

Version 2.17.0.23

Yeastar Technology Co., Ltd.



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1. Introduction

MyPBX – IP-PBX for Small Businesses/Home Office

MyPBX is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations (1-100 users per site). MyPBX also offers a hybrid solution (a combination of VoIP applications using legacy telecom equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

Applications

1.1 Features

Auto-provision	• Firewalls
Blind Transfer	Follow me
BLF Support	• Interactive Voice Response (IVR)
Blacklist	Intercom / Zone Intercom
Call Detail Records(CDR)	Music On Hold
Call Forward	Music On Transfer
Call Parking	• Paging / Zone Paging
Call Recording	• PIN Users
Call Pickup	• Queue
Call Routing	• QOS
Call Transfer	Ring Group
Call Waiting	Route by Caller ID
• Caller ID	• Skype Integration (Skype Connect)
Call Back	Three-way Calling
Conference	Mobility Extension
SMS to Mail	• External Storage
Mail to SMS	• DDNS
Speed Dial	OpenVPN
Define Office Time	• T.38
• Direct Inward System Access(DISA)	Voicemail
• DIDs	• VLAN
Distinctive Ringtone	• WAN
• Do Not Disturb(DND)	• PPPoE
Dial by Name	Static Route
• Alert	• integrated built-in packet capture tools
 multiple administrators 	• spy function



Note: the features with asterisk (*) will be supported in next version.

1.2 Hardware Specifications

1.2.1 Exterior Appearance

A. MyPBX Standard

Front Side



Figure 1-1 MyPBX Front Panel Picture

No.	Identifying
1	Green LED: Indicates correct power is being supplied to the unit
2	Green LED: Indicates the MyPBX is fully functional.
3	Green LED: Indicates stable WAN Port connection
(4)	Green LED: Indicates stable LAN Port connection
5	Red LED: Indicates presence of an FXO/GSM port.
	Orange LED: Indicates presence of a BRI port.
	Green LED: Indicates presence of an FXS port.
	LED Blinking- Red blinking: No connection between FXO port and
	PSTN
	LED Alternating - Red and Green (slow blink): FXO port is receiving
	an incoming call.
	LED Alternating - Red and Green (fast blink): FXO port is in use
	LED Dual - Green and Red (slow blink): FXS port is ringing
	LED Dual - Green and Red (fast blink): FXS port is in use



B. MyPBX Pro

Front Side

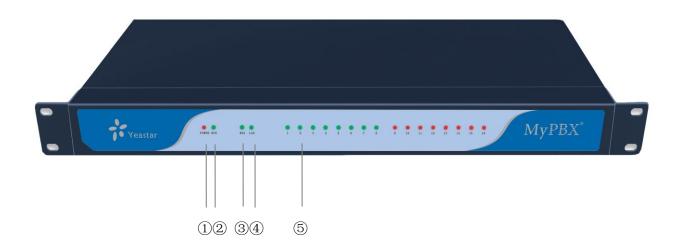


Figure 1-2 MyPBX Pro Front Panel Picture

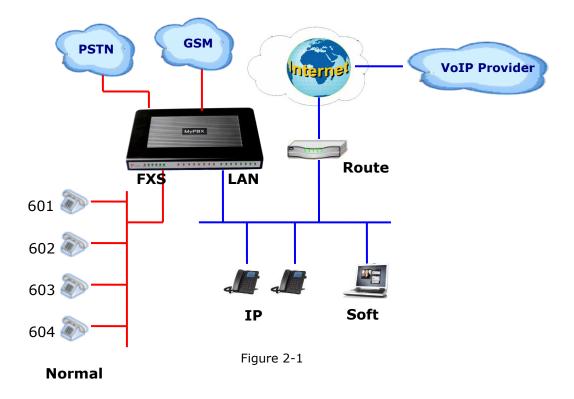
No.	Identifying
1	Green LED: Indicates correct power is being supplied to the unit
2	Green LED: Indicates the MyPBX Pro is fully functional.
3	Green LED: Indicates stable WAN Port connection
4	Green LED: Indicates stable LAN Port connection
5	Red LED: Indicates presence of an FXO/GSM port.
	Orange LED: Indicates presence of a BRI port.
	Green LED: Indicates presence of an FXS port.
	LED Blinking- Red blinking: No connection between FXO port and
	PSTN
	LED Alternating - Red and Green (slow blink): FXO port is receiving
	an incoming call.
	LED Alternating - Red and Green (fast blink): FXO port is in use
	LED Dual - Green and Red (slow blink): FXS port is ringing
	LED Dual - Green and Red (fast blink): FXS port is in use



2. System set up

2.1 Connection Drawing

A. MyPBX Standard





B. MyPBX Pro

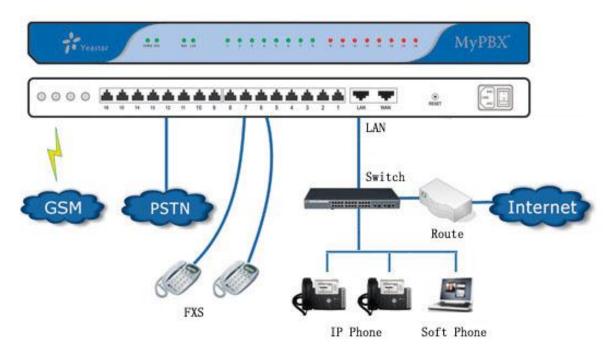


Figure 2-2

2.2 Connecting Ethernet Line

MyPBX provides two 10/100M Ethernet ports with RJ45 interface and LED indicator. Plug Ethernet line into MyPBX'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

MyPBX utilizes the high-performance switch power supply, which supplies the required power for the unit.

AC Input: 100~240V DC Output: 12V, 5A

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

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



3. Managing MyPBX

3.1 Administrator Login

From your web browser, input the IP address of the MyPBX server. If this is the first time you are configuring MyPBX, please use the default settings below (your PC should be in the same local network with MyPBX): IP Address: http://192.168.5.150 Username: admin , Password: password

Note: MyPBX supports multiple administrators in hierarchical mode (Administrator, General Manager, CDR Manager)

Administrator

have all authority . Username: admin , Password: password

·General Manager

have basic authority; without the advanced authority to create VoIP trunks, reset, update, backup and restore MyPBX. Username: admin , Password: password

·CDR Manager

only have the authority to check the call recordings.

🚳 http://192.168.5.186/ -	Microsoft Internet Explorer	
<u>File Edit V</u> iew Favorites	Toop Heb	- 1
	📓 🏠 🔎 Search 👷 Favorites 🤣 🎯 - 📚 🚍	
Address 🙆 http://192.168.5.18	6/ 🔽 🔁 G	
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	<u>^</u>
	MyPBX Configuration Panel	
Yeastar	User Login	
	User Name:	
	Password:	
	Language: English	
		Ξ
	Login	
E Done	🖉 Internet	~
C DOIR	S internet	



3.2 Status Monitor

3.2.1 Line Status

MyPBX	Embedded H	ybrid IP-PBX f Ø	or Small I	Businesses					Logout
Status Monitor Chine Status Basic		Free	a Bu		e nsions Iold	🔊 Unavaila	able 🔊 Ri	nging	
Extensions									
Trunks Outbound Routes	Ĩ	<u>500</u> (SIP)		3	502 (SIP)	Ŋ	<u>503</u> (SIP)		
Phone Provisioning	<i>a</i>	<u>505</u> (SIP)	्र <u>601</u> (FXS) 🧿	602 (FXS)	Ŋ	<u>603</u> (FXS)	<u>604</u> (FXS) (FXS)	
IVR Queues	Ø	<u>605</u> (FXS)	≫ <u>606</u> (FXS)					
Custom Prompts Ring Groups				Т	runks				
Inbound Routes	Status	Signal		Trunk Name	Туре	User Name P	ort/Hostname/II	P Reachability	
Blacklist	Registered		<u>305</u>	@192.168.4.138	SIP	305	192.168.4.138	OK	
Internal Settings	OK (1006 ms)			192.168.5.138	SP-SIP		192.168.5.138	OK (1006 ms)	
Internal Settings © Options	Failed			aa	SP-IAX		111111	UNKNOWN	
Business Hours	Idle			pstn4	FXO		Port 4		
Feature Codes	Disconnected			pstn5	FXO		Port 5		
SIP Settings	Disconnected			pstn6	FXO		Port 6		
IAX Settings	Disconnected			pstn16	FXO		Port 16		
Voicemail Settings	Enilod	Diagon Incost Of	1 Card	LIMITCO	LIMTO		Dort 0		

Figure 3.2.1

MyPBX Status Description:

Extensions:

- 1) 🔍 : Extension is unregistered
- 2) 🔊 : Extension is idle
- 3) 💐: Extension is ringing
- 4) $\overline{\mathbf{a}}$: Extension is busy in a call
- 5) **(a)**: Extension is on hold

Trunks: VOIP Trunk:

Status

Unregistered: Trunk registration failed. Registered: Succeed registration, trunk is ready for use. Request Send: Registering. Waiting: Waiting for authentication.



FXO Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

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

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

GSM Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

Signal

♥ : No signal.

¶ : Poor.

Y₁ : Average.

TII : Good.

Yul: Excellent.

BRI Trunk:

Status

Ok: The ports connect correctly. Disconnected: The port hasn't connected the PSTN line.

Service Provider:

Status

OK: Succeed registration, trunk is ready for use. Unreachable: The trunk is unreachable. Failed: Trunk registration failed.

3.3 Basic

3.3.1 Extension

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

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子 Back 🝷 🕥 🕤 💌	🔁 🏠 🔎 Se	earch 🤺 Favorites 🤞	🛛 🔗 - 😓 🖂				
dress 🕘 http://192.168.5.18	%/cgi/WebCGI?1000						✓ → Go 链
MyPBX	Embedded H	Hybrid IP-PBX for :	Small Businesses				Logout
Status Monitor	Extension	φ					
Line Status				Ext	ension List		
Basic 🔅	FXS Extension	ons					
Basic 🍣	Port	Extension	Full Na	me	Caller ID		
Trunks	3	601	601		601	😼 Edit 🗴 Delete	
Outbound Routes	4	602	602		602	🔊 Edit 🗴 Delete	
Auto Provision	5	603	603		603	🔊 Edit 🕺 Delete	
Inbound Call Control (2) IVR Queues	VoIP Extensi Create Ne		Multiple Extensions	Modify Selected Exte	ensions 🗴 Delete Sele	cted Extensions	
Custom Prompts		Extension	Туре	Full Name	Caller ID		
Ring Groups		500	SIP	500	500	🐕 Edit 🕺 Delete	
Inbound Routes Blacklist		501	SIP	501	501	🐕 Edit 🛛 🗶 Delete	
		502	SIP	502	502	🐕 Edit 🛛 🗶 Delete	
Internal Settings 🔹 🔕		503	SIP	503	503	🐕 Edit 🛛 🗶 Delete	
Options Business Hours		504	SIP	504	504	🐕 Edit 🛛 🗶 Delete	
Business Hours Feature Codes		505	SIP	505	505	🐕 Edit 🛛 🗶 Delete	
SIP Settings							
IAX Settings Voicemail 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. 'Bob Jones' .

·Caller ID

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



2) Voicemail

·Enable Voicemail

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

•Voicemail Access PIN

Voicemail Password for this extension, i.e. '1234' .

3) Mail Setting

•Enable Send Voicemail

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

·Send Voicemail to Email Address

This option defines whether or not voicemails/Fax is sent to the Email address as an attachment.

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

4) Flash

·Hook Flash Detection

Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MyPBX before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MyPBX 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.

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 MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MyPBX 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 SettingsRxgain: The Volume sent to FXS extension.Txgain: The Volume sent out by the FXS extension.

9) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX 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 MyPBX.

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 filed. 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

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

·spy modes

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



Normal spy: you can only hear the call, but can't talk

Whisper spy: you can hear the call, and can talk with the monitored extension Barge spy: you can hear the call and talk with them both

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

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

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

11)Call Duration Settings

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

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



General				
			D	
	on: 602		Port: 7	
Nam	ne0: 602	С	aller ID0: 602	
Voicemail				
Enable Voi	icemail 🛈 🛛 Voicemail	Access PIN #0	602	
Mail Setting	11.4			
Email Address	end Voicemail			
		'SMTD Sattings for	. Voice and silling the	"Voicemeil Cettinge"
	ensure that the section ' perly configured before u	-	r voicemaii (in the	voicemail Settings)
	,	J		
Flash				
Hook Flash De	etection 🛈 : 1000	ms		
Group				
Pickup Group	▼			
F _1				
Follow me				
Follow me:	Always No answer	Transfer to:	Voicemail	
rollow me.	☑ No answer ☑ When Busy	Transfer to:	C Number	
	I vvnen Dusy			
Other Option:	S			
Call Waitin	ng 🛈 🗖 🗖 DND 🛈	User V	Web Interface 🛈	Ring Out 0: 30
Volume Setti	nas			
	xgain 🛈 : 🕬 🔻		Txgain 🛈 :	40% 🔻
	-			
Mobility Exte			•	
Enable Mo	bility Extension	Mobility Extension	n Number 🙂 :	
Spy Settings				
	Allow Being S	Spied	Spy Modes 🛈 :	+
Call Duratia	Sotting			
Call Duration				
	tion 🙂 : 👘 s			

Figure 3.3.1.1



3.3.1.2 VOIP Extension

A VOIP extension is a SIP/IAX Account that allows an IP Phone or an IP Soft-Phone client to register on MyPBX.

1. Add VOIP Extension

Go to Extensions \rightarrow VOIP Extensions \rightarrow Create New Extension

1) General

·Type

Extension type: SIP, IAX or SIP/IAX.

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

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: 'Bob Jones'

·Caller ID

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

2) Voicemail

•Enable Voicemail

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

•Voicemail Access PIN

The voicemail Password for this extension, i.e. '1234' .

3) Mail Setting

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

•Enable Send Voicemail

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

·Email Address

Email address used to receive the voicemail or Fax. **Note**: Please ensure that the section 'SMTP Settings For Voicemail'(in the



'Voicemail Settings') have been properly configured before using this feature.

4) Group

·Pickup Group

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

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

5) Follow me (Call Forwarding)

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

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

6) Other Options

.Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the 'When busy' follow me options will not be available. The call waiting function of IP phone has higher priority than MyPBX'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 MyPBX User Web interface, which can be used to check voicemail and extension recordings. Users may login to MyPBX 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

MyPBX 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

•Enable SRTP

Enable extension for SRTP (RTP Encryption).

·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. More details for the system security configuration, please refer to **APPENDIX B**



MyPBX Security Configuration Guide.

9) IP Restriction

•Enable IP Restriction

Check this option to enhance the VoIP security for MyPBX. 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. More details for the system security configuration, please refer to <u>APPENDIX B</u> <u>MyPBX Security Configuration Guide</u>.

Permitted 'IP address/Subnet mask'

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

10) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX 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 MyPBX.

•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 filed. 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.



Extension - 500
General Type: SP Name : 500 Caller ID : 500
Voicemail Image: Second state of the second state of th
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
Follow me Image: Always Follow me: Image: No answer Transfer to: Image: Number Image: When Busy Image: Number
Other Options Call Waiting DND User Web Interface Ring Out : 30
Spy Settings Allow Being Spied Spy Modes ①:
Optional Settings ≈
VoIP Settings NAT①: Qualify: Image: Constraint of the set of t
IP Restriction Enable IP Restriction Permitted 'IP address/Subnet mask' 1 Permitted 'IP address/Subnet mask' 2 Permitted 'IP address/Subnet mask' 3 Permitted 'IP address/Subnet mask' 4 Permitted 'IP address/Subnet mask' 4
Mobility Extension Enable Mobility Extension Mobility Extension Number
Call Duration Setting Max Call Duration

Figure 3.3.1.2



2. Add Multiple Extensions

Go to Extensions \rightarrow VOIP Extensions \rightarrow Add Multiple Extensions.

- 1) Select the number of extensions that you would like to create.
- 2) Select the type of extension that you would like to create.
- 3) Fill in the starting extension number.

