

IP PBX GLIDERVOX ZX-30x

User's Manual

V1.2



TABLE OF CONTENTS

Chapter1 Brief Introduction	3
Chapter2 Safet y Notice	4
Chapter3 ZX30x Specification	5
3.1 Apearance&Model	5
3.2 System Features	5
3.3 Interface&Panel	6
3.4 Default configuration	8
3.5 Default Feature Key	8
Chapter4 Login in Home Page	9
Chapter5 Basic Configuration	11
5.1 Configure Extensions	11
5.2 Trunk	12
5.3 Outbound Routers	14
Chapter6 Inbound Control	16
6.1 Inbound Routers	16
6.1.1 General Settings	16
6.1.2 Zap Channel DIDs	17
6.1.3 VoIP Channel DIDs	18
6.2 IVR (Interactive Voice Response)	19
6.3 IVR Prompts	19
6.4 Ring Groups	20
Chapter7 Advanced Configuration	22
7.1 Options	22
7.2 Voice mail	22
7.2.1 General Settings	23
7.2.2 SMTP settings	24
7.2.3 Email settings	25
7.3 Conferencing	25
7.4 Music On Hold	26
7.5 Music On Ringback	27
7.6 Call Parking	27
7.7 DISA Settings	28
7.8 Follow Me	29
7.9 Paging and Intercom	30
7.10 Monitor	31
7.11 Time Based Rules	32
Chapter8 S tatus Display	33
8.1 Monitor List	33
8.2 Call Logs	33
8.3 Register Status	34
8.4 System Info	34



Chapter9 S ystem Management	35
9.1 Network and Country	35
9.2 DDNS&VPN	35
9.2.1 DDNS Settings	35
9.2.2 VPN Settings	36
9.3Time Settings	36
9.3.1 NTP Settings	37
9.3.2 Manual Time Settings	37
9.4 Management	37
9.5 Backup	38
9.6 Upgrade	38
Chapter10 Operating Instruction	40
10.1 How to link the $ZX30x$ IP PBX to the interwork	40
10.1.1 IP PBX behind the Router	40
10.1.2 IP PBX behind the Modem	40
10.2 How to log in the IP PBX system	41
10.3 How to make a internal call	42
10.4 How to make an outbound call	43
10.4.1 Make call via GSM trunk	43
10.4.2 Make call via VoIP trunk	45
10.5 How to make an incoming call	47
10.6 How to Set an incoming call to IVR based time rule	47
10.7 How to link two $ZX30x$ IPPBX in the same network	
10.8 How to link two IPPBX in different network	54
10.9 How to resolve problems about hearing only on one side	56
Chapter11 How to use Skype account in $ZX30x$	58
11.1 Register for Skype Manager	58
11.2 Create a SIP Profile and buy a Channel Subscription	58
11.3 Allocate Skype Credit to the SIP Profile	59
11.4 Configure your Skype for SIP certified PBX for outbound calls	60
11.5 Make an outbound call	61
11.6 Configure your Skype for SIP certified PBX for inbound calling	61
11.7 Set up a business account to test inbound calls from people with Skype	61
11.8 Make a test inbound call from Skype	
11.9 Assign an Online Number to receive calls from landlines and mobile phones	3.62
11.10 Make a test inbound call from a landline or mobile phone	62



Chapter1 Brief Introduction

Thank you for your pur chasing the ZX30x series of IP PBX.The all-in-one ZX30x IP PBX can not only provide the traditional basic PBX features(call hold, call forwarding, call waiting and so on) as well as enhanced features such as visual voice mail, music on hold, and auto attendant. In addition, the ZX30x IP PBX supports innovative functionality like private VoIP networking, remote access, superior VoIP voice quality with ad vanced audio processing, and the revolutionary ability to traverse a NAT and firewall. With VoIP solutions, SMEs can quickly deploy V oIP networks to connect multiple branch locations over the Internet without the need to change the current equipment or dial plan. By using the ZX30x IP PBX, an S ME can t ake advant age of the V oIP services provided by the ITSPs(Internet T elephony Service Provide rs) or traditional telephony services, reduce intra-company telephon y expenses, and allow VoIP remote a ccess anywhere via the internet.



Chapter2 Safety Notice

Please read the following safety notices before installing or using this IP PBX. They are crucial for a safe and reliable operation of the device.

- Please use the external power supply which is included in the package. Other power supplies may cause damage to the device, affect the performance or induce noise.
- Before using the external power supply in the package, please check with residential power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, otherwise, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not place the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposing the device to high temperat ure, below -10°C or high humi dity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling to the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock or breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this device in an ill-ventilated place.
- You are in a situation that could caus e bodily injur y. Before you wo rk on any equipment, be aware of the hazar ds involved with electric al circuitry and be familiar with standard practices for preventing accidents.



Chapter3 ZX30x Specification

3.1 Apearance&Model

ZX-304 Series IPPBX product line include **ZX304-A4, ZX308-A8,ZX304-AG4,ZX30-AG42**, **ZX30-G4**, so far, since they have almost the same software and structure so we will use ZX304-AG42 as the demo unit on this article.

M	lodel	FXS	FXO	GSM	E1
ZX304-A4	A404		4		
ZA304-A4	A422	2	2		
ZX304-A8	A808		8		
28304-80	A826	2	6		
ZX304-AG42	AG4204		4	2	
ZX304-AG42	AG4222	2	2	2	
ZX304-G4	G4			4	
ZX304	AE4 104		4		1
2/304	AE4 122	2	2		1

3.2 System Features

ZX30x se ries of IPPBX is an embedded ip pbx b ased on industry st andard for Home&SMEs, which is not only a PBX, but al so as a voice mail Server , IVR server , conferencing server. With excellent echo cancellation function, it can meet most of the customers' requirement.

- Up to 30 concurrent calls.
- Above 100 registers
- Configuration by Web
- Built-in SIP/IAX Server
- Build in Voice Mail Server
- Codec: G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX
- SIP/IAX Extensions(connect with IP Phone)
- Zap Extensions(connect with Analog Phone
- Call Forward/Call Hold/Call Transfer/Call Waiting/Caller ID
- Flexible Dial Plan
- Ring group
- Conference Room
- IVR and Auto attendand
- Multimedia Music On Hold and Ring Back



- Call Monitoring
- Video Call
- DISA setting
- Call parking
- Call Paging and intercom
- Follow me Setting
- Call Logs check and download
- Support IP Phone with Key function
- BLF(Busy Lamp Field)
- Static/DHCP/PPPoE network access
- System backup and store
- Set system time manually
- VPN Client (support N2N)
- DDNS Client (support Dyndns.org)
- Codec Negotiation/Echo cancelation/VAD.etc
- FAX T.38

3.3 Interface&Panel

Here, we take ZX30xG4 as the sample to show the interfa ce and the i ndicators at the back and frond panel.

1) Back panel



- 4 * GSM Antenna
- 2 * Network Interface (RJ45)
- 1 * Power port (DC 12V 2A)
- 1 * Reboot Button
- 2) Frond Panel





Mark	Function	Status	Description
PWR	Power Status	On	Power On
FVVR	Power Status	Off	Power Off
SYS	System Status	On	System working
313	System Status	Off	System Failed
WAN	WAN interface Status	Wink	Data exchanging
VVAIN	WAN INTENACE Status	Off	No Data exchanging
LAN	LAN Interface Status	Wink	Data exchanging
LAN	LAN IIILEHACE SIAIUS	Off	No Data exchanging
G1~G4	GSM Modules Status	Red	GSM channel
61~64	GSW Would's Status	Off	Failed
		Green	FXS channels
*1-4	Analog Modules Status	Red	FXO channels
		Off	Failed

3) Hard ware

- 32bit embedded RISC DSP
- 1G Onboard Nand Flash
- 128M Onboard SDRAM

4) envir onmental requirements:

- temperature: -10 °C -45 °C
- Storage temperature: -30 °C -65 °C
- humidity: 10-80% no dew
- Power: AC 100~240V

5) Packing List

- IPPBX 1 Unit
- GSM Antennas 4 Unit
- Power Adapter 1 Unit



3.4 Default configuration

- 1. WAN port IP address: http://192.168.1.100:9999
- 2. LAN port IP address: http://192.168.10.100:9999
- 3. LAN port super IP: 169.254.1.254/255.255.0.0
- 4. Web GUI username: admin
- 5. Web GUI password: admin

3.5 Default Feature Key

- 1. Press '**11' Playback the IP Address of WAN port
- 2. Press '**12' Playback the IP Address of LAN port
- 3. Press '600' Get into the Voicemail Box
- 4. Press '900' Get into the Meeting
- 5. Press '#' Blind Transfer
- 6. Press '*2' Attended Transfer
- 7. Press '*' Disconnect Call



Chapter4 Login in Home Page

After connecting the IP PBX to the local area network. Launch the web browser on a computer which is in this local area network. Enter the IP address for the system (W AN port IP address http://192.168.1.100:9999, LAN port IP address http://192.168.10.100:9999). The start web page will appear like this:



Enter Username and Password (default username is **admin**, password is **admin**), set Language, then click "login". Once the login is successful, the home page will be display: **Noted:** you have to add a network segment same with the WAN ports if your PC is not at 192.168.1.XXX.

		Activate Changes Logo
lome	Home	Move the mouse over to a field to see tooltips
Basic		
Inbound Control	System Info	
Advanced	Network	
tatus	WAN IP 192.158.1.100	
ystem	LAN IP 192.168.10.100 Storage	
	Flash Total: 1016.0M Used: 3.8M Ext Disk Total: N/A Used: N/A	
	Analog Chancels	
	1 2 3 4 5 6 7 8 CSM N/A CSM N/A FXO FXO FXS FXS	
	Device Info	
	ZX30 IP PBX	
	Model No.: ZX30-AG42 System Version: v1.0	
	Run Time:3 min Refresh Reboot Factory Defaults	



With the PBX GUI, you can configure extensions, conference, voicemail, Dial Plan and etc. Each page of the GUI has three columns:

The left column present all the options t ab that you can program the system. Click the tab to go to setting page of different options.

The middle column contains the primary content for each page.

The right column of the user interface contains T ooltips. This area provid es brief description for any options of the GUI

The home page is used for logout, Reboot and Factory Defaults.

- Logout: To log out the PBX GUI.
- Reboot: Reboot the IP PBX system
- Factory Defaults: Restore all settings to factory default.
- Activate change: Made the chang e active for the curr ent configuration after you make a configuration on some page.



Chapter5 Basic Configuration

5.1 Configure Extensions

Click the Extension tab and you will see the extensions setting, your created users are in this page. There are 30 users in your extensions list as default setting, you can add new extensions or remove the existing extensions.

