# S50 IP-PBX User Manual



XonTel Technology



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## **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

<ul> <li>Auto-provision</li> </ul>	Follow me
Blind Transfer	• Interactive Voice Response (IVR)
• BLF Support	Intercom / Zone Intercom
• Blacklist	Music On Hold
<ul> <li>Call Detail Records(CDR)</li> </ul>	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	<ul> <li>Skype Integration (Skype Connect)</li> </ul>
• Caller ID	Speed dial
Conference	<ul> <li>Three-way Calling</li> </ul>
Define Office Time	<ul> <li>Do Not Disturb(DND)</li> </ul>
<ul> <li>Direct Inward System Access(DISA)</li> </ul>	Voicemail
Distinctive Ringtone	• VLAN
• Firewalls	



## **1.2 Hardware Specifications**



## **1.2.1 Exterior Appearance**

## Front Side

Figure1-1 S50 Front Panel Picture

No.	Identifying
0	Green LED: Indicates the power connections normal.
٢	Green LED: Indicates the server system is in working order
3	Green LED: Indicates the system is ready.
۲	Green LED: Indicates the internet interface is in use
6	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

Reports Logout
Reachability

Figure 3.2.1

## **S50 Status Description:**

## **Extensions:**





## Trunks: VOIP Trunk:

← → C 🗋 192.168	3.5.150/cgi/WebCGI?1000						۵ 🗣 😒 ۴ ک
KonTe	50				Status	System PBX	Reports Logout
Line Status	Trunk Status						
Extension Status	Status	Signal	Trunk Name	Туре	User Name	Port/Hostname/IP	Reachability
Trunk Status	Disconnected		pstn1	FXO		Port 1	
System Status	Disconnected		pstn2	FXO		Port 2	
System Info							
Network Status							

## **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).

← → C 🗋 192.168.5.1	50/cgi/WebCGI?1000							6	। 🗘 🗣 🙆
KonTel					Status	System	PBX	Report	s Logout
Extensions	FXS/VoIP Extensions								
EXS/VolP Extensions	FXS Extensions								
Phone Provisioning				No FXS F	xtensions Defined				
Trunks				1017102					
Physical Trunk	VoIP Extensions								
VoIP Trunk	Add Extension	Add Bulk Extension	ons 🥖 Edit t	he selected Extensions	K Delete the selected Extensions		Total: 6	Show: 1-6	View: 25 🔻
Outbound Call Control		Extension	Туре	Register Name	Name	Caller ID			
Outbound Routes		300	SIP	300	300	300			X
Speed Dial Settings	- 0	301	SIP	301	301	301			X
		302	SIP	302	302	302			×
Inbound Call Control		303	SIP	303	303	303			×
IVR		304	SIP	304	304	304			×
Ring Groups		305	SIP	305	305	305			×
Queues							< First < Prev	01 Nex	t > Last >>
Conferences									
Inbound Routes									
Audio Settings									

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

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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 hookflash 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.

Extension - 300		
General Other Settings		
Type: SIP	Extension : 300	Password : pincode300
Name : 300	Caller ID: 300 Re	gister Name 🛈 : 300
Voicemail	√oicemail Access PIN #€0 : 300	
Mail Setting —		
Enable Send Voicemail	0	
Email Address :	a sastion 'SMTD Sattings for Voicoma	I'/in the Voicemail Settings")
have been properly configu	red before using this feature.	in the volcentar oettings)
Group		
Pickup Group 🛈 : 💶 🔻		
Call Duration Setting		
Max Call Duration	S	
VoIP Settings		
NAT:	Qualify: 🗹	Enable SRTP 0 :
Transport: UDP V	DTMF Mode : RFC2833 V	Register Remotely 🛈 : 🔲
	Save 🔀 Cance	4

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  $\rightarrow$  VOIP Extensions  $\rightarrow$  Create New Extension

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

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## ·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" 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.



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xtension - 300	
eneral Other S	Settings
General	
Type: SIP	Extension (): 300 Password (): Gulfsip 123
Name : 300	Caller ID : 300 Register Name : 300
/oicemail	
Enable Voicema	il 🛈 Voicemail Access PIN # 🛈 : 300
Mail Setting	
Enable Send Vo	picemail
mail Address0:	
ote: Please ensure ave been properly	that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') configured before using this feature
Proup	
Rickup Group	
ionale oronia - [1	*
all Duration Settin	
lax Call Duration	S S
/oIP Settings	
NAT 🛈 : 🗹	Qualify: C Enable SRTP
Transport: UDP	DTMF Mode 0: RFC2833 ✓ Register Remotely 0: ☑
Extension - 300	
Extension - 300 General Other Other Options	Settings
Extension - 300 General Other Other Options Call Waiting	Settings
Extension - 300 General Other Other Options Call Waiting Follow me	Settings DND User Web Interface Ring Out
Extension - 300 General Other Other Options Call Waiting Follow me	Settings
Extension - 300 General Other Other Options Call Waiting Follow me	Settings DND User Web Interface Ring Out : 30 Always Voicemail
Extension - 300 General Other Other Options Call Waiting Follow me Follow me:	Settings DND User Web Interface Ring Out : 30 Always No answer Transfer to:
Extension - 300 General Other Other Options Call Waiting Follow me	Settings DND User Web Interface Ring Out : 30 Always No answer Transfer to: When Busy
Extension - 300 General Other Other Options Call Waiting Follow me Follow me: IP Restriction	Settings          DND I User Web Interface Ring Out I: 30         Always         No answer         Transfer to:         Number         When Busy
Extension - 300 General Other Other Options Call Waiting Follow me Follow me: IP Restriction Enable IP Rest	Settings          DND <sup>(1)</sup> User Web Interface <sup>(1)</sup> Ring Out <sup>(1)</sup> : 30         Always <ul> <li>Voicemail</li> <li>No answer</li> <li>Transfer to:</li> <li>Number</li> </ul> When Busy         riction <sup>(1)</sup> :
Extension - 300 General Other Other Options Call Waiting Follow me Follow me IP Restriction Enable IP Rest Permitted 'IP addre	Settings          DND       User Web Interface       Ring Out       30         Always       Image: Setting of the set of the
Extension - 300 General Other Other Options Call Waiting Follow me Follow me IP Restriction Enable IP Rest Permitted 'IP addre	Settings          DND <sup>1</sup> User Web Interface <sup>1</sup> Ring Out <sup>1</sup> : 30         Always         Always         No answer         Transfer to:         When Busy         riction <sup>1</sup> :         ess/Subnet mask' 1 <sup>1</sup> :
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Enable IP Rest Permitted 'IP addre Permitted 'IP addre	Settings          DND       User Web Interface       Ring Out       30         Always       Image: Setting of the set of the
Extension - 300 General Other Other Options Call Waiting Follow me Follow me IP Restriction Enable IP Rest Permitted 'IP addre Permitted 'IP addre	Settings         DND <sup>●</sup> ✓ User Web Interface ●       Ring Out ●: 30         Always       ● Voicemail         No answer       Transfer to:       ○ Number         When Busy       Image: Constraint of the set of th
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Enable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre	Settings          DND I       User Web Interface I       Ring Out I: 30         Always       Image: Voicemail         No answer       Transfer to:       Number         When Busy       Number       Image: Voicemail         riction I:
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Renable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre Nobility Extension	Settings          DND       User Web Interface       Ring Out       : 30         Always       • Voicemail       • Voicemail         No answer       Transfer to:       • Number         When Busy       • Voicemail       • Settings         riction       • • • • • • • • • • • • • • • • • • •
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Enable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre Mobility Extensior Enable Mobility	Settings          DND I       User Web Interface I       Ring Out I       30         Always       Image: Subject of the set o
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Renable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre Mobility Extension Ring Simultane	Settings           DND <sup>(1)</sup> User Web Interface <sup>(1)</sup> Ring Out <sup>(1)</sup> : 30         Always       Image: Voicemail         No answer       Transfer to:       Number         When Busy       Number       Number         riction <sup>(1)</sup> :       ess/Subnet mask' 1 <sup>(1)</sup> :       Image: Voicemail         ess/Subnet mask' 2 <sup>(1)</sup> :       Image: Voicemail       Image: Voicemail         y Extension       Mobility Extension Number <sup>(1)</sup> :       Image: Voicemail         Outbound Prefix:       Image: Voicemail       Image: Voicemail
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me IP Restriction Enable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre Mobility Extension Enable Mobility Ring Simultane Spy Settings	Settings          DND I       User Web Interface       Ring Out       30         Always       Image: Voicemail       Image: Voicemail       Image: Voicemail         No answer       Transfer to:       Number       Image: Voicemail         When Busy       Image: Voicemail       Image: Voicemail       Image: Voicemail         when Busy       Image: Voicemail       Image: Voicemail       Image: Voicemail         riction I:       Image: Voicemail       Image: Voicemail       Image: Voicemail         when Busy       Image: Voicemail       Image: Voicemail       Image: Voicemail         riction I:       Image: Voicemail       Image: Voicemail       Image: Voicemail         when Busy       Image: Voicemail       Image: Voicemail       Image: Voicemail         riction I:       Image: Voicemail       Image: Voicemail       Image: Voicemail         ss:/Subnet mask' 1       Image: Image: Voicemask' 1       Image: Image: Voicemask' 1       Image: Image: Voicemask' 1         w Extension       Mobility Extension Number I:       Image: Im
Extension - 300 General Other Other Options Call Waiting Follow me Follow me Follow me: IP Restriction Renable IP Rest Permitted 'IP addre Permitted 'IP addre Permitted 'IP addre Nobility Extension Ring Simultane Spy Settings	Settings          DND <sup>(1)</sup> User Web Interface <sup>(1)</sup> Ring Out <sup>(1)</sup> : 30         Always       (1)       (1)         Always       (1)       (1)         No answer       Transfer to:       (1)         When Busy       (1)       (1)         riction <sup>(1)</sup> :       (1)       (1)         ess/Subnet mask' 1       (1)       (1)         ess/Subnet mask' 2       (1)       (1)         ess/Subnet mask' 3       (1)       (1)         y Extension       Mobility Extension Number <sup>(1)</sup> :       (1)         o       Outbound Prefix:       (1)         Allow Being Spied       Spy Modes <sup>(1)</sup> :       (1)