Add Multiple Extensions X
Create 5 💟 SIP 💟 extensions starting from: 506 Tip: After pressing the 'Create Extensions' button, you may use the 'Modify The Selected Extensions' button to modify the properties for these extensions.
Create Extensions

Figure 3.3.1.3

3. Edit VOIP Extension

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

3.3.2 Trunk

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

My PBX	Embedded Hybrid IP-PBX for S	Small Businesse				Logout
Status Monitor	🕨 Manage Trunks 🌣					
Line Status			Trun	klist		
	Analog Trunk					
Basic 🙁	Trunk Name		Port/Hostr			
Extensions						_
Trunks	pstn1		1		🔊 Edit	
Outbound Routes	pstn2		2		🔂 Edit	=
Auto Provision	pstn6		6		🔊 Edit	
Inbound Call Control 🔕	GSM Trunk					
IVR	Trunk Name		Port/Hostr	name/IP		
Queues	GSM11		11		N Edit	
Custom Prompts					<u> </u>	
Ring Groups	BRI Trunk					_
Inbound Routes	Trunk Name		Port/Hostr	name/IP		
Blacklist	BriTrunk7		7		😼 Edit	
Internal Settings	BriTrunk8		8		🔊 Edit	
Options	VolP Trunk					
Business Hours	+ New VolP Trunk					
Feature Codes						_
SIP Settings	Provider Name	Туре	Hostname/IP	User Name		
IAX Settings	8032	SIP	192.168.5.99	8032	🔊 Edit 📁 🗶 Delete	
Voicemail Settings	Service Provider					
DISA						
Conferencing	+ New Service Provider					
Paging Groups PIN User Settings	Provider Name	Туре	Hostname/I	Р		
Speed Dial Settings	192.168.5.190	SIP	192.168.5.19	90	🔊 Edit 😕 Delete	
Music On Hold Prompts						~
🙆 Done					Internet	

Figure 3.3.2



3.3.2.1 Analog Trunk (FXO)

1. Edit Analog Trunk (FXO)

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

1) General

•Trunk Name

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

·Volume Setting

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

2) Busy Detection

·Busy Detection

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

·Busy Count

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

·Busy Interval

The busy detection interval

·Busy Pattern

If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many Countries, it is 500 msec on, 500 msec off. Without Busy Pattern specified, MyPBX will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MyPBX 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 MyPBX to clarify Caller ID incoming calls.



Edit Analog Trunk - pstn4			х
	Trunk Name 🛈 :	pstn4	
	Volume Setting	40% 🔹	
Busy Detection			
	Busy Detection 🛈 :	Yes 🔻	
	Busy Count	4	
	Busy Interval 🛈 :	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	

Figure 3.3.2.1

3.3.2.2 GSM Trunk

1. Edit GSM Trunk

1) General

•Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'GSM9'

·Volume Setting

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

·Pin Code

Please enter your SIM card pin code here if your card has a pin code.



Edit GSM Trunk - GSM13	х
General	
Trunk Name 🛈 : GSM13	
Volume Setting 🛈 : 40% 🔻	
PIN Code:	
Warning: Be careful. If you failed to enter your correct PIN code 3 times in succession, SIM card will be blocked.	
Save X Cancel	

Figure 3.3.2.2

3.3.2.3 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. 10000000 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 MyPBX can dial this switch using overlap digits . If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then MyPBX 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

Echocancel 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.



Edit BRI Trunk - BriTrunk	7					Х
Trunk Name 🛈 :	BriTrunk7					
Signaling:	BRI-CPE	•	Switch Type 🛈 :	euroisdn	•	
PRI Dialplan 🛈 :	unknown	•	Reset Interval 🛈:	never	▼ s	
PRI Local Dialplan 🛈 :	unknown	•	Overlap Dial 🛈 :		-	
PRI Indication 🛈 :	Inband	•	Enable Facility 🛈 :		-	
Nsf🛈 :		•	Echo Cancellation 🛈 :	Off	-	
Hide Caller ID 🛈 :	No	•	Codec:	alaw	•	
Caller ID Prefix						
International Pref	īx:		National	Prefix:		
Local Pref	ix:		Private	Prefix:		
Unknown Pref	ix:					
DOD Settings						
Global DOD:						
DOD :		Asso	ciated Extension : 601 🔻		↑Add DOD	
Save X Cancel						

Figure 3.3.2.3

3.3.2.4 VOIP Trunk

1. 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: 'yeastar'.

·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. Set as 0 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 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.

Edit VoIP Trunk - test	2	۲
Provider Name:	test	
Hostname/IP:	192.168.4.136 : 5060	
Domain:	192.168.4.136	
User Name:	1111	
Authorization Name:	1111	
Password:	••••	
From User:		
Online Number 0:		
Maximum Channels 🛈 :	0	
Caller ID 🛈 :		
	Enable Outbound Proxy Server	
Codecs :	First: a-law V Second: u-law V Third: GSM V	
	Fourth: None Fifth: None	
Transport:	UDP Enable SRTP	
DTMF Mode:	rfc2833 🔹	
DOD Settings		
DOD:	Associated Extension: 601 - Add DOD	
	Save Cancel	

Figure 3.3.2.4.1



2. Add IAX Trunk

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

1) General

·Type

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

•Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar2'.

·Hostname/IP

Service provider's hostname or IP address. 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

·Username

Username of IAX account . Used for IAX trunk registration.

Password

Password of IAX account .

.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. Set as 0 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 setting in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

2) DOD Setting

DOD

Direct Outward Dialing Number.



·Associated Extension

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

Add VoIP trunk	Х
Туре:	
Provider Name:	
Hostname/IP:	: 4569
User Name:	
Password:	
Online Number 🛈 :	
Maximum Channels 🛈 :	0
Caller ID 🛈 :	
DOD Settings	
DOD:	Associated Extension: 601 ▼ ↑Add DOD
	Save Cancel

Figure 3.3.3.4.2

3.3.2.5 Service Provider

1. Add Service Provider

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

1) General

∙Туре

SIP or IAX

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

IAX - Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.



·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, 4569 is the standard port number used by IAX 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 SPS 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	
Codecs :	First: GSM 🔻 Second: a-law 🔻 Third: u-law 💌	
	Fourth: None Fifth: None	
Transport:	UDP -	
Qualify:		
DTMF Mode:	rfc2833 🔹	
DOD Settings		
Global DOD:		
DOD :	Associated Extension : 601 ▼ ↑Add DOD	
	Save Cancel	
	Figure 3.3.2.5	

3.3.3 Outbound Routes

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

MyPBX	Embedded Hy	/brid IP-PBX for Small	Businesses		Logout
Status Monitor	🕨 Manage Out	oound Routes 🔅			
Line Status	+ New Outbo	und Route	Outbound Routes		
Basic 🙁		Route Name	Dial Pattern		E
Extensions	⇒ ±	pstnout	9.	🔊 Edit 🛛 🗶 Delete	
Trunks	₹ \$	to190	х.	🔊 Edit 🛛 🗴 Delete	
Outbound Routes					
Auto Provision					
Inbound Call Control 🙁					_
IVR					
Queues					
Custom Prompts					
Ring Groups					
Inbound Routes					
Blacklist					
Internal Settings					
Options					
Business Hours					
Feature Codes					
SIP Settings					
IAX Settings					
Voicomail Cattinge					<u>×</u>
🛃 Done				🥑 Interr	net 🦪



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: **1NXXNXXXXX** 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.

·T.38 Support:



Enable T38 fax in this outbound route (Only for SIP Trunk).

·Rrmemory Hunt

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

·Member Extensions

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

Member Trunks

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

New Outbound Route	x
Route Name 🛈 :	
Dial Pattern 🛈 :	
Strip 🛈 : digits fr	om front
Prepend these digits	before dialing
Password:	
T.38 Support [©] : No	
Rrmemory Hunt	
Member Extensions	
Available Extensions	Selected
	Selected
10(SIP) >>> 100(SIP) → 101(SIP) → 102(SIP) → 500(SIP) ← 501(SIP) ← 503(SIP) ✓	
Member Trunks	
Available Trunks	Selected
1234(FXO) >>> pstn5(FXO) → pstn8(FXO) → pstn8(FXO) → pstn8(FXO) → BriTrunk9(BRI) ← BriTrunk10(BRI) ✓ Mexico(SIP) ≪	
🗸 Save 🔀 Car	ncel

Figure 3.3.3.1



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:

Yealink (T12,T18,T20,T22,T26,T28,T32,T38,VP530,VP-2009) 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 Yealink and Snom IP phone, MyPBX is not needed to be set as the only DHCP server any more.

			Phone Provisioning			
General Setti	ngs for Yealink					
<u>General Setti</u>	ngs for Aastra					
Phone Book						
Configured P	hone					
+ Add Phone	+ Add Bulk Phones	S Configure the Selected Phones	Celete the Selected Pho	nes		
	MAC Address	Manufacturer	Dhana Tura		Extension	Enabled
	WAC AUTIESS	manuracturer	Phone Type	Name	Extension	Ellabled
Not Configure		Manufacturer	Phone Type	Name	Extension	Ellabled
Not Configure	ed Phone	Refresh	Phone Type	Name	Extension	Enabled
Not Configure	ed Phone		Phone Type Manufacturer			Enabled e Type
Not Configure	ed Phone he Selected Phones	Refresh			Phone	
Not Configure	ed Phone he Selected Phones ID	Refresh MAC Address	Manufacturer		Phone	э Туре
Not Configure t	ed Phone he Selected Phones	Refresh MAC Address 001565113844	Manufacturer Yealink		Phone	э Туре
Not Configure t	ed Phone he Selected Phones ID 1 2	Refresh MAC Address 001565113844 001565114094	Manufacturer Yealink Yealink		Phone - -	э Туре - -
Not Configure t	ed Phone he Selected Phones	Refresh MAC Address 001565113844 001565114094 0015651be4a4	Manufacturer Yealink Yealink Yealink Yealink		Phone	э Туре - - -
Not Configure t	ed Phone he Selected Phones	Refresh MAC Address 001565113844 001565114094 0015651be4a4 001565132139	Manufacturer Yealink Yealink Yealink Yealink Yealink		Phone	э Туре - - -
Not Configure t	ed Phone he Selected Phones 1 1 2 3 4 5	MAC Address 001565113844 001565114094 0015651be4a4 001665132139 00156534136e	Manufacturer Yealink Yealink Yealink Yealink Yealink Yealink		Phone	э Туре - - - -

Figure 3.3.4

3.3.4.1 General Settings for Yealink

In this page, you can configure it before provisioning Yealink IP phones, including the items like general preferences, codecs, remote phone book and



firmware upgrade

· General Settings for reality *

Note: if firmware download server is enabled, IP phone will update the firmware automatically according the version and server you have configured during the provision process.

Concerned Producesces			Go Back to Phone Provisionin
Web servings embersion Adam Samede Finde Time Samede Finde Standard VITP Same conditional Time Finde conditional Values conditional Time Finde conditional Values conditional Time Finde conditional Values conditional Values conditional Values conditional Adam Codes Fire T12.111/20.122.128 conditional Codes Ad conditional Adam Codes Fire T12.111/20.122.128 (TRUE T28.128 conditional Codes Ad conditional Values Codes Fire V9508 or VP-2009 conditional Codes Codes Fire V9508 or VP-2009 conditional Values Codes conditional Values Codes Fire V9508 or VP-2009 conditional Time Firence Time condital 2	General Preferences		
Adam Planework # Freed		Language 0 :	English
Second Process for the Second Proces		Web server Type:	HTTP&HTTPS •
Time See: 40 - 40 - 40 - 40 - 40 - 40 - 40 - 40 -		Admin Password:	Fixed Prefix
Premay MP Bane: Secondary MP Bane: Depoint 0 org Secondary MP Bane: Depoint 0 org Dep Form: WW W 0 0 - Vecenal: WW W 0 0 - PHP UR: * Ausmatic * Dutom Conces A			admin
Secondary INP Searce: on good day ong Day light Samp Time: in wind in it is now in the in it is now in it is now in the in it is now in		Time Zone:	+8 China(Beijing)
Secondary INP Searce: on good day ong Day light Samp Time: in wind in it is now in the in it is now in it is now in the in it is now in		Primary NTP Server:	cn.pool.ntp.org
Daylett Samp Time: Saated Time Finanz: "WWW WW 100 Date Finanzi: "Vacenzi: Cotics # Audio Cotec: For 172 178 120 122 120 120 120 120 120 120 120 120			
Out of Forme: WWW uses 0 Voide 2001 Control 1 Prof UBL: Audio Code: 1212 118 120 122 728 728 178 178 178 1213 118 120 122 728 728 178 178 178 1213 118 120 122 728 728 178 178 179 1213 118 120 122 728 728 178 178 179 1213 118 120 122 728 728 178 179 179 1213 118 120 122 728 728 178 179 179 1213 118 120 122 728 728 178 179 179 1213 118 120 122 728 728 178 179 179 1213 118 120 120 128 178 179 129 129 129 129 129 129 129 129 129 12			
Vocenalit Yes PIP URL Automatic Cotecs # Automatic Ublable Codes Enable Codes UPU URL Provide UPU URL<		Time Format:	12 Hour 👻
Cotes # Custom 4udio Code: For 112,118,172,172,172,172,172,178,178 178,9 178,8 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9 178,9		Date Format:	WWW MMM DD 🔹
Codecs A Audio Code: For 112,118,120,122,125,123,138 Enable Code: Enable Code: 0721,5,9 0722,5,9 0722,7 0722,		Voicemail:	Yes 👻
Audo Code: For 172,118,703 172,218; 173 173 Disable Code:s Finable Code:s 0778,40 0772 0778,40 0774 0778,40 0774 0774,40 0774 <t< th=""><th></th><th>PNP URL:</th><th>Automatic Custom</th></t<>		PNP URL:	Automatic Custom
Image: Codes Image: Codes Image: Codes	Codecs 🕿		
Immune Download Server /> Emote Phone Book /> Firmware Download Server /> Enable: Details: Details: Details: Details: Details: Details: Details: Disable: Disable: <th></th> <th>Audio Codec For T12,T18,T20,T22,T26,T28,T32,T38</th> <th></th>		Audio Codec For T12,T18,T20,T22,T26,T28,T32,T38	
Image: Section of the sectio		Disable Codecs	Enable Codecs
Immune Download Server /P Enable Codes Immune Download Server /P Enable Download Server /P Enable Codes Immune Download Server /P Enable Download Server /P		G723_53	>>> PCMU
Addo Codec Far VP530 and VP-2009 Disable Codecs 0727 0728 0728 0729 0729 0720 0720 0721 0722 0722 0723 0725 0725 0725 0725 0726 0726 0725 07205		G726-16	G729
Audio Coder, Fiv VP-300 Disable Coders Coder, Fiver,		G726-32	G722
Audio Code: For VP:30 and VP:2009 0723 0723 0724 0727 <		G726-40	<u></u>
Audio Code: For VP:30 and VP:2009 0723 0723 0724 0727 <			««
Image: Disable Codecs Image: Disable Codecs Image: Disable Codecs Video Codec: For VP530 and VP-2009 Disable Codecs Image: Disable			
Provide Codec Provide Codec Udeo Codec Provide Codec Disable Codecs Provide Codec Provide Codec Provide Codec Provide Codeco			Enable Codest
Prove Book ★ Funnvare Download Server *>			
ACLC □ LCC □ Video Codec For VP530 and VP-2009 □ Disable Codecs Enable Codecs □ □ Perfect Perfect Perfect Perfe		G723	PCMA CZ20
Video Code: For VP530 and VP.2009 Disable Code:: Prest-es:: Prest-es:: Phonebook VRL Phonebook Name 1: 2: 3: 4: 5: T12 Firmware Name: T13 Firmware Name: Firmware Name: T13 Firmware Name: T13 Firmware Name: Firmware Name: Firmware Name: Firmware Name:		AACLC	
Video Codec For VP530 and VP-209 Disable Codecs Passed Passed Phone Book 念 Phone Book Name 1 2 3: 4: 5: Finavere Download Server ♪ Enabled: Disable firmware Name: 12 Firmware Name: 120 Firmware Name:<		ILBC	-
Video Codec For VP530 and VP-2009 Disable Codecs Phane Implement			
Disable Codecs Pione Book \$ Phonebook URL Phonebook Vame 1 2 3 3 3 4 5 Firmware Download Server \$ Enabled 12 Firmware Name: 120 Firmware Name:			
Remote Phone Book 冬 Phonebook URL Phonebook Name 1		Video Codec For VP530 and VP-2009	
Remote Phone Book 秦 Phonebook URL Phonebook Name 1:		Disable Codecs	
Remote Phone Book * Phonebook URL Phonebook Name 1 2 3: 4: 5: Firmware Download Server * Enabled: Daskied T12 Firmware Name: T22 Firmware Name: T22 Firmware Name: T22 Firmware Name: T23 Firmware Name: T33 Firmware Name: T35 Firmware Name: T37 Firmware Name: T38 Firmware Name: T39 Firmware Name: T30 Firmware Name: T30 Firmware Name: T37 Firmware Name: T38 Firmware Name: Firmware Name: T39 Firmware Name:			H263
Remote Phone Book Phonebook URL Phonebook URL Phonebook Name 1: 2: 3: 3: 4: 5: Firmware Download Server Enabled: Deabled T12 Firmware Name: T22 Firmware Name: T22 Firmware Name: T22 Firmware Name: T38 Firmware Name: T39 Firmware Name: T30 Firmware Name: T30 Firmware Name:			→ mp4v-es
Remote Phone Book 余 Phonebook URL Phonebook Name 1:			
Remote Phone Book 余 Phonebook URL Phonebook Name 1:			
Phonebook URL Phonebook Name 1:			88
Phonebook URL Phonebook Name 1:	Remote Phone Book ☆		
1:		Phonebook U	RL Phonebook Name
2:			
3: 4: 5: Firmware Download Server ◆ Enabled: Daabled T12 Firmware Name: T18 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T38 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP5200 Firmware Name: VP5200 Firmware Name: Server Type: ftp			
4: 5: Firmware Download Server Enabled: Dsabled T12 Firmware Name: T18 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T28 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP52009 Firmware Name: VP-2009 Firmware Name: Server Type: ftp			
S: Firmware Download Server ≈ Enabled: Dsabled T12 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP5300 Firmware Name: VP52009 Firmware Name: Server Type: ftp			
Firmware Download Server ≈ Enabled: Disabled T12 Firmware Name: T18 Firmware Name: T20 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T38 Firmware Name: T38 Firmware Name: Server Type: C ftp C http			
Enabled Disabled T12 Firmware Name: T18 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP530 Firmware Name: Server Type: Fitp			
T12 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP5200 Firmware Name: Server Type: ftp ftp	Firmware Download Ser		
T18 Firmware Name: T20 Firmware Name: T22 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP530 Firmware Name: VP5200 Firmware Name: Server Type: ftp ftp		Enabled:	Disabled
T20 Firmware Name: T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP530 Firmware Name: Server Type: ftp http		T12 Firmware Name:	
T22 Firmware Name: T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T33 Firmware Name: VP530 Firmware Name: VP530 Firmware Name: VP52009 Firmware Name: Server Type: ftp Fittp		T18 Firmware Name:	
T26 Firmware Name: T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP52009 Firmware Name: Server Type: Ctp Mttp		T20 Firmware Name:	
T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP52009 Firmware Name: Server Type: C ftp C http		T22 Firmware Name:	
T28 Firmware Name: T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP52009 Firmware Name: Server Type: C ftp C http		T26 Firmware Name:	
T32 Firmware Name: T38 Firmware Name: VP530 Firmware Name: VP-2009 Firmware Name: Server Type: C ftp C http			
T38 Firmware Name: VP530 Firmware Name: VP-2009 Firmware Name: Server Type: C ftp C http			
VP530 Firmware Name: VP-2009 Firmware Name: Server Type: C ftp C http			
VP-2009 Firmware Name: Server Type: C ftp C http			
Server Type: C ftp C http		VP530 Firmware Name:	
		VP-2009 Firmware Name:	
HTTP URL:		Server Type:	C ftp C http
		HTTP URL:	