			Activate Changes Logout
Home	Extension Configuration		DTMFMode: The Dual-Tone Multi-Frequency mode to be
Basic	Extensions:	Extensions Setting:	used is specified here and can be changed if necessary. The
Extensions	801 -	Extension:	default is rfc2833.
Trunks	802 803	Name:	
Outbound Routes	804 805	Password:	
Inbound Control	806 807	Caller ID:	
Advanced	808	VM Password:	
Status	809 810	E-mail:	
System	811 812	Analog Phone: No Analog lines detected.	
	813 814	Dial Plan: Dial Plan1 -	
	815 816	Advance Options:	
	817		
	818 819	Г SIP Γ IAX	
	820 821	☐ Call Waiting ☐ 3-Way Calling	
	822 823	I NAT Pickup Group 1 -	
	824	Delete VMail DTMFMode RFC2833 =	
	825 826	☐ Video Support	
	827 828 -	Codecs Configure:	
	New Delete		
		Save (tame)	

Extensions Setting include:

- Extension The extension is assigned to the defined user.
 - Name The full name of the individual assigned to this extension.
- Password The password is used to Extension registered
- VM Password
 The password is used to access voicemail for the specified Extension
- E-mail Set the user's E-mail
- Caller ID Identifies the Caller ID presented when the listed extension dials out
- Analog Phone A drop-down menu is available to identify the analog phone port which this extension will access.
 - Dial PlanYou can choice dial plan based on the extensions' need, this
option references the Dial Rules option on the left tool bar.

There are also several advanced extension options available. The advanced options establish the connections from the listed extension to ot her systems within the IPPBX system server. These advanced options include the following:

- Voicemail The extension support voicemail
- SIP The extension support SIP protocol
- IAX The extension support IAX protocol
- Call Waiting The extension support Call Waiting function
- 3-Way Calling The extension support 3-Way Calling functions



- Pickup Group Select pickup group of the extension
- Delete VMail If this option is set, then voicemails will not be checkable using a Phone. Messages will be sent via e-mail, only. Note: You need to have an smtp server configured for this functionality.
- Video Call Enable/Disable Video support function for this extension.
- Codecs Click here, you can set the extension's codec (default support: alaw, ulaw and G.729).

5.2 Trunk

If you want to make external call, you must register with a Trunk in order to connect to the Public Switched Telephone Network (PSTN) or other VoIP service provider. Through the web page you can add a trunk.

There are three Trunk categories: Analog Trunk, VolP Providers, Peer.

	Add T	runk	Х
Provider Type: Analog Trunk C Custom Trunk P Peer	Lines:	Analog Port #1 Analog Port #3 Analog Port #5 Analog Port #6	

Analog Trunk

Select the Analog ra dio button to define the analog ports you have access to as a service provider. This will give you the ab ility to place calls thr ough the IP PBX utilizing analog lines. The analog port s available will be displayed when you select this option. Choose one or more analog ports by selecting their associated checkbox. You will not be able to create an analog service provider if you don ot have any analog port s available.

Custom Trunk

The Custom VoIP option allows you to create a custom VoIP definition. To create the custom VoIP provider definition you will need to complete the following:

	Description:
Provider Type: Apples Type:	Protocol: SIP 🔻
C Analog Trunk	DialPlan: default 🝷
Custom Trunk	Register:
C Peer	Host:
	Without Authentication
	Username:
	Password:

- Description The description should be used as the name of the custom VoIP definition
- Protocol Specify either a IAX or SIP protocol
- DialPlan Select a DialPlan for this trunk.
- Register Enable/Disable server register. Registering is not required for all providers



- Host The IP address of your service provider
- Username The user name associated with your provider account
- Password The password associated with your provider account
- Without Authentication if you connect to Voip server without Authentication, pls selected this.

Peer

The Peer option allows you to create a custom VoIP Peer.

	Add Trunk	Х
Provider Type: C Analog Trunk C E1 Trunk C VoIP Trunk Peer	Peer Name: Protocol: SIP DialPlan: default Host: dynamic NAT: Without Authentication Username: Password:	

- Peer Name Defines a peer name for this peer.
- Protocol Specify either a IAX or SIP protocol
- DialPlan Select a DialPlan for this peer
- Host dynamic | hostname | IP Address
- NAT Disable/Enable the NAT function
- Without Authentication if you connect to the PBX without Authentication, pls selected
- Username Defines the peer username
- Password Defines the peer password

Once you have added a VoIP Trunk it will appear on the list of Trunk on the Trunk page. There is an Options dr op-down list asso ciated with each Trunk listing. The Options drop-down list allows you to edit or delete the Trunk definition, as well as further refine the definition by choosing sever all ad vance options. Select eith er Cod ecs or Advanced to further refine the definition.

- Edit Edit you select the trunk.
- Codecs Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the internet.
- Advanced The following advanced options are available to further refine your trunk.

Advanced Settings			
trunkname:	trunk_1		
insecure:	very		
port:	5060		
caller ID:			
fromdomain:	192.168.1.100		
fromuser:	test		
contact:			
qualify:	yes		

• Trunkname Specify a trunk name if you want to refer to the service provider



- Insecure definition as something other than specified in Comment
 Insecure This option specifies how connects to a service provider (host) should be handled. Valid options are very/yes/no/invite/port. (Default is "very")
 Port The register request is sent through the port. (Default is SIP:5060,IAX:4569)
- Caller ID The caller ID will be set to the value specified in this field
- Fromdomain Sets default from: domain in SIP messages when acting as a SIP client.
- Fromuser Sets default from: user in SIP messages when acting as a SIP client
- Contact Specifies a primary extension for call routing

5.3 Outbound Routers

The Dial Rules t ab on the left to olbar allows you to u se basic p attern matching to differentiate outbound calls and route them accordingly (create different DialPlan).

					Activate Changes	Logout
Home	Outbound Routes				Ex: Delete 1 digits t front and prepend 2	from the
Basic	List of DialPlans:			1	dialing	50 belore
Extensions		Delete				
Trunks Outbound Routes	DialPlan1 - New	Delete				
Inbound Control	List of Dial Rules: Add	a Dial Rule				
Advanced	S.No RuleName Dial Pattern		all Using	Options		
Status	1 Call_GSM Begins with 9 and followed by mo digits	re than 3	Ports 1,3	Edit Delete		
System						

Click on Add a Dial Rule to define a new DialPlan. The following dialog will be displayed.



Home	Outbound Routes		Ex: Delete 1 digits from the front and prepend 256 before
Basic	List of DialPlans:		dialing
Extensions			
Trunks	DialPlan1 - New Delete		
Outbound Routes			
Inbound Control	List of Dial Rules:		
Advanced	S.No RI	X ns	
Status	1 Ca Rule Name: Call_GSM	slett	
System	Place this call through : Ports 1,3 🔹		
	PIN Set: Dialing Rules : If the number begins with 9 and followed by (more than) 3 digits (define a custom pattern) Delete 1 digits from the front and auto-add digit before dialing		

A DialPlan is comprised of the following items:

- Rule Name Set a rule name
- Place this call through Select a Trunk through which the call should be made
- Failover Select a trunk Failover
- PIN Set Set a password when you dial base the Dial rule.
- Dialing Rules The Dialing Rule gives you the ability to use basic pattern matching to differentiate calls and route them accordingly. For instance, if a number begins with 9256 followed by 7 or more digits, that would define a call within the state of Alabama. If a call began with 9 followed by 7 digits, it would be a local call that probably did n't require a long dist ance charge. Instead of adding a rule for every extension or phone num ber you call, specify the pattern in this rule similar to the example.
- Define a custom pattern Set a custom pattern by yourself.

Custom Pattern:	_9XXX.
	(define a Basic Pattern) Z Any digit from 1 to 9
	 N Any digit from 2 to 9 X Any digit from 0 to 9 Any number of additional digits

- N Any digit from 2 to 9
- Z Any digit from 1 to 9
- X Any digit from 0 to 9
 - Any number of additional digits

Example: "_9ZNXXX." mean first number is 9, second number is any digit from 1 to 9, third number is any digit from 2 to 9 and each "X" is any digit from 0 to 9. The "." is more.

 Delete This option gives you the opportunity to remove specified digits from the call being dialed and r eplace them with the digits needed to make the call. You can also prepend digits to the beginning.



Chapter6 Inbound Control

6.1 Inbound Routers

6.1.1 General Settings

		Activate Changes	Logout
Home	Inbound Routes	Move the mouse ov field to see tooltips	er to a
Basic			
Inbound Control	General Zap Channel DIDs VoIP Channel DIDs		
Inbound Routes	List of Incoming Call Rules		
IVR	S.No Incoming Rule Options		
IVR Prompts	State Interming Kale Options 1 1,2,3,4,5,6' to 'TimeRule – Time Based Rules' Edit Delete		
Ring Groups	1 1,2,3,4,5,6' to 'TimeRule Time Based Rules'		
Advanced			
Status			
System			
	And the Control of th		
	Add an Incoming Rule		

The same pattern-matching logic used for processing outbound calls can also be employed for inbound calls. The two defaults define routing based on whether a n incoming call matches or doesn't match a pattern you define.

from Trunk Ports 1,3 Destination working time IVR Menu	Route A	All Unmatched incoming calls 👻
Destination working time IVR Menu 🔻		
	Destination	working time IVR Menu 🔻

There are only a few options you need to configure

- Route Make a selection from the drop-down list to choose how the calls will be routed. You can select from All Unmatched Calls or Calls which Match
- From Provider Select from the list of providers which you previously configuration
 - To Extension The previously configuration extension which should receive the Call.



6.1.2 Zap Channel DIDs

	Saved Successfull!	Activate Changes	Logout
Home	Zap Channel DIDs	Move the mouse ov field to see tooltips	er to a
Basic	Council Zee Channel DID: USID Channel DID:	neiu to see toorupa	
Inbound Control	General Zap Channel DIDs VoIP Channel DIDs		
Inbound Routes	List of Zap Channels		
IVR	S.No Channel DID Extension Options		
IVR Prompts	1 1 806 Edit Delete		
Ring Groups			
Advanced			
Status			
	Adıt Zap Channei		
nis page used	to set Zap channel DID.		
	Add Zap Channel	x	
	Channel: DID Extension:		
	Save Cancel		

• Channel

•

DID Extension

Set Inbound Zap channel.(eg: Use channel 1, you should set 1) Set a local extension



6.1.3 VoIP Channel DIDs

				Ac	tivate Changes	Logout
lome	VoIP Channel DIDs			Mor fiel	ve the mouse ov d to see tooltips	er to a
Basic	A silveral	a al lam	with all with the second	10	a to ove toortpo	
nbound Control	General	Zap Channel DIDs	VoIP Channel DIDs			
Inbound Routes		List of VoIP Cha	nnels			
IVR	S.No DID Number	DID Extension		otions		
IVR Prompts	1 _85337096	801		Delete		
Ring Groups		10-T	and and a second se			
Advanced						
Status						
		Add VolP Co	tauel			
s page used	to set Zap cha	annel DID.				
				Х		
	DID Num DID Exte	nsion:				
	DID Exte		ancel			

- DID Number Set a V oIP DID Numb er (The value, you can se t in adva nced settings of VoIP trunk).
- DID Extension Set a local extension



6.2 IVR (Interactive Voice Response)

Through the we b p age, you can create In teractive V oice Response (I VR). IVR are designed to allow for more efficient routing of calls from incoming callers.

