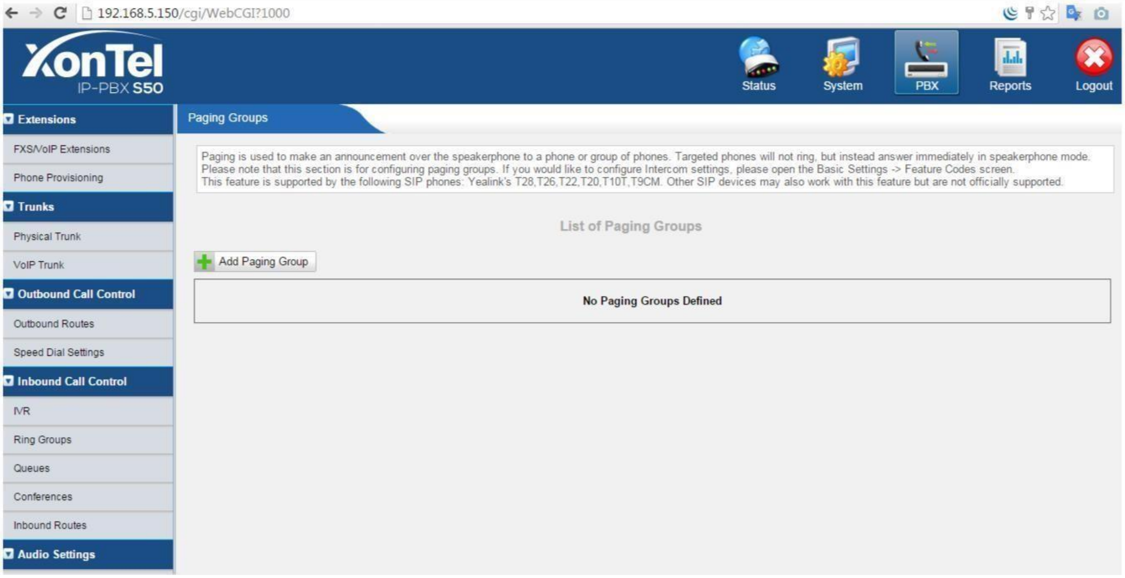


Figure 3.5.8.1

·Paging Group Number
Defines the numbered extension that may be dialed to reach this group.

·Duplex
Paging is typically one way for announcements only. Checking this will make paging duplex, allowing all users in the group to talk and be heard by all.

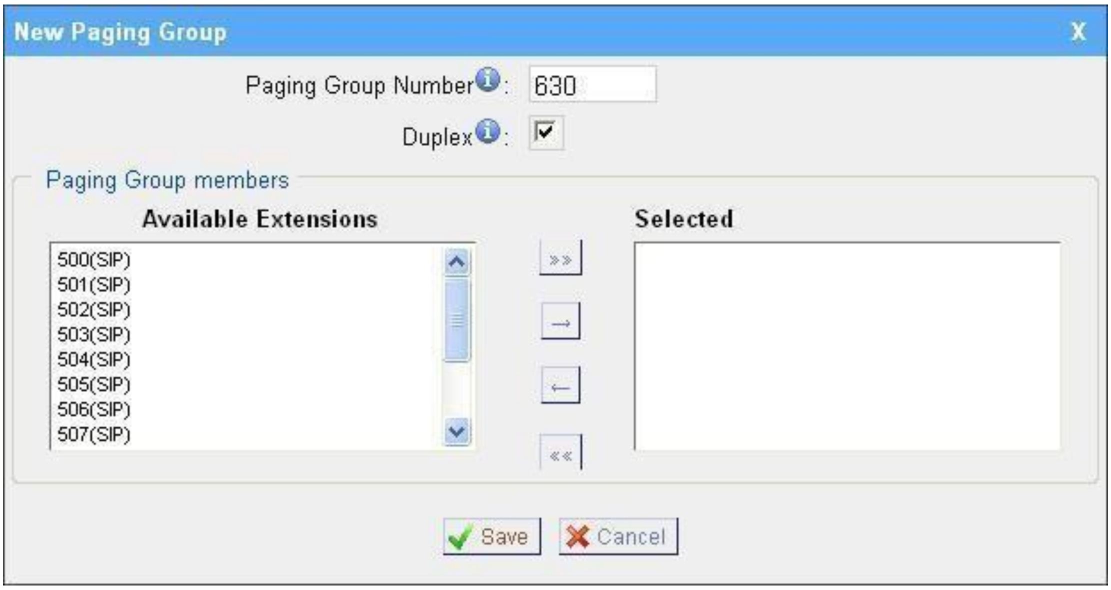
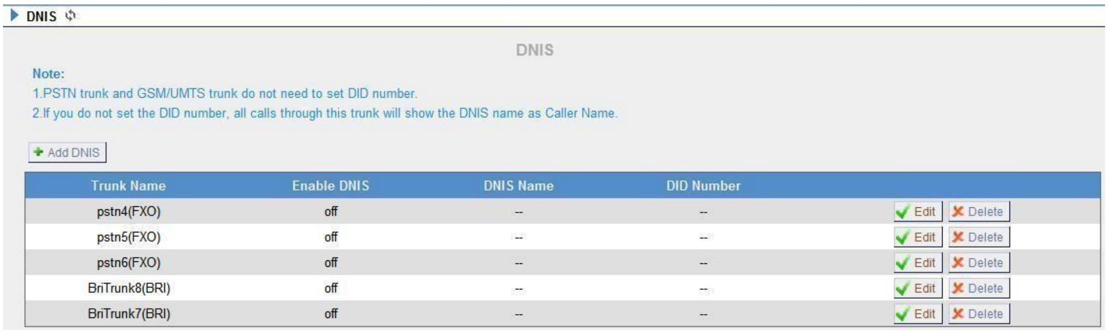


Figure 3.5.8.2

3.5.9 DNIS Settings

DNIS (Dialed Number Identification Service) is a telephone service that identifies for the receiver of a call the number that the caller dialed.

Note:
1. DID number is not available in PSTN/GSM/UMTS trunks
2. If DID is not configured here, all the calls via this trunk will show the DNIS instead of the original caller ID.



Trunk Name	Enable DNIS	DNIS Name	DID Number	
pstn4(FXO)	off	--	--	Edit Delete
pstn5(FXO)	off	--	--	Edit Delete
pstn6(FXO)	off	--	--	Edit Delete
BriTrunk8(BRI)	off	--	--	Edit Delete
BriTrunk7(BRI)	off	--	--	Edit Delete

Figure 3.5.9

3.5.10 PIN User settings

PIN User are used to manage lists of PINs that can be used to access restricted features such as Outbound Routes. If user use PIN User call out, system will auto record the PIN in the call detail records.

1) Options
·Access Code
.Dial this code to access PIN.

·Prompt for Entry
Prompt caller enter the PIN Number.
·Prompt for Entry Failure
Prompt the caller when an invalid PIN is entered.

2) PIN User
S50 can store a number of PIN Users. PIN Users may be used to keep track of calls in relation to particular activities or clients. They can also be used to keep track of calls by particular users or sets of users.
• When a PIN is entered while making a call or during a call, that PIN is included in the call records output by the system.
• PIN entered are checked against those stored by the system. If an invalid PIN is entered, the PIN is requested again.
• The system administrator can configure certain numbers or types of numbers to require entry of a PIN before you can continue making a call to such a number.
• The system administrator can also configure you to have to enter a PIN before making any external call.

·Name

A character-based name for this PIN list, i.e. 'XontelPIN'.

·PIN List

Enter a list of one or more PINs, One PIN per line.

·Available Outbound Routes

The outbound routes which can be used by the pin user.

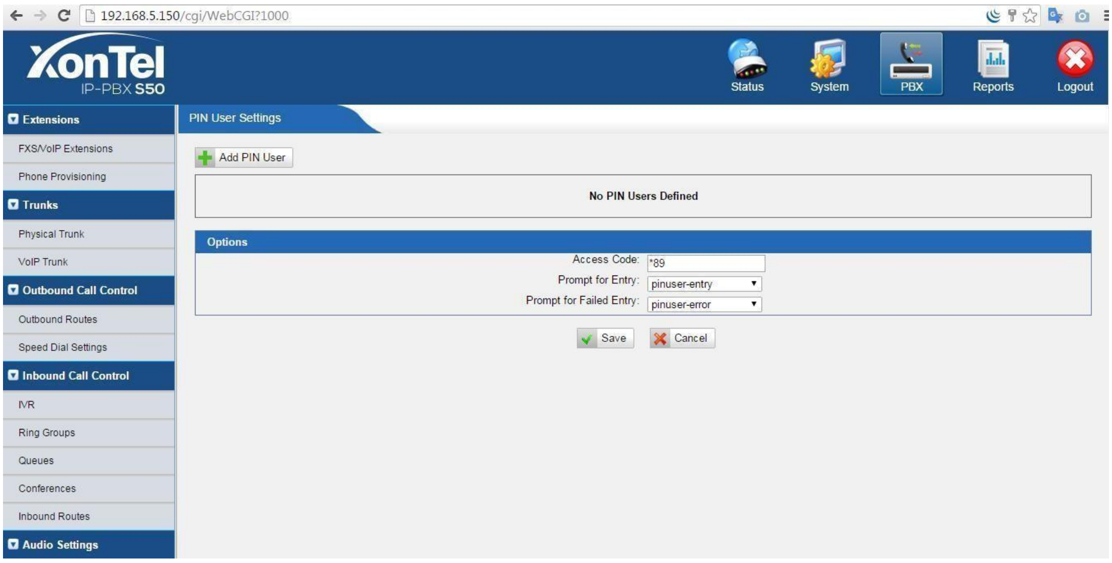


Figure 3.5.10.

3.5.11 Speed Dial Settings

1) Options

·The prefix of speed dial

The prefix should be dialed before the speed dial number. Default is *99.

2) Add new speed dial.

·Source Number

The speed dial number.

·Destination Number

The number you want to call.

e.g. The source number is"123". The destination number is 5503305.The prefix number is*99.You can use an extension with any type to dial *99123, then it will call to number 5503305.

Note: Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

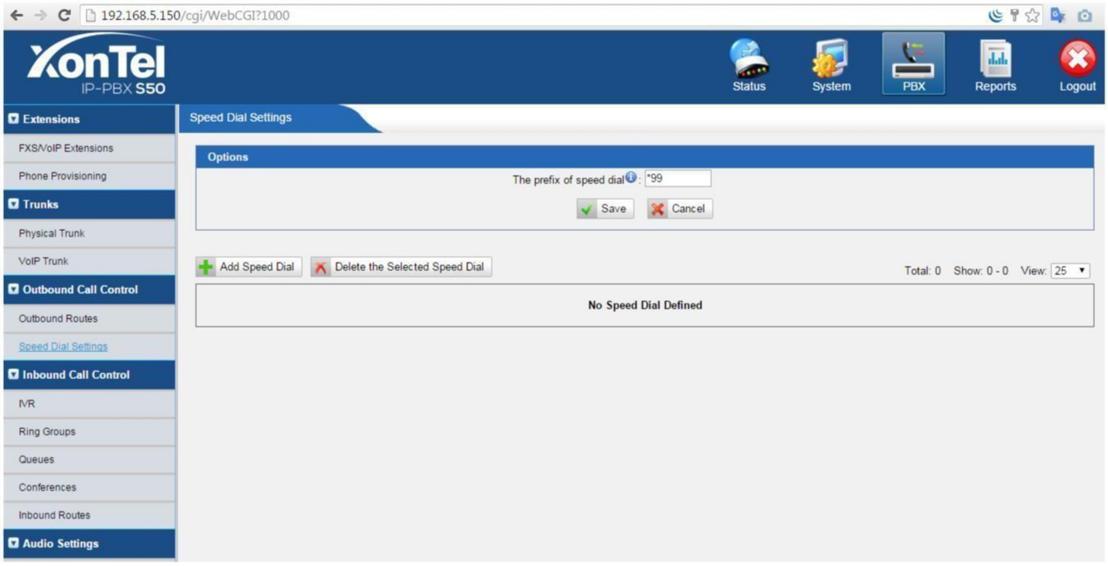


Figure 3.5.11

3.5.12 Music on Hold Prompts

The administrator can upload on hold music as follows:

1) Click 'Upload Music on Prompt '.