Figure 3.3.4.1

3.3.4.2 General Settings for Aastra

In this page, you can configure it before provisioning Aastra IP phones, including the items like local dial plan, time and date setting programkeys configuration and so on.

General Preferences					
	Local D	tial Plan			
		rminator 🗖 Enable			
Time and Date Setting	Send Dial Plan Ter	minator 📋 Enable			
Time and Date Setting					
		Server1			
	Time	Server2			
Auto-Resync					
	-	ic Mode none		•	
	Resy	nc Time 00:00	•		
Programkeys Configuration 奈					
	Кеу Туре			Line	
	1 None	· ·	Global		
	2 None		Global	-	
	3 None		Global		
	4 None	_	Global	-	
	5 None		Global		
	6 None	_	Global	*	
	7 None	·	Global	-	
ľ	8 None	v	Global	*	
Softkeys Configuration 奈					
		Bottom Keys			
Show All Botto Key	mkeys Type	Label	Value	Line	
□ 1	None 👻	Laber	Value	Global	
	None 👻			Global	
□ 2 □ 3	None 👻			Global	
4	None 👻			Global	
5	None			Global	
6	None 🔻			Global	
		Top Keys			
Show All Topke	eys				
Key	Туре	Label	Value	Line	
L 1	None 👻			Global	
□ 2	None 👻			Global	
3	None 👻			Global	
4	None 👻			Global	
5	None 👻			Global 🔻	

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 🗘
ŕ	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 3.3.4.3

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.

Type:	None	•	Gr	oup: None	<u> </u>
Nick Name			Favorit	e0: No	•
Organization 0:			Titl	eO:	
Email ⁰ :			Birthda	y @ :	
First Name			Fa	e 0 :	
Office Number:	-		Mobile Num	nber:	
Home Number:					
0 I N I 0					
Sub Number					
Sub Name:		S	ub Number:		↑Add Sub
Sub Name:		S	ub Number:		↑Add Sub

Figure 3.3.4.3.1



2) Upload Phonebook

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

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

3.3.4.4 configure phone

Let's take provisioning yealink as an example Create New Phone have two modes, Create New phone in webpage and Upload the IP Phone's configure file.

1. Add new phone via webpage

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

Enabled

Choose yes or no to enable or disable this extension

NewConfig

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

MAC address

Input the MAC address of IP phone

•Name

Put the name of this Phone here.

Manufacturer

You can choose the Manufacturer of IP phone

·Phone Type

Choose the model of your phone. Only for snom phone

·Call Waiting

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

•Key as Send

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



•Auto redial

Enable the auto redial for IP Phone

•Auto answer

Configure if auto answer is allowed for IP phone

•Phone book

Enable the feature of phone book of IP phone

·Line

Extension: Selected the extension number for IP Phone.

Label: It is shown on the LCD for users to identify the account.

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

Add Phone					Х
General					_
Enabled:	Yes 🔻	NewConfig	Yes	-	
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 Ex	xtension:	Label:	L	ine Active:	
Line2 Ex	xtension:	Label:	L	ine Active:	
Line3 Ex	xtension:	Label:	L	ine Active:	
Line4 Ex	xtension:	Label:	L	ine Active:	
Line5 Ex	xtension:	Label:	L	ine Active:	
🗆 Line6 Ex	xtension:	Label:	L	ine Active:	

Figure 3.3.4.4.1

2) Audio codec

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

Audio Codec As General 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 Sett	ings 🛠					
Memory Key						
Key	Туре		Value	Line	Extensio	n
DSS Key1	N/A	-		line1	•	
DSS Key2	N/A	•		line1	•	
DSS Key3	N/A	-		line1	•	
DSS Key4	N/A	-		line1	•	
DSS Key5	N/A	-		line1	-	
DSS Key6	N/A	-		line1	•	
DSS Key7	N/A	-		line1	•	
DSS Key8	N/A			line1	•	
DSS Key9	N/A			line1	v	
DSS Key10	N/A	•		line1	•	
Line Keys Setting	~ ^					
Line Keys Setting	-					
Key	Туре	Value	Labe	Line	e Exten	sior
Line Key 1	N/A 🔻			Line1		
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, MyPBX 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.



Add Phone				Х
General				
Enabled:	Yes	•		
MAC Address:	001565113844	Name:		
Manufacturer:	Yealink	Phone Type:	T12 🔻	
Call Waiting:	Enabled	 Key As Send: 	# •	
Auto Redial:	Disabled	Auto Answer:	Disabled -	
Phone Book:	Enabled	•		
Line				
Line1 E	xtension:	✓ Label:	Line	Active:
Audio Codec As Genera Custom	-			
	Disable Codecs		Enable Codecs	
G723_53 G723_63 G726-16 G726-24 G726-32 G726-40		 »» »» PCMA PCMU G729 G722 ← ≪« 		
Line Keys Settings ≫				
		Save X Cancel		

Figure 3.3.4.5

3.3.4.6 Upload a file

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

Note: the file format must be .cfg

Please edit the configuration files in advance before uploading.

	Upload a file	х
1	Choose a File to Upload:	浏览
And a second sec	Upload	X Cancel

Figure 3.3.4.6



3.4 Inbound Call Control

3.4.1 IVR

When there's an inbound call aims at Auto Attendant, MyPBX will play an IVR recording and route the caller to the requested destination (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 DTMF digits inputted by the user.

🗿 MyPBX - Embedded Hyb	rid IPPBX for Small	Businesses - /	Aicrosoft Internet Explorer			-
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites	<u>T</u> ools <u>H</u> elp				//	1
🚱 Back 🝷 🕥 🐇 💌	😰 🏠 🔎 Searc	:h 🤺 Favorite	s 😧 🍃 🌺 🚍			
Address 🕘 http://192.168.5.18	86/cgi/WebCGI?1000			✓	▶ Go 链接	alt.
MyPBX	Embedded Hyt	orid IP-PBX	for Small Businesses		Logout	
Status Monitor 🔹	🕨 Manage IVR 🕸)				
Line Status	+ Create New	IVR	IVR			
Basic 🙁	Name	Number	Allow Dialing of Other Extensions			1
Extensions	welcome	660	Yes	😒 Edit 🛛 🗶 Delete		
Trunks	L				_	
Outbound Routes						
Auto Provision						
Inbound Call Control 🔹						
<u>IVR</u>						
Queues						
Custom Prompts						
Ring Groups Inbound Routes						
Blacklist						
Internal Settings 🔹						
Options						
Business Hours Feature Codes						
SIP Settings						
IAX Settings						
Voicomail Sattingo						2
🙆 Done				🥑 Internet		.:

Figure 3.4.1

3.4.1.1 Create IVR

Click 'Create New IVR'.

1) General

·Number

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

•Name

A name for the IVR

Prompt

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



·Repeat Count

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

·Key Timeout

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

·Allow Dialing of Other Extensions

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

2) Key Press Events

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

·Key

The Key pressed when the callers hear the IVR prompt.

Action

When the callers press the corresponding key, the action MyPBX executes.
No Action: Do nothing
Connect to Extension: Connect the call to an extension.
Connect to Voicemail: Connect the call to the voicemail of an extension
Connect to RingGroup: Connect the call to a ringgroup.
Connect to IVR:Connect the call to an IVR.
Connect to Conference Room: Connect the call to a conference room.
Connect to DISA:Connect the call to a DISA.
Connect to Faxes:Connect the call to a queue.
Connect to Faxes:Connect the call to Faxes of extensions.
Dial by Name:The callers can dial the name of an extension to connect to the corresponding extension.
Hung up: Hung up the call.

Destination

Where will MyPBX route the call when the action occurs.

•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.



Edit l	VR - 660		Х
	Number 🛈 :	660	
	Name 🛈 :	660	
	Prompt ¹ :	aa Custom Prompts	
F	Repeat Count 🛈 :	3 💟	
	Key Timeout 🛈 :		
		Allow Dialing of Other Extensions	
	– 🛈 Keypress Ev		
	Key	Action Destination	
	0	No Action	
	1	Connect to Extension V Extension 101	
	2	Connect to Voicemail 🔽 Voicemail 101	
	3	Connect to RingGroup	
	4	Connect to IVR 🔽 IVR 660	
	5	Connect to Conference Room 🔽 Conference Room 640 💟	
	6	Connect to DISA VIEW DISA test	
	7	Connect to Queues V Queues 680	
	8	Connect to Faxes V Faxes 101	
	9	Dial by Name	
	#	Hangup	
	*	No Action	
	Timeout	No Action	
	Invalid	No Action	
		Save Cancel	

Figure 3.4.1.1

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. This field allowed to be blank empty, it means the agents can login to this queue without password.

•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. Selecting 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 MyPBX to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

Frequency



How often to announce queue position and estimated hold time. **Note:** '0 seconds' means disable the announcement

4) Periodic Announcements

·Prompt

Select a prompt file to play periodically.

Frequency

How often to announce a prompt to the caller.

5) Events

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

6) Failover-Destination

Defines the failover action. A failover occurs after the user reach the Queue max wait time.

7) Others

•Music On Hold

Select the 'Music on Hold' Class for this Queue.

·Leave When Empty

This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.

Yes: Callers are forced out of a queue when no agents are logged in. No: Callers will remain in a queue with no agents.

·Join Empty

This option controls whether callers can join a call queue that has no agents. There are two options,

Yes: Callers can join a call queue with no agents or only unavailable agents No: Callers cannot join a queue with no agents The default option is No.

•Agent Announcement

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

·Join Announcement

Announcement played to callers once prior to joining the queue.

·Retry

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

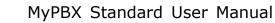


·Wrap-up time

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

New Queues	x
Queue Name 🛈 :	692
Queue Number 🛈 :	692
Queue Password 👀:	
Queue Agent Timeout	
Queue Max Wait Time 1	
Queue Ring Strategy	
Agents	
Available Agents	Selected
500(SIP) 501(SIP)	>>
502(SIP) 503(SIP)	→
504(SIP) 505(SIP)	←
506(SIP) 507(SIP)	
	**
Caller Position Announcements	
Announce Position	Yes 💌
Announce Hold Time 🛈	
Frequency 🛈	30 seconds 💌
Periodic Announcements	
Prompt	
Frequency 0	30 seconds 👻
Events	
Кеу	
Action	End Call
Destination	
Failover-Destination	
Action	End Call
Destination	~
C Others	
Music On Hold 🛈	
Leave When Empty 🛈	
Join Empty	
Agent Announcement	
Join Announcement	
Retry	
Wrap-up time 🛈	: 30
V Sav	e 🔀 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
	File Name: welcome Dial extension: 500 v to record a new voice prompt Record X Cancel	

Figure 3.4.3.1

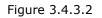
2. Upload Prompt

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, 1Kb/s.

Upload Prompt					Х
The file size	ze must no	t be larger than 1	8MB!		
WAV format: gsm 6.10 8kHz,Mono,1	Kb/s √ alaw	/ulaw 8kHz,Mono	,1Kb/s√ pcm 8kH	z,Mono,16Kb/s	
Choose a File to Upload 🛈 :			Browse		
	Upload	X Cancel			





3.4.4 Ring Groups

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

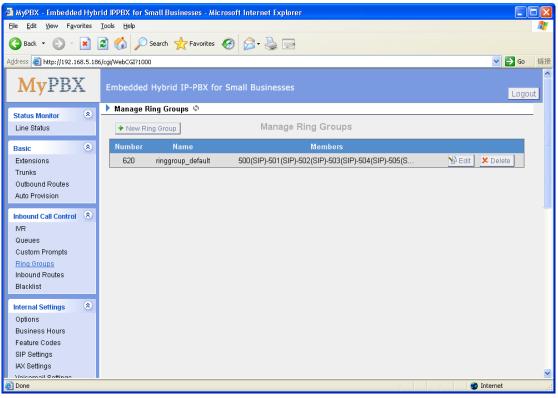


Figure 3.4.4

3.4.4.1 Create Ring Group

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

1) General

·Ring Group Name

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

•Ring Group Number

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

Strategy

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

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



•Seconds to ring each member

1. If the strategy is 'Ring All Simultaneously', it means set the number of seconds to ring this group before routing the call according to the 'Destination if No Answer' settings.

2. If the strategy is 'Ring Sequentially', it means set the number of seconds to ring a single extension before moving onto the next one.