			Activate Changes Logout
Home	IVR		Move the mouse over to a field to see tooltips
Basic	IVR Menu:	IVR Setting	
Inbound Control	IVR - working time	Name: working time Extension:	
Inbound Routes	IVR - closed time	Welcome Massage	
IVR			
IVR Prompts		Please Select: welcome -	
Ring Groups		Dial other Extensions?	
		Keypress' Events	
Advanced		Key Action	
Status		0 Goto Extension - 801 -	
System		2 Disabled V	
		3 Disabled V	
		4 Disabled -	
		5 Disabled 👻	
		6 Disabled 👻	
		7 Disabled 👻	
		8 Disabled 👻	
		9 Disabled -	
	New Delete	Save Caucel	
		STATES STATES	

Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

- Name Set a IVR name
- Extension Set a IVR connect number
- Welcome Massage Select a welcome massage voice from record
- Dial other Extensions Enable/Disable allow dial other extensions.

6.3 IVR Prompts

In the event that one wants to record custom IVR prompts for the IP PBX, which can be used in a IVR, the Record may be used.

				Activate Changes Logo	ut
Home	IVR Prompts(Record Voices For	Custom IVR)		Move the mouse over to a field to see tooltips	
Basic		List of Recorded voices		Their to ace tobility?	
Inbound Control	S.No Name		tions		
Inbound Routes	1 closed.gsm	Record Again	Play Delete		
IVR					
IVR Prompts	2 welcome.gsm	Record Again	Play Delete		
Ring Groups					
Advanced					
Status					
System					
		Record a new voice			



A list of previously recor ded menus is displayed. Here, t he user may modify sever al options

- Record Again Clicking this button allows the user to make another attempt at recording and replacing an existing custom sound file
- Play
 Clicking this button brings up a dialog entry box to allow the input of an extension that System will dial and play the prompt over
- Delete Clicking this button will delete the selected prompt

There are two options under "Record a new voice"

Record a new Voice	х
File Name: Extension used for recording:]

- File Name This text entry box specifies the saved name of the file that is to be recorded.
- Extension Used for Recording This drop-down select box allows the user to choose which extension will dial to wait for the user to speak the prompt

6.4 Ring Groups

A ring group is a group of users assigned to answer incoming call to a single extension. When a caller dials a ring group extension, all of the phones of the users in the ring group will ring together, the call is answered when any one of the users in the group pick up the call.You can configure Ring Groups through the web page



Define Ring Groups to Dial more than one extension



Add I	Ring Group			Х
Name:	Strategy	: Ring all	•	
	→ SIP, SIP, SIP, SIP, SIP, SIP, SIP,	'801 User1 '605 User2 '803 User3 '804 User4 '805 User5 '806 User7 '808 User8	▲ Ⅲ	
Ring Group Members		Available Chan	nels	
Extension for this	ring group(Opti	on) :		
Ring (each/all) for	these many se	conds : 20		
If not answered				
C Goto an Extension				
C Goto an Extension Voicem	nail			
C Goto a RingGroup				
C Goto an IVR menu				
HangUp				
Sav	e Cancel			

- Name Set a Ring Group name
- Strategy There is a drop-down list, you can choose Ring all or Ring in order.
- Ring Group Members Add Ring Group member from Available channels.

If the Ring Group no answered you can choose to Goto an Extension, Goto an Extension Voicemail, Goto a RingGroup, Goto an IVR menu,HangUp.



Chapter7 Advanced Configuration

7.1 Options

		Activate Changes Logout
Home	Options	Music On Ringback: Enable music on ringback.
Basic	Local Extension Settings	mase on migback
Inbound Control		
Advanced	Local Extensions are Varying 💌	
Options	Operator Extension :	
Voicemail	Global Ring Time Set: 30 (seconds)	
Conferencing	Music On Ringback: 🔽	
Music on hold	☐ Allow analog phones to be assigned to multiple extensions	
Music on Ringback	□ Allow extensions to be AlphaNumeric (SIP/IAX users)	
Call Parking		
DISA	Default Settings for a New User	
Follow Me	🔽 Voicemail 🗆 🗆 NAT	
Paging and Intercom	SIP TIAX	
Time Based Rules	🔽 Call Waiting 🕅 3-Way Calling	
Status	1234 VoiceMail Password 🖵 Delete VMail	
System	Save (Game)	

- Local Extensions are
- Operator Extension
- Global Ring Time Set
- Music On Ringback
- Default Settings for a New User

Set up the digit of local extensions Set up Operator Extension. (you can dial "0" go

- to the extension at any time)
- Set default each extension ring time.

Enable/Disable the Music On Ringback function Set up the Default Settings for a New User, when You create a new extension will use the configuration.

7.2 Voice mail

The ZX30x provid es Voice mail for it s end us ers as an optional f eature.End users can retrieve their voice mails and change thei r p assword.The relationship between the extension and the voice mail is established in the User Extension section of the GUI. You can configure the voicemail through this page.



7.2.1 General Settings

			Activate Changes Logout
Home	Voicemail Configuration		Minimum message Time: This select box sets the
Basic	General SMTP Setti	ngs Email Settings	minimum duration of a voicemail message. Messages
Inbound Control	General SMIP Setu	ngs Email Settings	below this threshold will be automatically deleted.
Advanced	VoiceMail Reference		automatically deleted.
Options	Extension for checking message	s: 600	
Voicemail	Max greeting (seconds)	60	
Conferencing	Direct to Voicemail:		
Music on hold	Dial '0' for Operator:		
Music on Ringback			
Call Parking	Voice Message Options		
DISA	Message Format:	WAV (16-bit) 🔻	
Follow Me	Maximum messages :	100 -	
Paging and Intercom	(per folder) Max message time;	5 minutes 👻	
Time Based Rules	Min message time:	1 second -	
Status			
System	Playback Options		
	Say message Caller-ID		
	☐ Say message duration		
	Play envelope		
	Allow users to review		
	Same	Cancel	

Standard configuration information is al so present, allowing you to confirm th e extension used to check messages as well as general parameters such as the following:

• Extension for Checking Messages This option defines the extension which Users

Max greeting(Seconds) With
 Attach recordings to a mail

this option, you specify the maximum amount of time available to record your

call in order to access their voicemail account.

- Attach recordings to e-mail
 Dial "0" for Operator
 voicemail greeting.
 Enable/Disable send recording file to you email by attachment
 Callers who are sent to voice mail can press "0"
 - Callers who are sent to voice mail can press "0" for the operator and be transferred either during the voice mail salutation, or after recording the message. If this option is not enabled, a caller's pressing "0" will be ignored.

There are several options that can be specified to define the voicemail message in the system.

•	Message Format	This option gives you the ability to choose the format in
		which messages will be mailed.

- Maximum Messages The maximum number of messages per voice mail box is set here.
- Maximum Message Time The maximum duration of a message left by a caller is set here
- Minimum Message Time The minimum duration of a message is dictated here.

There are several playback options that can be specified.

• Say Message Caller-ID The Say Message Caller ID option reads the caller ID before the voice mail message is played



- Say Message Duration This option identifies exactly how long the message lasted.
- Play Envelop
- Allow Users to Review

related to a voice mail. This option provides incoming callers the option to review their message before it is saved and can be played back by the owner of the voice mail extension. Standard options are presented to you, allowing you to discard the message or re-record it if you aren't happy with it.

The envelope provides the date, time, and caller ID

7.2.2 SMTP settings

		Activate Changes Logout
Home	Voicemail Configuration	Password: input password of your email.
Basic	General SMTP Settings Email Settings	you chief
Inbound Control	General SHIP Settings Email Settings	
Advanced	SMTP Settings:	
Options	Smtp server :	
Voicemail	Port : 25	
Conferencing	SSL/TSL : T	
Music on hold	Enable ssmtp Authentication	
Music on Ringback	Username : Password :	
Call Parking DISA	, as word .	
Follow Me	Save Cancel	
Paging and Intercom		
Time Based Rules		
Status		
System		
mtp server	The IP address or hostname of an SMTP may connect to, in order to send e-mail no	-
	•	
cemail;	eg:mail.yourcompany.com	
ort	The port number on which the SMTP servent of the se	ver is running; generally
SL/TSL	Enable use SSL/TLS to send secure mes	sages to server.
nable SMTP A	,	· •
	anable COMTD Authentication as t an	d configure th e fo llo
	enable SSMTP Authentication se t, and information	
sername	-	

• Password input password of your email.



7.2.3 Email settings

		Activate Changes Logout
Home	Voicemail Configuration	Attach recording to e- mail: This option defines
Basic	General SMTP Settings Email Settings	whether or not voicemails are sent to the Users' e-mail
Inbound Control	General SMIP Settings Email Settings	addresses as attachments.
Advanced	Template for Voicemail Emails	
Options	Burners states and	
Voicemail	✓ Attach recordings to e-mail	
Conferencing	Sender Name IPPBX Server	
Music on hold	From username@mailserver.com	
Music on Ringback	Subject you've a voicemail from \${VM_CALLERID}	
Call Parking	Message Dear \${VM_NAME}, you have a new voicemail from \${VM_CALLERID}, the message time is \${VM_DUR}.	
DISA		
Follow Me		
Paging and Intercom		
Time Based Rules		
Status		
System	Salve Cancel	
	Template Variables:\t: TAB \${VM_NAME} : Recipient's firstname and lastname \${VM_DAB : The duration of the voicemail message \${VM_MAILBOX} : The recipient's extension \${VM_MCALLEND} : The caller id of the person who left the message \${VM_CALLEND} : The caller id of the person who left the message \${VM_DATE} : The date and time the message was left	
ender Name rom Set	Set the name for sender the from email	

- Subject Set the email title
- Massage Input the matter in your email.

7.3 Conferencing

Every company reaches the point of needin g more people on a ca II than it can effectively include through three-way calling. conference bridges allow you to include more people as well as project a professional image.

		Activate Changes Logou
Home	Conference Room Configuration	Move the mouse over to a field to see tooltips
Basic	Conference Number	
Inbound Control		
Advanced	Room Extension: 901	
Options	Conference Password	
Voicemail	Conterence Password	
Conferencing	PIN Code: 1232	
Music on hold	Admin PIN Code: 2345	
Music on Ringback	Real Anna Article	
Call Parking	Conference Options	
DISA	Play hold music for first caller	
Follow Me	🔽 Enable caller menu	
Paging and Intercom	Announce callers	
Time Based Rules	Record conference Quiet Mode	
Status		
System	Save Lancel	

The configuration of the conference room and standard features is very straightforward.