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.

Physical Trunk			
BRI Trunk			
	No BRI	Trunks Detected	
Analog Trunk			7.
Trunk Name	Port		
pstn1	1		
pstn2	2		R
pstn3	3		
pstn4	4		
pstn5	5		
pstn6	6		
GSM/UMTS Trunk			
Trunk Name	Port	Туре	
GSM9	9	GSM	

Figure 3.3.2





## 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.

Edit Analog Trunk - pstn1	x
Trunk Name	pstn1
Volume Setting	<u>:</u> 40% ▼
Busy Detection	
Busy Detection	Yes 💌
Busy Count	: 4
Busy Interval	t: 1
Busy Pattern	
Frequency Detection	· No
Busy Frequency	:
Polarity Detection	No 🔻
Advanced Options	
Caller ID Start : Ring -	Caller ID Signaling : Bell - USA
Caller ID Detection : Yes 🔻	
Save	X Cancel
4-0	

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 Dialin (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.

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Signaling: BRL-CPE Switch Type 1: euroisdn    PRI Dialplan 1: unknown Reset Interval 1: never s   PRI Local Dialplan 1: unknown Overlap Dial 1: no    PRI Indication 1: Inband Enable Facility 1: Enabled    PRI Indication 1: none Echo Cancellation 1: Off    Hide Caller ID 1: No Codec: alaw    Caller ID Prefix International Prefix: Private Prefix:    Local Prefix: Private Prefix:     DOD Settings Slobal DOD:	rium rumo	BriTrunk	7				
PRI Dialplan 1: unknown   PRI Local Dialplan 1: unknown   PRI Local Dialplan 1: unknown   PRI Indication 1: Inband   Nsf11: none   Nsf11: none   Hide Caller ID 11: No   Caller ID Prefix   International Prefix:   Local Prefix:   Unknown Prefix:   DOD Settings	Signaling:	BRI-CPE	•	Switch Type	euroisdn	•	
PRI Local Dialplan Inknown   PRI Indication Inband   Nsf none   Nsf none   Echo Cancellation Off   Hide Caller ID No   Caller ID Prefix   International Prefix:   Local Prefix:   Unknown Prefix:   DOD Settings Blobal DOD:	PRI Dialplan 🛈 :	unknown	•	Reset Interval 🛈 :	never	•	s
PRI Indication ①: Inband   Nsf①: none   Echo Cancellation ①: Off   Hide Caller ID ①: No   No Codec:   alaw   Caller ID Prefix   International Prefix:   Local Prefix:   Unknown Prefix:   DOD Settings Blobal DOD:	PRI Local Dialplan 0:	unknown	•	Overlap Dial 0 :	no	•	
Nsf@: none   Hide Caller ID @: No   Caller ID Prefix   International Prefix:   Local Prefix:   Unknown Prefix:   DOD Settings Slobal DOD:	PRI Indication 🛈:	Inband	•	Enable Facility	Enabled	•	
Hide Caller ID   • No   Caller ID Prefix   International Prefix:   Local Prefix:   Unknown Prefix:     DOD Settings     Blobal DOD:	Nsf0:	none	•	Echo Cancellation	Off	•	
Caller ID Prefix International Prefix: Local Prefix: Unknown Prefix: DOD Settings Blobal DOD:	Hide Caller ID	No	•	Codec:	alaw	•	
	Unknown Pref Unknown Pref DOD Settings Global DOD:			Private	Prefix:		

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.



Provider Name:	192.168.4.136			
Hostname/IP:	192.168.4.136		: 5060	
Domain:	111			
User Name:	111			
Authorization Name:	111			
Password:				
From User:				
Online Number				
Maximum Channels	0			
Caller ID 🛈 :	111			
	Enable Outbound Pro	xy Server		
Codecs :	First: a-law 🔻 S	econd: u-law	▼ Third: GSM	
	Fourth: None	Fifth: None	•	
Transport:	UDP - Qualify:	~		
DTMF Mode:	rfc2833 •			
OD Settings				
				1

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.



Edit Service Provider Trunk-S	PS-192.168.4.138	Х
Туре:	SIP 💌	
Provider Name:	192.168.4.138	
Hostname/IP:	192.168.4.138 : 5060	
Maximum Channels 0:	0	
Codecs :	First:     GSM     Image: Second:     a-law     Image: Third:     u-law       Fourth:     None     Image: Fifth:     None     Image: Third:     Units of third:	
Transport:	UDP -	
Qualify:	<u>र</u>	
DTMF Mode:	rfc2833 🔹	
DOD Settings		
Global DOD:		
DOD :	Associated Extension : 601  Add DOD	
	Save K Cancel	

Figure 3.3.2.4

## 3.3.3 Outbound Routes

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

← → C 🗋 192.168.5.150	0/cgi/WebCGI?1000							<b>७</b> ¶☆	🗣 🙆 🗉
KonTel IP-PBX \$50	0			Status	System	PB		Reports	Cogout
C Extensions	Outbound Routes								
FXS/VoIP Extensions	+ Add Outbound Route	K Delete the selected Route							
Phone Provisioning		Route Name	Dial Pattern						
🔽 Trunks		pstnout	9.				X		
Physical Trunk									
VolP Trunk									
Outbound Call Control									
Outbound Routes									
Speed Dial Settings									
Inbound Call Control									
IVR									
Ring Groups									
Queues									
Conferences									
Inbound Routes									
Audio Settings									



## 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: **1NXXNXXXX** 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**

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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.