2) Ring Group Members

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

3) Destination If No Answer

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

Add Ring Group	X	ζ
Ring Group Name 🛈 :	620	
Ring Group Number	620	
Strategy 🛈 :	Ring all simultaneously	
Seconds to ring each member 🛈 :	60	
Ring Group members (1) Available Extensions	Selected	
500(SIP) ▲ 501(SIP) ■ 502(SIP) ■ 503(SIP) ■ 504(SIP) ■ 505(SIP) ● 602(FXS) ▼		
Destination If No Answer:		
6	End Call	
С	Extension 602 -	
С	Voicemail Voicemail 602 -	
Destination:	IVR VR welcome -	
C	Ring Group 🔻	
C	Conference Room 640 -	
C	Queues	
Save	e X Cancel	

Figure 3.4.4.1



3.4.5 Inbound Routes

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

🗿 MyPBX - Embedded Hybi	rid IPPBX for Small Businesses	- Microsoft Internet Exp	olorer	
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Address 🗿 http://192.168.5.18	6/cgi/WebCGI?1000			✓ ● Go 链接
My PBX	Embedded Hybrid IP-PB	X for Small Business	ses	Logout
Status Monitor 🔹	Manage Inbound Routes	φ		
Line Status	+ New Inbound Route	Manag	e Inbound Routes	
Basic 🔊 Extensions Trunks Outbound Routes Auto Provision	Route Name pstnin	DID Number	Caller ID Number	Edit 🗶 Delete
Inbound Call Control IVR Queues Custom Prompts Ring Groups Inbound Routes Blacklist				
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Viologmail Softings				
Done				🔮 Internet

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: 'pstncallin' 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.

For more information, please refer to **Appendix G How to Use DID.**

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

MyPBX 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.

How do I configure distinctive ring tones? Please refer to <u>APPENDIX E</u>.



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 Yealink and Snom phone.

•Enable Callback

You can enable the callback function of this inbound route. If you want to configure the callback function, please refer to <u>chapter 3.5.12</u>

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.

Business Days

Define where the calls will be routed during Business Days.

•Office Days

Select one defined business days office days.

·Office Hours Destination

Configure where to route the incoming calls during office hours.

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

•Extension Route the incoming calls to a specific extension.

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

·IVR Route the incoming calls to a specific IVR.

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

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

•DISA Route the incoming calls to a specific DISA.

•Queues Route the incoming calls to a specific Queue.



Faxes

Route the incoming faxes to a specific extension's mail address. Note: This function only supports T.38 faxes.

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 MyPBX 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 which 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.

6) Fax Detection

Configure if detecting faxes in this inbound route.

Destination

Configure where the faxes will be routed when faxes are detected.

•No detect

Do not detect faxes.

•Custom Email

Customize an E-mail address to receive the faxes. You should first configure the 'Voicemail Settigns->SMTP Settings for Voicemail' correctly before you use this option.

Faxes

Send faxes to an extension. If choosing a FXS extension here, the fax will be



sent to the FXS port selected, you should connect a fax machine to this FXS port. If Choosing a VoIP extension, the fax will be sent to the extension's voicemail as an attachment.

Note: If you receive faxes with custom Email address, the 'SMTP settings' of 'Voicemail Settings' should be configured successfully in advance. If you receive faxes with E-mail address configured in VOIP extension voicemail, you should first make sure the tested email to your email address works fine.

Create New Inbound Rou	ıte		Х
General			
	Route Name 🛈 :		
	DID Number 🛈 :		
	Extension 🛈 :		
c	Caller ID Number 🛈 :		
Dist	tinctive Ringtone 🛈 :		
	Enable Callback : 🔤	Callback Settings	
⊂ Member Trunks			
Available	• Trunks	Selected	
456(FX0) pstn5(FX0) pstn6(FX0) GSM9(GSM) BriTrunk7(BRI) BriTrunk8(BRI)	>>> 		
 Business Days 			
Office Days :	default 🔽		
Destination :	End Call		
Non-office Hours			
Destination :	End Call		
During Holidays			
Holiday :			
Destination :	End Call		
Fax Detection			
Destination :	No Detect		
	🖌 Save 🕻	Cancel	

Figure 3.4.5.1



3.4.6 Blacklist

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

🕘 MyPBX - Embedded Hybri	d IPPBX for Small Businesses - Microsoft Internet Explorer	
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>I</u>	jools <u>H</u> elp	
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Address 🕘 http://192.168.5.186/	cgi/WebCGI?1000	✓ → Go 链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Logout
Status Monitor 🔹 -	🕨 Manage Blacklists 🕸	
Line Status	+ New Blacklist Manage Blacklists	
Basic 🙁	Blacklist	
Extensions	10000	🗴 Delete
Trunks	10086	X Delete
Outbound Routes	1234567	X Delete
Auto Provision	13850050500	X Delete
Inbound Call Control IVR Queues Custom Prompts Ring Groups Inbound Routes Blacklist		
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Melsomail Settings		M
🛐 Done		🥑 Internet 🛒

Figure 3.4.6

3.4.6.1 Create Blacklist

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

	Create New Blacklist X
	Blacklist Number :
	Type : Inbound Save Save Cancel
l	

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 .

•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 before transferring the call. Otherwise, the call will be transferred directly without any prompt. Default: Yes.

·Virtual Ring Back Tone

It's only for GSM trunk. Once enabled, when the caller call out with GSM trunks, the caller will only hear the virtual ring back tone generated by the system before callee answers the call.



· 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: To display IVR's name, please press the key instead of the extension number directly.

·Follow Me Prompt

Configure whether to play a prompt 'please wait while trying to look at the person you are calling' when transfer a call by follow me settings.

•Music on hold for Follow Me

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

·Invalid Phone Number Prompt

Configure the prompt when the dialed phone number is invalid.

Busy Line Prompt

Configure the prompt when the dialed phone number is busy.

·Dial Failure Prompt

Configure the prompt when dial failed due to conjunction no-available channel.

2) Extension Preferences •User Extensions

The default value is 500 to 616

•Ring Group Extensions

The default value is 620 to 629

·Paging Group Extensions

The default value is 630 to 639

·Conference Extensions

The default value is 640 to 659

·IVR Extensions

The default value is 660 to 679

·Queue Extensions

The default value is 680 to 689



General Preferences	
Ring Timeout 🛈 :	30 s
MAX Call Duration 🛈 :	6000 s
Maximum Concurrent Calls	0
Music On Hold:	calmriver 💌
Tone Region 🛈 :	United States/North America
HTTP Bind Port	80
FXO Mode 🛈 :	
Virtual Ring Back Tone	
Distinctive Caller ID	
Follow Me Prompt 0:	
Music on hold for Follow Me	
Invalid Phone Number Prompt	
Busy Line Prompt ¹	
Dial Failure Prompt	•
xtension Preferences	
User Extensions :	
Ring Group Extensions :	
Paging Group Extensions :	630 to 639
Conference Extensions :	640 to 659
IVR Extensions :	660 to 679
Queue Extensions :	680 to 689
Res	set to Defaults

Figure 3.5.1

3.5.2 Business Hours

1) General

- •Enable Business Hours
- **·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 enalbed for the device whatever the general setting is.

·Disable Office (closed) timing

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

3) Business Days You can setup the business hours here.

4) Holidays

You can setup the holidays here.

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



Business Hours 🔅

General			
C Enab	le Business Hours	tours	
Others			
*81	Enable Office Closed Timing	Office Closed Timing State: Disabled	
*82	Enable Office Timing	Office Timing State: Disabled	
*081	Disable Office (Closed) Timing		
Business Da			
Name		Details	
default	Mon 08:30-12:00/14:00-18:00/1	9:00-22:00 Tue 08:30-12:00/14:00-18:00/19:00-22:00 Sat 08:30-12:00/00:00-00:00/00:00-00:00	🔊 Edit 🗴 Delete
Holidays			
Note : Beside		oliday could be set as well. Those holidays would be treated as non-office time.	
	Name	Details	

Figure 3.5.2

3.5.3 Feature Codes

1) General

•One Touch Record

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

·Extension for Checking Voicemail

Users can 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

Users can go to the main menu by dialing *02 (*02 is default setting).

Attended Transfer

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

·Attended Transfer Timeout

The time out of transferring a call

·Blind Transfer



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

·Call Pickup

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

•Extension Pickup

Users may pick up 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)

General			
	🔽 One Touch Record	*1	
	Check Extension Voicemail	*2	
	Voicemail for Extension	#	
	🔽 Voicemail Main Menu 🛈	*02	
	Attended Transfer	*3	
	Attended Transfer Timeout	15	s
	Blind Transfer	*03	
	Call Pickup	*4	
	Extension Pickup	*04	
	✓ Intercom	*5	
	Vormal Spy	*90	
	✓ Whisper Spy ⁽¹⁾	*91	
	✓ Barge Spy <i></i>	*92	
Call Parking Preferences			
	Call Parking	*6	
	Extension range used to park calls	690-699	(Ex: 690-699)
	Number of seconds a call can be parked for $f 0$	60	
Call Forwarding Preferences			
	Reset to Defaults	*70	
	Enable Forward All Calls	*71	
	Disable Forward All Calls	*071	
	Enable Forward When Busy	*72	
	Disable Forward When Busy	*072	
	Enable Forward No Answer	*73	
	Disable Forward No Answer	*073	
	Forward to Number	*74	
	Forward to Voicemail	*074	
	Enable Do Not Disturb	*75	
	Disable Do Not Disturb	*075	

Figure 3.5.3

3.5.4 SIP Settings

1) General

·UDP Port

Port use for sip registrations, Default is 5060.

•TCP Port

Port use for sip registrations, Default is 5060.

•TLS Port

Port use for sip registrations, Default is 5061.

·RTP Port Start

Beginning of RTP port range

•RTP Port End End of RTP port range

·DTMF Mode

Set default mode for sending DTMF. Default setting: rfc2833

Max Registration/Subscription Time

Maximum duration (in seconds) of a SIP registration . Default is 3600 seconds.

Min Registration/Subscription Time

Minimum duration (in seconds) of a SIP registration . Default is 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, MyPBX 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 useragent parameter of asterisk, the default is 'MyPBX', 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. More details for the system security configuration, please refer to APPENDIX B MyPBX Security Configuration Guide.

Pedantic

Enable pedantic parameter. Default: no.

Session -timers

Enable session-timer mode. Default: yes.

Session-expires

The max refresh interval.

Session-minse

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

Session-refresher

Choose session-refersher, the default is Uas.



UDP Port♥: 5060 Enable TCP Port♥: 5060 Enable TLS Port♥: 5061 RTP Port Start: 10000 RTP Port End: 12000 DTMF Mode♥: frc2833 ▼ Max Registration/Subscription Time♥: 3600
Enable TCP Port 0: 5060 Enable TLS Port 0: 5061 RTP Port Start: 10000 RTP Port End: 12000 DTMF Mode 0: rfc2833 ~ Max Registration/Subscription Time 0: 3600 Min Registration/Subscription Time 0: 60
Enable TLS Port ①: 5061 RTP Port Start: 10000 RTP Port End: 12000 DTMF Mode ①: rfc2833 ▼ Max Registration/Subscription Time ①: 3600 Min Registration/Subscription Time ①: 60
RTP Port Start: 10000 RTP Port End: 12000 DTMF Mode 0: rfc2833 ▼ Max Registration/Subscription Time 0: 3600 Min Registration/Subscription Time 0: 60
RTP Port End: 12000 DTMF Mode rfc2833 • Max Registration/Subscription Time 3600 Min Registration/Subscription Time 600
DTMF Mode 0 : rfc2833 Max Registration/Subscription Time 0 : 3600 Min Registration/Subscription Time 0 : 60
Max Registration/Subscription Time 0 : 3600 Min Registration/Subscription Time 0 : 60
Min Registration/Subscription Time 0: 60
Default Incoming/Outgoing Registration Time 🕄 🛛 120
Register Attempts 0: 8
Register Timeout 0: 20
Calling Channel Codec Priority 0: Yes
Video Support 0 : Yes 🔹
Max Bit Rate 0 : 384 kb/s
DNS SRV Look Up 1 . No
User Agent 0 :
User Agent 4.
Note: Configuration of this section is only required when you use remote extensions.
Enable STUN:
STUN Address:
STUN Port:
External IP Address
External Host [©] :
External Refresh Interval [®] :
Local Network Identification 0:
NAT Mode 0: yes 🔹
Allow RTP Re-invite 0 : yes 🔹
Codecs
Available Codecs Allowed Codecs
G722 G726 GSM
ADPCU G729A, H263
MPEGA
88
G.729 License Key :
Note: If you would like to use G.729, please enter your license key above.
005 ♠
Tos SIP. C53 Cos SIP. 3
Tos Audio: EF Cos Audio: 5
Tos Video: AF41 Cos Video: 4
Advanced Settings 🛪
From Field: From
To Field: INVITE •
10000000000000000000000000000000000000
180 Ringing:
180 Ringing: □ Remote Party ID: □ Send □ trust
180 Ringing: □ Remote Party IDO: □ send □ trust Allow GuestO: No
180 Ringing: □ Remote Party ID: □ Send □ trust
180 Ringing: □ Remote Party IDO: □ send □ trust Allow GuestO: No
180 Ringing: Remote Party ID Allow Guest Pedantic No ▼
180 Ringing: Remote Party ID 0: Send T trust Allow Guest 0: No ▼ Pedantic 0: No ▼ Session-timers 0: Accept ▼

Figure 3.5.4



3.5.5 IAX Settings

1) General

·UDP Port

Port use for IAX2 registrations, Default is 4569.

Bandwidth

Low/medium/high with this option you can control which codec to be used.

·Min Registration/Subscription Time

Minimum duration (in seconds) of a IAX2 registration. Default is 60 seconds.

Max Registration/Subscription Time

Maximum duration (in seconds) of a IAX2 registration. Default is 1200 seconds.

2) 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.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.

Note: If you would like to use G.729, please enter your license.



General	
Codecs	
Allowed Codecs: Vu-law Va-law VGSM SPEEX	G726 CADPCM CG729A CH261 CH263 CH263P CH264

Figure 3.5.5

3.5.6 Voicemail Settings

- 1) General Voicemail Settings
- a) Message Options

·Max Messages per Folder

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

•Max Message Time

Set the maximum length of a single voicemail message.

•Min Message Time

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

•Ask Caller to Dial 5

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

·Delete Voicemail

If this option is set, the voicemail will be deleted from the server after notification.

·Operator Breakout from Voicemail

If this option is set, the caller can jump out of the voicemail and go to the destination (IVR) you set by dialing $0^{"}$.

b) Greeting Settings

·Busy Prompt

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



·Unavailable Prompt

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

·Leave a Message Prompt

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

c) Playback Options

·Announce Message Caller ID

If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

·Announce Message Duration

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

. Announce Message Arrival Time

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

. Allow Users to Review Messages

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

2) SMTP Settings for Voicemail

Note: If you want to send voicemail messages as email attachments, please configure this section.

·E-mail Address

The E-mail Address that MyPBX 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 MyPBX will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

•Port

SMTP Port: the default value is 25.



·Use SSL/TLS to send secure message to server

If the server of sending email needs to authenticate the sender, you need to select the check box.

Note: Must be selected for Gmail or exchange server.

After filling out the above information, you can click on the 'Test Account Settings' button to check whether the setup is OK.

1) If the test is successful, you can use the email safely.

2) If test failed, please check the above information is correct or network is proper.

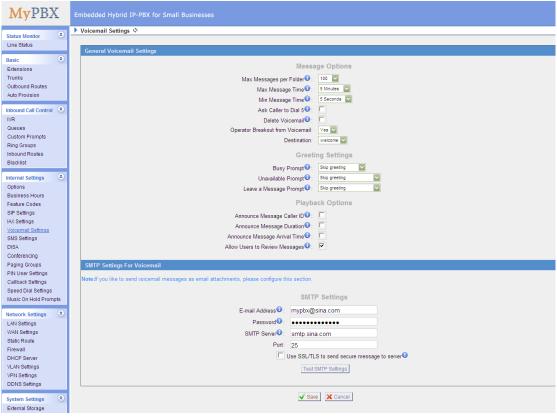


Figure 3.5.6

3.5.7 SMS Settings

MyPBX supports SMS to Email and Email to SMS. (Only for GSM trunks)

1) Enable SMS to Email

If you enable this, as soon as the GSM trunks receive small messages, MyPBX will send the text of this message to the email addresses listed on the Email List. You can add email addresses to the Email List.



lears@yeastar.com	0
Email Address:	† Add Email

Figure 3.5.7.1

2) Enable Email to SMS

If you enable this, you can use MyPBX to send out message by sending an email to the specified address.