The conference room use def ault extension 900, but you c an always change it to any extension number you w ant. After establishing the extension for the room, you n eed to specify the p assword settings for the c onference. Assign the PIN Co de used b y participants to enter the conf erence as well as the Administrator PIN Code used by the moderator of the conference to open the conference room.

7.4 Music On Hold

				Activate Changes	Logout
Home	Music on Hold			Move the mouse ov field to see tooltips	er to a
Basic	Music On Hold Reference	a			
Inbound Control					
Advanced		Music: music1 -			
Options		Save Causel NOH Re	alond		
Voicemail					
Conferencing	Upload Music File				
Music on hold		he Music File Name:	(*.asm)		
Music on Ringback		Note: Please use .gsm format v			
Call Parking		Server IP address:	once me.		
DISA		lect Music directory: music1			
Follow Me		iccertable an eccory. Inducer			
Paging and Intercom		Update			
Time Based Rules		opone			
Status					
ist of Music C	n Hold Dis	splay Music On I	Hold class list		
lass Se	t	Music On Hold	class name		
lusic	Se	lect music. (you	can replace mu	sic file throug	gh th
	up	date page.)			
nter The Mus	ic File Name	Set you want u	upgrade music fi	ile name	
FTP Server II	P address	Set the TFTP	server IP		

• Select Music directory Select directory that you want saved music file.



7.5 Music On Ringback

Basic Music C Inbound Control Advanced Options Voicemail Conferencing Music on Nold Music on Ringback Call Parking	Ringback n Hold Reference Music: music2 ~ Save Emicel MOH Reload	Move the mouse over to a field to see tooltips
Inbound Control Advanced Options Voicemail Conferencing Music on hold Music on Ringback Call Parking	Music: music2 👻	
Inbound Control Advanced Options Voicemail Conferencing Music on hold Music on Ringback Call Parking	Music: music2 👻	
Options Voicemail Conferencing Music on hold Music on Ringback Call Parking		
Voicemail Conferencing Music on hold Music on Ringback Call Parking	Save Francel MOH Reload	
Conferencing Music on hold Music on Ringback Call Parking		
Music on hold Music on Ringback Call Parking		
Music on Ringback Call Parking		
Call Parking		
and the second se		
1.0.000		
DISA		
Follow Me		
Paging and Intercom		
Time Based Rules		
Status		
System		

• Music Select a music for Music On Ringback

Notice: You must enable Music On Ringback function.(In Options Page)

7.6 Call Parking

		Activate Changes Log
Home	Call Parking Preferences	Move the mouse over to a field to see tooltips
Basic	Call Packing Reference	
Inbound Control		
Advanced	Extension to Dial for Parking Calls: 7000 What extensions to park calls on: 7001-7200 (Ex: '701-720')	
Options	Number of seconds a call can be parked for: 45	
Voicemail	Pickup Extension: *8	
Conferencing	Pickup Specified Extension: *7	
Music on hold	Blind Transfer: #	
Music on Ringback	Attended Transfer: *2	
Call Parking	Disconnect Call: *	
DISA	Timeout for answer on attended transfer: 15	
Follow Me	Save Cancel	
Paging and Intercom		
Time Based Rules		
Status		
System		
tension to D	ial for Parking Calls: Set Call Parking	number

- What extensions to park call on: Set the Call Parking get number (eg:701-720)
- Number of seconds a call can be parked for: Set the second call time
- Pickup Extension:

- Set Pickup Extension
- Pickup Specified Extension
- Set Pickup Specified Extension
- Blind Transfer
 allows unattended or blind transfers. It works like this:



While on a conversation with another p arty, you dial the bli ndxfer seque nce. the system says "T ransfer" then gi ves you a di al tone, while putting the other party on hold. You dial the transferee number and t he caller is put through to that number immediately. Your line drop s. The caller ID displayed to the person receiving the transferred call is exactly the same as the caller ID presented to you.

- Attended Transfer allows attended transfer or s upervised transfer. It works like this: While on conversation with another party, you dial the atxfer key sequence. the system says "Transfer" then gives you a dial tone, while putting the other party on hold. You dial the transferee number and talk with the transferee to introduce the call, then you can hang up and the ot her party will be connected with the transferee. In case the transferee does not want to ans wer the call, he/sh e simply hangs up and you will be back to your original conversation. Press the disonnect key sequence, set to * by default, to return yourself to the original caller.
- Disconnect Call Disconnect the current transfer call(for Attended transfer).
- Timeout for answer on attended transfer: Set the answer timeout value.

7.7 DISA Settings

		Activate Changes Logout
Home	DISA Settings	Extension for this Disa (Option): If you want this
Basic	List of Disa	DISA to be accessible by dialing an extension, you can
Inbound Control	S.No DISA Name Options	define an extension number for this DISA.
Advanced	1 Test 1 Edit Delete	for this DISA.
Options		
Voicemail		
Conferencing		
Music on hold		
Music on Ringback		
Call Parking		
DISA		
Follow Me		
Paging and Intercom		
Time Based Rules		
Status		
System		
	New DISA	
List of DISA	 DISA name are listed in the table. 	

New DISA Create a new DISA.



		Add a Disa X
	DISA N	ame:
		PIN: Without PIN
	Response Timeou	ut(s): 5
	Digit Timeou	
	Exter	nsion for this Disa(Option):
		Allow Outbound Route
		Select DialPlan DialPlan1 👻
		Save Cancel
		Save
DISA N	lame	Set a name for DISA
• PIN		Set a password for DISA
 Responsion 	nse Timeout(s)	Set effective time for inputing a password
 Digit Ti 	meout(s)	After you input the right password, the inte
bet	wee	n digits that you need dial.
Extension for	or this DISA(Option	n) Set a number connect DISA
Select Dial	Plan	Select your DialPlan for calling out

7.8 Follow Me

		Saved Successfull!		Activate Changes Logout
Home	Follow Me			Destinations: Set your followme numbers with fixed
Basic		List of Follow Me		format : " <number></number>
Inbound Control	S.No Extensions	State Forward No.	Options	[& <number>],<ringtime>" • for example:</ringtime></number>
Advanced	1 804	BN 806	Edit Delete	809,10 810,10
Options		100 100		8068803,20 9013542125751,30
Voicemail				3013342123731/30
Conferencing				
Music on hold				
Music on Ringback				
Call Parking				
DISA				
Follow Me				
Paging and Intercom				
Time Based Rules				
Status				
System				
		New Fallow Me		

List of Follow MeNew Follow Me

Call Forward extensions are listed in the table. Create a new Call Forward



		Add a Follow Me		x
	Extension:	•		
		Ring lasting for 20 s	econds	
	Sta	atus: 🗆 Always 🗆 Busy I	No answer	
		orward number		
	• Forward a l	ocal Extension: 🔿 Forwa	rd a Outside Number:	
	Select forward	extension		
		Save Cance	1	
Extension		Select a need to ca	I forward extensio	n
Ring Time		Set the extension ri	ng time	
State		Set state of the exte	0	wavs F
olulo		answer)		Mayo, I
Select forv	vard extension	Select a call forward	d to extension	
hen vou sele	ct "Forward an	Outside Number" the	follow page will be	e displa
		Add a Follow Me		Х
	Extension:	▼		
		Ring lasting for 20 s	econds	
		itus: 🗆 Always 🗆 Busy 🛙	No answer	
		orward number		
		.ocal Extension: Forwa	rd a Outside Number:	
	Select DialPlan			
	Set forward ou	tside number		
		Save Cancel		

- Select DialPlan Select a Call forward to outside number using dialingrules
- Set forward outside number Input a Call forward to outside number. (Notice: This number must be consistent with the corresponding DialPlan)

7.9 Paging and Intercom

				Activate Changes Logout
Home	Paging and Intercom			Duplex: Paging is typically one way for announcements
Basic	ti	st of Paging Groups		only. Checking this will make the paging duplex, allowing a phones in the paging group to be able to talk and be heard
Inbound Control	S.No Paging Group	n or rugnig croups	Options	phones in the paging group to be able to talk and be beard
Advanced	1 500 Market		Edit Delete	by all. This makes it like an "instant conference".
Options			Retaining Residences	madane comercinee .
Voicemail				
Conferencing				
Music on hold				
Music on Ringback				
Call Parking				
DISA				
Follow Me				
Paging and Intercom				
Time Based Rules				
Status				
System				
		Add Paging Groun		



• List of Paging Groups

Call Forward extensions are listed in the table.

Add Paging Group Ci

Create a new Call Forward

Add Pa	iging Group	Х
Paging Extension: Group Description:		
	← SIP/801 User1 SIP/605 User2 → SIP/803 User3 SIP/804 User4 SIP/805 User5 SIP/806 User6 SIP/807 User7 SIP/808 User8	+ III +
Paging Group Members Duplex:	Device List	

Save Cancel

- Paging Extension
- Set a extension for the Paging Group.
- Group Description Provide a descriptive title for this Page Group.
- Paging Group Members Selected Device(s) in this Page.
- Device List Select Device(s) to Page.
- Duplex Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to t alk and be heard by all. This makes it like an "instant conference".

7.10 Monitor

Advanced 1 821 Options 2 810 Voicemail 2 810 Conferencing 3 3 Music on hold 3 3 Music on Ringback 2 3 Call Parking 3 3 DISA 5 5	ension Record Time Always	Monitoring Ext Inbound Disable i Enable	Outbound Enable Enable	Options Edit Delete Edit Delete	field to see tooltips
Advanced 1 821 Options 2 810 Voicemail Conferencing Music on hold Music on Ringback Call Parking DISA Follow Me	ension Record Time Always	Inbound Disable	Outbound Enable	Edit Delete	
Advanced 1 821 Options 2 810 Voicemail 2 810 Conferencing 3 3 Music on hold 3 3 Music on Ringback 3 3 Call Parking 3 3 DISA 5 3	Always	Disable	Enable	Edit Delete	
Options 2 810 Voicemail Conferencing Music on hold Music on Ringback Call Parking DISA Follow Me					
Voicemail Conferencing Music on hold Music on Ringback Call Parking DISA Follow Me					
Music on hold Music on Ringback Call Parking DISA Follow Me					
Music on Ringback Call Parking DISA Follow Me					
Call Parking DISA Follow Me					
DISA Follow Me					
Follow Me					
and the second					
And a second					
Paging and Intercom					
Monitor					
Time Based Rules					
Status					
Network Settings					
System					

• List of Monitoring Extension

Create

Add Monitor

Monitoring extensions are listed in the table. a new Monitor



Add Monitor	х
Extension: 810 🔻	
Monitor Time	
Always Monitor:□ Start Time: 09 • : 00 • End Time: 17 • : 30 • Start Day: Mon • End Day: Fri •	
Monitor Settings	
Inbound Record: 🗹 Outbound Record: 🗹	
Save Cancel	