Add Outbound Route			Х
Route	Name: pstnout		
	Password:	V PIN Settings	
T.38 S	upport <sup>1</sup> : No	~	
Rrmemor	v Hunt <sup>®</sup> · No		
0ff	ice Hours :		
Dial Patterns 🕕		•	
Dial Pattern	Strip	Prepend	
	Sup		
9.	1	Prepend	×.
+ Add			
Member Extensions		Selected	
		200(SIP)	•
	^ <u></u>	301(SIP)	
		302(SIP)	
		303(SIP)	
		304(SIP)	
		305(SIP)	
	**	306(SIP)	
	×	307(SIP)	*
Member Trunks			
Available Trunks		Selected	
22204086(SIP)		pstn1(FXO)	~
22204148(SIP)	22	pstn2(FXO)	
KSAbranch(SPS)			
KSA_Second(SPX)			
ToUAE(SPX)	-		
	4C 4C		
	× .		×
	🖌 Save 🎽	Cancel	

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, www.sahabtec.com

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#### 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	12
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	122

Figure 3.3.4.2

## 3.3.4.3 Phone book

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

Phone Book 🌵	
◆ Create New Contact	Go Back to Auto Provision
Contacts	
No Contact Defined	
Deny List	
No Deny List Defined	
Upload Phonebook	
Note: All the existing phonebooks of the IP phone would be deleted automatically if the phonebooks are configured in this way.	
No Phonebook Uploaded	

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.



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Type.	None 👻	Group:	None 👻
Nick Name		Favorite	No 👻
Organization 0:		Title <sup>()</sup> :	
Email <sup>0</sup> :		Birthday	
First Name		Family Name®:	
Office Number:		Mobile Number:	
Home Number:			
Sub Number			
Sub Name	9	Sub Number	Add Sub
Sub Name:	S	Sub Number:	↑Add Sub
Sub Name:	S	Sub Number:	↑Add Sub
Sub Name:	S	Sub Number:	↑Add Sub
Sub Name:	9	Sub Number:	↑Add Sub

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.

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

neiai			
Enabled:	Yes 🗸	NewConfig 🛈 :	No
MAC Address:	001565	Name:	
Manufacturer:	Yealink 🗸	Phone Type:	T28 🗸
Call Waiting:	Enabled 🗸	Key As Send:	# 🗸
Auto Redial:	Disabled 🗸	Auto Answer:	Disabled 🗸
Phone Book:	Enabled 🗸		
Line1 Ext	ension:	Label:	Line Active:
Line3 Ext	ension:	Label:	Line Active:
] Line3 Ext ] Line4 Ext	ension:	Label:	Line Active:
] Line3 Ext ] Line4 Ext ] Line5 Ext	ension:	Label:	Line Active:

Figure 3.3.4.4.1

# 2) Audio codec

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

Custom	
Disable Codecs	Enable Codecs
G723_53 G723_63 G726-16 G726-24 G726-32 G726-40	>>> PCMA PCMU G729 G722 ←

Figure 3.3.4.4.2

# 3) Line keys settings

Configure the DSS keys/Function Keys



Memory Key						
Key	Тур	e	Value	Line		Extension
DSS Key1	N/A			[ine1	*	
DSS Key2	N/A.			[ine1	*	
DSS Key3	N/A.	*		line1	¥ [	
DSS Key4	N/A.	*		line1	-	
DSS Key5	N/A	*		line1	*	
DSS Key6	N/A	*		line1	-	
DSS Key7	N/A	w		line1	-	
DSS Key8	N/A.	*		line1	¥.	
DSS Key9	N/A	.*		line1	-	
DSS Key10	N/A	*		line1	-	
Line Keys Setting Line Keys Setting Key	gs Type	Value	Labe	1 14	Line	Extension
Line Key 1	NA	*		Line 1	*	
Line Key 2	N/A	*		Line2	*	
Line Key 3	N/A	*		Line3	*	
Line Key 4	N/A	*		Line4	*	
Line Key 5	N/A	¥		Line5	÷	
Line Key 6	N/A	-		Line6	*	

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

XonTel			S50	IP-PB	X User Manua	al
Add Phone						x
General						
Enabled:	Yes	•	NewConfig	Yes	1.¥	
MAC Address:	001565		Name:			
Manufacturer:	Yealink	•	Phone Type:	T28		
Call Waiting:	Enabled	•	Key As Send:	#		
Auto Redial:	Disabled	•	Auto Answer:	Disabled	•	
Phone Book:	Enabled	•				
Line						
Line1 E	xtension:	*	Label:		Line Active:	
Line2 E	xtension:	Ŧ	Label:		Line Active:	
Line3 E	xtension:	-	Label:		Line Active:	
Line4 E	xtension:	*	Label:		Line Active:	
Line5 E	xtension:	*	Label:		Line Active:	

Figure 3.3.4.5

Label:

Line Active:

# 3.3.4.6 Upload a file

Line6

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

-

Note: the file format must be .cfg

Extension:

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.



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← → C 🗋 192.168.5.15	0/cgi/WebCGI?1000						<mark>હ ૧</mark> ર	2 🗣 🙆
KonTel IP-PBX S50				Status	System	PBX	Reports	COS Logout
Extensions	Inbound Routes							
FXS/VoIP Extensions	Add Inbound Route	K Delete the selected	Route					
Phone Provisioning		Route Name	DID Number	Caller ID Number				
🔽 Trunks		pstnin					×	
Physical Trunk								
VoIP Trunk								
Outbound Call Control								
Outbound Routes								
Speed Dial Settings								
Inbound Call Control								
IVR								
Ring Groups								
Queues								
Conferences								
Inbound Routes								
D Audio Settings								

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 EventsA 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.

New IVR						×
Number 🛈 :	660					
Name 🛈 :	welcome					
Prompt 🛈 :	default 💌	Custom IVR Prompts				
Repeat Count :	3 💌					
WaitExten 🛈 :	3 🗸					
	Allow Dialing	) Other Extensions				
C KeyPres	ss Events				1	
	Кеу	Action		Destination		
	0	Connect to Extension		User Extension 500	•	
	1	Connect to Extension	•	User Extension 501	•	
	2	Connect to RingGroup	•	RingGroup ringgroup_de	əfa	
	3	No Action	•		Y	
	4	No Action	•		¥.	
	5	No Action			<b>T</b>	
	6	No Action			Y	
	7	No Action			T	
	8	No Action			Y	
	9	No Action			Y	
	#	No Action	•		Y	
	*	No Action	•		Y	
TimeOu	t 🛈	Connect to Extension		User Extension 500	•	
Invalio	d 🛈	Hangup	•			
		🖌 Save	X Cancel			

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 <u>RingAll</u>: Ring All available Agents simultaneously until one answers. <u>LeastRecent</u>: Ring the Agent which was least recently called. <u>FewestCalls</u>: Ring the Agent with the fewest completed calls. <u>Random</u>: Ring a Random Agent. <u>RRmemory</u>: 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

# KonTel

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

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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.



# S50 IP-PBX User Manual

New Queues		x
G	)ueue Name🛈:	692
Qu	eue Number 🛈 :	692
Queu	e Password 🛈:	
Queue Ag	ent Timeout 🛈:	30
Queue Ma	x Wait Time 🛈 :	1800
Queue R	ing Strategy 0:	ringall 🗸
Agents		
Available Agent	5	Selected
500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP) 506(SIP) 507(SIP)		>>  
Caller Position Announcements Annou Announ	unce Position () : ce Hold Time () : Frequency () :	Yes V Yes V 30 seconds V
Periodic Announcements	Prompt <sup>1</sup> : Frequency <sup>1</sup> :	30 seconds
Events		
	Key: Action:	End Call
	Destination:	
Eailover Dectination		
Failover-Desuitation	Action:	End Call
	Destination:	
Others		
M	usic On Hold 0:	calmriver
Leave	When Empty 0:	Yes 💌
	Join Empty	No
Agent Ar	nnouncement	
Join Ar	nnouncement	
	Retry U:	30
1	/Vrap-up time 🔍 :	30
	V Save	a 🗶 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.

Record New Prompt	x	A STATE
File Name: Dial extension: Reco	welcome 500 v to record a new voice prompt rd X Cancel	

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.

Upload Prompt			Х
The file si: WAV format: .gsm 6.10.8kHz Mono 11	ze must not be larger than '	1.8MB! o 1Kb/s , pcm 8kHz I	Mono 16Kb/s
Choose a File to Upload		Browse	
	Upload X Cancel		

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.