•Enable Country Code

If you want to add country code before the dialed numbers, please tick this.

·Country Code

The country code to be add before the dialed numbers.

·Receive mails every

The intervening time of receiving mails from POP3 server.

•Access Code

This PIN code is used to verify the subject of the emails received. If the form of email passes the verification, it will be send out by SIM card. If not, this email will be deleted immediately.

3) Email Settings

Note:

1. If you want to use "SMS to Email", please configure POP3 setting.

2. If you configure the POP3 setting, MyPBX will download emails from the mail server regularly. Once downloaded, the emails will be deleted from the mail server.

·Email Address

This email address will be used to:

1. Send email to the addresses listed on "SMS to Email" setting.



2. Receive email and send the text of the email to the target mobile number by SMS.

Note: If you use gmail, just put your user name here. E.g. email address: test@gmail.com, you just put "test" here.

Password

Input the password of this email here.

- •SMTP Server (SMTP) •SMTP Server Port •Receive Server (POP3)
- **·Receive Server Port**

If you want to know more about Email to SMS, please refer to APPENDIX F

•	SMS Settings 🚸				
			SMS Se	ettings	
Þ	Enable SMS To Email				
	GSM Trunk Name	GSM Port		Email List	
	GSM9	9	chengeng930@163	.com chengeng530@gmail.com chenge	🔊 Edit 🔽 Delete
	GSM11	11		-	🕅 Edit 🛛 🖊 Delete
	GSM13	13		-	Y Edit K Delete
	Enable Email To SMS				
	Email To SMS				
			Enable Country Code:	Γ	
			Country Code:	China +86	
			Receive mails every	60	
			Access Code		
•	Email Settings				
	Email Settings				
	Note: 1. If you want to use "SMS	to Email" please co	nfigure POP3 setting		
				ail server regularly. Once downloaded, the	emails will be deleted from the mail
	server.				
			Email Address	chengeng530	
			Password 0:	•••••	
			SMTP Server (SMTP):	smtp.gmail.com	
			SMTP Server Port:	587	
			Receive Server (POP3):	pop.gmail.com	
			Receive Server Port:	995	
				Jse SSL/TLS for security on this server(SMT	(P) ⁽¹⁾
			Test SMTP Settin	gs Test POP3 Settings	
			🖌 Sav	e XCancel	

Figure 3.5.7.2

3.5.8 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 a PIN number, followed by the pound sign (#). If the PIN number is correct, the user will hear dial tone 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.

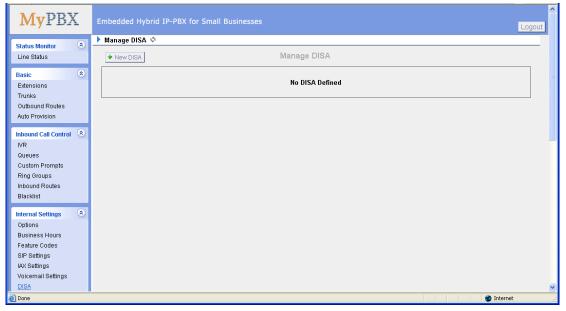
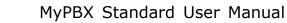


Figure 3.5.8.1

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

New DISA		x
General		
Na	me 🛈 :	
PI	N #🛈 :	
Response Time	out 🛈 : 10	
	out 🛈 : 5	
Member Outbound Routes		
Available Outbound Routes		Selected Outbound Routes
pstnout	>>	
	\rightarrow	
	←	
	««	
	🖌 Save 🔀 Cancel]

Figure 3.5.8.2





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 .

3.5.9 Conferencing

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. Conference agents can dial * to access to the settings options and the admin can kick the last user out and can lock the conference room.

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 #10:	
Save X Cancel	

Figure 3.5.9



3.5.10 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.

🗿 MyPBX - Embedded Hyb	rid IPPBX for Small Businesses - Microsoft Internet Explorer
<u>File Edit View Favorites</u>	Iools Help 🥂 👔
	🖹 🏠 🔎 Search 🤺 Favorites 🛷 🔗 ዿ 🖂
Address 🗃 http://192.168.5.18	6/cg/WebCG721000 🕑 🏹 🙆 🏙
MyPBX	Embedded Hybrid IP-PBX for Small Businesses
Status Monitor 🔹	Paging Groups 🗄
Line Status	Paging Groups
Basic Extensions Trunks Outbound Routes Auto Provision	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.
Inbound Call Control 🔹	+ New Paging Group
IVR Queues Custom Prompts Ring Groups	No Paging Groups Defined
Inbound Routes	
Blacklist	
Internal Settings 💲	
Options Business Hours	
Feature Codes	
SIP Settings	
IAX Settings	
Voicemail Settings	
DISA	New York Contraction of the Second Se
e	🔮 Internet

Figure 3.5.10.1

·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 Se	lected
500(SIP)	
501(SIP) 502(SIP)	
503(SIP) →	
504(SIP) 505(SIP)	
505(SIP) 506(SIP)	
507(SIP)	
🖌 Save 🔀 Cancel	

Figure 3.5.10.2

3.5.11 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

φ		DNIS		
	nk do not need to set DID number. , all calls through this trunk will sho Enable DNIS	ow the DNIS name as Caller Nam DNIS Name	e. DID Number	
pstn4(FXO)	off		-	🗸 Edit 🗴 Delete
pstn5(FXO)	off	-	-	🖌 Edit 🔰 Delete
pstn6(FXO)	off	-	-	🗸 Edit 🔰 Delete
	off			🖌 Edit 🔰 Delete
BriTrunk8(BRI)	οπ			- Lan

Figure 3.5.11

3.5.12 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.

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

MyPBX 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.

• 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. 'YeastarPIN'

·PIN List

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

•Outbound Route

PIN User can use those outbound route to make call out.

MyPBX	Embedded Hybrid IP-PBX for S	mall Businesses				Lo	gout _
Status Monitor 🛞	PIN User Settings I I I I I I I I I I I I I I I I I I I						
Line Status		PIN User Se	ttings				
Basic	+ New PIN User						
Extensions	Name	PIN List		10.1	1. I		
Trunks Outbound Routes	PINNumber	0592-0596-0599		🐕 Edit	🗴 Delete		
Auto Provision	Options						
		Access Code:					
Inbound Call Control 🙁		Prompt For Entry:	pinuser-entry				
Queues		Prompt For Entry Failure:	pinuser-error				
Custom Prompts		🗸 Save	🔀 Cancel				
Ring Groups Inbound Routes							
Blacklist							
Internal Settings 🔹							
Options							
Business Hours							
Feature Codes							
SIP Settings IAX Settings							
Voicemail Settings							
DISA							~
🙆 Done						🥑 Internet	

Figure 3.5.12



3.5.13 Callback Settings

MyPBX allows caller A to dial an inbound route number, and after hearing the ring, A can hang up the call or wait for MyPBX to cut off the call, then MyPBX will call A with this number. When A pick up the call, A can dial the number he wants to call; MyPBX will call the number with its outbound route.

Note:

1. If you'd like to use callback feature, please make sure if it's enabled on the inbound route setting panel.

2. No callback rules needed to be set if the trunk supports call back with the caller ID directly.

·Allow All Numbers

If you want to apply Callback function to all incoming numbers, please tick Allow All numbers.

Follow the step to use this function.

Step 1: Enable Callback.

Inbound Routes – Choose "Yes" on" Enable Callback" to enable this function. Refer to **<u>chapter 3.4.5.1</u>**

create New Inbou	ind Route	х
General		
	Route Name 🛈 :	
	DID Number 0 :	
	Extension 🛈 :	
	Caller ID Number 🛈 :	
	Distinctive Ringtone 🛈 :	
	Enable Callback : Yes Callback Settings	

Figure 3.5.13.1

Step 2: Create Callback number

New Callback	Х
Callback Number:	
✓ Save X Cancel	
Figure 3.5.13.2	

Step 3: Create Callback Rules

You will need to create callback rules when the system should strip or add digits.



•Trunk Name

Choose the trunk with callback rules

•Strip digits from front

Define how many digits will be stripped from the call in number before the callback is placed. For example, when you call from number 123456789 into MyPBX, the caller ID is 0123456789, but you can only call 123456789 successfully from MyPBX trunk. You should configure number 0123456789 as the call back number and strip 1 digit before the callback is placed

•Prepend before dialing

Define digits added before a callback number before the callback is placed. For example, the call in number (Caller ID) is 123456789, MyPBX need to send 9123456789 to its trunk when call to this number. You should configure 123456789 as the call back number and add 9 before the callback is placed. You can add 'w' for analog trunks for some delay too.

Create Callback Rules	х
Trunk Name : 1234(FX0)	
Strip 🛈 : digits from front	
Prepend 0: before dialing	
Save X Cancel	

Figure 3.5.13.3

3.5.14 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.



🗟 MyPBX - Embedded Hybrid IPPBX for Small Businesses - Microsoft Internet Explorer							
Ele Edit View Fgyvorkes Iools Help							
🚱 Back + 🕥 - 💌 😰 🏠 🔎 Search 👷 Favorites 🤣 🎓 😓 🚍							
Address 🕘 http://192.168.5.18	6/cgi/WebCGI?1000		✓ ➡ Go 链接				
My PBX	Embedded Hybrid IP-PBX for Small Bus	sinesses	Logout				
Status Monitor 🔹	Speed Dial Settings I have a setting the set of the set o						
Line Status	+ New Speed Dial	Speed Dial Settings					
Basic 🌲	Source Number	Destination Number					
Trunks	123	5503305	🗴 Delete				
Outbound Routes							
Auto Provision	Options						
Inbound Call Control 🙁		The prefix of speed dial 0: *99					
IVR		Save X Cancel					
Queues							
Custom Prompts							
Ring Groups							
Inbound Routes Blacklist							
Internal Settings 🔹							
Options Business Hours							
Eusiness Hours Feature Codes							
SIP Settings							
IAX Settings							
Voicemail Settings							
DISA			•				
🙆 Done			🔮 Internet 🛒				

Figure 3.5.14

3.5.15 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, 8.000kHz, Mono, 1kb/sec; Alaw/Ulaw 8.000kHz, Mono, 1kb/sec; PCM 8.000kHz, Mono, 1kb/sec. The file size must not be greater than 1.8MB.

Upload Music on Hold Prompt	х
	st not be larger than 1.8MB! alaw/ulaw 8kHz,Mono,1Kb/s 、pcm 8kHz,Mono,16Kb/s
Choose a file to upload 🛈 :	Browse
Uplo	ad X Cancel

Figure 3.5.15



3.6 Network Settings

3.6.1 LAN Settings

·DHCP

If this option is set, MyPBX 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 MyPBX.

•**IP Address** Set the IP Address for MyPBX.

•Subnet Mask Set the subnet mask for MyPBX.

•Gateway Set the gateway for MyPBX.

•**Primary DNS** Set the primary DNS for MyPBX.

•Secondary DNS Set the secondary DNS for MyPBX.

•**IP Address2** Set the second IP Address for MyPBX.

•Subnet Mask2 Set the second subnet mask for MyPBX.



🗿 MyPBX - Embedded Hyb	rid IPPBX for Small Businesses - Microsoft Internet Explorer
<u>F</u> ile <u>E</u> dit ⊻iew F <u>a</u> vorites	Iools Help
😋 Back 🔹 🐑 🐇 💌	🗟 🏠 🔎 Search 👷 Favorites 🔣 🔗 - 😓 🥽
Address 🎒 http://192.168.5.18	is/cgi/WebCGI71000
MyPBX	Embedded Hybrid IP-PBX for Small Businesses
Status Monitor 🔹	LAN Settings &
Line Status	
	LAN Settings
Basic 🙁	DHCP: No 🗸
Extensions	Enable SSH: Yes Y Port: 8022
Trunks Outbound Routes	
Auto Provision	Hostname: MyPBX
Autor romoton	IP Address: 192.168.5.196
Inbound Call Control 🛛 🙈	Subnet Mask : 255.255.0
IVR	Gateway: 192.168.5.1
Queues	Primary DNS: 192.168.5.1
Custom Prompts Ring Groups	Secondary DNS :
Inbound Routes	
Blacklist	IP Address2:
Internal Settings	Subnet Mask2:
Options Business Hours	✓ Save X Cancel
Feature Codes	
SIP Settings	
IAX Settings	
Voicemail Settings	
DISA	
🙆 Done	🍅 Internet

Figure 3.6.1

3.6.2 WAN Settings

Note: Only have WAN port's hardware version support this functions.

It support three connection types: DHCP (obtain an IP automatically), PPPoE, Static IP Address.

·DHCP

.If your ISP says that you are connecting through DHCP or a dynamic IP address from your ISP, perform these steps:

Step1: Select **DHCP** as the WAN Connection Type.

Step2: Click **Save** button to save the settings.

Step3: Reboot the device.

Step4: Check the WAN's Status (System Info \rightarrow WAN \rightarrow Status).

-Static IP Address

If your ISP says that you are connecting through a static or fixed IP address from your ISP, perform these steps:

- Step1: Select **Static IP Address** as the WAN Connection Type.
- Step2: Enter the IP Address.
- Step3: Enter the Subnet Mask.
- Step4: Enter the Gateway Address.
- Step5: Enter the Primary DNS and Secondary DNS.
- Step6: Click the **Save** button to save the settings.



Step7: Reboot the device.

Step8: Check the WAN's Status (System Info \rightarrow WAN \rightarrow Status).

·PPPoE

If your DSL provider says that you are connecting through PPPoE or if you normally enter a user name and password to access the Internet, perform these steps:

Step1: Select **PPPoE** as the WAN Connection Type.

Step2: Enter the User Name.

Step3: Enter the Password.

Step4: Click the **Save** button to save the settings.

Step5: Reboot the device.

Step6: Check the WAN's Status (System Info \rightarrow WAN \rightarrow Status).

Use WAN:
CDHCP
Static IP Address
IP Address:
Subnet Mask:
Default Gateway:
Primary DNS:
Secondary DNS:
C PPPoE
User Name:
Password:

Figure 3.6.2

3.6.3 Static Route

MyPBX 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 MyPBX to force it goes out through different gateway when access to different internet. The default gateway priority of MyPBX from high to low is OpenVPN \rightarrow WAN port \rightarrow LAN port.

1) Route table The current route rules of MyPBX

Destination

The destination network to be accessed to by MyPBX

·Subnet Mask

Specify the destination network portion.



Gateway

Which gateway MyPBX 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.

Static Route Settings Ø									
Routing 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					
Static Route Rules									
Destination : Subnet Mask: Gateway: Metric : Interface : LAN • + Add									
No Static Routes Defined									

Figure 3.6.3

3.6.4 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. More detailed configuration and other safety related configuration, please refer to <u>APPENDIX B MyPBX Security Configuration</u> <u>Guide</u>.



		II. .common rule with the 'action' is 'accept',		Firewall has starte	d successfully		
		No Common Rules Defi	ned				
Auto Defense Add Rule							
		No Auto Defense Rules De	fined				
IP Blacklist							
+ Add Rule				IE	P Blacklist Manage		
Port	Protocol	Rate					
5060	UDP	120/60s	🔊 Edit	🗶 Delete			
5060	UDP	40/2s	🔊 Edit	🗶 Delete			
8022	TCP	5/60s	🔊 Edit	🗴 Delete			
Other Settings Oisable Ping Orop All							

Figure 3.6.4

1) Enable Firewall

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

2) Common Rules

Name

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

Description

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

Protocol

The protocols for this rule .