Extension	Select a Monitoring extension
Monitoring Time	Set always Monitor or select a Monitoring time
Monitoring Settings	Set inbound record and outbound record

7.11 Time Based Rules

Home	Time Base	ed Rules		Move the mouse over to a field to see tooltips
Basic		List of Ring Groups		Held to see toolups
Inbound Control	S No. E	RuleName	Options	
Advanced	1 In		e	
Options		Edit Time Rule	x	
Voicemail		Rule Name : Incoming (Ex: July4)		
Conferencing		Time & Date Conditions		
Music on hold		Start Time: 09 ▼ : 00 ▼ End Time: 17 ▼ : 30 ▼		
Music on Ringback		Start Day: Mon 🔻 End Day: Fri 💌		
Call Parking		Start Date: 01 V End Date: 31 V		
DISA		Start Month: January		
Follow Me		Destination		
Paging and Intercom		if time matches: VoiceMenu working time 🝷	-	
Time Based Rules				
Status		Save Cancel		
System				
		idew Time Whie		

On this page, Define call routing rules based on date and time



Chapter8 Status Display

8.1 Monitor List

					Activate Changes	Logou
lome	Monitor				Move the mouse ov field to see tooltips	er to a
Basic	Extension: 804	• Delete				
inbound Control	Date: Aug 🔻 24	4 - 2010 - Go				
Advanced		and the second design of the second	Monitoring File			
Status	S.No Caller ID	Destination	Date	Options		
Monitor List	1 810	804	2010/08/24 15:00:03	Delete 🐸		
Call Logs						
Register Status						
System Info						
Network Settings						
System						

This web page will display Monitor info for each extension

8.2 Call Logs

lome	Call Logs											
Basic	Start Date:	Jul	+	26 -	2010	÷	Field: Caller ID	-		Filter	Download	Delete
Inbound Control	End Date:									-		
Advanced	Call	Start			C	aller	ID	D	estination	Duratio	on (sec)	Disposition
Status												
Call Logs												
Register Status												
System Info												
System				1								
							No log messa	ges fo	und			
							No log messa		-			
							No log messaj	ges rou	na.			

This web page will display call logs

- Download download the call logs file
- Delete delete the call logs file

www.vaidsys.ru



8.3 Register Status

In this page, you can check SIP/IAX Users and Trunks Status.

						Activate Changes	Log
Home	Register Status						
Basic	SIP Users Status	IAX2 Users St	strue	SIP Trunks Sta	tue TAY	? Trunks Status	
Inbound Control		IMAZ USEIS DL	acus	STP TIONKS Sta	LUS IMAZ	TIONKS Status	
Advanced	SIP Users Status:						
	Name/username	Host		Nat ACL Port	Status		
Status	830	(Unspecified)	D	0	UNKNOWN		
Call Logs	829	(Unspecified)	D	0	UNKNOWN		
	828	(Unspecified) (Unspecified)	D	0	UNKNOWN		
Register Status	826	(Unspecified)	D	0	UNKNOWN		
System Info	825	(Unspecified)	D	0	UNKNOWN		
	824	(Unspecified)	D	0	UNKNOWN		
System	823	(Unspecified)	D	0	UNKNOWN		
	822	(Unspecified)	D	õ	UNKNOWN		
	821	(Unspecified)	D	0	UNKNOWN		
	820	(Unspecified)	D	O	UNKNOWN		
	819	(Unspecified)	D	D	UNKNOWN		
	818	(Unspecified)	D	D	UNKNOWN		
	817	(Unspecified)	D	Q	UNKNOWN		
	816	(Unspecified)	D	O	UNKNOWN		
	815	(Unspecified)	D	D	UNKNOWN		
	814	(Unspecified)	D	O	UNKNOWN		
	813	(Unspecified)	D	D	UNKNOWN		
	812	(Unspecified)	D	D	UNKNOMN		
	811	(Unspecified)	D	D	UNKNOWN		
	810	(Unspecified)	D	D	UNKNOMN		
	809	(Unspecified)	D	D	UNKNOWN		
	808	(Unspecified)	D	D	UNKNOWN		
	807	(Unspecified)	D	Q	UNKNOWN		
	806	(Unspecified)	D	D	UNKNOWN		
	805	(Unspecified)	D	Q	UNKNOWN		
	804	(Unspecified)	D	0	UNKNOWN		
	603	(Unspecified)	D	D	UNKNOWN		
	802	(Unspecified)	D D	0	UNKNOWN		
	30 sip peers [Monitored: 0	(Unspecified)			UNKNOWN	and the second	

8.4 System Info

In this page it will display nonce system info

		Activate Changes Logout
Home	System Information	Move the mouse over to a field to see tooltips
Basic	General Resources	
Inbound Control	dener un incesources	
Advanced	OS Version:	
Status	Linux IP PBX 2.6.22.18	
Call Logs	Uptime: 18:06:54 up 2 days, 8:58,	
Register Status	Load Average: 0.09, 0.14, 0.11	
System Info	Firmware Version: System v1.0	
System	Server Date & TimeZone: Wed, 02 Jun 2010 18:06:54 -0400 Refresh Synchronize Hostname: IPPEX	



Chapter9 Sy stem Management

9.1 Network and Country

On this page you can set WAN, LAN interface information and the country of Tone Zone.

				Activate Changes	Logoui
Home	Network & Country Settings			DNS: Specify a name to resolve domain name	server
Basic	WAN Port Setup			to resolve domain nar	nes
Inbound Control		-			
Advanced		Static 👻			
Status	Hostname:				
System		192.168.1.72 255.255.255.0			
Network & Country		192.168.1.1			
DDN5 & VPN		61.139.2.69			
Time Settings					
Management	LAN Port Setup				
Backup		192.168.10.100			
Update	Subnet mask:	255.255.255.0			
	Country setting				
	Tone Zone: CN - China		•		
	Save	Gancel			

• IP Assign:

you can select STATIC, DHCP and PPPoE three mode Set your Country, and use the Country Tone

• Tone Zone:

9.2 DDNS&VPN

9.2.1 DDNS Settings

Home	DDNS Settings		Domain: Set Domain.
Basic	DDN	S VPN	
Inbound Control			
Advanced			
Status	Dyndns.org DDNS		
System	DDNS Enable:	v	
Network & Country	Username:	IP556	
DDNS & VPN	Password:		
Time Settings	Domain:	IP555, dyndns.org	
Management		Sale	
Backup			
Update			

On this page, you can set DDNS reference.


Notice: Now, it only supports Dyndns.org server. More other servers, you can customize based on your requirement

9.2.2 VPN Settings

		Activate Changes Logout
Home	VPN Settings	Local IP: Set IP that in the VPN address.
Basic	DDNS VPN	viii dudi cust
Inbound Control	DONS VEN	
Advanced		
Status	N2N VPN	
System	VPN Enable:	
Network & Country	Server IP:	
DDNS & VPN	Port:	
Time Settings	Local IP:	
Management	Username:	
Backup	Password:	
Update	Save Carcel	

On this page, you can set VPN reference.

Notice: Now, it only supports N2N VPN. More other VPN, you can customize based on your requirement.

9.3Time Settings

		Activate Changes Logout
Home	Time Settings	Save: Save the settings
Basic	Time Settings	
Inbound Control		
Advanced	ONTP O Manual Time Set	
Status	NTP Server: pool.ntp.org	
System		
Network & Country	Time Zone: (GMT+08:00)Beijing,Hong Kong,Urumqi 🚽 🗸 🗸	
DDNS & VPN		
Time Settings		
Management		
Backup		
Update		
	Sarje Guncal	



9.3.1 NTP Settings

	⊙ NTP	C Manual Time Set
NTP Server:	pool.ntp.org]
Time Zone:	(GMT+08:00)Beij	iing,Hong Kong,Urumqi 🛛 👻

- NTP Server Specify the NTP server that you wish to use. You may type ei ther the domain name or the IP address of the server, and it may be either remote or local. The default server is pool.ntp.org. Be aware that the PBX needs to be able to access a NTP server in order to function properly.
- Time Zone Select your time zone so that the system will set time base on the time zone.
- 9.3.2 Manual Time Settings

O N	TP 🔍 Ma	nual Time	Set
Year:		(YYYY, eg	: 2010)
Month:		(MM, eg:	05)
Day:		(DD, eg:	08)
Hour:		(HH, eg:	09)
Minute:		(MM, eg:	30)
Syn	chronize current	PC time	Sync

• Synchronize current PC time Click the button ,the current PC time synchronization.

9.4 Management

		Activate Changes Logout
Home	Management	Set Voice Language: Set the system language voice.
Basic	Change Password	
Inbound Control		
Advanced	Enter New Password:	
Status	Retype New Password:	
System	Apply	
Network & Country		
DDNS & VPN		
Time Settings	Set Language	
Management	See any any a	
Backup	Set Voice Language: English 👻	
Update		
	Save	
	(Show Advanced Options)	

- Change Password On this p age, you can c hange the administrator p assword (Default password: admin)
- Set Language Set the system language voice

And you can also set the advanced options about SIP and Zap protocol in the "Show



Advanced Options" list, that is useful when you set connect two ippbx in different network.

9.5 Backup

Home	Backup / Restore Conf	gurations		Move the mouse over to a field to see tooltips
Basic		List of Configuration Back	cups	
Inbound Control	S.No Name	Date	Options	
Advanced	1 backup1	Jun 03, 2010	Restore Delete 🕈	
Status	Contraine.	(101 CA 01 CA 1		
System				
Network & Country				
DDNS & VPN				
Time Settings				
Management				
Backup				
Update				

On this page, clicking the "Take a Backup" button, you can backup once configuration

9.6 Upgrade

Home	date	Enter The Sound File
Basic	in the second	Name: Pis enter the Sound file name, that you need
Inbound Control	Upgrade System Package	upload to IVR Prompts. Pls use .gsm format file.
Advanced	Enter The Package Name:	
Status	TFTP Server IP address:	
System	Update	
Network & Country		
DDNS & VPN	Upload IVR Prompts	
Time Settings	Enter The Sound File Name: (*	*.gsm)
Management	Note: Please use .gsm format voice file.	
Backup	TFTP Server IP address:	
Update	Upload	
	Upload Backup File	
	Enter The Backup File Name:	
	Note: Don't change the backup file name.	
	TFTP Server IP address:	
	Upload	

In this page you can upgrade system package

- Enter The Package Name Set system package name
- TFTP Server IP address Set TFTP server IP

Unzip the file you download, you will get a TFTP server and an upgrading packet.







Run the TFTP server, you will see below:

Current Directory	D:VZX50_Update	Browse
Server interfaces	192.168.1.71	→ Show Di
Titp Server Titp	Client DHCP server Syslog serve	er
Current Action	Listening on port 69	

Enter the configuration page, then upgrading page;

Enter The Package Name, hereby it's uImage-md5

Enter TFTP Server IP address, hereby it's

192.168.1.71

After done, click Update to update, then the system will reboot automatically.