2) Click 'Browse' to choose the desired audio file.

3) Click 'Upload' to upload the selected file.

Note: The sound file format should be as follows:

GSM 6.10 8kHz, Mono, 1Kb/s;

Alaw/Ulaw 8kHz, Mono, 1Kb/s;

PCM 8kHz, Mono, 16Kb/s.

Figure 3.5.12

3.6 Network Settings

3.6.1 LAN Settings

·DHCP

If this option is set, S50 will use DHCP to get an available IP address from your local network. Not recommended.

·Enable SSH

This is the advance way to access the device, you can use the putty software to access the device.In the SSH access, you can do more advance setting and debug.

·**Port**: the default is 8022,

·Hostname

Set the host name for S50.

·IP Address

Set the IP Address for S50.

·Subnet Mask

Set the subnet mask for S50.

·Gateway

Set the gateway for S50.

·Primary DNS

Set the primary DNS for S50.

·Secondary DNS

Set the secondary DNS for S50.

·IP Address2

Set the second IP Address for S50.

·Subnet Mask2

Set the second subnet mask for S50.

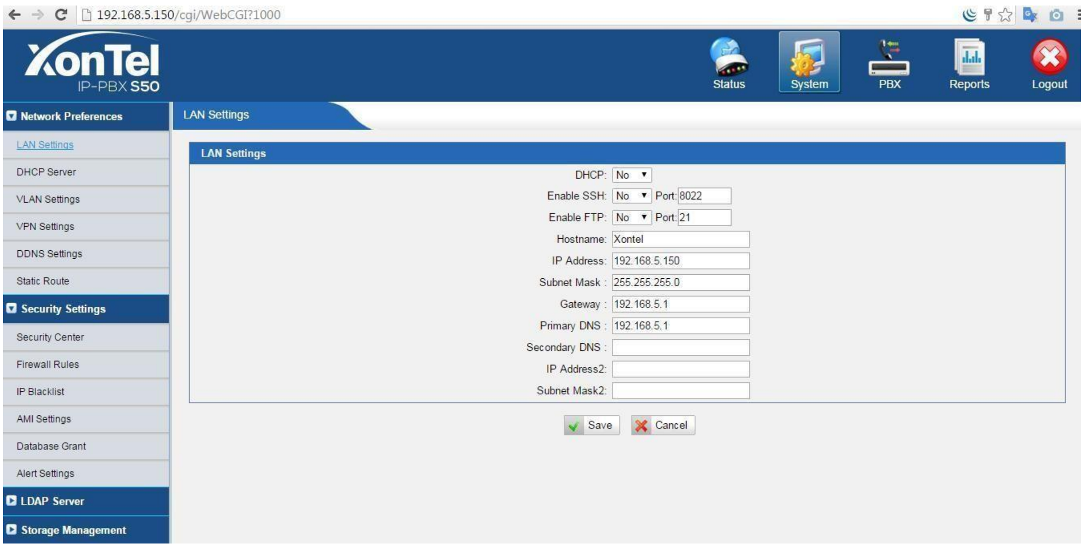


Figure 3.6.1

3.6.2 Static Route

S50 will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for S50 to force it goes out through different gateway when access to different internet. The default gateway priority of S50 from high to low is OpenVPN → WAN port → LAN port.

1) Route table

The current route rules of S50

·Destination

The destination network to be accessed to by S50

·Subnet Mask

Specify the destination network portion.

·Gateway

Which gateway S50 will go through when access to the destination network.

·Metric

The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.

·Interface

Define which internet port to go through.

2) Static Route Rules

You can add new static route rules here.

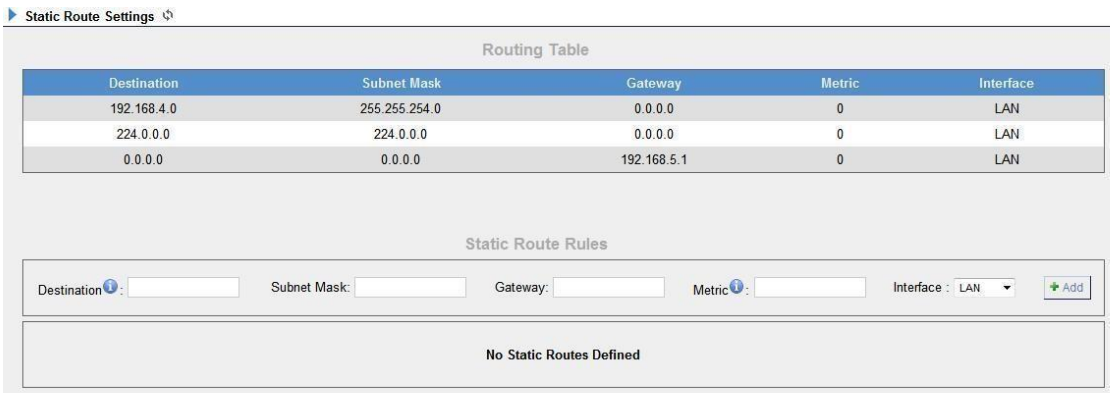


Figure 3.6.2

3.6.3 Firewall

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.
In order to ensure the safe operation of the system, we suggest to enable and configure firewall.

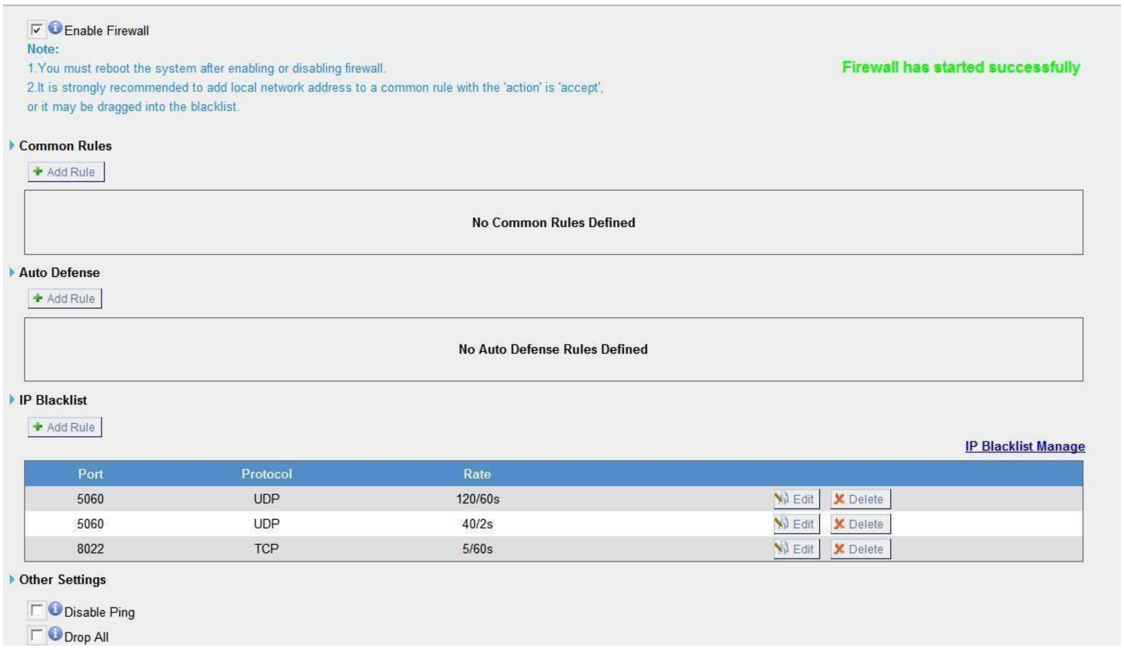


Figure 3.6.3

1) Enable Firewall
Enable the firewall to protect the device. You should reboot the device to let the firewall run successfully.

2) Common Rules

·Name

A name for this rule , e.g. 'HTTP'.

·Description

Simple description for this rule. E.g.: Accept the specific host to access the web interface for configuration.

·Protocol

The protocols for this rule.

·Port

Initial port should be on the left and end port should be on the right. The end port must be equal to or greater than start port.

·IP

The IP address for this rule. The format of IP address is: IP/mask
Ex:192.168.5.100/255.255.255.255forIP 192.168.5.100
Ex:216.207.245.47/255.255.255.255 for IP 216.207.245.47
Ex:192.168.5.0/255.255.255.0 for IP from 192.168.5.0 to 192.168.5.255 .

·MAC Address

The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.

·Action

Accept: Accept the access from remote hosts.
Drop: Drop the access from remote hosts.
Ignore: Ignore the access.

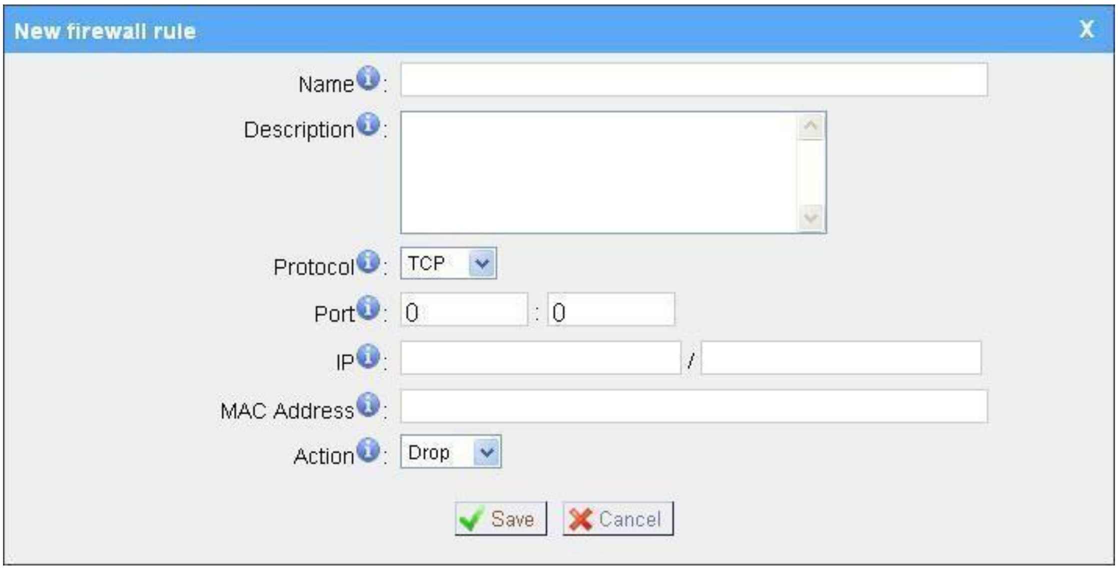


Figure 3.6.3.1

3) Auto Defense

·Port

Auto defense port, e.g.: 8022.

·Protocol

Auto defense protocol, TCP or UDP.

·Rate

The maximum packets or connections can be handled per unit time.
E.g.: (Port: 8022 Protocol: TCP Rate: 10/minute) means maximum 10 TCP connection to port 8022 can be handled per minute, the eleventh connection will

be refused directly.

Figure 3.6.3.2

4) IP Blacklist

You can set some packets accept speed rules here. When a IP address which hasn't been accepted in common rules sends packets faster than the allowed speed, it will be set as black IP address and blocked automatically.

a) New Rule

·**Port**

Auto defense port

·**Protocol**

Auto defense protocol. TCP or UDP.

·**IP Packets**

Allowed IP packets number in the specific time interval.

·**Time interval**

The time interval to receive IP packets. For example, IP packets 90,time interval 60 means 90 IP packets are allowed in 60 seconds.

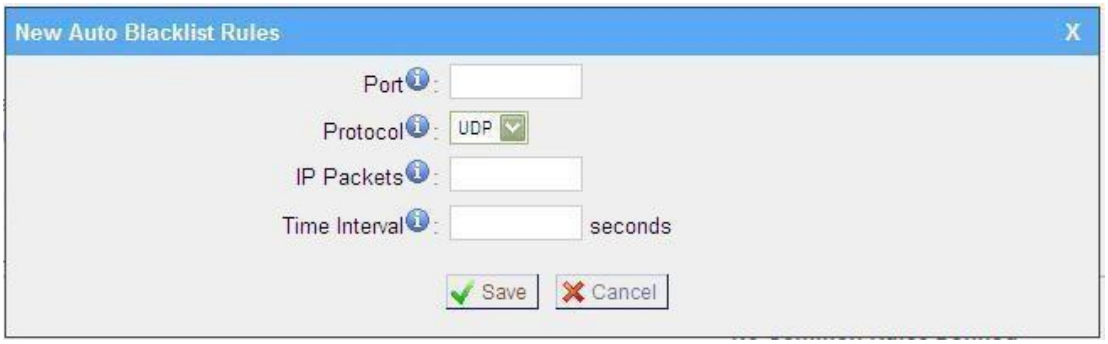


Figure 3.6.3.3

b) IP Blacklist Manage

You can manage the IP addresses which are blocked automatically here.