S50 IP-PBX			Status System	PBX	Reports	COS Logout
Extensions	Ring Groups					
FXS/VoIP Extensions	+ Add Ring Gro	up				
Phone Provisioning	Number	Namo	Mombors			
Trunks	620	ringgroup default	300(SIP)-301(SIP)-302(SIP)-303(SIP)-304(SIP)-305(S			
Physical Trunk						
VolP Trunk						
Outbound Call Control						
Outbound Routes						
Speed Dial Settings						
Inbound Call Control						
IVR				2		
Ring Groups						
Queues						
Conferences						
Inbound Routes						
Audio Settings						
Basic Settings						

#### 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.

KonTel	S50	IP-PBX User Manual
Add Ring Group		х
Ring Grou Ring Group Seconds to ring each Mobility Extension Rings Simu	Ip Name <sup>(1)</sup> : 621 Number <sup>(1)</sup> : 621 Strategy <sup>(1)</sup> : Ring all simul member <sup>(1)</sup> : 60 ultaneously: No ~	taneously ~
Ring Group members 0 Available Extensions		Selected
300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 306(SIP) 307(SIP)		
Destination If No Answer:	End Call	
Destination:	OExtension OVoicemail OIVR ORing Group OConference Room OQueues	Extension 300 Voicemail 300 Voicem
~	Save 🔀 Cancel	

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.

← → C 🗋 192.168.5.150	0/cgi/WebCGI?1000						<b>6</b> 75	3 💁 🙆
KonTel IP-PBX \$50				Status	System	PBX	Reports	CO Logout
Extensions	Inbound Routes							
FXS/VoIP Extensions	+ Add Inbound Route	X Delete the selected	I Route					
Phone Provisioning		Route Name	DID Number	Caller ID Number				1
🔽 Trunks	0	pstnin					X	
Physical Trunk								
VoIP Trunk								
Outbound Call Control								
Outbound Routes								
Speed Dial Settings								
Inbound Call Control								
IVR								
Ring Groups								
Queues								
Conferences								
Inbound Routes								
Audio Settings								

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.

 $\cdot \mathrm{IVR}$  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.

	Route Name	: pstnin		
	DID Number 🛈	- [		
	Extension 🛈	-		
C	aller ID Number 🕠			
0.		*L		
Disti	nctive Ringtone			
Nember Trunks 🛈				
Available	Trunks		Selected	
pstn1(FXO)		2020	192.168.4.136(SIP)	
pstn2(FXO)				
		←		
		««		
Business Davs		**		
Business Days Office Hours :	default	««		
Business Days Office Hours : Office Hours	default	««		
Business Days Office Hours : Office Hours Destination :	default IVR	•	IVR welcome	
Business Days Office Hours : Office Hours Destination : Non-office Hours	default IVR	««	MR – welcome	•
Business Days Office Hours : Office Hours Destination : Non-office Hours Destination :	default IVR IVR	«« •	IVR welcome	•
Business Days Office Hours : Office Hours Destination : Non-office Hours Destination : During Holidays	default IVR IVR	× «	IVR – welcome	*
Business Days Office Hours : Office Hours Destination : Non-office Hours Destination : During Holidays Holiday :	default IVR IVR	<ul> <li>««</li> <li>•</li> <li>•</li></ul>	IVR welcome IVR welcome	•
Business Days Office Hours : Office Hours Destination : Non-office Hours Destination : During Holidays Holiday :	default IVR IVR		IVR – welcome	*
Business Days Office Hours : Office Hours Destination : Non-office Hours Destination : During Holidays Holiday : Destination :	default IVR IVR End Call		IVR welcome IVR welcome	*

Figure 3.4.5.1



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.

← → C 🗋 192.168.5.150	0/cgi/WebCGI?1000						<b>6</b> 7 5	3 🗣 🙆
KonTel IP-PBX S50				Status	System	PBX	Reports	CO Logout
Extensions	Blacklist							
FXS/VoIP Extensions	+ Add Blacklist						Total: 0	Show: 0-0
Phone Provisioning								
Trunks			No Blacklists Defined					
Physical Trunk								
VolP Trunk								
Outbound Call Control								
Outbound Routes								
Speed Dial Settings								
Inbound Call Control								
I/R								
Ring Groups								
Queues								
Conferences								
Inbound Routes								
Audio Settings								

Figure 3.4.6

# 3.4.6.1 Create Blacklist

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

Create New Blacklist	х
Blacklist Number : Type : Inbound	
Save Save	

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 ExtensionsThe 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



Status Monitor	P General Preterences G
Line Status	
	General Preferences
Basic 🙁	Ring Timeout 9 30 s
Extensions	
Trunks	MAX Call Duration U: 6000 s
Outbound Routes	Maximum Concurrent Calls
Phone Provisioning	Music On Hold: calmiver -
Inbound Call Control	Tone Region 🕖 : United Kingdom 👻
IVR	HTTP Bind Port 9: 80
Queues	FXO Mode : FCC V
Custom Prompts	Distinctive Caller ID 0 · No •
Ring Groups	Follow Ma Promet: Yes
Rlacklist	
	invalid Phone Number Phonps 0.
Internal Settings 🙁	Busy Line Prompt U:
Options	Dial Failure Prompt 🕛 :
Business Hours	Extension Preferences
Feature Codes	User Extensions : 500 to 616
SIP Settings	Ring Group Extensions 620 to 629
DISA	
Conferences	Paging Group Extensions: 630 to 639
Paging Groups	Conference Extensions: 640 to 659
DNIS Settings	IVR Extensions : 660 to 679
PIN User Settings	Queue Extensions : 680 to 689
Speed Dial Settings	Reset to Defaults
Music on Hold Prompts	
Network Settings 🙁	Save X cancel

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 DaysYou 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.



# • Reviews Haars 0 Constrict © Datable Derivate Haars \* Datable Office (Derivat) Torrag \* Datable \* Datable Office (Derivat) Derivation \* Datable Derivation

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

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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)

Feature Codes 0			
and the second se			
General	and a second second second	Colorest.	
	Gree Tauch Record O	*1	
	Chark Extension Voicemail	*2	
	Vocerual Main Menu@	702	
	<ul> <li>Attended Transfer</li> </ul>	13	
	Attended Transfer Timecut 🔍	15	3
	P Bird Traveler	103	
	Cat Pickey	N	
	Extension Pickup Q	*04	
	F Marcan	-15	
	G Nerreal SpyQ	-50	
	17 Whisper Bay O	191	
-	F Barga Soy Q	192	
Call Furking Professions	and the second	- 200	
	Call Parking	76	
	Estansian range used to park calls.	690-699	(E.s. 696-696)
	Number of accords a call can be parted for	60	
Call Forwarding Protorenza			
	Reset to Defaults	*70	
	Enable Forward All Calls	-71	
	F Death Fernard All Calls	1071	
	- Enable Forward When Bulls	*72	
	F Disable Forward When Busy	1072	
	Enable Forward No Answer	*73	
	Clisable Forward Na Annuw	*073	
	Farward to NumberO	*74	
1	Farward to Voicemail	1074	
	Enable Do Not Distarts	*75	
	F Distable Do Nat Distarts	1075	

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**

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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 60 seconds.

# ·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;

'10.0.0/255.0.0.0' : Also RFC1918;

'172.16.0.0/12': Another RFC1918 with CIDR notation;

'169.254.0.0/255.255.0.0' : Zero conf local network.

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-refersher, the default is Uas.





SIP (Session Initiation Protocol) Configuration 🔅	
General	
: UDP Port	5060
Enable TCP Port :	5060
Enable TLS Port	5061
RTP Port Start:	10000
RTP Port End:	12000
DTMF Mode 🚺 :	rfc2833 👻
Max Registration/Subscription Time	3600
Min Registration/Subscription Time	60
Default Incoming/Outgoing Registration Time	120
Register Attempts 🛈 :	8
Register Timeout 🛈 :	20
Calling Channel Codec Priority	Yes 💌
Video Support 👀:	Yes 🔻
Max Bit Rate	384 kb/s
DNS SRV Look Up	No
User Agent 10:	
Enable STUN: STUN Address: STUN Port: External IP Address • External IP Address • External Refresh Interval • Local Network Identification • NAT Mode • Allow RTP Re-invite • SPEEX G722 G728 ADPCM G729A MPEG4	yes ▼ yes ▼ Allowed Codecs S U=Bw B=Bw B=Bw SSM H283P H284
C 700 License Veu :	
Note: If you would like to use G.729, please e	nter your license key above.
QUS 🕿	
Tos SIP: CS3	
Tos Audio: EF 🔻	Cos Video:
Advanced Settings A	
Automeen Johnnys A.	From
From Field.	
10 Field:	
Remote Party ID	
Session-timers •	1900
Session-expires V:	
Session-refresher	
Cossidirfeiteater.	