•Port

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

·IP

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

•MAC Address



The format of MAC Address is XX:XX:XX:XX:XX:XX, X means $0 \sim 9$ 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	x
Name():	
Description 🛈 :	
	v
Protocol 🤨 :	TCP 💌
Port🤨:	0 : 0
IP🛈:	
MAC Address0:	
Action 🛈 :	Drop 💌
	Save Cancel

Figure 3.6.4.1

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 be refused directly.

New auto defense rule	X
Port ¹ : 0	
Protocol 🕖 : TCP 💌	
Rate 🛈: / Second 💌	
Save Save	

Figure 3.6.4.2



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 ¹ :	
Protocol 🛈 : UDP 🔽	
IP Packets	
Time Interval 0: seconds	
✓ Save X Cancel	

Figure 3.6.4.3

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	Protocol	Attacked Port	Source IP address	
1970-Jan-819:58	UDP	5060	203.117.31.243	🗶 Delete

Figure 3.6.4.4

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.

3.6.5 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 MyPBX 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 3.6.5

3.6.6 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 MyPBX VLAN over Lan.

·VLAN Subnet Mask



Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX 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 MyPBX VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Lan.

2) VLAN Over Wan

•NO.1

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

·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 MyPBX VLAN over Wan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

•NO.2

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

·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 MyPBX VLAN over Wan.



·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

Vian Over Lan	
NO.1:	
VLAN Number:	
VLAN IP Address:	
VLAN Subnet Mask:	
Default Gateway:	
NO.2:	
VLAN Number:	
VLAN IP Address:	
VLAN Subnet Mask:	
Default Gateway:	
Vlan Over Wan	
NO.1:	
VLAN Number:	
VLAN IP Address:	
VLAN Subnet Mask:	
Default Gateway:	
NO.2:	
VLAN Number:	
VLAN IP Address:	
VLAN Subnet Mask:	
Default Gateway:	

Figure 3.6.6

3.6.7 VPN Settings

A virtual private network (VPN) is a method of computer networking--typically using the public internet--that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. MyPBX supports OpenVPN.

•Enable VPN

·Import VPN Config

Import configuration file of OpenVPN. Don't configure 'user' and 'group' in the 'config' file.



VPN Settings III

VPN Settings			
	Enable VPN:		
	Import VPN Config	Browse	
	Import		
	Save X Cancel		

Figure 3.6.7

3.6.8 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

DDNS Settings 🖗
DDNS Settings
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 <u>dyndns org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com</u>
DDNS is not running
Enable DDNS:
DDNS Server: dyndns.org
User Name:
Password:
Host Name:
V Save X Cancel

Figure 3.6.8



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, MyPBX 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.

How to configure External Storage, please see APPENDIX B How to Configure Autobackup.

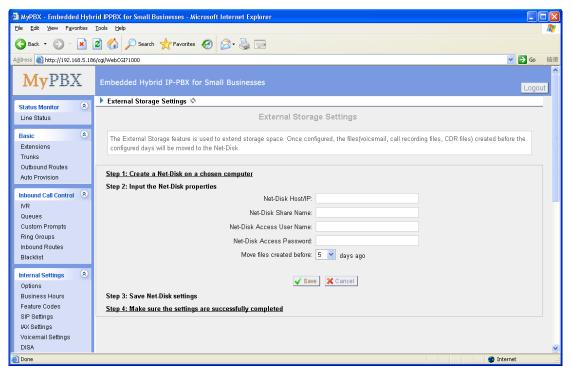


Figure 3.7.1

3.7.2 Password Settings

1) Change Password

•User

MyPBX support 3 levels users, choose one of them to change its password.

•Enter Old Password



The default password is '**password**'.

•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 设置

Whether enable CDR User.

Change Password 🖓
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

MyPBX have prompts of many languages. You can download the appropriate language you need. MyPBX can support American English, Australian English, Chinese, Dutch, French, Canadian French, German, Greek, Hungarian, Italian, Polish, Portuguese, Brazilian Portuguese, Russian, Spanish, Mexican Spanish, Turkish, Thai, Korean currently.

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.

My PBX	Embedded Hybrid IP-PBX for Small Businesses	
Status Monitor 🛞	▶ System Prompts Settings ♦	
Line Status	System Prompts Download	
Basic 🛞		
Extensions	Prompts Download	=
Trunks	Note:	
Outbound Routes	Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server,	
Phone Provisioning	please contact the local dealer for the prompts.	
Joining	Lord Provide Table	
Inbound Call Control 🔹	Local Prompts: English	
IVR	Download Mode: Auto Detection	
Queues	Download mode, Auto beledabit	
Custom Prompts		
Ring Groups	Prompts: Engish 👻	
Inbound Routes		
Blacklist	S Download	
Internal Settings		
Options		
Business Hours		
Feature Codes		
SIP Settings		
IAX Settings		
Voicemail Settings		
SMS Settings DISA		
DISA Conferences		
		-
Paning Groups		

Figure 3.7.3



3.7.4 Date and Time

Set the date and time for MyPBX.

•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 MyPBX will update the time automatically

•Set Date & Time Manually

•	▶ Date & Time ♥		
	Date & Time		
	Server Time: Sun Jun 10 4:43:11 2012		
	Time Zone: 0 United Kingdom (London)		
	Daylight Saving Time: Automatic		
	Automatically Synchronize With An Internet Time Server		
	NTP Server: pool.ntp.org		
	C Set Date & Time Manually		
	Date		
	Time Time AM *		

Figure 3.7.4



3.7.5 Backup and Restore

🗿 MyPBX - Embedded Hybr	id IPPBX for Small Businesses - Microsoft Internet Explorer
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites	Tools Helb
🚱 Back 👻 🐑 💌 📘	🖸 🏠 🔎 Search 🧙 Favorites 🚱 🔗 - 🌺 🚍
Address 🕘 http://192.168.5.186	(cg)/WebCGI71000
MyPBX	Embedded Hybrid IP-PBX for Small Businesses
Status Monitor 🛞	Backup/Restore Configuration Information
Line Status	Manage Configuration Backups
Basic 🔹	Create a New Backup Upload a Backup
Trunks	List Of Previous Configuration Backups:
Outbound Routes	Name Date Options backup 2011mar28 161735 2011mar28 tar Mon Mar 28 16:26:43 2011 Download from System Restore Previous Configuration from Backup
Auto Provision	1 backup_2011mar28_161735_2011mar28.tar Mon Mar 28 16:26:43 2011 Download from System Restore Previous Configuration from Backup
Inbound Call Control (*) IVR Queues Custom Prompts Ring Groups Inbound Routes Blacklist	
Internal Settings (*) Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings DISA	~
🙆 Done	Internet

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.



🗿 MyPBX - Embedded Hybrid IPPBX for Small Businesses - Microsoft Internet Explorer					
Elle Edit View Favorites Iools Help 🧗					
🚱 Back 👻 🕑 🕤 💌	🔇 Back 👻 😰 😪 💋 Search 👷 Favorites 🔣 🔗 🖓 层				
Address 🕘 http://192.168.5.18	\$/cgi/WebCGI?1000				
MyPBX	Embedded Hybrid IP-PBX for Small Businesses				
Status Monitor 🔹	Reset and Reboot Options III				
Line Status	Reboot System				
Basic 🔹	Reboot System				
Trunks Outbound Routes Auto Provision	Warning:Rebooting the system will terminate all active calls!				
Inbound Call Control 🔹	Reboot				
Queues	Reset to Factory Defaults				
Custom Prompts					
Ring Groups	Reset to Factory Defaults				
Inbound Routes					
Blacklist	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				
Internal Settings 🔹	 rease to not in system on the system on the rook right begins on king. Any power menophon during this time could cause damage to the system. 				
Options					
Business Hours	Reset to Factory Defaults				
Feature Codes					
SIP Settings IAX Settings					
Voicemail Settings					
DISA	×				
🕘 Done	🔮 Internet				

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.

More Information, please see http://www.yeastar.com/download/MyPBX/

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.



A MyDRY - Embedded Hyb	rid IPPBX for Small Businesses - Microsoft Internet Explorer	
Ele Edit View Favorites		
		~
	🗟 🏠 🔎 Search 🤺 Favorites 🛛 🔗 - 🌺 🚍	
Address 🕘 http://192.168.5.18	6/cgi/WebCGI?1000	✓ Go 链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	
Status Monitor 🔹	Vpdate System Firmware φ	
Line Status		
Basic 🔹	Firmware Download Source:	
Extensions		
Trunks	HTTP URL TFTP Server	
Outbound Routes	HTTP URL:	
Auto Provision	Reset Configuration to Factory	
Inbound Call Control	Defaults:	
IVR	+ Start	
Queues	• otari	
Custom Prompts		
Ring Groups		
Inbound Routes		
Blacklist		
Internal Settings 🔹 🔕		
Options		
Business Hours		
Feature Codes		
SIP Settings IAX Settings		
Voicemail Settings		
DISA		~
🙆 Done		😮 Internet 🛛 🖓

Figure 3.7.7

3.7.8 Alert Settings

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

More details for the system security configuration, please refer to <u>APPENDIX B</u> <u>MyPBX Security Configuration Guide</u>.

▶ 警报设置 ◇			
攻击类型	电子邮件通知	电话通知	
IPATTACK	yes	yes	≫ 编辑
WEBLOGIN	yes	yes	12 编辑

Figure 3.7.8.1

1.IPATTACK

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

1) Phone Notification Settings

•PHONE Notification

Whether enable phone notification.

·Number



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

Attmtpts

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

Interval

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

Prompt

Users will hear the prompt while receiving the phone notification.

2) E-mail Notification Settings

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

·E-mail Notification

Whether enable E-mail Notification

·Recipient' Name

The recipients for the alert notification, and multiple email addresses are allowed, please separate them by `;'.

e.g. ;

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)

IPAT	ТАСК					Х
	Phone Notificatio	n Settings hone Notification: Number(): Attempts(): Interval(): Prompt:	Yes ▼ 500;81234 3 ▼ 10 s default ▼	Custom Prom	<u>pts</u>	
	-	n Settings mail Notification: pient's Name()): Subject:	Yes ▼ sandy@yeasta	ir.com;1708		
	0	pbx hostname:\$ attack source ip attack dest mac attack source po attack source po attack source po attack occurred	address:\$(SOU ::\$(DESTMAC) ort:\$(DESTPOF rotocol:\$(PROT	RT)	*	

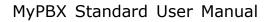
Figure 3.8.2

2.WEBLOGIN

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

WEBLOGIN	x					
Phone Notification Settings Phone Notification: Number(): Attempts(): Interval(): Prompt:	Yes ▼ 500;913003923337 3 ▼ 10 s default ▼ <u>Custom Prompts</u>					
E-mail Notification Settings E-mail Notification:	Yes 🔻					
Recipient's Name() : Subject:	sandy@yeastar.com;1708; hoho					
pbx hostname:\$(HOSTNAME) login ip address:\$(SOURCEIP) login username:\$(USERNAME) login occurred:\$(DATETIME)						







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.

Search Cond	lition									
Start Date: 1	9 Nov 2	.012 E	End Date: 19	Nov 2012 C	aller/Callee:			Trunk: All	•	
Ouration 🛈 :		Billing Du	iration 0:		Status:	All	Communicat	tion Type: All	¥	Start Searching
🕏 Download ti	he record	s 🗴 Delete th	e records					Show:1-5 Total: (5	View: 25 🔻
Time	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Type	Pin User	
2012-11-19 01:30:05	500	913003923337		GSM13	13	0	NO ANSWER	Outbound	123	🗴 Delete
2012-11-19 01:29:57	500	*89			8	8	ANSWERED	Internal	123	🗴 Delete
2012-11-19 01:28:28	500	913003923337		GSM13	13	0	NO ANSWER	Outbound		🗴 Delete
2012-11-19 01:28:21	500	501			15	15	ANSWERED	Internal		🗴 Delete
2012-11-19 01:28:17	500	503			1	0	NO ANSWER	Internal		🗴 Delete

Figure 3.8.1

3.8.2 System Logs

You can download and delete the system logs of MyPBX.

1) Options

•Enable Hardware Log

Save the infomation of hardware; (up to 4 log files)

·Enable Normal Log

Save the prompt information; (up to 16 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)

2) Packet Capture Tool

This feature always used to capture packets for technician. Integrate packet



capture tool 'Wireshark' in MyPBX.

Users also could specify the destination IP address and port to get the packets.

·IP

Specify the destination IP address to get the packets.

•Port

Specify the destination Port to get the packets.

	▶ System Logs ♦								
				S	ystem Logs Operations				
		he Selected Logs	Y Doloto The Se	locted Loca					
	Download The Selected Logs Delete The Selected Logs								
		Name		Туре					
		web.log		Web		The second secon			
	Options								
	Enable Hardware Log			[Enable Normal Log	Enable Debug Log			
		Enable Web	o Log🛈						
	Packet Capture Tool								
	Packet Capture Stopped								
					IP:				
Port:									
<u></u>					Start Stop 😽 Download				

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 .



General 余 Hardware Version: MyPBX-Standard V3

Firmware Version: 2.17.0.21-beta02

Uptime: 18:47:13 up 15:04, load average: 1.00, 1.00,

18:47:13 up 15:04, load average: 1.00, 1.00, 1.00	
LAN A	
Hostname: MyPBX	
MAC Address: 00:15:65:55:0e:12	
IP Address: 192.168.4.141	
Subnet Mask: 255.254.0	
Gateway: 192.168.5.1	
Primary DNS: 192.168.5.1	
Secondary DNS:	
Disk Usage 🛠	
Note: If there is not enough disk space on the system, the oldest voicemail messages, call record files and call log files will be automatically deleted as n Disk Usage:	ecessary.
Filesystem 1k-blocks Used Available Use% Mounted on /dev/mtdblock0 33127 2954 3173 90% / /dev/mtdblock2 491520 6428 485092 1% /persistent	
Memory Usage 🛠	
Memory Usage:	
total used free shared buffers Mem: 87972 33356 54616 0 140	





4. Access MRI

MRI (MyPBX 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.

Edit Extension - 501	х
General Type: SP Extension 501 Password pincode501 Name : 501 Caller ID : 501	
Voicemail Finable Voicemail Voicemail Access PIN # 501	
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	
Follow me: Image: No answer Transfer to: Image: Number Image: When Busy Other Options	
Call Waiting ● □ DND ● □ User Web Interface ● Ring Out ●: 30 Optional Settings ≫	
Save X Cancel	

Figure 4-1

4.2 User login

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



network administrator.

🗿 http://192.168.5.186/	- Microsoft Internet Explorer
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites	Iools Heb
🕒 Back 🝷 🕥 🕤 💌	🔰 🏠 🔎 Search 👷 Favorites 🤣 🔝 💿
Address 💩 http://192.168.5.18	6/
MyPBX	Embedded Hybrid IP-PBX for Small Businesses
	MyPBX Configuration Panel
Yeastar	User Login
	User Name: 501
	Password: •••
	Language: English
	Language, jargan 💼
	Login Reset
	Copyright @ 2011 Yeastar Technology, Co., Ltd. All Rights Reserved.
Done Done	internet

Figure 4-2

4.3 Voicemail

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

MyPBX	Embedded Hybrid I	P-PBX for Small Businesses				Logout
	Voicemail Ø					
Voicemail Call Recordings	Folder: New Viewing 1-1	Solder:	New			
Voicemail Settings Settings		Caller ID	Date	Duration	Options	
Customer Feedback						

Copyright $\ensuremath{\mathbb{C}}$ 2011 Yeastar Technology Co., Ltd. All Rights Reserved.

Figure 4-3





4.4 Record

Users can play, delete and download recorded files here.

MyPBX	Embedded Hybrid IP	P-PBX for Small Businesses			Logout
Voicemail Call Recordings	Call Recordings				
Voicemail Settings Settings		Caller ID	Time	Duration	Options

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 Faxex will be sent to email as an attachment.

My PBX	Embedded Hybrid IP-PBX for Small Businesses	ogout
	► Voicemail Settings Φ	
Voicemail	Voicemail Settings	
Call Recordings Voicemail Settings		
Settings	Voicemail Password:	
	Enter again to confirm:	
	Email Address 0 :	
	Enable Voicemail 🛈 : 🔽	
	Enable Send Voicemail®:	
	✓ Save X Cancel	
88		
Customer Feedback		
	Copyright © 2011 Yeastar Technology Co., Ltd. All Rights Reserved.	