Figure 3.6.3.4

5) Other Settings

·Disable Ping

Enable this item, net ping from remote hosts will be dropped.

·Drop All

When you enable 'Drop All' feature, system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one 'TCP' accept common rule must be created for port used for SSH access, port used for HTTP access and port sued for CGI access

3.6.4 DHCP Server

Dynamic Host Configuration Protocol (DHCP) is a network protocol that enables a server to automatically assign an IP address to a computer from a defined range of numbers (i.e., a scope) configured for a given network. You can set a local network NTP server for S50 here too.

Note: When using 'Phone Provisioning' for Grandstream IP phone, Enter the IP address of the server directly, e.g.:192.168.5.150; for other phones using the default configuration.

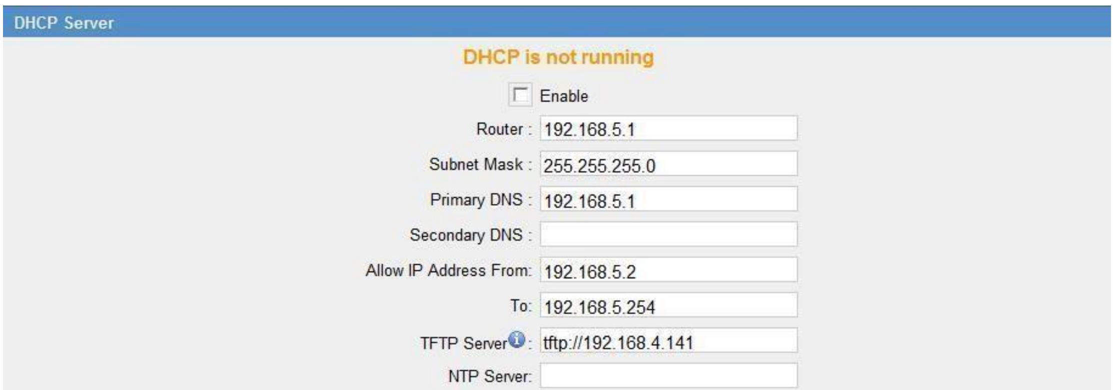


Figure 3.6.4

3.6.5 VLAN Settings

A VLAN(Virtual LAN) is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

1) VLAN Over Lan

·NO.1

Click the NO.1 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for S50 VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for S50 VLAN over Lan.

·Default Gateway

Set the Default Gateway for S50 VLAN over Lan

·NO.2

Click the NO.2 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for S50VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for S50 VLAN over Lan.

·Default Gateway

Set the Default Gateway for S50 VLAN over Lan.

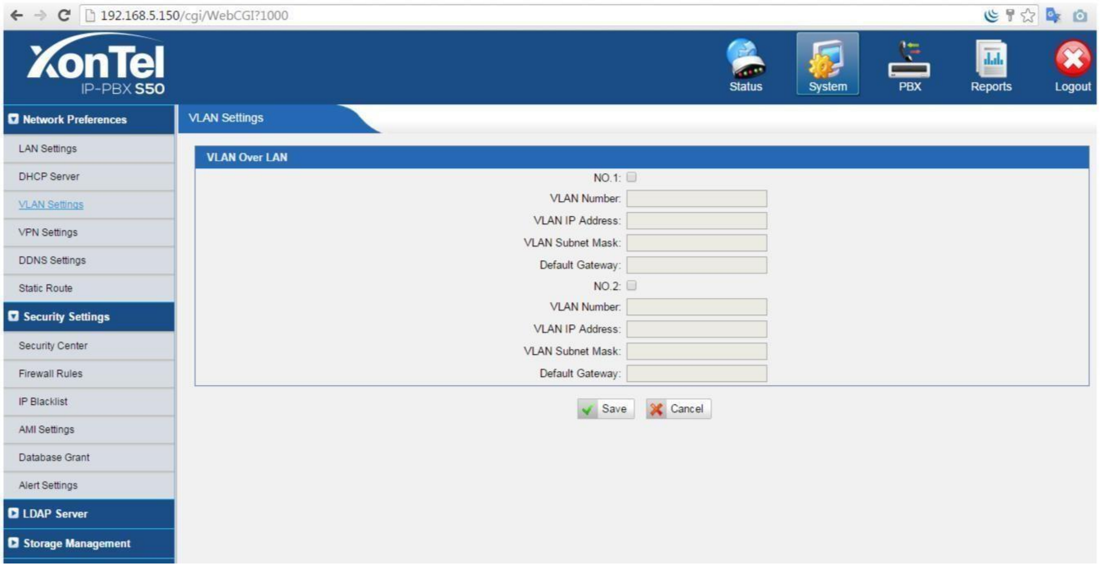


Figure 3.6.5

3.6.6 DDNS Settings

DDNS(Dynamic DNS) is a method / protocol / network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.

·Enable DDNS

·DDNS Server

Select the DDNS server you sign up for service.

·User Name

User name the DDNS server provides you.

·Password

User account's password.

·Host Name

Note: DDNS allows you to access your network using domain names instead of IP address. The service manages changing IP address and updates your domain information dynamically. You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com

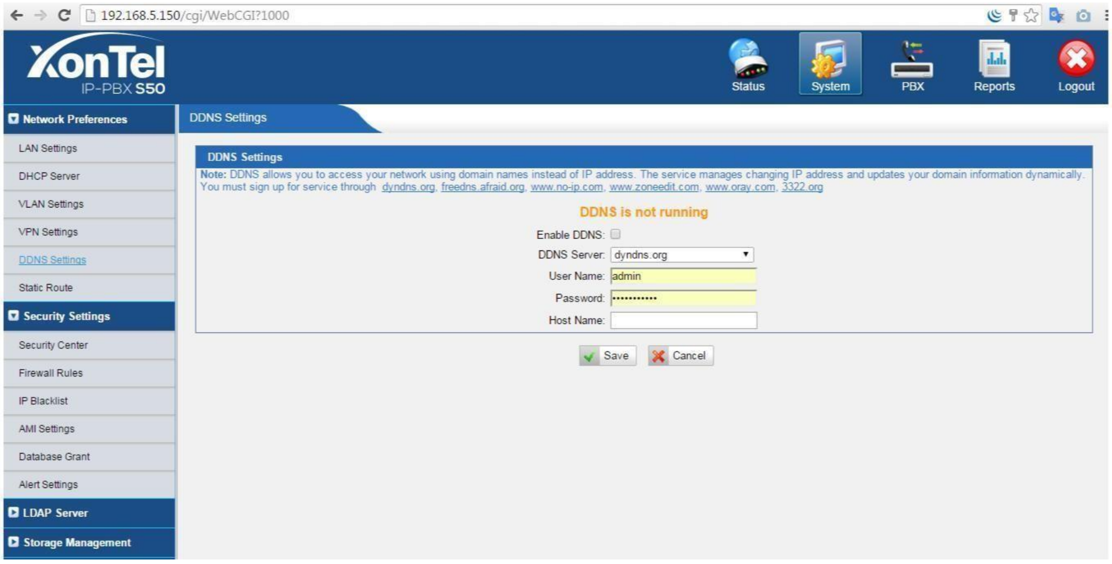


Figure 3.6.6

3.7 System Settings

3.7.1 External Storage

The External Storage feature is used to extend storage space. Once configured, the files (voicemail, call recording files, CDR files) created before the configured days will be moved to the Net-Disk.

Auto-Backup extends the allocated disk space for backing up critical files. When properly configured, S50 will move all qualified files to a Windows PC every 30 minutes. For the voicemail files and recoding files, they must be created before Auto-Backup has been configured.

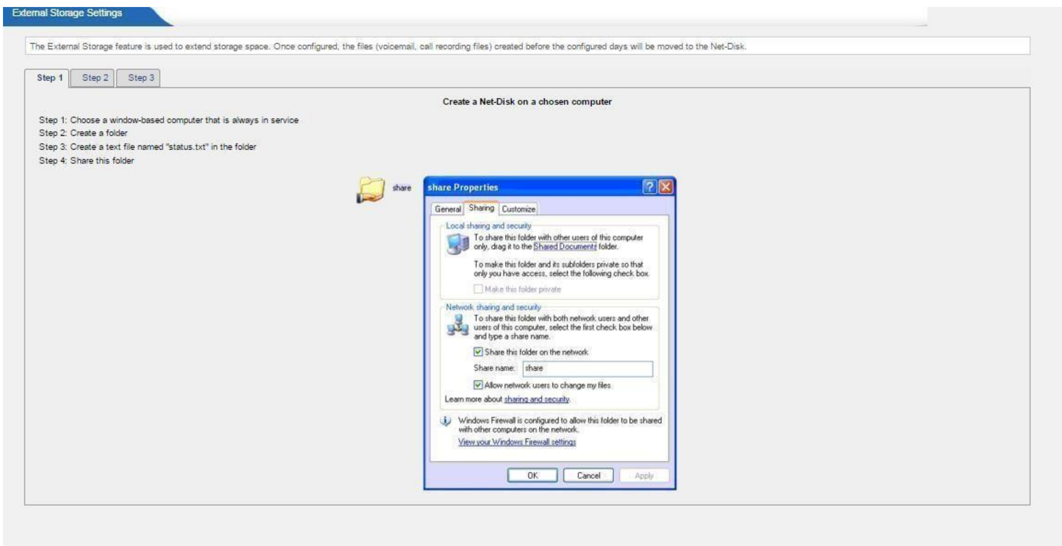


Figure 3.7.1

3.7.2 Password Settings

1) Change Password

·User
S50 support 3 levels users, choose one of them to change its password.

·Enter Old Password
The default password is '**xontel**'.

·Enter New Password

·Retype New Password
To change the password, enter the new password and click update.
The system will then prompt you re-login using your new password.

2) CDR setting
Whether enable CDR User.

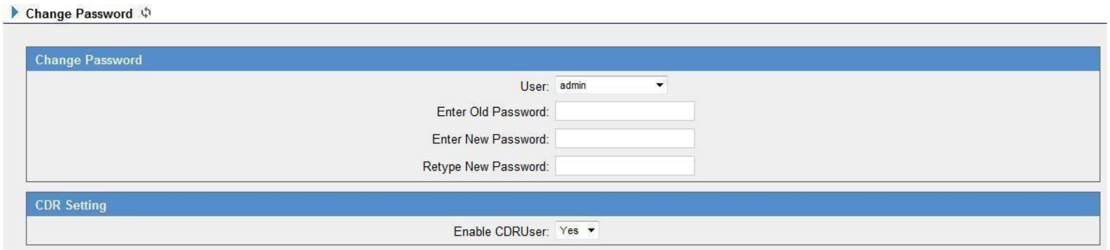


Figure 3.7.2

3.7.3 System Prompts

S50 have prompts of many languages. You can download the appropriate language you need.

Note:
Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server, please contact the local dealer for the prompts.