SFigure 3.5.4

# 3.5.5 Voicemail Settings

1) General Voicemail Settings

a) Message Options

# •Max Messages per Folder

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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  $0^{0}$ .

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.



← → C 🗋 192.1	68.5.150/cgi/WebCGI?1000	96	7☆	۵
C Extensions	Voicemail Settings			
FXS/VoIP Extensions	General Volcemail Settions			
Phone Provisioning	Message Options			
Trunks	Max Messages per Folder 1 10 *			
	Max Message Time(): 5 Minutes 🔻			
Physical Trunk	Min Message Time 0: 5 Seconds 🔻			
VolP Trunk	Ask Caller to Dial 50:			
Outbound Call Control	Delete Voicemail 🛛 🔅			
Outbound Routes	Operator Breakout from Voicemail 🕖 : No 💌			
Sneed Dial Settings	Destination: verticome			
- Contraction of the Contraction	Greeting Settings			
Inbound Call Control	Busy Prompt : Play busy greating •			
IVR	Unavailable Prompt U: Play unavailable greatings *			
Ring Groups	Leave a Message Prompt : Skip greating *			
Queues	Prayback Options			
Contemporar	Amoune Messee Destination			
Companyes	Amount Mission Amount			
Inbound Routes	Allow Users to Review Messages 0:			
Audio Settings	SMTP Settings For Voicemail			
Custom Prompts	Note:If you would like to send voicemail messages as email attachments, please configure this section.			
Music on Hold Prompts	SMTP Settings			
Curtan Donate Cattings	E-mai Address 0: [admin]			
Gystem Prompts Gettings	Password 0:			
Basic Settings	SMTP SeverU:			
General Preferences	Pott			
Business Hours	List out is to set set and the set of the se			
Feature Codes	eet Ski P Setings	_		
Universal Sections	Save 🗶 Cancel			
TANGGUNG SAMASO				

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.



v DISA	
General	
Name 💷 :	
PIN #10 :	
Response Timeout 🛈 : 1	0
Digit Timeout 🛈 : 5	
Member Outbound Routes Available Outbound Routes	Selected Outbound Routes
pstnout	33
	<u></u>
	**
- Cove	Cancel
Save	Cancel

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.

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#### 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).

Edit Conference Room 640	Х
Extension : 640	
Admin 🛈 :	
PIN #00:	
Save 🔀 Ca	ncel

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.


← → C 🗋 192.168.5.150	/cgi/WebCGi?1000	(응 위 ☆ ) 💁 (의)
KonTel	Status System PBX	Reports Logout
Extensions	Paging Groups	
FXS/VoIP Extensions	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	in speakerphone mode.
Phone Provisioning	Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the Basic Settings -> Feature Codes 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 of	screen. fficially supported.
Trunks		
Physical Trunk	List of Paging Groups	
VolP Trunk	+ Add Paging Group	
Outbound Call Control	No Paging Groups Defined	
Outbound Routes		
Speed Dial Settings		
Inbound Call Control		
NR		
Ring Groups		
Queues		
Conferences		
Inbound Routes		
Audio Settings		

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.

New Paging Group	X
Paging Group Number	630
Duplex 🛈 :	
Paging Group members	
Available Extensions	Selected
500(SIP)         501(SIP)         502(SIP)         503(SIP)         504(SIP)         505(SIP)         506(SIP)         507(SIP)	>>>         <
🖌 Sav	/e X Cancel

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.

DNIS 🕸				
		DNIS		
Note:				
1.PSTN trunk and GSM/UMTS trun	k do not need to set DID number.			
2.If you do not set the DID number,	all calls through this trunk will sho	w the DNIS name as Caller Name	e.	
+ Add DNIS				
Trunk Name	Enable DNIS	DNIS Name	DID Number	
pstn4(FXO)	off	-	-	🧹 Edit 🔰 Delete
pstn5(FXO)	off	122		🖌 Edit 🔰 Delete
pstn6(FXO)	off	-	<u>19</u> 3	🖌 Edit 🛛 🗶 Delete
BriTrunk8(BRI)	off	1591		🖌 Edit 🛛 🗶 Delete
BriTaunk7(BDI)	off.			Calit M 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.

← → C 🗋 192.168.5.15	50/cgi/WebCGI?1000					<mark>७ न</mark> જ	r 🗣 🙆 🗄
KonTel IP-PBX S50			Status	System	PBX	Reports	Cogout
Extensions	PIN User Settings						
FXS/VoIP Extensions	+ Add PIN User						
Phone Provisioning							
Trunks		No PIN U	sers Defined				
Physical Trunk	Options						
VolP Trunk		Access Code	*89				
Outbound Call Control		Prompt for Entry Prompt for Failed Entry	pinuser-entry				
Outbound Routes			pindser-entri				
Speed Dial Settings	-	🖌 Save	X Cancel				
Inbound Call Control							
IVR							
Ring Groups							
Queues							
Conferences							
Inbound Routes							
Audio Settings							

#### 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.



← → C 🗋 192.168.5.150	/cgi/WebCGI?1000					<b>۳</b> کا	ය් 💁 🙆
KonTel IP-PBX S50			Status	System	PBX	Reports	Logout
Extensions	Speed Dial Settings						
FXS/VoIP Extensions	Options						
Phone Provisioning		The prefix of speed dial 0 : *99					
Trunks		🖌 Save 🔀 Cancel					
Physical Trunk							
VolP Trunk	+ Add Speed Dial 📉 Delete the Selected Speed Dial				Total: 0	Show: 0 - 0 V	iew: 25 🔻
Outbound Call Control							
Outbound Routes		No Speed Dial Defined					
Speed Dial Settings							
Inbound Call Control							
IVR							
Ring Groups							
Queues							
Conferences							
Inbound Routes							
Audio Settings							

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.

Upload Music on Hold Prompt			Х
The file siz wav format: gsm 6.10 8kHz,Mono,1K Choose a file to upload	te must not be larger than 1.6 b/s \ alaw/ulaw 8kHz,Mono,1 Upload X Cancel	MB! Kb/s v pcm 8kHz, Browse	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.

← → C 🗋 192.168.5.15	0/cgi/WebCGI?1000						€ ¶ ☆	r 💁 🙆 🗄
KonTel IP-PBX \$50				Status	System	PBX	Reports	COS Logout
Network Preferences	LAN Settings							
LAN Settings	LAN Settings							
DHCP Server		DHCP:	No 🔻					
VLAN Settings		Enable SSH:	No • Port: 8022					
VPN Settings		Enable FTP:	No V Port: 21					
DDNS Settings		Hostname:	Xontel					
DDIVG Settings		IP Address:	192.168.5.150					
Static Route		Subnet Mask :	255.255.255.0					
Security Settings		Gateway :	192.168.5.1					
Security Center		Primary DNS :	192.168.5.1					
Firewall Rules		Secondary DNS :						
ID Disabilitat		IP Address2:						
IP Blacklist		Subnet Mask2.						
AMI Settings		🖌 Save	Cancel					
Database Grant								
Alert Settings								
LDAP Server								
Storage Management								

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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.

	Routin	ng Table		
Destination	Subnet Mask	Gateway	Metric	Interface
192.168.4.0	255.255.254.0	0.0.0.0	0	LAN
224.0.0.0	224.0.0.0	0.0.0.0	0	LAN
0.0.0.0	0.0.0.0	192.168.5.1	0	LAN
	outo N			
~	Subnet Mask: Gatew	av: Metric 🔍	Inte	rface:LAN 🔻 🕂 🗛
Destination 🛈 :		, mons [		

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.