Figure 4-5



4.6 Settings

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

bedded Hybrid IP-PBX for Small Businesses	out
iettings Φ	
Settings © Settings Follow Me: Voicemail Voicemail Voicemail Voicemail Voicemail Call Waiting Voicemail Call Waiting Voicemail Call Waiting On DND Voicemail Number Call Waiting Call Waiting Call Voicemail Voicemail Number Save Cancel	_
ie	ttings © ettings ettings Follow Me: O No Answer Transfer To: O Voicemail O Number Call Waiting Out 30 Mobile Extension Number :

Copyright $\ensuremath{\mathbb{C}}$ 2011 Yeastar Technology Co., Ltd. All Rights Reserved.

Figure 4-6



5. Access CDR

CDR (MyPBX Recording Interface). Users may access CDR to research call logs. Username: cdr

Password: password

5.1 Call Logs

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



Figure 5-1

5.2 Password Settings

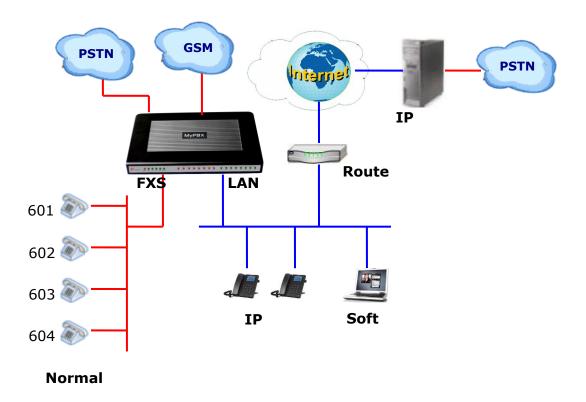
Change the password for cdr user.

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

Figure 5-2



6. Use MyPBX



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 MyPBX, connected to a local PSTN.

•GSM Trunk: GSM ports of MyPBX, connected to GSM Network.

•BRI Trunk: BRI ports of MyPBX, 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.

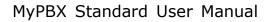
Edit Outbound Route			х
Route Name 🛈 :	pstnout		
Dial Pattern 🛈 :	9.		
Strip 🛈 :	1 digits from	front	
Prepend these digits		before dial	ing
Password:			
Member Extensions			
Available Extensions			Selected
		500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP)	
Member Trunks			
Available Trunks			Selected
8032(SIP) 192.168.5.190(SPS) pstn6(FXO) GSM11(GSM) BriTrunk8(BRI) BriTrunk7(BRI)		ostn1(FXO) ostn2(FXO)	
5	Save X Cance	el	

Figure 6-1

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

Before we do add this, please make sure you have a VoIP Trunk account. Trunks \rightarrow VoIP Trunk \rightarrow SIP Trunk

Enter your account information on this page, and click Save.

Created New VoIP trunk		X
Туре:	SIP 💌	
Provider Name:	voipprovider	
Hostname/IP:	voip.6699.org	: 5060
Domain:	voip.6699.org	
User Name:	16885885	
Authorization Name:	16885885	
Password:	•••••	
From User:	16885885	
Online Number 🛈 :		
Maximum Channels 🛈 :	1	
Γ	Enable Outbound Proxy Server	
Transport:	UDP 🗹 Enable SRTP 🛈 : 🗖	
Caller ID 🛈 :		
DOD Setting		
DOD:	Associated Extension: 500 💌	↑Add DOD
	Save X Cancel	

Figure 6-2

2. Add Outbound Routes

As we can see from the Outbound Route of 'voipout', all phone numbers



starting with 0 will have their first digit stripped off (digit 0) and will be sent to the SIP Trunk.

New Outbound Route			х
Route Name🛈:	vopiout		
Dial Pattern 🛈 :	0.		
Strip 🛈 :	1 digits from	front	
Prepend these digits		before dialir	ng
Password:			
Member Extensions			
Available Extensions			Selected
	→ 65	500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP)	
Member Trunks Available Trunks			Selected
pstn1(FXO) pstn2(FXO) pstn6(FXO) GSM11(GSM) BrTrunk7(BRI) BrTrunk8(BRI) 192.168.5.190(SPS)		voipprovider(SI 3032(SIP)	P)
5	Save X Cance	el	

Figure 6-3

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 500, and digit 1 to route the



call to extension 501.

1. Add IVR

To add a new IVR, go to IVR \rightarrow Create New IVR

Edit IVR welcome					×
Number 🛈 : 660					
Name 🛈 : 🛛 welcome					
Prompt 🛈 : 🛛 default 💌	Custom IVR Prompts				
Play times 🛈 : 💽					
WaitExten 🛈 : 💽					
🔽 🛈 Allow Dialir	ng Other Extensions				
🚺 KeyPress Events —					
Кеу	Action		Destination		
0	Connect to Extension	•	User Extension 500	•	
1	Connect to Extension	•	User Extension 501	•	
2	No Action	•		~	
з	No Action	•		Y	
4	No Action	•		T	
5	No Action	•	[T	
6	No Action	•	[~	
7	No Action	•		~	
8	No Action	•		V	
9	No Action	•		V	
#	No Action	•		V	
*	No Action	•		V	
TimeOut 🛈	Connect to Extension	•	User Extension 500	•	
Invalid 🛈	Connect to Extension	•	User Extension 500	•	
	🗸 Save	🗙 Cancel			

Figure 6-4

2. Add Inbound Routes

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



General	A III		
	me0 : allin		
DID Numb			
Extensi			
Caller ID Numb	er 🛈 :		
Distinctive Rington	ne 🛈 :		
Nember Trunks			
Available Trunks		Selected	
	»» [pstn1(i	FXO)	~
	pstn2()		
	GSM11	(GSM) Ik7(BRI)	
		k8(BRI)	_
		ovider(SIP)	~
Ouring Office Hours			
	C End Call		
	C Extension	Extension 500	~
	O Voicemail	Voicemail 500	~
Destination:	• IVR	IVR welcome	*
	RingGroup	RingGroup ringgroup_de	fi 💙
	C Conference Room	Conference Room 640	*
	C DISA	DISA	*
	C Queues	Queues	*
	C Faxes	Faxes 500	~
	C Outbound Routes	Route Name pstnout	~
Outside Office Hours			
	C End Call		
	C Extension	Extension 500	~
	C Voicemail	Voicemail 500	~
Destination:	IVR	IVR welcome	~
	C RingGroup	RingGroup ringgroup_de	fi 💙
	C Conference Room	Conference Room 640	~
	C DISA	DISA	~
	C Queues	Queues	~
	C Faxes	Faxes 500	~
	Outbound Routes		~
		- P	

Figure 6-5



APPENDIX A FAQ

Q1. How to Register SIP device?

A1:

1) Register SIP soft phone Download the x-lite softphone from counterpath website www.counterpath.com After install the x-lite, right click the panel and select the SIP Account setting and then configure it. Display Name: 500 User Name: 500 Password: 500 Authorization Name: 500 Domain: 192.168.5.150 2) Register IP Phone (for example, Yealink's T28 IP Phone) a) Connect the T28's Internet port to the switch. And it can get the IP from your route. b) Press the 'OK' key on T28 to get the IP of T28. c) Put the IP on web browser then you can enter the T28 configure page through this IP. d) Put the SIP extensions info on the T28 IP phones. **Display Name: 501** User Name: 501 Register Name: 501 Password: 501 SIP Server: 192.168.5.150

Use the same method register another T28 to other extension.

Q2. How do I reset MyPBX back to the factory default settings?

A2: To perform a reset, please follow steps below:

Step 1: Hold down the 'Reset' button on the back of the unit for 5 seconds and watch the LEDs on the front of the MyPBX. When the status LED turns red, let go of the reset button.

Step 2: When the RUN status LED starts blinking, MyPBX will be set back to factory defaults.

Step 3: To access the configuration page, navigate to 192.168.5.150 using a web browser. Make sure that you are on the 192.168.5.0 subnet before



doing this.

Step 4: Login to the device with the username 'admin' and the password 'password', in order to begin reconfiguring the device.

APPENDIX B MyPBX Security Configuration Guide

VoIP attacks, although it is not an everyday occurrence, it does exist. While using VoIP, system security is undoubtedly one of the issues we care about most. But with the appropriate configuration, and some basic safety habits, we can improve the security of the telephone system. Moreover, the powerful built-in firewall function in MyPBX is adequate to enable the system to run safely and stably.

This guide will introduce the highest defense level in MyPBX, and we strongly recommend that you configure firewall and other security options according to this guide, to prevent the attack fraud and the system failure or calls loss.

Note: In this guide, the configuration options marked with `*' only exist in 2.17.XX.XX and above versions, namely, 3.2 guest calls option, 3.3 remote registered option, and 5 alarm settings.

1. Security Configuration for Web GUI

1.1 Change the default access port for HTTP on Options page

Select Internal Settings→Options→General Preferences→HTTP Bind Port



General Preferences	
Ring Timeout 🛈 :	30 s
MAX Call Duration 🛈 :	6000 s
Maximum Concurrent Calls	0
Music On Hold:	calmriver 🔻
Tone Region 🛈 :	Czech Republic 🗸
HTTP Bind Port	80
FXO Mode	CZECH 🗸
Virtual Ring Back Tone 🛈 :	No 🔻
Distinctive Caller ID	No 💌
Follow Me Prompt	Yes •
Music on hold for Follow Me 🛈 :	Default 🔻
Invalid Phone Number Prompt 🛈 :	•
Busy Line Prompt 🛈 :	•
Dial Failure Prompt	•

Figure 1-1

1.2 Change the default password for the web GUI

Select System Settings→Change Password

•	Change Password 🔅		
	Change Password		
	User:	admin 🔻	
	Enter Old Password:		
	Enter New Password:		
	Retype New Password:		

Figure1-2

2. Disable SSH on LAN Settings Page

2.1 Disable SSH

Select LAN Settings \rightarrow Enable SSH. If external debugging isn't required, please select "No".



LAN Settings	
DHCP	No 🔻
Enable SSH	No ▼ Port: 8022
Hostname	MyPBX
IP Address	192.168.4.151
Subnet Mask	255.255.254.0
Gateway	192.168.5.1
Primary DNS	192.168.5.1
Secondary DNS	:
IP Address2	:
Subnet Mask2	:

Figure 2-1

2.2 Change the default password for SSH

We can use the Linux command passwd to change root password of MyPBX.

1. Login via putty.exe

😵 PuTTY Configurat	ion 🔀
Category:	
 Session Logging Terminal Keyboard Bell Features Window Appearance Behaviour Translation Selection Colours Connection Data Proxy Telnet Rlogin SSH Serial 	Basic options for your PuTTY session Specify the destination you want to connect to Host Name (or IP address) Port 192.168.5.139 8022 Connection type: Baw Paw Telnet Rogin SSH Save or delete a stored session Saved Sessions Default Settings Load Default Settings Delete Close window on exit: Only on clean exit
<u>A</u> bout	<u>Open</u> <u>C</u> ancel

Figure 2-2

2. The default username is root and the default password is ys123456.



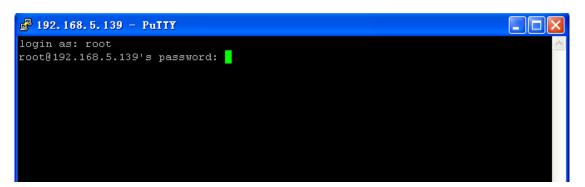
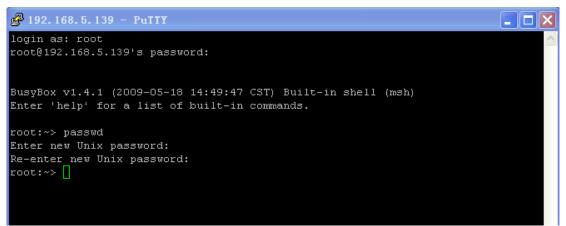


Figure 2-3

3. Step 2 use command passwd to change the root's password





3. Security Configuration for Extensions

3.1 Change the default SIP Port

Select SIP settings→General→UDP Port

General		
	UDP Port 0:	5060
Enable	TCP Port 0:	5060
Enable	TLS Port 0:	5061
	RTP Port Start:	10001
	RTP Port End:	10200
	DTMF Mode 🛈 :	rfc2833 🔹
Max Registration/Subs	scription Time 🛈 :	3600
Min Registration/Subs	scription Time 🛈 :	60

Figure 3-1

3.2* Disable guest calls

Select SIP settings→Advanced Settings→Allow Guest



Advanced Settings 🛠	
From Field	From •
To Field	INVITE -
180 Ringing	:
Remote Party ID	send trust
Allow Guest	No 👻
Pedantic	No 🔻
Session-timers 🛈	Accept -
Session-expires	: 1800 s
Session-minse	: 90 s
Session-refreshe	: Uas 🔻

Figure 3-2

3.3* Security Configuration for remote extensions

If remote registration isn't required, please disable it.

Optional Settings	*	
VoIP Settings		
NAT 🛈 : 🗖	Qualify: 🔽	Enable SRTP 🛈 : 🗖
Transport: UDP 🔻	DTMF Mode : RFC2833 -	Register Remotely 🛈 : 🗖

Figure 3-3

3.4 Set an enhanced password and enable IP restriction for

extensions

1) Set a new extension password at the higher security level, e.g. AjK5Up1G.

Edit Extension - 500			X
General			7
Type: SIP 💟	Extension 🛈: 500	Password 🛈: AjK5Up1G	
Name 0: 500	Caller ID: 500		



2) Enable IP restriction and enter the permitted "IP address/Subnet mask", e.g. 192.168.5.136.



VoIP Settings		
NAT 🛈 : 🗖	Qualify: 🔽	Enable SRTP 🛈 : 🗖
Transport: UDP 🔻	DTMF Mode 🛈 : RFC2833 🔻	
IP Restriction		
Enable IP Restric	tion 🛈 :	
Permitted 'IP address	/Subnet mask' 1 🛈: 192.168.5.	136/255.255.255.255
Permitted 'IP address	/Subnet mask' 2 🛈:	
Permitted 'IP address	/Subnet mask' 3 🛈:	
	/Subnet mask' 4 🛈:	

Figure 3-5

4. Set up Proper Firewall Rules

Note: Please backup the configurations on backup and restore page before you go ahead. In the case that you lock the device, you can reset to factory default and restore the previous configurations. Below example rules works with MyPBX firmware version 2.15.xx.xx or higher versions.

Step 1. Enable firewall on firewall page of MyPBX.

Step 2. Add a common rule to accept local network access.

Create a common rule to allow the all the IP addresses of the local phones to access MyPBX server. For example, if the IP addresses of the local network are 192.168.5.1-254, the configurations could be as below:

Name: LocalNetwork Protocol: BOTH Port: 1:65535 IP: 192.168.5.0/255.255.255.0 Action: Accept



Edit Firewall Rule	х
Name : LocalNetwork	
Description : Accept_any_local_hosts	*
	Ŧ
Protocol 🛈 : BOTH 👻	
Port ¹ : 1 : 65535	
IP1: 192.168.5.0 / 2	55.255.255.0
MAC Address 🛈 :	
Action 1: Accept -	
Save X Cancel	

Figure 4-1

Step 3. Create common rules to accept remote extensions or remote administrators, if you use SIP trunk, please accept the provider's host as well.

Note: If there are no remote extensions, the rule is not required.

1) Set up the firewall rule to allow the public IP address of remote extensions to access MyPBX server. e.g.110.111.132.6, the configurations could be as below:

Name: Remote Extension Protocol: BOTH Port: 1:65535 IP: 110.111.132.6/255.255.255 Action: Accept

Add Firewall Rule		х
Name 🛈 : F	RemoteExtension	
Description 🛈 :		
Protocol 🛈 : 🗄	вотн 💌	
Port0: 1	1 : 65535	
IPÛ: 1	110.111.132.6 / 255.255.255.255	
MAC Address0:		
Action 🛈 : 🗸	Accept 💌	
	Save Cancel	

Figure 4-2



Step 4. Configure auto blacklist rules

Auto blacklist rules: the Server would add the IP address to the blacklist automatically if the number of the packets it sends exceed the rule you configured.