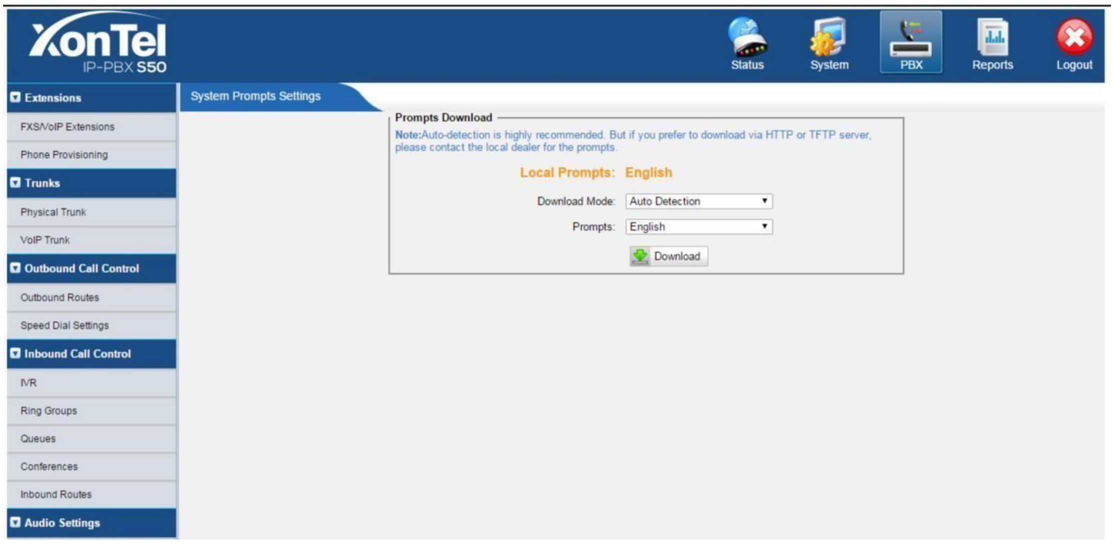


Figure 3.7.3

3.7.4 Date and Time

Set the date and time for S50.

·Time Zone

You can choose your time zone here.

·Daylight Saving Time

Set the mode to Automatic or disabled

·Automatically Synchronize With an Internet Time Server

Input the NTP server so that S50 will update the time automatically.

·Set Date & Time Manually

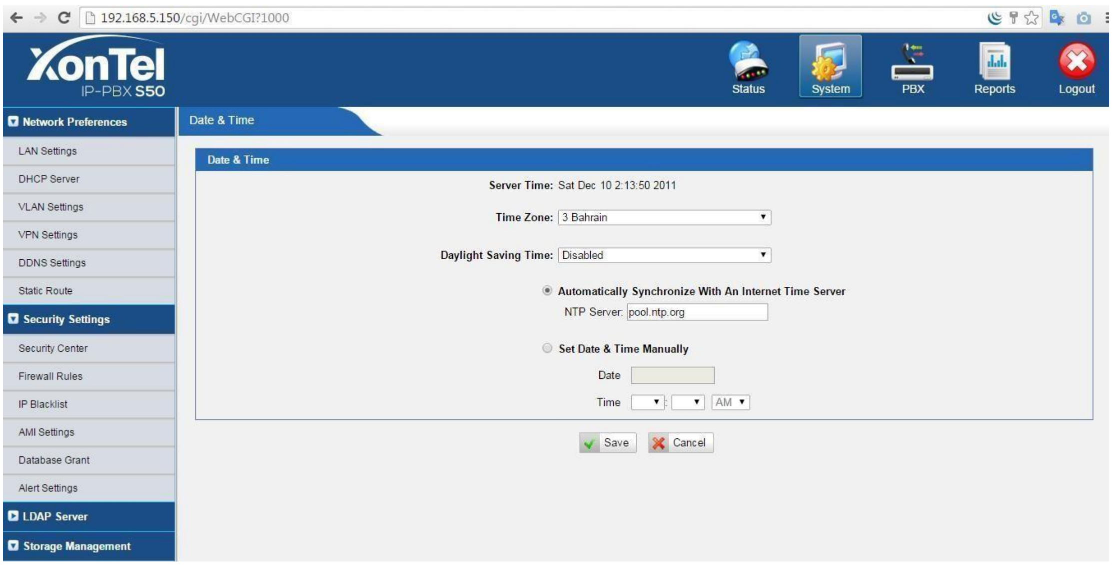


Figure 3.7.4

3.7.5 Backup and Restore

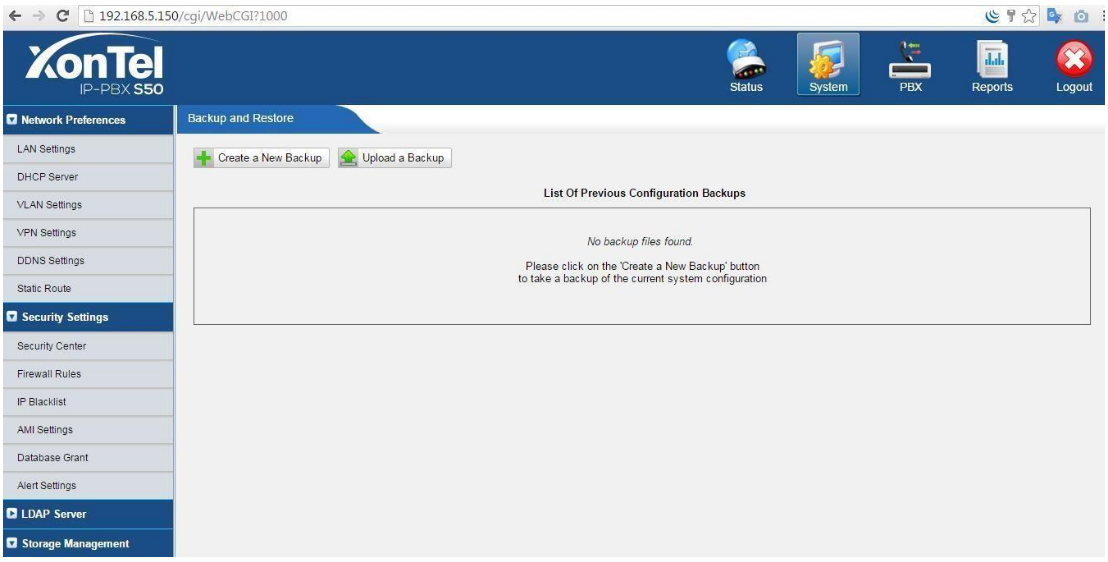


Figure 3.7.5

3.7.6 Reset and Reboot

·Reboot System
Warning: Rebooting the system will terminate all active calls!

·Reset to Factory Defaults
Warning: A factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.

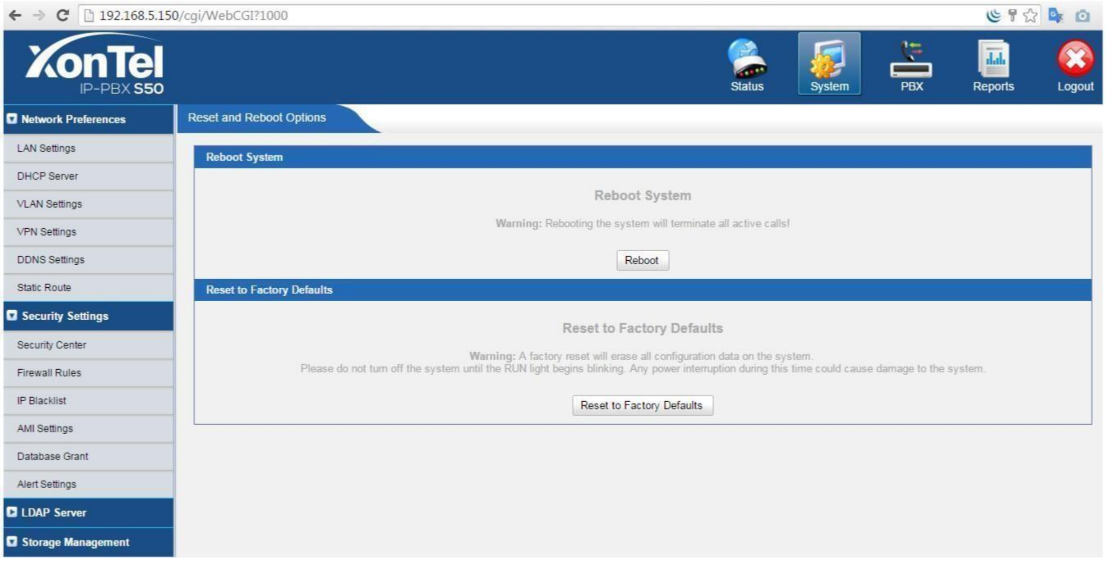


Figure 3.7.6

3.7.7 Firmware Update

Upgrading of the firmware is possible through the Administrator web interface using a TFTP Server or an HTTP URL.
Enter your TFTP Server IP address and firmware file location, then click start to update the firmware.

Note:

1. If enabled 'Reset configuration to Factory Defaults', System will restore to factory default settings.
2. When update the firmware, please don't turn off the power.

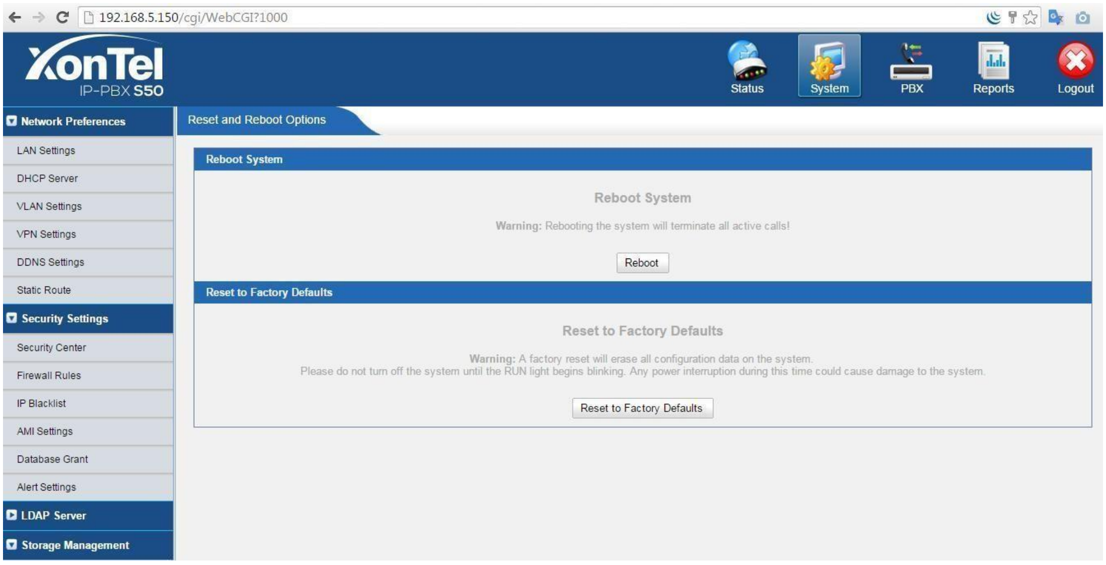


Figure 3.7.7

3.7.8 Alert Settings

If the device is attacked, the system will notify users the alert via call or E-mail. The attack mode include IP attack and Web Login.

Alert Settings

Attack Type	Phone Notification	E-mail Notification	
IPATTACK	Yes	Yes	
WEBLOGIN	Yes	Yes	

Figure 3.7.8.1

1. IPATTACK

When the system is attacked by IP address, the firewall will add the IP to auto IP Blacklist and notify the user if it match the protection rule.

1）Phone Notification Settings

·PHONE Notification

Whether enable phone notification.

·Number

The numbers could be set for alert notification, users can setup multiple extension and outbound phone numbers. Please separate them by ';'.
e.g. '500;9911', if the extension has configured Follow Me Settings, the call would go to the forwarded number directly.

·Attmtpts

The attempts to dial a phone number when there is no answer.

·Interval

The interval between each attempt to dial the phone number. Must be greater than 3 seconds, the default value is 10 seconds.

·Prompt

Users will hear the prompt while receiving the phone notification.

2）E-mail Notification Settings

Note: Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

·E-mail Notification
Whether enable E-mail Notification

·Recipient' Name
The recipients for the alert notification, and multiple email addresses are allowed, please separate them by ';'.
e.g. 123@163.com; basel@xontel.com .

·Subject
The subject of the alert email.

·Email Content
Text content support for predefined variables. Variable names and corresponding instructions are as follows:

$(HOSTNAME) Host name
$(LOCALIP) Local IP address
$(SOURCEIP) Attack source IP address
$(DATETIME) Occurred
$(USERNAME) User name（WEBLOGIN effective）
$(DESTMAC) Attacks destination MAC（IPATTACK effective）
$(DESTPORT) Attacks destination Port number（IPATTACK effective）
$(PROTOCOL) Protocol type（IPATTACK effective）
$(INTERFACE) Network interface name（IPATTACK effective）

2. WEBLOGIN
Web Login Alert Notification ：Enter the password incorrectly five times to login S50 Web interface will be as attack, the system will limit the IP login within 10 minutes and notify the user.