C C Enable Firewall			
Note:			
1.You must reboot the sy	stem after enabling or disabling firewa	II.	Firewall has started successful
2.It is strongly recommen	ded to add local network address to a	common rule with the 'action' is 'accept',	
or it may be dragged into	the blacklist.		
Common Rules			
+ Add Rule			
		No Common Rules Defined	
uto Defense			
+ Add Rule			
		No Auto Defense Rules Defined	
P Blacklist			
+ Add Rule			
			IP Blacklist Mana
Port	Protocol	Rate	
5060	UDP	120/60s	🔊 Edit 🔀 Delete
5050	UDP	40/2s	📢 Edit 🗶 Delete
5000		5/00	

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

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# KonTel

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, X means  $0 \sim 9$  or  $A \sim F$  in hex, the  $A \sim F$  are not case sensitive.

#### Action

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

New firewall rule	×
Name 🛈 :	
Description 🛈 :	<u>^</u>
	<u>v</u>
Protocol 🛈 : 🛛	TCP
Port <sup>©</sup> :	0 : 0
IP:	/
MAC Address 🛈 :	
Action 🛈 : [	Drop 🔽
	Save XCancel

Figure 3.6.3.1

3) Auto Defense•PortAuto 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

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be refused o	directly.
	,

New auto defense rule	X
Port: 0	
Protocol 📵 : TCP 💌	
Rate 0 : 🛛 🗸 Second 💌	
Save Save	

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.

New Auto Blacklist Rules	Х
Port®:	
Protocol 🛈 : UDP 💟	
IP Packets	
Time Interval	
Save X Cancel	

Figure 3.6.3.3

b) IP Blacklist Manage

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

		IP Blacklist		
				Go Back to Firewall Rules Settings
Attacked Time		Attacked Port	Source IP address	
1970-Jan-819:58	UDP	5060	203.117.31.243	× Delete



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.

DHCP Server	
DHCP is	s not running
	Enable
Router :	192.168.5.1
Subnet Mask :	255.255.255.0
Primary DNS :	192.168.5.1
Secondary DNS :	
Allow IP Address From:	192.168.5.2
To:	192.168.5.254
TFTP Server	tftp://192.168.4.141
NTP Server:	

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.



← → C 🗋 192.168.5.15	0/cgi/WebCGI?1000					<mark>() 위</mark> ☆	<b>a</b> O
KonTel IP-PBX S50			Status	System	РВХ	Reports	Logout
Vetwork Preferences	VLAN Settings						
LAN Settings	VLAN Over LAN						
DHCP Server		NO.1:	0				
VLAN Settings		VLAN Number:					
VPN Settings		VLAN IP Address:					
The country of		VLAN Subnet Mask:					
DDNS Settings		Default Gateway:					
Static Route		NO.2:					
Security Settings		VLAN Number:					
Security Center		VLAN IP Address.					
Firmurall Dulor		Default Cateway:					
rilewall Rules		Delauli Galeway.					
IP Blacklist		🖌 Save	Cancel				
AMI Settings							
Database Grant							
Alert Settings							
LDAP Server							
Storage Management							

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



← → C 🗋 192.168.5.150	0/cgi/WebCGi?1000 🕲 🖁 🏠 💺 🙆 🗄
KonTel IP-PBX S50	Status System PBX Reports Logout
Vetwork Preferences	DDNS Settings
LAN Settings	DDNS Settings
DHCP Server	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 dyndris org, freedris afraid org, www.norje.com, www.zoneedit.com, www.cay.com, 3322.org
VLAN Settings	DDNS is not running
VPN Settings	Enable DDNS: 🗐
DDNS Settings	DDNS Server: dyndns.org
Static Route	Passwort
Security Settings	Host Name:
Security Center	🖌 Save 🔀 Cancel
Firewall Rules	
IP Blacklist	
AMI Settings	
Database Grant	
Alert Settings	
LDAP Server	
Storage Management	

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.

Change Password	
User:	admin 👻
Enter Old Password:	
Enter New Password:	
Retype New Password:	
CDR Setting	
Enable CDRUser:	Yes 💌

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.



IP-PBX 550				Status	System	PBX	Reports	Logout
Extensions	System Prompts Settings							
FXS/VoIP Extensions		Prompts Download Note:Auto-detection is highly recommended. B	ut if you prefer to dow	nload via HTTF	or TFTP server.			
Phone Provisioning		please contact the local dealer for the prompts.						
Trunks		Local Prompts:	English					
Physical Trunk		Download Mode:	Auto Detection	۲				
VoIP Trunk		Prompts:	English	۲				
Outbound Call Control			Download					
Outbound Routes								
Speed Dial Settings								
Inbound Call Control								
ľ√R								
Ring Groups								
Queues								
Conferences								
Inbound Routes								
Audio Settings								

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

← ⇒ C 🗋 192.168.5.150	0/cgi/WebCGI?1000						<b>€</b> ¶☆	D 0
KonTel IP-PBX S50				Status	System	PBX	Reports	CO Logout
Network Preferences	Date & Time							
LAN Settings	Date & Time							
DHCP Server		Server Time	: Sat Dec 10 2:13:50 2011					
VLAN Settings		Time Zone	: 3 Bahrain	•				
VPN Settings								
DDNS Settings		Daylight Saving Time	: Disabled	*				
Static Route		0	Automatically Synchroniz	e With An Internet	Time Server			
Security Settings			NTP Server: pool.ntp.org					
Security Center		(	Set Date & Time Manually					
Firewall Rules			Date					
IP Blacklist			Time :	▼ AM ▼				
AMI Settings	-		Save X Can	cel				
Database Grant								
Alert Settings								
LDAP Server								
Storage Management								

www.sahabtec.com



Figure 3.7.4

# 3.7.5 Backup and Restore

← → C 🗋 192.168.5.150	0/cgi/WebCGI?1000						<b>٤ 위</b> ☆	<b>Dx</b> (D)
KonTel IP-PBX S50				Status	System	PBX	Reports	Logout
Network Preferences	Backup and Restore							
LAN Settings	+ Create a New Backup	全 Upload a Backup						
DHCP Server			List Of Bandana Card	number Dealers				
VLAN Settings			List Of Previous Config	gurauon backups				
VPN Settings			No backup file	s found				
DDNS Settings			Please click on the 'Create	a New Backup' button				
Static Route			to take a backup of the currer	nt system configuration				
Security Settings								
Security Center								
Firewall Rules								
IP Blacklist								
AMI Settings								
Database Grant								
Alert Settings								
LDAP Server								
Storage Management								

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.



← → C 🗋 192.168.5.1	i0/cgi/WebCGI?1000 🕲 🕏 🏠 📴 🙆
KonTel IP-PBX S50	Statius System PBX Reports Logout
Network Preferences	Reset and Reboot Options
LAN Settings	Reboot System
DHCP Server	
VLAN Settings	Reboot System
VPN Settings	Warning: Rebooting the system will terminate all active calls!
DDNS Settings	Reboot
Static Route	Reset to Factory Defaults
Security Settings	Deet to Cartony Defaulte
Security Center	Warning: A fectory meet will area all configuration data on the sustem
Firewall Rules	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.
IP Blacklist	Reset to Factory Defaults
AMI Settings	
Database Grant	
Alert Settings	
LDAP Server	
Storage Management	

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.