(Note: the upgrading will set your system as default, please make backup before you do it.)



Chapter10 Operating Instruction

10.1 How to link the ZX30x IP PBX to the interwork

10.1.1 IP PBX behind the Router

If your of fice access the public network with router, you can put the IPPBX behind the router. You should connect the Wan port of the IPPBX to the Lan ports of the router, and you also can connect HU B or Switch to the Lan ports of the IPPBX to let some PC or IP Phone to access the public network.



10.1.2 IP PBX behind the Modem

If you have the public IP and want the IPPBX a ccess the public network directly without router, then you should c onnect the Wan port of the IPPBX to the public network and connect HUB or Switch to the Lan port s of the IPPBX to let your PC access the public network...(If you want to access the public network through Modem, then you should use the PPPOE function of the IPPBX and let the IPPBX dial-up to connect the public network)





10.2 How to log in the IP PBX system

After connecting the ippbx to the loca I area ne twork. Launch the web b rowser on a computer that is in this local area n etwork. Enter the IP address for the system (default: Wan port IP address is http://192.168.1.100:9999, Lan port IP address is http://192.168.1.100:9999, Lan port IP address is http://192.168.

Username: Password:	
Please login	

Enter Username and password (default username is admin, password is admin), then click login. Once the login is successful, the home page will be display:

		Activate Changes	Logout
Home	Home	Move the mouse ov field to see tooltips	er to a
Basic	and the second se	held to see toolops	
Inbound Control	System Info		
Advanced	Network		
Status	WAN IP 192,168.1.100 LAN IP 192,168.10.100		
System	Storage		
	Flash Total: 1016.0M Used: 3.8M Ext Disk Total: N/A Used: N/A		
	Analog Chancels		
	1 2 3 4 5 6 7 8 CSM N/A CSM N/A FXO FXO FXS FXS		
	Device Info		
	ZX30 IP PBX		
	Model No.: ZX30-AG42 System Version: v1.0		
	Run Time:3 min Refresh Reboot Factory Defaults		

With the $PBX\,$ GUI, you can configure extensions, conference, voicemail, Outbound Routers and etc. Each page of the GUI has three columns:

The left column present all the options tab that you can program the system. Click the tab to go this kind of option setting page.

The middle column contains the primary content for each page.



The right column of the user interface c ontains Tooltips. This area pr ovides brief description for any options of the GUI

The home page is used for logoff, Reboot and Factory Defaults.

- Logout: To log out the PBX GUI.
- Reboot: Reboot the ZX30x system
- Factory Defaults: Restore all settings to factory default.
- Activate change: Made the chang e active for the curr ent configuration after you make a configuration change on some page.

10.3 How to make a internal call

Making internal calls are the base requirement for a teleph ony system. Below are the settings for this usage. It is base on ZX30x but setting is the same in other ZX30x products.



There are 30 default users, the extensions number are 801~830 Set user, Extension is 803,Name, Password and Caller ID, etc Select Dial Plan is DialPlan1 Set Extension 804 as the same way



Use a IP Phone based SIP protocol registered with the user. Then you can use 803 call 804 successfully.

10.4 How to make an outbound call

To make an outbound call, we need to add a trunk first. There are two types of Trunk:Analog Ports:GSM/FXO ports of ZX30x connect to GSM/PSTN lines.VoIP Trunk:SIP or IAX trunk, connect to remote SIP/IAX serverI am u sing ZX30xG4, the port1-4are configured as GSM ports. When a port isconfigured a s GSM/FXO port,the corresponding LED showsRED. When a port is

configured as FXS port, the corresponding LED shows $\ensuremath{\mathsf{GREEN}}$.

10.4.1 Make call via GSM trunk

You can use the GSM trunking to make outgoing call via your out si line. The set up is as per below:



Add Analog Trunk

Trunks -> Add a Trunk:



		Lines: Individual lines of the
Home	Trunk	PBX Ex: Analog Port #3: The third analog port of the PBX,
Basic		third analog port of the PBX.
Extensions	Add Trunk X	
Trunks	Lines: 🔽 Analog Port #1	
Outbound Routes	Provider Type: Analog Port #3	
Inbound Control	Analog Trunk Analog Port #5	
Advanced	Custom Trunk	
Status System	C Peer	
	Save Ennot	
	ann a teane	

Add Outbound Routers

In Outbound Routers -> add a Dial rule as below Dial Rules

		Activate Changes Logout
Home	Outbound Routes	Ex: Delete 1 digits from the front and prepend 256 before
Basic	List of DialPlans:	dialing
Extensions		
Trunks	DialPlan1 V New Delete	
Outbound Routes		
Inbound Control	List of Dial Rules:	
Advanced	S.No R	
Status	1 Ca Rule Name: Call_GSM	
System	Place this call through : Ports 1,3 👻	
	Failover : None PIN Set: Dialing Rules : If the number begins with 9 and followed by (more than) 3 digits (define a custom pattern) Delete 1 digits from the front and auto-add digit before dialing	

We have now added a Dial rule "OUT_GSM" in the "DialPlan1".

As we can see from the dialing rule of "OUT_GSM", all numbers start with 9 will be cut the first digit ('9') and sent to GSM (port1 or port3).

Choose Dial Plan for extensions:

On the User page, edit the extensions to choose DialPlan1.



Home	Extension Configuration		Analog Phone: If this user is attached to an analog port on
Basic	Extensions:	Extensions Setting:	the system, please choose the port number here,
Extensions Trunks Outbound Routes Inbound Control Advanced Status System	805 User5 807 User6 807 User7 808 User8 809 User8 810 User10 811 User11 812 User12 813 User13 814 User14 815 User15 5	Extension: 801 Name: User1 Password: 801 Caller ID: 801 VM Password: 801 E-mail: Analog Phone: None * Dial Plan: DialPlan1 * DialPlan1 *	
	816 User16 817 User17 818 User18 819 User19 820 User20 821 User21 822 User22 823 User23 824 User24 825 User25	Voicemail Can Reinvite SIP IAX Call Waiting 3-Way Calling NAT Pickup Group 1. Delete VMail DTMFMode RFC2833. Codecs Configure: Statemark	
	826 User26 827 User27 828 User28 New Delete	disallow: all allow:alaw,ulaw,g729	

After we have done a bove, in the extension we can dial 9 + local number to dial out via GSM line.

10.4.2 Make call via VoIP trunk

Via the voip trunking we can dial call via the voip service to reduce our cost when making international calls.



Trunk -> Add a Trunk: Add a Custom Trunk.



							Activate Changes Logout
Home	Trunk				_		Analog/Voip Trunks/Peer Analog lines are attached to
Basic	List of 1	Trunk					analog interfaces of the PBX using FXO cards. Voice over
Extensions	S.No		Add Trunk		×	phan	IP (VoIP) connections are provided by an Internet
Trunks						ititine 🔻	Telephony Trunk (ITSP).
Outbound Routes			Description: VoIF				
Inbound Control		Provider Type: C Analog Trunk	Protocol: SIP				
Advanced		Custom Trunk	DialPlan: def: Register: 🔽	ault 🔻			
Status		CPeer	Host:	.dyndns.or			
System			T Without Auth	entication			
			Username:	test			
			Password: ••••	•			
			Save Cantel				
			-				

Add Dial Rule

In Dial Rules -> add a new calling rule as below *Dial Rules*

		Activate Changes Logout
Home	Outbound Routes	Dialing Rules: Ex: If the number begins with '256' and
Basic	List of DialPlans:	followed by 7 digits or more
Extensions	DialPlan1 + Wrw Delete	
Trunks	DialPlant + Delete	
Outbound Routes		
Inbound Control	List of Dial Rules:	
Advanced	S.No Rt X hs	
Status	1 Ca Rule Name: Out_VoIP	
System	Place this call through : Custom - VoIP • Failover : None • PIN Set: Dialing Rules : If the number begins with 0 and followed by (Image many 1 digits (define a custom pattern) Delete 1 digits from the front and auto-add digit before dialing	

Now we have added a new calling rule "Out_VoIP" in the "DialPlan1".

As we can see from the "Out_VoIP" dialing rule, all numbers start with 0 will be cut the first one digits ('0') and sent to my sip service provider.

The Out_GS M is in the same DialPlan1. Since we have a dded this dial plan to the extensions in above, we don't need to add dial plan again.

So when we have added two calling rules, any call start with 9 will be route to GSM, and call starts with 0 will be route to VoIP.



10.5 How to make an incoming call

Add an Incoming call.

					Activate Changes	Logout
Home	Inbound Routes				Add a Incoming Ru Define a new Rule fo	ile:
Basic			10.5		Incoming calls based and/or the number c	on trunk
Inbound Control		oming Calling Rule is not			and/or the number of	alleu,
Inbound Routes		ck on 'Add a incoming R add a new incoming call				
IVR.			seren.			
IVR Prompts						
Ring Groups				×		
Advanced	2.25		11.102			
Status		Unmatched incoming c Trunk Ports 1,3	alls 💌			
System	Destination		-			
		807 - User7 808 - User8 809 - User8 809 - User9 810 - User10 811 - User11 812 - User12 813 - User13 814 - User14 815 - User16 817 - User16 817 - User17 818 - User18 820 - User29 821 - User21 822 - User22 823 - User22 824 - User24 825 - User25 824 - User25 824 - User25 825 - User27 825 - User27 828 - User28 829 - User28 829 - User29 829 - User30	B			

Select Route "All Unmatched incoming calls"

From provider "Port 1, 3"

To extension "801 – User1" (here, you can select a extension, a IVR or others) Then, if there is incoming call from Port1 or port3 channel, the extension 801 will ring.

10.6 How to Set an incoming call to IVR based time rule

Add record a custom voice

Record -> Record a new voice

Home	IVR Prompts(Re	ecord Voices For Cust	om IVR)		Extension used for recording: Select a device
Basic		List	of Recorded voices		through which this voice men will be recorded.
Inbound Control	5.No Name		Option	is.	will be recorded.
Inbound Routes	1			10	
IVR		Record	a new Voice	x	
IVR Prompts	4		trends many	te	
Ring Groups	2	File Name:	welcome	te	
dvanced			ecording: 801 User1 🔹		
itatus		Reco	rd Canca		
System					
		-	land a ball him		

Set the record name is "Welcome"



Choose a extension used for recording, here we use EXT 801

Click Record button

Then, the extension 801 will ring

Pick up the phone record "Welcome" message

Then hangup and finish the record .