3.8 Reports

3.8.1 Call Logs

The call Log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by filter the call logs by call date, caller/callee, trunk, duration, billing duration, status, communication type and pin user.

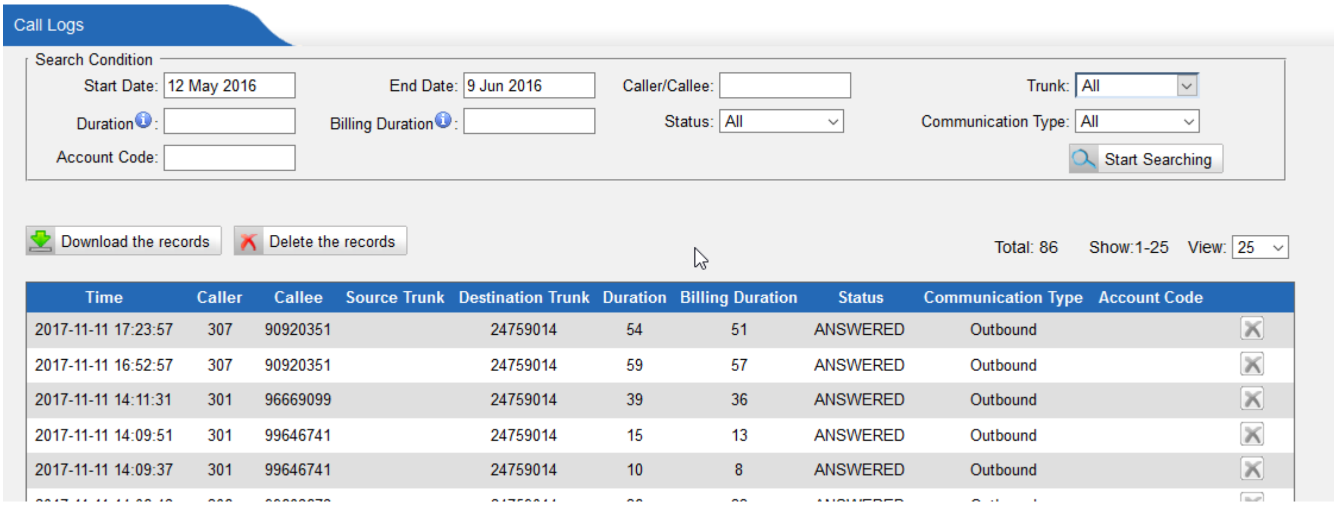


Figure 3.8.1

3.8.2 System Logs

You can download and delete the system logs of S50.

·Enable Hardware Log
Save the information of hardware; (up to 4 log files)

·Enable Web Log
Save the history of web operations (up to 2 log files)

·Enable Debug Log
Save debug information (up to 2 log files)

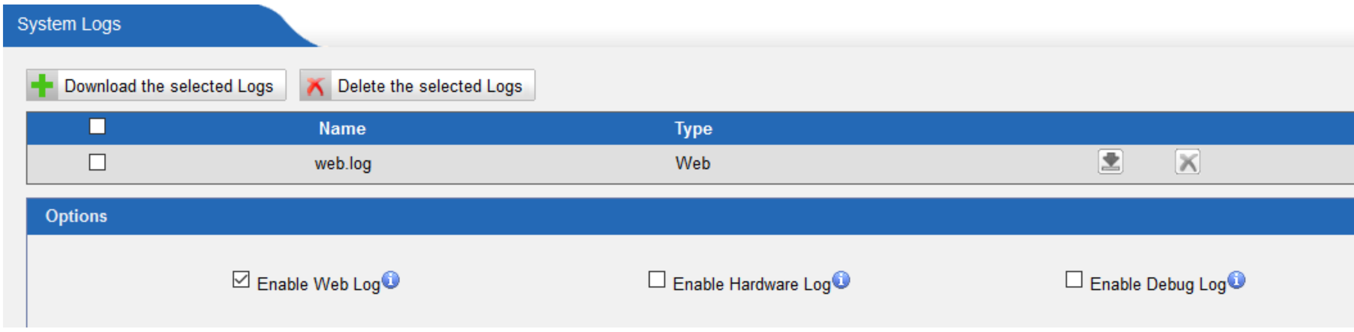


Figure 3.8.2

3.8.3 System Info

General:
Information about hardware version, firmware version and system uptime.

LAN:
Information about hostname, MAC address, IP address, gateway, Primary DNS and Secondary DNS.

Disk Usage:
Disk usage information.

Memory Usage:
Memory usage information.

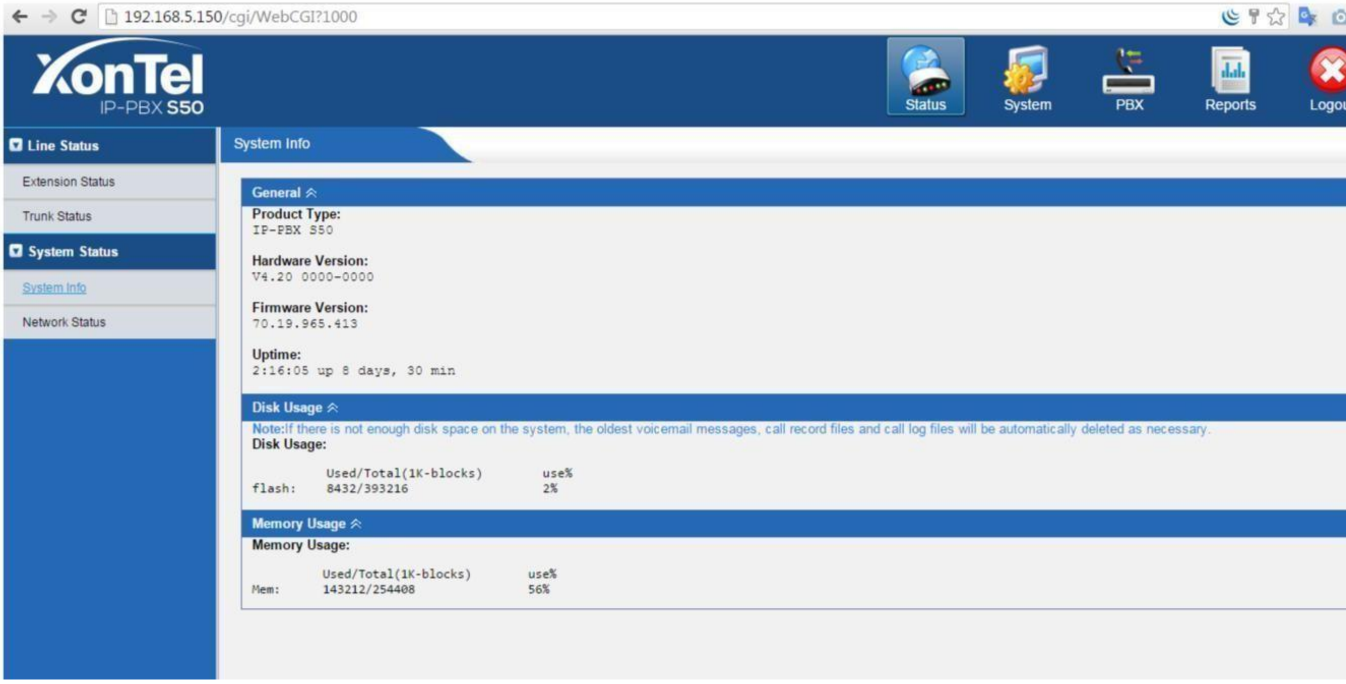


Figure 3.8.3

4. Access MRI

MRI (S50 Recording Interface).Users may access MRI by logging into the MRI web interface with their username (extension number) and voicemail password.

4.1 Allow users to access MRI

The extension's 'User Web Interface' option must be checked before the associated user can log into MRI.

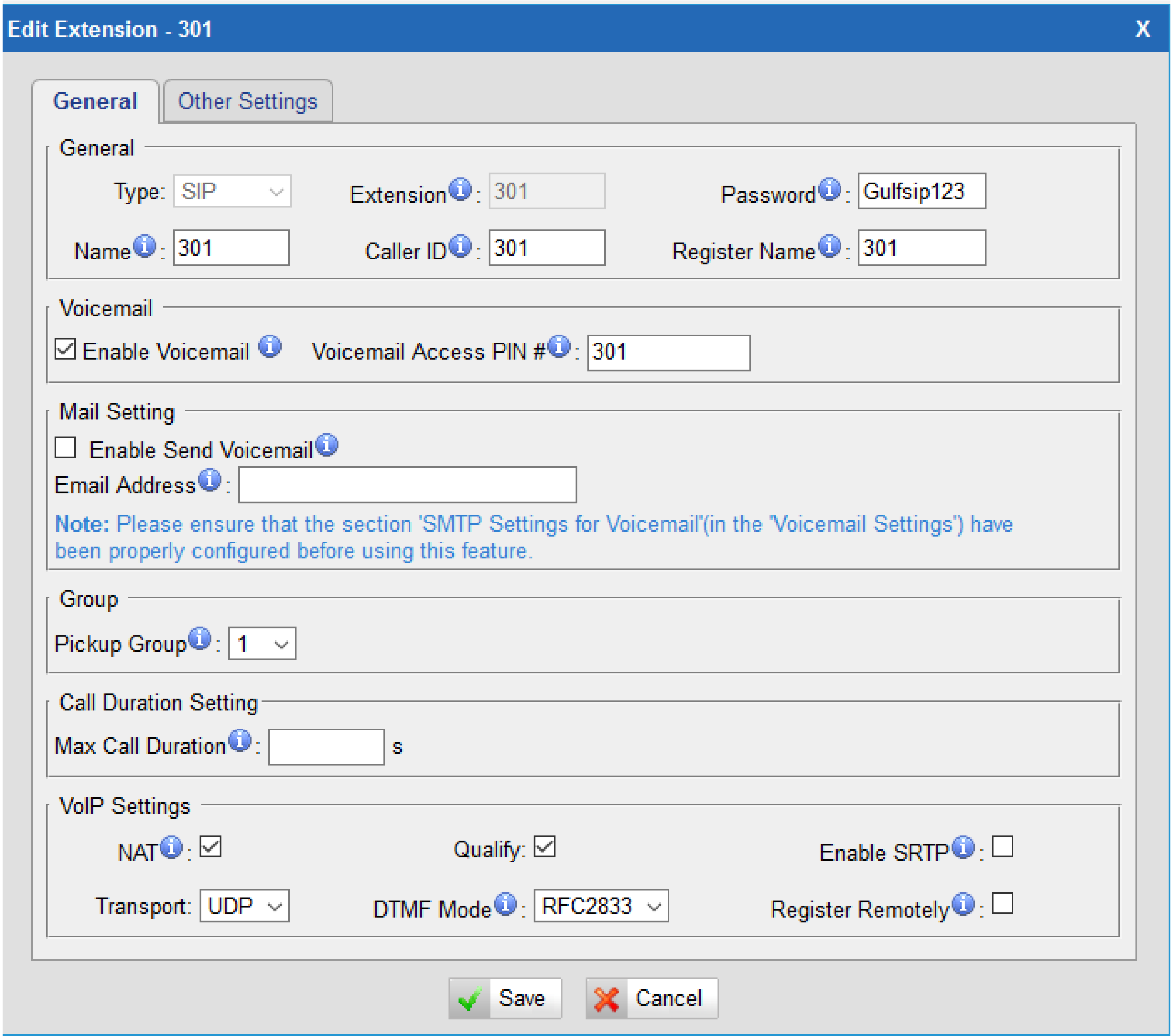


Figure 4-1

4.2 User login

Users can access the MRI web interface by navigating to the S50 IP address using a web browser. If you are unsure of this address, please contact your network administrator.