← → C 🗋 192.168.5.15	0/cgi/WebCGI?1000 👋 🛱 🏠 📭 🙆
KonTel IP-PBX \$50	Status System PBX Reports Logout
Network Preferences	Reset and Reboot Options
LAN Settings	Reboot System
DHCP Server	
VLAN Settings	Reboot System
VPN Settings	Warning: Rebooting the system will terminate all active calls!
DDNS Settings	Reboot
Static Route	Reset to Factory Defaults
Security Settings	Reset to Factory Defaults
Security Center	Warning: A factory reset will erase all configuration data on the system.
Firewall Rules	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.
IP Blacklist	Reset to Factory Defaults
AMI Settings	
Database Grant	
Alert Settings	
LDAP Server	
Storage Management	

Figure 3.7.7



# 3.7.8 Alert Settings

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

A	lert Settings			
	Attack Type	Phone Notification	E-mail Notification	
	IPATTACK	Yes	Yes	
	WEBLOGIN	Yes	Yes	P



#### 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.

XonTe	e				S	50 IP-F	PBX Us	ser Manua	al	
Call Logs										
Search Condition	May 2016		End Date	9 Jun 2016	Caller/C	allee:		Trunk:	All	~
Duration <sup>(1)</sup> :		E	Billing Duration 🛈		s	tatus: All	~	Communication Type:	All 🔍 Start Sear	∽ ching
2 Download the reco	ords 📉	Delete the	e records			5		Total: 86	Show:1-25	View: 25 ~
Time	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Typ	e Account C	ode
2017-11-11 17:23:57	307	90920351		24759014	54	51	ANSWERED	Outbound		×
2017-11-11 16:52:57	307	90920351		24759014	59	57	ANSWERED	Outbound		$\times$
2017-11-11 14:11:31	301	96669099		24759014	39	36	ANSWERED	Outbound		×
2017-11-11 14:09:51	301	99646741		24759014	15	13	ANSWERED	Outbound		×
2017-11-11 14:09:37	301	99646741		24759014	10	8	ANSWERED	Outbound		×
				A.7500.1.1	~~			<b>.</b>		

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.



← → C 🗋 192.16	3.5.150/cgi/WebCGI?1000			(영무☆ 💁 🖸
KonT		Status	System PBX	Reports Logo
Line Status	System Info			
Extension Status	General ☆			
Trunk Status	Product Type: IP-PBX S50			
System Status	Hardware Version:			
System Info	V4.20 0000-0000			
Network Status	Firmware Version: 70.19.965.413			
	Uptime: 2:16:05 up 8 days, 30 min			
	Disk Usage 佘			
	Note: If there is not enough disk space on the system, the older Disk Usage:	st voicemail messages, call record files and call log files wi	Il be automatically deleted as	necessary.
	Used/Total(1K-blocks) use% flash: 8432/393216 2%			
	Memory Usage ☆			
	Memory Usage:			
	Used/Total(1K-blocks) use% Mem: 143212/254408 56%			

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.



General       Other Settings         General       Type: SIP ▼ Extension ●: 301 Password ●: Gulfsip123         Name ●: 301       Caller ID ●: 301 Register Name ●: 301         Voicemail       ✓         ✓ Enable Voicemail ● Voicemail Access PIN # ●: 301         Mail Setting         □ Enable Send Voicemail ●         Voicemail         ✓ Enable Voicemail ●         Voicemail         ✓ 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 ●: s         VolP Settings         NAT ●:        Qualify: ✓         Enable SRTP ●: □         Transport: UDP ▼       DTMF Mode ●: RFC2833 ▼         Register Remotely ●: □	it Extension - 301		
General       Type:       SIP       Extension 1: 301       Password 2: Gulfsip123         Name 1: 301       Caller ID 1: 301       Register Name 1: 301         Voicemail       Voicemail Access PIN # 1: 301         Mail Setting       Enable Voicemail Voicemail Access PIN # 1: 301         Mail Setting       Enable Send Voicemail 1: Settings         Mail Setting       Enable Send Voicemail 2: Settings for Voicemail 7: Settings for Voicemail 7: Settings for Voicemail 7: Settings for Voicemail 7: Settings 1: Settings 1: Settings 1: Setting	General Other Settings		
Type:       SIP       Extension 0:: 301       Password 0:: Gulfsip123         Name 0::       301       Caller ID 0:: 301       Register Name 0:: 301         Voicemail       ✓       Image: Sip: 301       Register Name 0:: 301         Voicemail       ✓       Voicemail Access PIN #0:: 301         Mail Setting       Image: Sip: 301       Image: Sip: 301         Note:       Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.         Group       Fickup Group 0::       1         Pickup Group 0::       1       Image: Sip: 301         Call Duration Setting       Image: Sip: 301       Image: Sip: 301         Max Call Duration 0::       s       Image: Sip: 301         VoIP Settings       Image: Sip: 301       Image: Sip: 301         NAT 0::       Image: Sip: 301       Image: Sip: 301         Image: Sip: 301       Image: Sip: 301       Image: Sip: 301         Image: Sip: 301       Im	General		
Name •: 301       Caller ID•: 301       Register Name •: 301         Voicemail       Voicemail Voicemail Access PIN #•: 301         Mail Setting       Instant Access PIN #•: 301         Mote: 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 ~         Pickup Group •: 1 ~       Instant Access PIN #•: Instant Access PIN #•: Instant Pickup Group •: 1 ~         Call Duration Setting       Instant Pickup Group •: 1 ~         Max Call Duration •: Instant Pickup Group •: Instant	Type: SIP 🗸 E	xtension 0: 301	Password 0 : Gulfsip123
Voicemail       ✓         ✓       Enable Voicemail         ✓       Save         ✓       Save	Name 🛈 : 301	Caller ID : 301	Register Name 🛈 : 301
☑ Enable Voicemail ●       Voicemail Access PIN #●: 301         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 ●:	Voicemail		
Mail Setting         □ Enable Send Voicemail         Email Address         Email Address         Image: Send Voicemail         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         Pickup Group         Image: Send Voicemail Settings         Call Duration Setting         Max Call Duration         Image: Send Voicemail Settings         VolP Settings         NAT         Image: Send Voicemail Settings         Image: Send Voicemail Settings         Image: Send Voicemail Settings         Image: Settings         Image: Send Voicemail Settings         Image: Setings         Ima	🗹 Enable Voicemail 🛈 🛛 Voice	mail Access PIN # 🛈 : 301	
□ Enable Send Voicemail         Email Address         Email Address         Image: Setting Setti	Mail Setting		
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 VolP Settings NAT : 2 Qualify: 2 Enable SRTP : Transport: UDP DTMF Mode : RFC2833 ~ Register Remotely : Save Xave Cancel	Enable Send Voicemail		
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 <sup>(1)</sup> : 1 ~         Call Duration Setting         Max Call Duration <sup>(1)</sup> : s         VolP Settings         NAT <sup>(1)</sup> : 1 ~         DTMF Mode <sup>(1)</sup> : RFC2833 ~         Register Remotely <sup>(1)</sup> : 1         Save	Email Address 🛈 :		
Group Pickup Group Call Duration Setting Max Call Duration VolP Settings NAT Call Duration Call Dur	Note: Please ensure that the se	ction 'SMTP Settings for Voicem	ail'(in the 'Voicemail Settings') have
Group       Pickup Group       :       1       ✓         Call Duration Setting       Max Call Duration       :       :       s         VolP Settings       NAT       :       ✓       Enable SRTP       :       :         Transport:       UDP       DTMF Mode       :       Register Remotely       :       :         ✓       Save       X Cancel       X Cancel       X Cancel	Deen property conligured before	using this leature.	
Pickup Group   Call Duration Setting   Max Call Duration   Max Call Duration   Image: Setting setti	Group		
Call Duration Setting Max Call Duration VolP Settings NAT : Qualify: Qualify: Enable SRTP : Transport: UDP DTMF Mode : RFC2833 Register Remotely : Cancel	Pickup Group		
Max Call Duration       s         VolP Settings       NAT         NAT       :       Qualify:         Transport:       UDP       DTMF Mode         Image: Save       Image: Save	Call Duration Setting		
VoIP Settings         NAT <sup>1</sup> :        Qualify:        Enable SRTP <sup>1</sup> :          Transport: UDP        DTMF Mode <sup>1</sup> : RFC2833        Register Remotely <sup>1</sup> :	Max Call Duration 🛈 :	s	
NAT <sup>①</sup> :        Qualify:        Enable SRTP <sup>①</sup> :          Transport: UDP ∨       DTMF Mode <sup>①</sup> : RFC2833 ∨       Register Remotely <sup>①</sup> :	VolP Settings		
Transport: UDP V DTMF Mode (): RFC2833 V Register Remotely (): C	NAT 🛈 : 🖂	Qualify: 🗹	Enable SRTP 🛈 : 🗆
Save Save	Transport: UDP 🗸	DTMF Mode : RFC2833 V	Register Remotely 🛈 : 🗆
		🖌 Save 🔀 Can	cel

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.