Use the same way to record "Closing" message

				Activate Changes	Logout
Home	IVR Prompts(Record Voices For C	Custom IVR)		Move the mouse ove field to see tooltips	er to a
Basic		List of Recorded voices			
Inbound Control	S.No Name		ions		
Inbound Routes	1 closed.gsm	Record Again	Play Delete	-	
IVR					
IVR Prompts	2 welcome.gsm	Record Again	Play Delete		
Ring Groups					
Advanced					
Status					
System					
	1 - 3	Record a new voice			

Add a Ring Group Ring Group -> New Ring Group

				Activate Changes Log
Home	Ring Groups			Goto an IVR menu: Sel goto an IVR menu if the
Basic				ringroup no answer.
Inbound Control		ting Group	x	
Inbound Routes		ung uroup	A	
IVR	Name: tech	Strategy; Ring all 🔹		
IVR Prompts	SIP/801 User1	+ SIP/805 - User5	*	
Ring Groups	SIP/802 User2 SIP/803 User3	SIP/806 User6 SIP/807 User7	111	
Advanced	SIP/804 User4	SIP/808 User8 SIP/809 User9		
Status		>>> SIP/810 User10 SIP/811 User11		
		SIP/812 User12	*	
System	Ring Group Members	Available Channels		
		ring group(Option) :		
		these many seconds : 20		
	If not answered			
	C Goto an Extension			
	🖓 Goto an Extension Voicem	IVR working time 🔻		
	C Goto a RingGroup	TWO WORKING LINE		
	Goto an IVR menu			
	C HangUp			
	Sav	re Committe		
	THE	w Ring Group		

Example:

Name the ring group "tech"

Choose the group members whose extensions are "801, 802, 803, 804"



			Activate Changes Lo
Home	IVR		Move the mouse over to field to see tooltips
Basic	IVR Menu:	IVR Setting	and a set of the
Inbound Control	IVR - working time IVR - closed time	Name: working time Extension:	
Inbound Routes	IVR - closed time	Welcome Massage	
IVR	5.4 M	Please Select: welcome -	
IVR Prompts		Dial other Extensions?	
Ring Groups		Keypress' Events	
Advanced		Key Action	
Status		0 Goto Extension - 801 -	
System		1 Goto Extension 802 2 Goto RingGroup tech	
		3 Disabled -	
		4 Disabled 👻	
		5 Disabled 👻	
		6 Disabled - 7 Disabled -	
		8 Disabled +	
		9 Disabled 👻	
	New Delete	* ~ 11.1	

Select IVR-working time, Set welcome massage is "Welcome" Set keypress' Events

Dial "0" go to extension 805

Dial "1" go to extension 806

Dial "2" go to ringgroup tech

Click Save button

Home	IVR		Move the mouse over to a field to see tooltips		
Basic	IVR Menu:	IVR Setting			
Inbound Control	IVR - working time IVR - closed time	Name: closed time Extension:			
Inbound Routes	IVR - closed time	Welcome Massage			
IVR					
IVR Prompts		Please Select: closed •			
Ring Groups		Dial other Extensions? Keypress' Events			
Advanced		A TRUCK AND A MORE			
		Key Action 0 Goto Extension - 801 - *			
Status	1				
System					
		3 Disabled 👻			
		4 Disabled 👻			
		5 Disabled 👻			
		6 Disabled			
		8 Disabled +			
		9 Disabled 👻			
	New Delete	* * 0.0			
		Save Cancel			

Then set IVR-closed time



Set welcome massage is "Closing" Add a Time Rule

Time Based Rules -> New Time Rule



Set a Rule Name, eg: incoming Set the Time & Date Conditions "If time matches" --- go to "working time" "If time not match" --- go to "closed time" Click the save button, saved the configuration

Add a Trunk

Trunks -> add a Trunk

					Activate Changes	Logout
Home	Trunk				Move the mouse ov field to see tooltips	er to a
Basic	List of Trunk				field to ace toolupa	
Extensions	S.No	Edi	tTrunk	x ption		
Trunks	1	Lines	- -	btions +		
Outbound Routes	2		Analog Port #1	itians 🔫		
nbound Control		ler Type: og Trunk	Analog Port #5			
dvanced		om Trunk	Analog Port #6			
itatus	C Peer					
System						
		Sav	e Cancel			
			18.7.1990			



Add an incoming router

Inbound routers -> add an incoming rule

		Activate Changes	Logou
Home	Inbound Routes	Move the mouse ov field to see tooltips	er to a
Basic	List of Incoming Call Rules	nuis to see toolope	
Inbound Control	S.No Incoming Rule Options		
Inbound Routes	Route all unnatched incoming calls from provider Custom - VgIP' to '802 - User2'		
IVR	VoiP' to '802 User2'		
IVR Prompts			
Ring Groups	X		
Advanced	Route All Unmatched incoming calls 👻		
Status	from Trunk Ports 1,3,5,6		
System	Destination Incoming Time Based Rule -		
	Sive		
	Add inc. Jocenning state		

Select Route: All Unmatched incoming calls From provider: Ports 3, 4

To extension: incoming—Time Based Rule

		Activate Changes	Logout
Inbound Routes			er to a
List of Incoming Call Rules			
	Options		
to 'Incoming Time Based Rule'			
Add an Incoming Bule			
	List of Incoming Call Rules S.No Incoming Rule Route all unmatched incoming calls from provider 'Ports 1,3,5,6' 1 to 'Incoming Time Based Rule'	List of Incoming Call Rules S.No Incoming Rule Options 1 Courte all unmatched incoming calls from provider 'Ports 1,3,5,6' Edit Delete Delete	Inbound Routes Move the mouse owned to see toolfips S.No Incoming Rule Options 1 to 'Incoming Time Based Rule' Delete

Then click Activate Changes, Made the change active for the current configuration

10.7 How to link two ZX30x IPPBX in the same network

We start from linking two the IP PBX in the same network and then try to expand to different network. Below is the structure of how to link two IPPBX in the same LAN:





Register the ZX30xA as an peer in ZX30xB(via IAX2 trunk),so the extensions in ZX30xA can make calls to ZX30xB's extensions via this "special" trunk. In above structure:

- 1. ZP302A registers ZX30x A as an extension 601.
- 2. ZP302B registers to ZX30xB as an extension 801.
- 3. All the extensions under ZX30x Aare in the format 6XX.
- 4. All the extensions under ZX30xB are in the format 8XX
- 5. Extensions under ZX30xA can make calls to extension under ZX30xB use format 8XX.
- 6. Extensions under ZX30xB can make calls to extension ZX30x A use format 6XX.

<u>Step 1</u>: Set up a peer 699 in ZX30xA

In the page Trunks \rightarrow Add a Trunk

			Add Trunk		х
	Provider 1 O Analog Tr O Custom 1 O Peer	runk	Peer Name Protocol: DialPlan: Host: □ Without	IAX default dynamic t Authentication	
			Username: Password:		
Peer Name: ZX	30xB	;			
Peer Username:	699	Acco	ount of this Pe	er	
Password:	699	IAX2	Log on pass	word	
Advance Options:	Select IAX	protocol			
Step 2: Set up an	IAX trunk ir	nZX30z	$\mathbf{x}\mathbf{B}$ to link to \mathbf{Z}	X30xA via thisZ	ZX302

In the page Trunks--> Add a Trunk



	Add Trunk	Х
Provider Type: O Analog Trunk O Custom Trunk O Peer	Description: Call_ZX Protocol: IAX DialPlan: default Register: Host: 192.168.1.100 Without Authentication Username: 699 Password: •••	

<u>Step 3</u>: Set Dial Rule in ZX30xB, all calls start with 6 will be sent to ZX30xA. In the page: Outbound Routers --> Add a Dial Rule

	х
Rule Name: Call_ZX	
Place this call through : Custom - Call_ZX 🔹	
Failover : None 🗸	
PIN Set:	
Dialing Rules : If the number begins with 6 and	
followed by (🔽 more than) 1 digits (define a custom pattern)	
Delete 0 digits from the front and auto-add digit before dialing)
Save Cancel	

<u>Step 4</u>: Set the user 601 and Dial Plan in ZX30xA. In the page: Extensions \rightarrow Dial Plan

Extensions Settin	g:
Extension:	601
Name:	User2
Password:	601
Caller ID:	601
VM Password:	601
E-mail:	
Analog Phone:	•
Dial Plan:	DialPlan1 💌 DialPlan1

Active the change and apply the test:

1. Register an IP phone ZP302B to ZX30xB with 801 extension.

2. Register an IP phone ZP302A to ZX30xA with 601 extension.

3. Use 801 to dial 601. And you can see 601 will ring and you can pick up the calls. Above is the way to router ZX30xB's call to ZX30xA,

Accordingly, if you want to call from ZX30xA to ZX30xB, continue as follow: <u>Step 5</u>: Set Dial Rule in ZX30xA all calls start with 8 will be sent to ZX30xB.



				Х
		Rule Name:	Call_ZX	
	PI	ace this call through :	Peer - ZX 🔹	
		Failover :	None 🔻	
		PIN Set:		
		Dialing Rules :	If the number begins with 8 and	
			followed by (🔽 more than) 1 digits	
			(define a custom pattern)	
<u>Step 6</u> : Set t	Delete the user		ront and auto-add digit before dialing Save Cancel or in ZX30x <i>B</i>	9
		Extensions Setting	j .	

Extensions Setting	:
Extension:	801
Name:	User1
Password:	801
Caller ID:	801
VM Password:	801
E-mail:	
Analog Phone:	▼
Dial Plan:	DialPlan1
	DialPlan1

Active the change and apply the test:

Use 601 to dial 801, and you can see 801 will ring and you can pick up the calls.

10.8 How to link two IPPBX in different network

The generally environment for two ZX30x in different location is: two the ZX30x IP PBX are both in the internet and using the public IP.



The configuration is same with above guide (10.7) "Link two z^{ZX30x} IP pbx in the same network but use the public IP address as the "HOST" settings, like the bellow: In the page Trunks of ZX30xB--> Add a Trunk



Provider Type: C Analog Trunk Custom Trunk Peer	Description: Call_ZX': Protocol: IAX • DialPlan: default • Register: Host: 218.200.12.1
	Without Authentication Username: 699 Password: •••

The generally environment for two ZPX30x IP PBX in different location and one or both two are both behind router and using the private IP.So, we need to do port forwarding in the router and make ZX30x IP PBX can reach to each other.