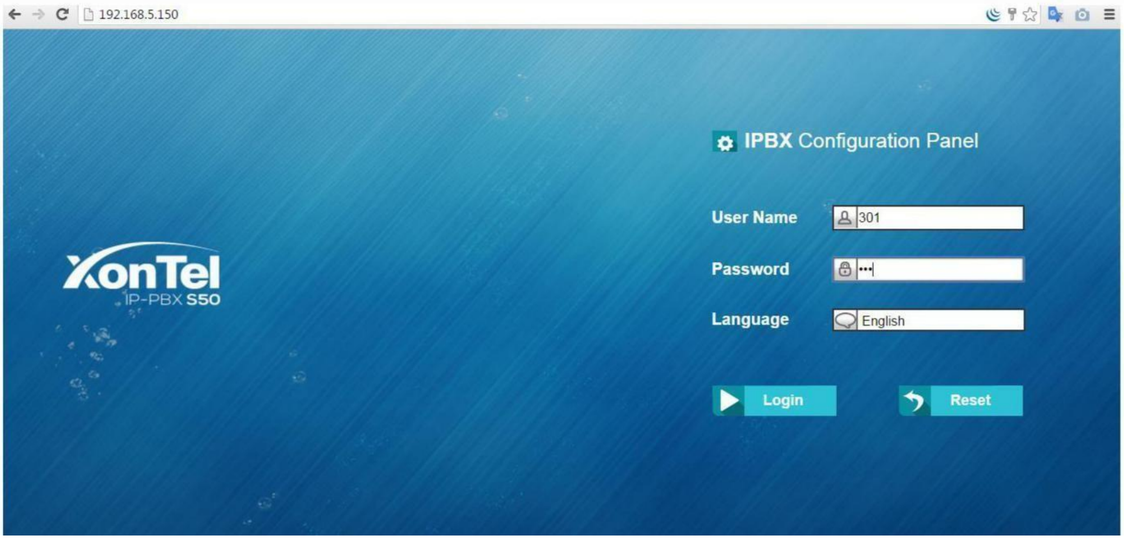


Figure 4-2

4.3 Voicemail

Users can check, delete, move and download voicemail files here.

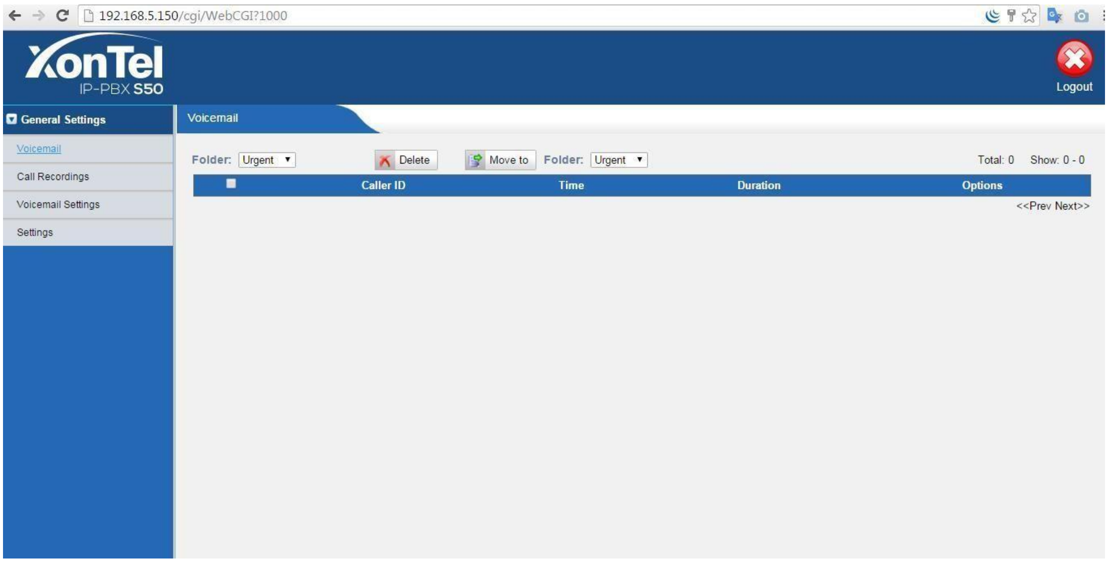


Figure 4-3

4.4 Record

Users can play, delete and download recorded files here.

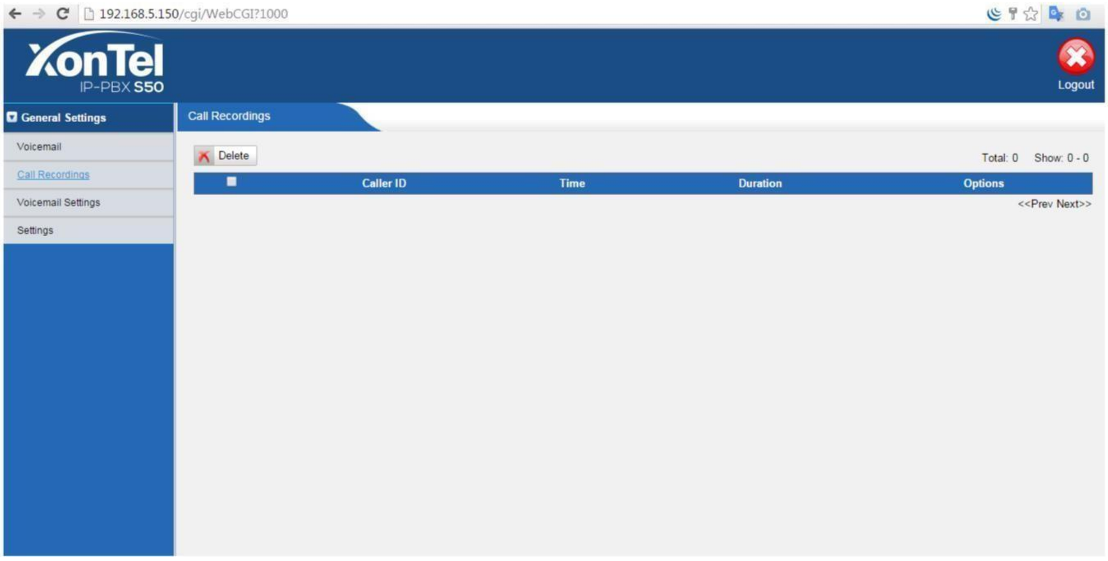


Figure 4-4

4.5 Voicemail Settings

·**Voicemail password**: new voicemail box password.
·**Enter again to confirm**: confirm new voicemail box password.
·**Email Address**: Email address use to receive the voicemail or Fax.
Note: Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.
·**Enable Voicemail**
Check this box if the user should have a voicemail account.
·**Enable Send Voicemail**
Once enabled, the voicemail or Faxes will be sent to email as an attachment.

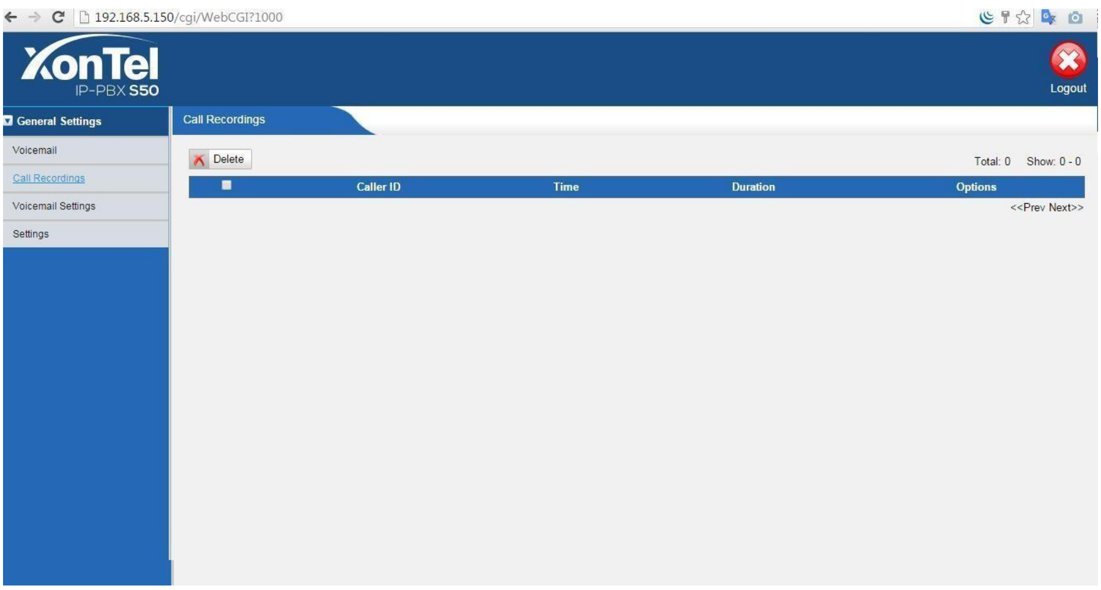


Figure 4-5

4.6 Settings

You can do some basic setting here. Such as call forwarding, DND, Mobile Extension Number.

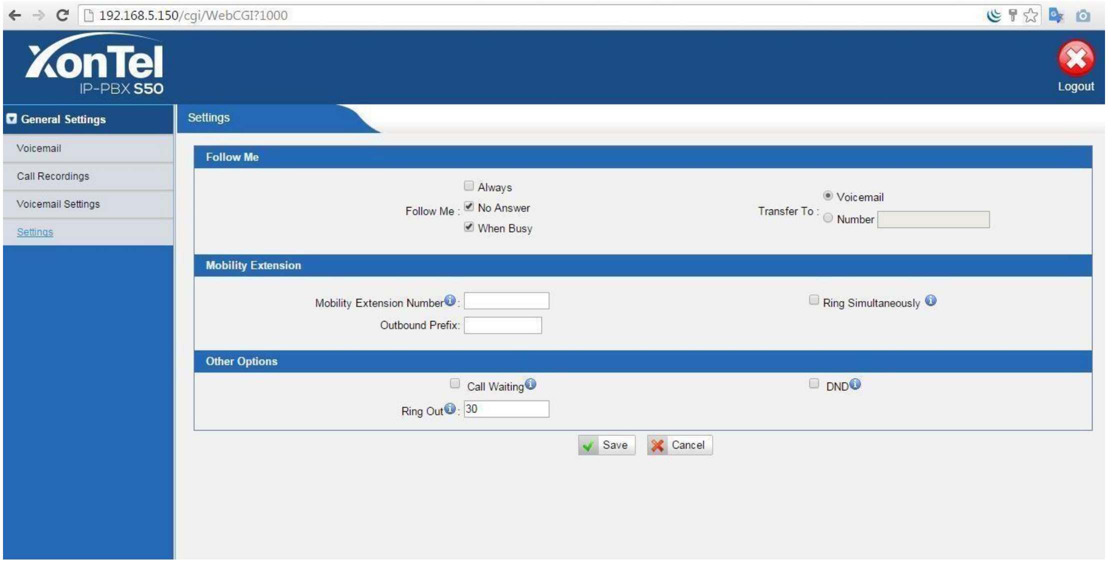


Figure 4-6

5. Access CDR

CDR (Call Detail Record).Users may access CDR to research call logs.

Username: cdr
Password: xontel

5.1 Call Logs

The call Log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by filter the call logs by call date, caller/callee, trunk, duration, billing duration, status, communication type.