← → C 192.168.5.150	(승무 ☆) 💁 💿 🚍
	IPBX Configuration Panel
P-PBX 550	User Name 🛕 301 Password 💩 📲 Language 📿 English
	Login Reset

Figure 4-2

# 4.3 Voicemail

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

← → C 🗋 192.168.5.150	/cgi/WebCGI?1000				(영 위 ☆ 💁 🙆
KonTel IP-PBX S50					Logout
General Settings	Voicemail				
Voicemail	Folder: Urgent V	K Delete	Move to Folder: Urgent •		Total: 0 Show: 0 - 0
Call Recordings		Caller ID	Time	Duration	Options
Voicemail Settings					< <prev next="">&gt;</prev>
Settings					

Figure 4-3

# 4.4 Record

Users can play, delete and download recorded files here.



← → C 🗋 192.168.	5.150/cgi/WebCGI?1000				🅲 🕈 🔂 💺 🙆
KonTe	50				Logout
General Settings	Call Recordings				
Voicemail	X Delete				Total: 0 Show: 0 - 0
Call Recordings		Caller ID	Time	Duration	Options
Voicemail Settings					< <prev next="">&gt;</prev>
Settings					

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.

← → C 🗋 192.168	.5.150/cgi/WebCGI?1000				🅲 🕈 🏠 💁 🧰
KonTe	50				Logout
General Settings	Call Recordings				
Voicemail	K Delete				Total: 0 Show: 0 - 0
Call Recordings		Caller ID	Time	Duration	Options
Voicemail Settings					< <prev next="">&gt;</prev>
Settings					



# 4.6 Settings

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

← → C 🗋 192.168.5.15	0/cgi/WebCGI?1000	🐸 🕈 😒 🔤
KonTel IP-PBX \$50		Coput
General Settings	Settings	
Voicemail	Follow Me	
Call Recordings	Always	
Voicemail Settings	Follow Me : C No Answer	Voicemail Transfer To :
Settings	🗹 When Busy	Number
	Mobility Extension	
	Mobility Extension Number : Outbound Prefix:	C Ring Simultaneously 🚱
	Other Options	
	Call Waiting Ring Out : 30	
	Save	Cancel

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.



← → C 🗋 192.168.5	.150/cgi/WebCGI?1000							<mark>٣ ٦</mark>	<b>a</b> O
KonTe	0				Status	System	PBX	Reports	Logout
Reports	Call Logs								
Call Logs System Logs	Search Condition Start Date: 10 Dec 20 Duration : Account Code:	Billing	End Date: 10 Dec 2011 Duration	Caller/Callee Status	All T	Communic	Trunk: All ation Type: All	T T	
	Download the records	Delete the reco	rds Destination Trunk	Duration Bil	ling Duration State	is Commu	Total: 0 S	Show:0 View:	25 <b>v</b>
							< <prev next="">&gt; Pa</prev>	age : 1 / 0	Goto

Figure 5-1

# 5.2 Password Settings

Change the password for cdr user.

Change Password	
Enter Old Password:	
Enter New Password:	
Retype New Password:	
	Save

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.

Route	Name <sup>(1)</sup> : pstnout		
I	Password:	<ul> <li>PIN Settings</li> </ul>	
T.38 S	Support <sup>©</sup> : No	~	
Rrmemo	ry Hunt🛈 : No	~	
Off	ce Hours :	~	
Dial Patterns 🛈			
Dial Pattern	Strip	Prepend	
9.	1	Prepend	×
🕂 Add			
Member Extensions			
Available Extensions	3	Selected	
	^ >>>	300(SIP) 301(SIP)	^
		302(SIP) 303(SIP)	
		304(SIP)	
		305(SIP)	
	~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~	300(SIP) 307(SIP)	
-		Corr(cm)	
Member Trunks 🛈			
Available Trunks		Selected	
22204086(SIP)	^	pstn1(FXO)	~
22204148(SIP)	>> >>	pstn2(FXO)	
KSA Second(SPX)			
ToUAE(SPX)			
	<del>~</del>		
	**		
	Y		~
	Save	X Cancel	

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  $\rightarrow$  VoIP Trunk  $\rightarrow$  SIP Trunk Enter your account information on this page, and click Save.

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 0:		
Maximum Channels 🛈 :	0	
Caller ID 🛈 :	22204086	
Realm0:		
	Enable Outbound Proxy Ser	/er
Codecs :	First: 🛛 a-law 🔍 Secon	d: u-law 🗸 Third: 🕻
	Fourth: None 🗸 Fift	h: None 🗸
Transport:	UDP 🧹 Enable SRTP	: Qualify:
DTMF Mode:	rfc2833 v	
DOD Settings		

Associated Extension: 300 🗸

🗸 Save

↑Add DOD

💢 Cancel

Figure6-2

†Add Bulk

#### <u>Manu</u>al Х

 $\sim$ 

DOD:



#### 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.

d Outbound Route			
	Route Name 🛈 : 🔽	ipout	
	Password:	PIN Settings	
	T.38 Support 0 : N	0 ~	
,	Rimemony Hunt	0 ~	
'	Office Hours :	<u> </u>	
	Onice Hours .	~	
Dial Patterns V	<b>0</b> 4-i-	Descent	
Dial Pattern	Strip	Prepend	
0.	1	Prepend	×
Add			
Member Extensions 🛈			
Available Ex	tensions	Selecte	d
		300(SIP)	^
		»» 301(SIP)	
		302(SIP)	
		303(SIP)	
		← 304(SIP)	
		305(SIF)	
	~	«« 300(SIP)	
		307(017)	•
Member Trunks			
Available	Irunks	Selecte	d
pstn1(FXO)	~	22204086(SIP)	~
pstn2(FXO)		>>>>	
22204148(SIP)			
KSAbranch(SPS)		→	
KSA_Second(SPX)			
ToUAE(SPX)		-	
		× ×	
	×		× .
	🗸 Sav	e 🔀 Cancel	
		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

lit IVP wolcomo			
IILTVK - WEICOITIE	0		
	Number <sup>(1)</sup> : 660		
	Name <sup>(1)</sup> : welcon	ne	
	Prompt <sup>(1)</sup> : welcon	ne <ul> <li>Custom Prompts</li> </ul>	
	Repeat Count 🛈 : 🛛 🗸		
	Key Timeout 🛈 : 🏾 🗸		
	🗹 🛈 Enable	Direct Dial	
Keypress Events			
Key	Action	Destination	
0	Connect to Extension	✓ Extension 300 ✓	
1	Connect to Extension	✓ Extension 301 ✓	
2	No Action	✓ Extension 300	
3	No Action	<b>~</b>	
4	No Action	✓	
5	No Action	✓	
6	No Action	<b>~</b>	
7	No Action	✓	
8	No Action	~	
9	No Action	<b>~</b>	
#	No Action	<b>~</b>	
*	No Action	✓	
Timeout	Connect to Extension	V Extension 300 V	

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.

Add Inbound Route			Х
General			
Rou	ite Name 🛈 : allin		
DID	Number 🛈 :		
E	xtension 🛈 :		
Caller ID	Number 🛈 :		
Distinctive F	Ringtone 🛈 :		
Enal	ble Callback : No 🤍	Callback Settings	
Member Trunks			
Available Trunk	(5	Selected	
	×× + + ×	pstn1(FXO) pstn2(FXO) 22204088(SIP) 22204148(SIP)	~
Business Davs			
Office Hours : d	lefault 🗸		
Office Hours Destination :	VR 🗸	IVR welcome	~
Non-office Hours Destination :	VR 🗸	IVR welcome	~
During Holidays			
Holiday :	~		
Destination : E	End Call 🗸		~
Fax Detection			
Destination :	lo Detect 🗸 🗸		~
	🖌 Save 🔀 C	Cancel	

Figure6-5



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