For the ZX30xA is behind the router, you need to forward the IAX2 port in your router, so all the packets received on the router WAN port (210.11.25.127:4569) will be forwarded to the ZX30xA (192.168.1.21:4569). Below is the setting page in a linksys router:



Applications = & Gaming	Setup	Security	1	Applica & Gam		dministration	Status	
	Port Range F	orwarding	1	Po	rt Triggering	UPni	P Forwarding	DMZ
UPnP Forwarding								UPnP Forwardin
	Application	Ext.Port	TCP	UDP	Int.Port	IP Address	Enabled	UPnP Forwarding can be i
	FTP	21	۲	0	21	192.168.1.0		to set up public services o your network. When users
	Teinet	23	0	0	23	192.168.1.0		the Internet make certain requests on your network
	SMTP	25	۲	0	25	192.168.1.0		Router can forward those requests to computers eq
	DNS	53	0	۲	53	192.168.1.0		to handle the requests. If, example, you set the port number 80 (HTTP) to be
	TFTP	69	0	۲	69	192.168.1.0		forwarded to IP Address 192,168,1.2, then all HTTP
	finger	79	0	0	79	192.168.1.0		requests from outside use be forwarded to 192.168.
	HTTP	80	۲	0	80	192.168.1. 199		is recommended that t computer use static IP
	POP3	110	0	0	110	192.168.1.0		address.
	NNTP	119	0	0	119	192.168.1 0		You may use this function establish a Web server or
	SNMP	161	0	۲	161	192.168.1.0		server via an IP Galeway, this format, Windows XP of
	ssh	2020	0	0	22	192.168.1. 235		used to configure this thro UPnP communication Be st
	httpl	8080	۲	0	80	192.168.1.29		that you enter a valid IP Address. (You may need
	http2	8090	0	0	80	192.168.1 209		establish a static IP addre with your ISP in order to
	IAX	4569	۲	0	4569	192.168.1.21		properly run an internet se For added security,
	IAX2	4569	0	0	4569	192.168.1.21		More

<u>Step 2</u>: Set up the Provider Host in ZX30xB

Set up the service provider and calling rule in ZX30-B to make it register to ZX30x A. This method is almost the same as above, EXCEPT you need to use the 210.11.25.127 as the service provider instead of 192.168.1.21.

Step 3: Set port forwarding in the router for ZX30xB

Use the same method as Step 1 do port forwarding in router-B for ZX30xB as above.

Setp4:Link twoZX30x and make calls

Accordingly, set the 601 users in ZX30xA and 801 users in ZX30xB, and build the correct dial rules as above, you can make calls between two the ZX30x IP PBX.

Noted: You can also apply a DDNS to get one fi xed domain for both ZX30x IP PBX and connect to each other rather than use the Port Forwarding in the router.

10.9 How to resolve problems about hearing only on one side

If your IPPBX behind the Router, you should bu ild a IP Address Map to resolve this problem as follow:

Management---->Show Advanced Options ----> Global SIP Settings

Global SIP Settings 📼

--->NAT Support



NAT Support		
	Extern ip:	
	Extern Host:	
	Extern Refresh:	
	Local Network Address:	
	NAT mode:	•
	Allow RTP Reinvite:	•
Extern IP	Replace with your external IP ad	dress this your public IP or doma

- Extern Host Replace with your external IP address this your public IP or domain
- Extern Refresh Set time for fresh, default 10
- Local Network Address Replace with your local network address and mask
- NAT mode If your IPPBX behind the Router, set default yes



Chapter11 How to use Skype account in ZX30x

11.1 Register for Skype Manager

1. V isit skype.com/business and click Skype Manager



- 2. Complete the on-screen instructions to register for Skype Manage r. You can eith er use your e xisting personal a ccount or create a new one spec ifically for your Skype Manager.
- 7. Please bear in mind that the account you use to r egister will be used to administer products and credit thro ughout you r busi ness. We therefor e recommend that you create a new Skype account using your business name.

11.2 Create a SIP Profile and buy a Channel Subscription

Note: You need to be signed into Skype Manager to access the Skype for SIP settings.

1. C lick **Features** in the toolbar



- 2. In the Features menu on the left, click Skype for SIP.
- 3. C lick Create a new profile.
- Give your SIP Profile a friendly name so it's easier to remember and click on Next. Your Profile's registration details, including its username and password are displayed. Make a note of these details so that you can set up and configure your PBX.



Authentication detail	S	
lease choose the method (of authentication needed for your PBX.	
Registration (Username/password)	or, IP Authentication 🥥	
SIP User	99050000015459	
Password	4j9x7i7Ybggv8g Generate a new password	
Skype for SIP address	sip.skype.com	
UDP Port	5060	

5. C lick **Profile settings**.

6. C lick Buy a channel subscription to activate this profile.

- 7. Enter the number of channels you require and click **Buy now**.
- 8. Channel subscriptions are the amount of concurrent calls you would like to use with your SIP Profile. These channels are charged on a monthly basis.
- 9. If you don't want to make outbound calls with Skype for SIP, please proceed to step 6.

11.3 Allocate Skype Credit to the SIP Profile

- 1. Click **View profile** next to the name of the SIP Profile to which you want to allocate credit.
- 2. C lick Set up outgoing calls.
- 3. Enter the amount of Skype Credit you want to allocate to the SIP Profile and click **Add credit**.



		E	E
Profile name	Profile 5		
Calling channels	Buy a channel subscript	on to activate this profile	
Outgoing calls	Set up outgoing calls		
	Add credit	Auto-Recharge settings	
	(5) € 10.00	Add credit	
Caller ID 🗐	Set up Caller ID		
Incoming calls	Add a number or busines	s account	

4. If you want to enable **auto-recharging**, click on the Auto-Recharge settings tab, enter the recharge amount and the minimum balance required before recharging, then click **Save changes**.

11.4 Configure your Skype for SIP certified PBX for outbound calls

In the trunk of our IPPBX setting:

	Description: skype
Provider Type:	Protocol: SIP -
C Analog Trunk Custom Trunk Peer	DialPlan: default 👻
	Register:
	Host: sip.skype.com
	Without Authentication
	Username: 9905000001545
	Password: •••••••••



Outbound setting of our IPPBX:



	x
Rule Name: Skype	
Place this call through : Custom - skype -	
Failover : None 🗸	
PIN Set: 🗖	
Dialing Rules : If the number begins with 00 and	
followed by (💌 more than) 5 digits	
(define a custom pattern)	
Delete 2 digits from the front and auto-add digit before dialing	J
Save Cancel	

11.5 Make an outbound call

After we have done ab ove, in the extension we can dial 00 + Countr y Code + City Area Code + local number to dial out via skype line

For example: Dial number 00862885337096 will contact our company.

11.6 Configure your Skype for SIP certified PBX for inbound calling

	Х
Route All Unmatched incoming calls 👻	
from Trunk Custom - skype -	
Destination working time IVR Menu 🔹	
Save Cancel	

Inbound Routing of our IPPBX:

11.7 Set up a business account to t est inbound calls from people with Skype

- 1. Create a new business account in Skype Manager. For more information on creating a new business account, please see the <u>Skype Manager User Guide</u>.
- 2. C lick **View profile** next to the name of the SIP Profile to which you want to add the business account.
- 3. C lick Add a number or business account.
- 4. In the **Add business account** tab, enter the newly created business.



Caller ID 😡	Set up Caller ID	
Incoming calls	Add a number or business account	1
	You can receive incoming calls on your SIP Profile via Skype Online Numbers and via Skype business accounts. When someone calls your Online Number or contacts your business account on Skype the calls ge forwarded to your SIP Profile.	t
	Add Online Number Add business account	
	Add an existing business account	
	Architects.Engineering Create a new account	
	Extension number (optional) 🤒	
	Confirm	
	Important: If a Skype account is attached to a SIP Profile it cannot be used to sign into Skype on your computer or any other device.	

5. Click Confirm.

11.8 Make a test inbound call from Skype

Call the business account's Skype Name you created in step 7 from Skype.

11.9 Assign an Online Number to receive calls from landlines and mobile phones

- 1. C lick **View profile** next to the name of the SIP Profile to which you want to assign an Online Number.
- 2. C lick Add a number or business account.
- 3. C lick Buy a new number

Caller ID 🕝	Set up Caller ID	
Incoming calls	Add a number or business account	9
	You can receive incoming calls on your SIP Profile via Skype Online and via Skype business accounts. When someone calls your Onlin contacts your business account on Skype the calls get forwarded to Profile.	ie Number or
	Add Online Number Add business account	
	Buy a new number	

11.10 Make a test inbound call from a landline or mobile phone

Call the Online Number associated with the SIP Profile from a landline or mobile phone.





You have now successfully set up Skype for SIP for use with your Skype for SIP certified PBX.

For more help with setting up and using Skype for SIP, please see <u>support.skype.com</u> or check the <u>skype for sip user guide</u>

Appendix A

Activate Changes Logout Move the mouse over to a Home Home field to see tooltips Basic System Info **Inbound Control** Network Advanced WAN IP 192.168.1.76 Status LAN IP 192.168.10.100 Storage System Flash Total: 1016.0M Used: 19.6M Ext Disk Used: Total: N/A N/A E1 Channels 1 2 3 4 5 6 8 9 10 11 12 13 14 15 16 **Analog Channels** 32 33 34 35 FXO FXO FXS FXS **Device Info** Model No.: ZX-304 System Version: v3.0 Run Time:6 min Refres Reboot **Factory Defaults**

Model with E1 port and 2FXS/2FXO

E1 trunks configuration

Basic Extensions Trunks Outbound Routes 1 2 Provider Type: C Analog © E1 Trunk C VoIP Trunk C Peer Lines: E1 Channel #1 E1 Channel #2 E1 Channel #3 E1 Channel #3 E1 Channel #4 Peer E1 Channel #5 E1 Channel #8 E1 Channel #8 Select All	Home	Trunks					
Trunks 1 2 Description: >tions > tions > ti	Basic			List of Tru	nk		
Trunks 1 Outbound Routes 2 Inbound Control 2 Advanced C Analog Status E1 Channel #1 System E1 Channel #3 C Peer E1 Channel #4 E1 Channel #5 E1 Channel #7 E1 Channel #8	Extensions	S.No		Add a Trunk		х	otions
Outbound Routes 2 Inbound Control Advanced Advanced © E1 Trunk Status © VoIP Trunk System E1 Channel #1 E1 Channel #4 Peer E1 Channel #5 E1 Channel #6 E1 Channel #8 Image: Status	Trunks	1		Description	<u>1</u>		otions 👻
Inbound Control C Analog E1 Channel #1 Advanced © E1 Trunk E1 Channel #3 Status C VoIP Trunk E1 Channel #4 System E1 Channel #5 E1 Channel #7 E1 Channel #7 E1 Channel #8 E1 Channel #8	Outbound Routes	2					otions 👻
Advanced	Inbound Control			Lines:		-	
Status ^C VoIP Trunk ^C Peer ^C Peer ^{E1} Channel #4 ^{E1} Channel #5 ^{E1} Channel #6 ^{E1} Channel #7 ^{E1} Channel #8 ^S ^{E1} Channel #8 ^S ^{E1} Channel #8 ^S ^{E1} Channel #8 ^S ^{E1}	Advanced				and the second se		
E1 Channel #6	Status		C VoIP Trunk		and the second se		
E1 Channel #6 E1 Channel #7 E1 Channel #8	System		C Peer		E1 Channel #5		
E1 Channel #8					and the second se		
						-	
Select All							
					Select All		
				Savo Canco			
Save Cancel				Save			