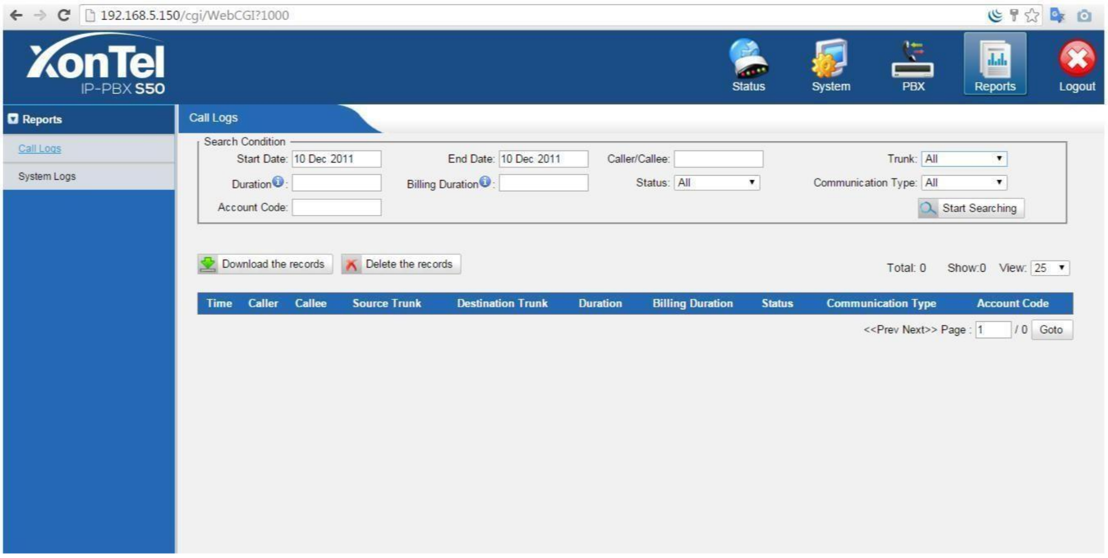


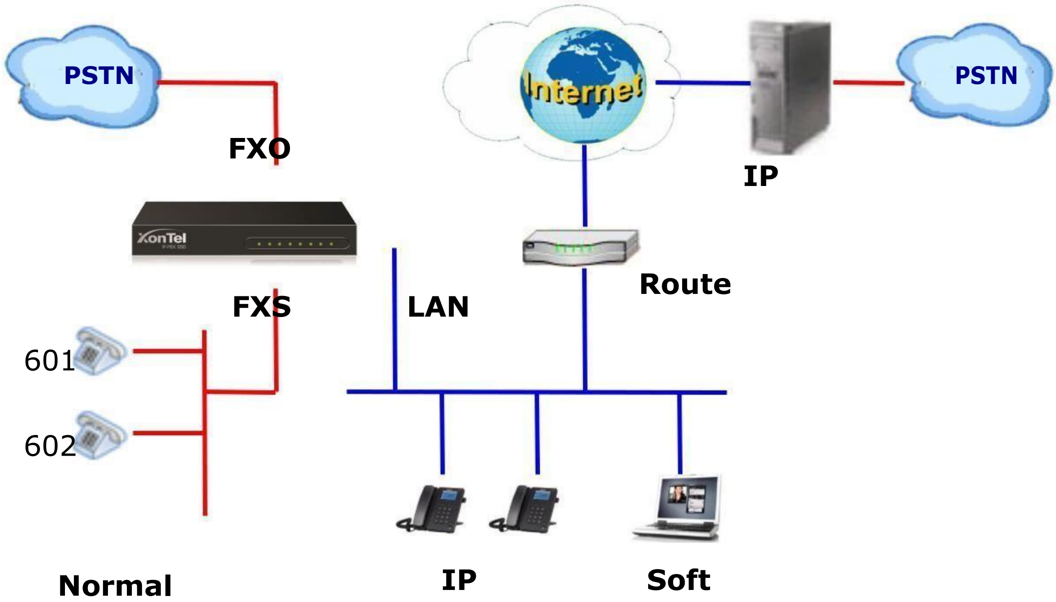
Figure 5-1

5.2 Password Settings

Change the password for cdr user.

Figure 5-2

6. Use S50



6.1 Make outbound call

To make an outbound call, we need to add trunk first. There are five types of VoIP Trunk:

·**Analog Trunk:** FXO ports of S50, connected to a local PSTN.
·**BRI Trunk:** BRI ports of S50, connected to a local PSTN.
·**VoIP Trunk:** Connected to remote VOIP service server.
·**Service Provider:** Connected to service provider server.

What are FXO and FXS?

FXS (Foreign exchange Station) is an interface which drives an analog telephone or FAX machine. FXS interfaces deliver power, provide ringing, and use FXO signaling. FXS interfaces are what allow you to hook telephones and other analog devices to your PBX

FXO (Foreign exchange Office) is an interface that connects to a phone line to supply your PBX with access to a public telephone network. FXO interfaces use FXS signaling.FXO interfaces allow you to connect your PBX to real analog phone lines.

6.1.1 Sample Routing via PSTN Trunk

Let's route all inside extensions through an analog trunk by dialing 9. In Outbound Routes, add a new outbound route as below.

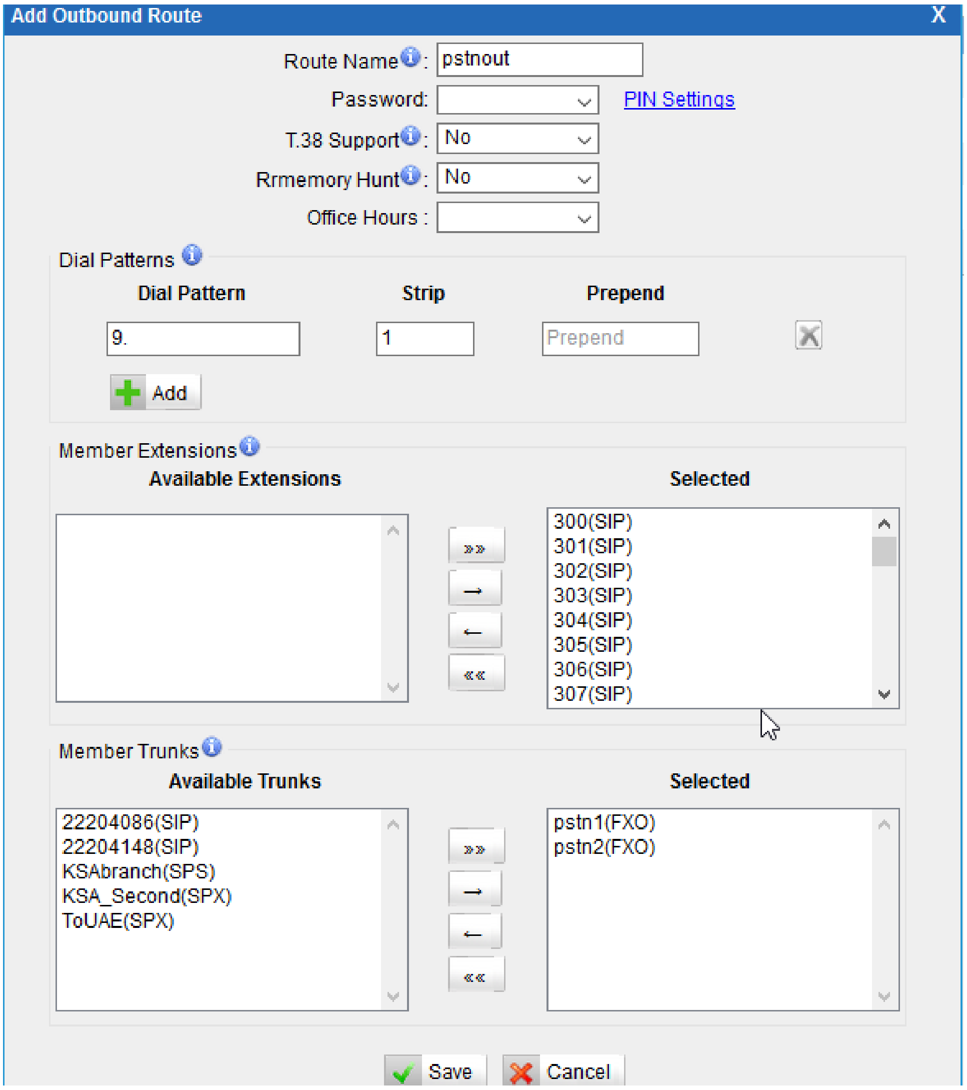


Figure6-1

As we can see from the outbound route of 'pstnout', all phone numbers starting with 9 will have their first digit stripped off (digit 9) and will be sent to the PSTN (port 1-2).

After we have configured the above, we can dial 9 + local number to dial out via a PSTN line.

Note: Setting number prefix to wild card X. and setting Strip to 0 digits from the front will allow all calls to go through this outbound route.

6.1.2 Sample Routing via VoIP Trunk

Let's configure all inside extensions to dial '0' through the VoIP Trunk.

1. Add VoIP service provider

Trunks→VoIP Trunk →SIP Trunk
Enter your account information on this page, and click Save.

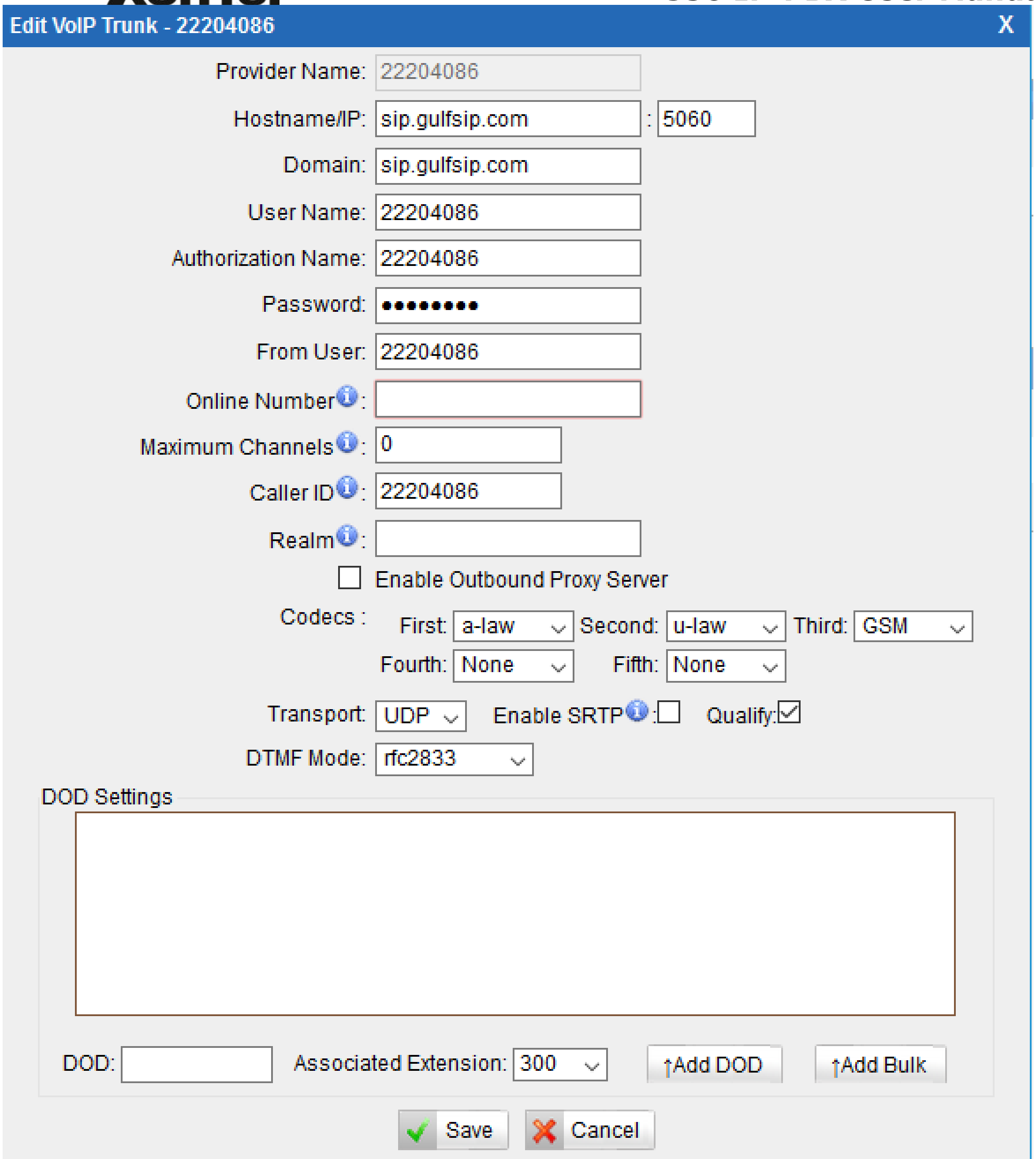

Edit VoIP Trunk - 22204086
X
Provider Name: 22204086
Hostname/IP: sip.gulfsip.com : 5060
Domain: sip.gulfsip.com
User Name: 22204086
Authorization Name: 22204086
Password: ●●●●●●●●●
From User: 22204086
Online Number:
Maximum Channels: 0
Caller ID: 22204086
Realm:
Enable Outbound Proxy Server
Codecs : First: a-law Second: u-law Third: GSM
Fourth: None Fifth: None
Transport: UDP Enable SRTP: Qualify:
DTMF Mode: rfc2833
DOD Settings
DOD: Associated Extension: 300 ↑Add DOD ↑Add Bulk
Save Cancel

Figure6-2

2. Add Outbound Routes

As we can see from the Outbound Route of 'voipout', all phone numbers starting with 0 will have their first digit stripped off (digit 0) and will be sent to the SIP Trunk.

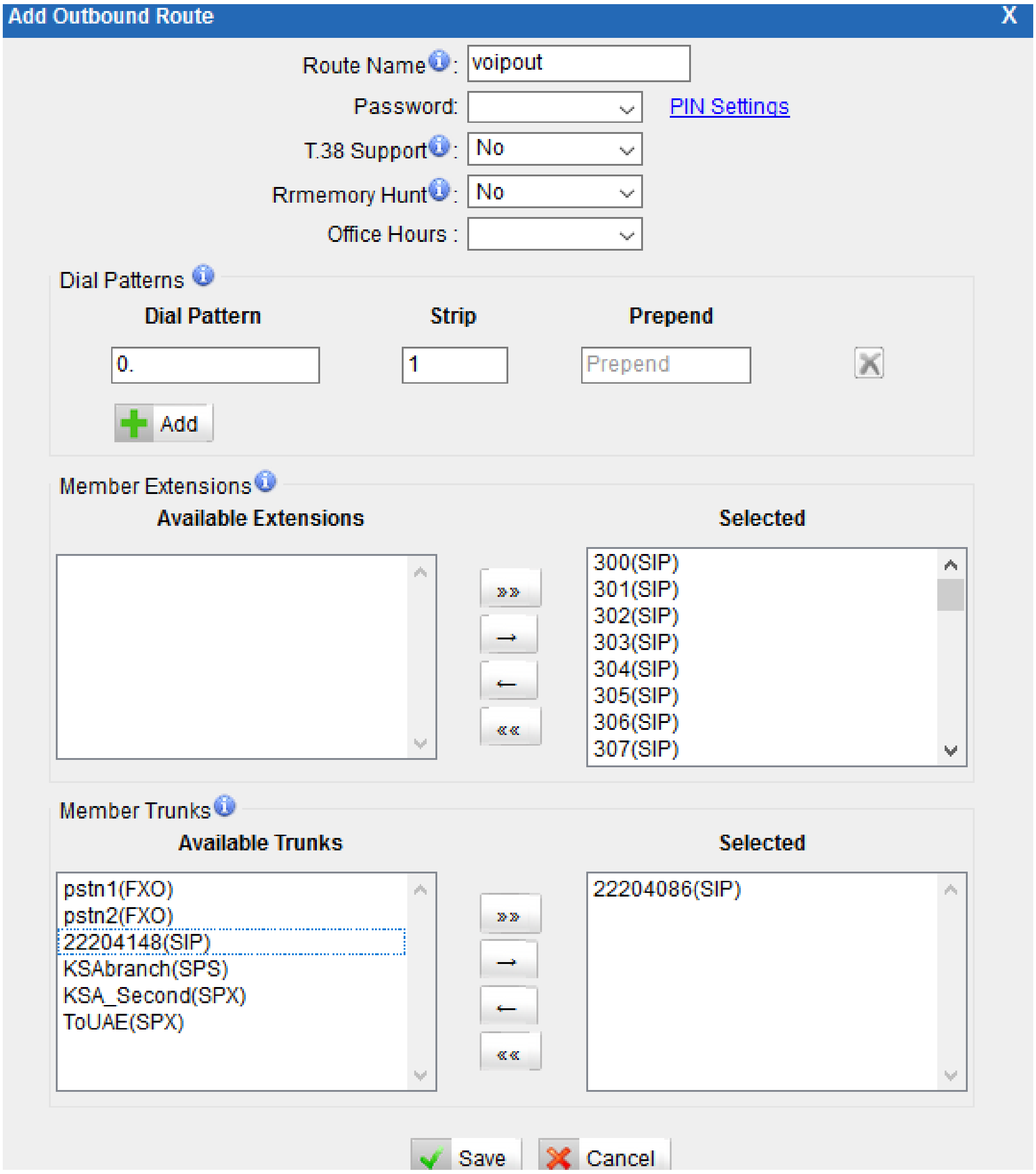


Figure6-3

Now that we have added two outbound dialing rules, any call starting with 9 will be routed to the PSTN, and any number starting with 0 will be routed to the SIP Trunk.

6.2 Incoming call

6.2.1 Sample Routing to an IVR

Let’s configure an incoming call to route to the IVR. In the IVR itself, let’s configure digit 0 to route the call to extension 300, and digit 1 to route the call to extension 301.

. Add IVR

To add a new IVR, go to IVR → Create New IVR

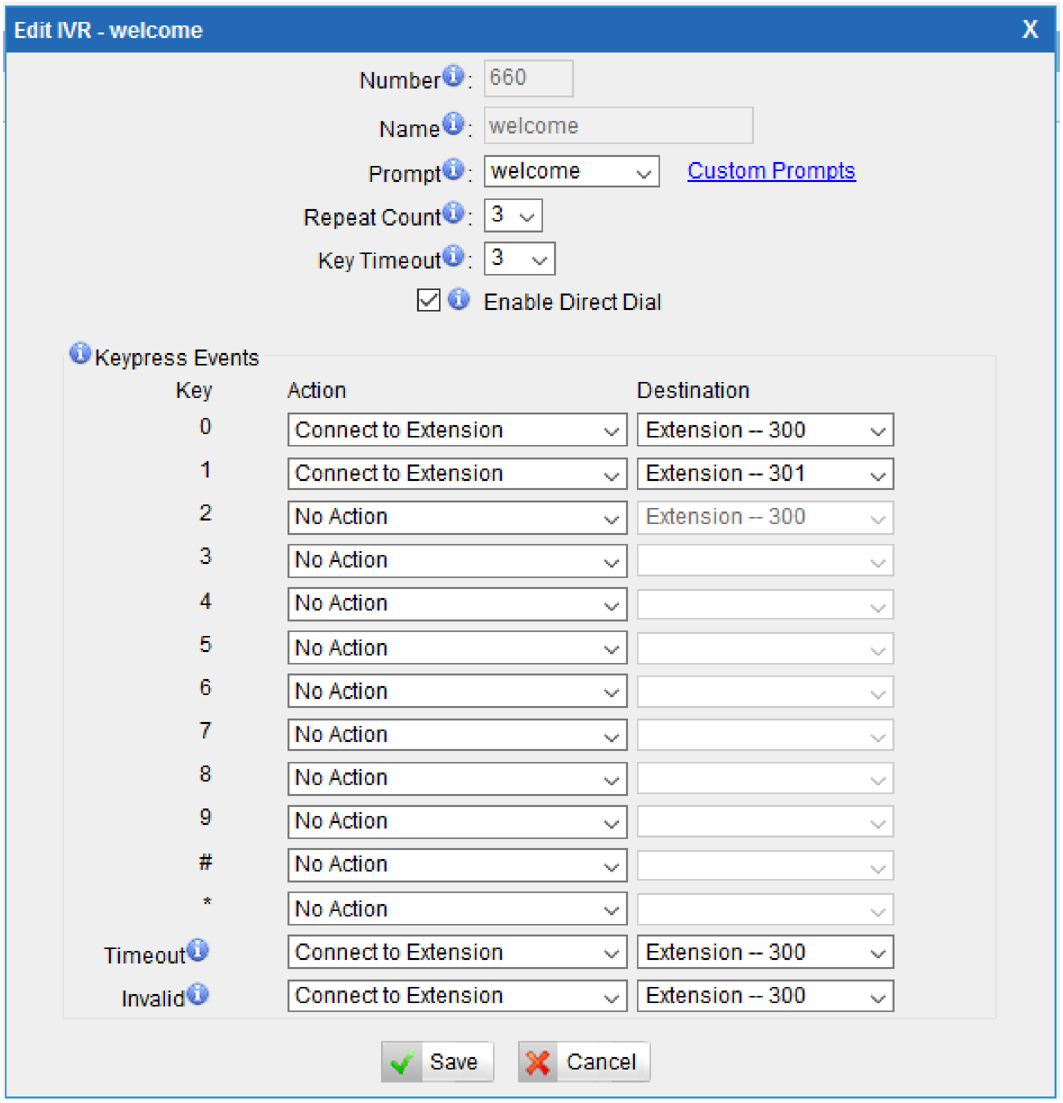


Figure6-4

2. Add Inbound Routes

As we can see from the Inbound Route of 'allin', all incoming calls will be sent to the IVR.

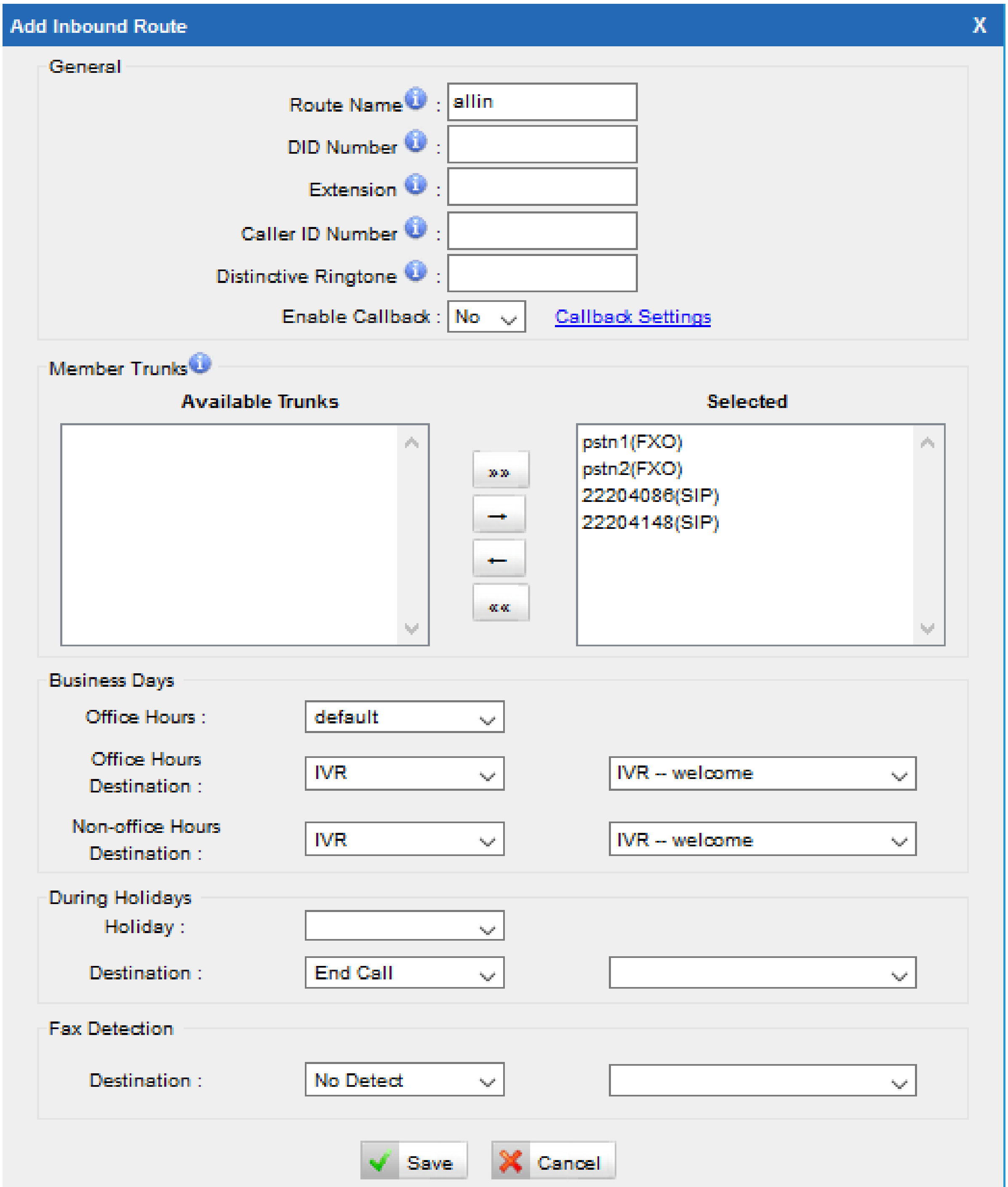


Figure6-5