1) Add two auto blacklist rules for port: 5060.

Rule No.1:

Port: 5060 Protocol: UDP IP Packets: 120 Time Interval: 60 seconds

Add Auto Blacklist Rules	х
Port ¹ : 5060	
Protocol 🛈 : UDP 💌	
IP Packets 🛈 : 120	
Time Interval 🛈 : 60 seconds	
Save X Cancel	

Figure 4-3

Rule No.2:		
Port : 5060		
Protocol: UDP		
IP Packets: 40		
Time Interval: 2 seconds		
Add Auto Blacklist Rules		х
Port ¹ :	5060	
Protocol 🛈 :	UDP 💌	
IP Packets	40	
Time Interval	2 seconds	
	Save Cancel	

Figure 4-4

2) Add an auto blacklist rule for Port:8022
Rule No.3
Port: 8022
Protocol: TCP
IP Packets: 5
Time Interval: 60 seconds



Add Auto Blacklist Rules	Х
Port 1: 8022	
Protocol 🛈 : TCP 💌	
IP Packets 0: 5	
Time Interval : 60 seconds	
Save X Cancel	

Figure 4-5

Step 5. Add a Firewall Rule for VoIP trunk registration

Note: If there is no VoIP trunk, this rule is not required. And if the RTP IP address of VoIP trunk and Registration IP address of the VoIP trunk are different, we need create a rule to accept the RTP IP address too.

Add a rule to accept the IP address of the VoIP trunk to access MyPBX server. For example: If the IP address of the VoIP trunk is 110.5.14.6, Protocol is UDP and Port is 5060, the configuration could be as below:

Name: VoIPTrunk Protocol: UDP Port: 5060: 5060 IP: 110.5.14.6/255.255.255 Action: Accept

Add Firewall Rule			Х
Name 🛈 :	VolPTrunk		
Description 🛈 :		~	
		<u>₩</u>	
Protocol ¹ :	UDP 💙		
Port ¹ :	5060 : 5060		
IPÛ:	110.5.14.6	/ 255.255.255.255	
MAC Address 🛈 :			
Action ⁽¹⁾ :	Accept 🗸		
	🗸 Save 🔀 Cancel		

Figure 4-6



Step 6. Add a firewall rule to accept the remote access of HTTP port. For example, if the remote access IP is 110.5.14.6, and the port is 80, the configuration could be as below.

Name: RemoteHTTP Protocol: TCP Port: 80:80 IP: 110.5.14.6/255.255.255 Action: Accept

Add Firewall Rule			х
Name 🛈 :	RemoteHTTP		
Description 🛈 :			
		<u></u>	
Protocol ¹ :	TCP 🔽		
Port ¹ :	80 : 80		
IPÛ:	110.5.14.6	/ 255.255.255.255	
MAC Address 🛈 :			
Action 🛈 :	Accept 🐱		
	Save X Cancel		

Figure 4-7

Step 7. Add a firewall rule to accept remote access of SSH port. For example: if the remote access IP is 110.5.14.6 and the port is 8022.

Note: If the remote access of SSH port is not needed, this rule is not required.

Name: RemoteSSH Protocol: TCP Port: 8022:8022 IP: 110.5.14.6/255.255.255 Action: Accept

Yeastar	MyPBX Standard User Manual
Add Firewall Rule	X
Name 🛈 :	RemoteSSH
Description 🛈 :	
Protocol ⁽⁾ :	TCP
Port ⁰ :	8022 : 8022
IP Û :	110.5.14.6 / 255.255.255.255
MAC Address	
Action 0:	Accept 🗸
	Save X Cancel

Figure 4-8

Step 8. Add other firewall rules by yourself. For example, if you are using features about email, you should add the firewall rules for the SMTP server and POP3 server.

Step 9. Enable Drop all (If this feature is enabled, all the packets and connection that do not match the rules would be dropped.)

Note: Before enable this feature, please add a rule to accept the local network access, or the server might not be accessed.

Other Settings		
🗆 🛈 Disable P	ing	
Drop All		

Figure 4-9

Step 10. The Configuration of firewall settings is completed. See as below figure.



				Firewall Settings			
	boot the syste	m after enabling or disa		mmon rule with the 'action' is 'accept', or i	it may be dragged int	o the blacklist	
Common Rule					e may be aragged me		
+ Add Rule							
	Action	Name	Protocol	IP	MAC Address	Port	
⇒ ±	ACCEPT	LocalNetwork	BOTH	192.168.5.0/255.255.255.0		1:65535	🔊 Edit 🗴 Delete
₹\$\$¥	ACCEPT	RemoteExtension	BOTH	110.111.132.6/255.255.255.255		1:65535	🔊 Edit 🗴 Delete
₹\$\$\$±	ACCEPT	VolPTrunk	UDP	110.5.14.6/255.255.255.255		5060:5060	🔊 Edit 🗴 Delete
₹\$\$\$±	ACCEPT	RemoteHTTP	TCP	110.5.14.6/255.255.255.255		80:80	🔊 Edit 🗴 Delete
t 7	ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	-	8022:8022	😼 Edit 🗴 Delete
Auto Defense Add Rule							
				No Auto Defense Rules Defined			
IP Blacklist							
+ Add Rule							
							IP Blacklist Manage
Port		Protocol		Rate			
5060		UDP		120/60s	No Ed		
5060		UDP		40/2s	No Ed		
8022		TCP		5/60s	🔊 Ed	it 🗴 Delete	
Other Settings	Ping						



Step 11. Define the authorized data transmitting rate and drop all other unauthorized hosts.

Note: If we setup IP blacklist rules here, those unauthorized hosts will still be able to send SIP packets into MyPBX if the packets transmitting rate is under the rate defined. But as soon as the transmitting beyond the define rate, the IP will be added into blacklist automatically. This is useful for remote extension who is using dynamic IP address.

Remove all the IP blacklist rules if you don't allowed any other unauthorized hosts access MyPBX. See as below figure.

Note: If 'drop all' is not enabled, please don't remove the IP blacklist rules in case the system security hole.



2.lt is strongly re	oot the system a ecommended to			Firewall Settings with the 'action' is 'accept',		Firewall	has started successfu
or it may be drag common Rules Add Rule	gged into the bla	cklist.					
	Action	Name	Protocol		MAC Address	Port	
7 ±	ACCEPT	Localnetwork	BOTH	192.168.5.0/255.255.255.0		1:65535	🔊 Edit 😕 Delete
₹ \$ \$ \$	ACCEPT	RomoteExtension	BOTH	110.111.132.6/255.255.255.255		1:65535	😼 Edit 📁 🗶 Delete
₹ \$ \$ \$	ACCEPT	VolPTrunk	UDP	110.5.14.6/255.255.255.255		5060:5060	😼 Edit 🔰 Delete
₹ \$ \$ \$	ACCEPT	RemoteHTTP	TCP	115.5.14.6/255.255.255.255		80:80	🔊 Edit 😕 Delete
¥ 3	ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	-	8022:8022	🐕 Edit 🛛 🗶 Delete
+ Add Rule				No Auto Defense Rules Defined			
PBlacklist							
 Add Rule 							IP Blacklist Ma
				No Auto Black IP Address			
ther Settings							
🗆 🛈 Disable P							

Figure 4-11

5*. Alert Settings

After enabling alert settings', if the device is attacked, the system will notify users the alert via call or e-mail. The attack modes include IP attack and Web Login.

5.1 IPATTACK

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

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

```
Phone Notification Settings:

Phone Notification: Yes

Number: 500;5503301

Attempts: 1

Interval: 60s

Prompt: default
```

Note: If there's outbound number to notify, the number should be fit with the dial pattern of outbound route.

E-mail Notification Settings:



E-mail Notification: Yes

To: alert@yeastar.com

Subject: IPAttack

IPAT	ГАСК		٢	K
	Phone Notification	n Settings hone Notification:	Yes 🔻	
		Number 🕕 :	500;5503301	
		Attempts 🕕 :	1 -	
		Interval 🚺 :	60 s	
		Prompt:	default Custom Prompts	
	E-mail Notification	n Settings ·mail Notification:	Yes 🔻	
		To():	alert@yeastar.com	
		Subject:	IPAttack	
	0	attack source ip attack dest mac attack source po	address:\$(SOURCEIP) :\$(DESTMAC) ort:\$(DESTPORT) rotocol:\$(PROTOCOL)	
		 ✓ 	Save X Cancel	
		F	igure 5-1	

5.2 WEBLOGIN

Enter the password incorrectly five times when logging in MyPBX Web interface will be deemed as attack, the system will limit the IP login within 10 minutes and notify the user.

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

Phone Notification Settings:

Phone Notification: Yes Number: 500;5503301 Attempts: 1 Interval: 60s



Prompt: default

Note: If there's outbound number to notify, the number should be fit with the dial pattern of outbound route.

E-mail Notification Settings:

E-mail Notification: Yes

To: alert@yeastar.com

Subject: WebLogin

WEBLOGIN	Х
Phone Notification Settings Phone Notification: Yes Number i: 500;5503301; Attempts i: 1 Interval i: 60 s Prompt: default <u>Custom Prompts</u>	
E-mail Notification Settings E-mail Notification: Yes To alert@yeastar.com Subject: WebLogin pbx hostname:\$(HOSTNAME) login ip address:\$(SOURCEIP) login username:\$(USERNAME) login occurred:\$(DATETIME)	
Save X Cancel	

Figure 5-2

6. Note

If the phenomena of toll fraud have been happened in your MyPBX system. We are really sorry about that, then please enhance the protection level of your firewall refer to the above steps.

In addition, please change the all password: Web GUI password, SSH password, and all extensions password.



APPENDIX C How to Configure Autobackup

Before Autobackup can be properly configured, an SMB share folder accessible from MyPBX must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

Step 1 Add a new folder, rename it, and set this new folder's share Properties according to Figure B-1

st 💭	hare	share Properties 🛛 🛛 🔀
		General Sharing Customize
		Local sharing and security
		To share this folder with other users of this computer only, drag it to the <u>Shared Documents</u> folder.
		To make this folder and its subfolders private so that only you have access, select the following check box.
		Make this folder private
		Network sharing and security To share this folder with both network users and other users of this computer, select the first check box below and type a share name.
		Share this folder on the network
		Share name: share
		Allow network users to change my files
		Learn more about sharing and security.
		Windows Firewall is configured to allow this folder to be shared with other computers on the network. View your Windows Firewall settings
		OK Cancel Apply

Figure B-1 Set up share Properties

Step 2 Enter the new folder and create a new text file, then rename this file to status.txt. This step is very important, DO NOT forget to create the status.txt file.

Step 3 Configure Autobackup settings on MyPBX to Figure B-2



External Storage Settings								
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.								
<u>Step 1: Create a Net-Disk on a chosen computer</u>								
Step 2: Input the Net-Disk properties								
Net-Disk Host/IP:	192.168.5.222							
Net-Disk Share Name:	share							
Net-Disk Access Username:								
Net-Disk Access Password:								
Move files created before:	1 🔽 days ago							
🗸 Sav	e 🔀 Cancel							
Step 3: Save Net-Disk settings								
Step 4: Make sure the settings are successfully complete	<u>ed</u>							

Figure B-2 Autobackup Setting

Net-Disk Host/IP: Change this to the IP address of the computer where backup files will be stored.

Net-Disk Share Name: Change this to the name of the shared folder where backups will be stored.

Net-Disk Share Username: The user name used to log into the network share. Leave this blank if it is not required

Net-Disk Share Password: The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to Figure B-3, then you have successfully configured Autobackup on the MyPBX unit.

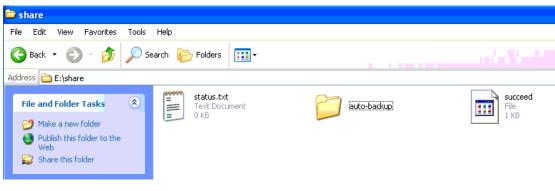


Figure B-3 Autobackup setting succeed



APPENDIX D How to Configure NAT setting

When MyPBX is behind a NAT(firewall), you need to configure NAT setting on MyPBX if you want to use a remote extension.

Please follow section **1** or **2** below depending on your network configuration. **1.** If MyPBX is connected to a local network, you must set up port forwarding on your router. Specifically, you must map port 5060 (default SIP port) and port 10001-10200 (default RTP port range) as UDP ports.

Next, go to the MyPBX web interface and configure the SIP settings according to Figure C-1:

External IP Address: your router's public IP address

External Host: your router's domain

External Refresh Interval: 20 seconds

Local Network Identification:192.168.5.0/255.255.255.0 (change this according to your network configuration)

NAT mode: Yes

Allow RTP Reinvite: No

NAT	
Note: Configuration of this section is only re	equired when using remote extensions.
Enable STUN:	
STUN Address:	
STUN Port:	
External IP Address 🛈 :	
External Host 🛈 : 🕚	yeastar.3322.org
External Refresh Interval 🛈 : 💈	20
Local Network Identification $\widehat{oldsymbol{v}}$:	192.168.5.0/255.25
NAT Mode🛈:	yes 👻
Allow RTP Reinvite 🛈 :	yes 💌

Figure C-1

Assuming that your router's host address is yeastar.3322.org, your local network is from 192.168.5.1-192.168.5.254, and the subnet Mask is 255.255.255.0, the MyPBX network settings should configured like Figure C-2



LAN Settings	
DHCP:	No 💌
Enable SSH:	Yes V Port: 8022
Hostname:	MyPBX
IP Address:	192.168.5.186
Subnet Mask :	255.255.255.0
Gateway :	192.168.5.1
Primary DNS :	192.168.5.1
Secondary DNS :	
IP Address2:	
Subnet Mask2:	

Figure C-2 MyPBX Network setting

2. If MyPBX has a public IP, (i.e. is connected directly to your internet service provider), the network settings should be configured according to Figure C-3:

LAN Settings	
DHCP:	No 💌
Enable SSH:	Yes Y Port: 8022
Hostname:	MyPBX
IP Address:	221.245.25.117
Subnet Mask :	255.255.255.0
Gateway :	221.245.25.1
Primary DNS :	202.101.103.54
Secondary DNS :	202.101.103.55
IP Address2:	
Subnet Mask2:	

Figure C-3

Next, you should configure the NAT settings according to Figure C-4

NAT	
Note: Configuration of this section is only	required when you use remote extensions.
Enable STUN	
STUN Address	
STUN Port	
External IP Address	221.245.25.117
External Host	
External Refresh Interval	
Local Network Identification	
NAT Mode	yes 💌
Allow RTP Reinvite	no 👻

Figure C-4

External IP Address: The public IP address of MyPBX External Host: Leave this blank if no domain has been configured External Refresh Interval: Leave this blank Local Network Identification: Leave this blank NAT mode: Yes Allow RTP Reinvite: No



APPENDIX E How to Use Auto Provision

Step1. Disable DHCP Server on your local network. E.g. Disable DHCP Server on Linksys Router.

A Division of Cisco Systems, Inc.							Firmware Version: 1.04.06
				Etherfa	st® Cable/DSI	L Router	BEFSR41
Setup	Setup		pplications	Administra	tion Stat	tus	
	Basic Setup	DDNS	& Gaming MAC Addre	ess Clone	Advanced Rout	ting	
Internet Setup						Basic	Setup
Internet Connection Type	Obtain an IP au	tomatically 💌				The Basic	Setup screen is
Optional Settings (required by some ISPs)	Host Name: Domain Name: MTU:	C Enable 💿 D	isable Size: 1	500		where back performed Service Pot that you e information be obtained you have	sic configuration is 1. Some ISPs (Internet roviders) will require nter the DNS n. These settings can tid from your ISP. After configured these rou should set a router
Network Setup						password	I from the
Router IP	Local IP Address:	192.168.	1.1			Administri screen.	ation->Management
	Subnet Mask:	255.255.255	.0 💙			Completin	g the Internet Setup
Network Address Server Settings (DHCP)	Local DHCP Server Start IP Address:	© Enable © Di 192.168.1. 100	sable			section is set up for Please loo	all that is required to your specific ISP. k at the table below to the Router for your
	Number of Address:	50				internet of	Jinecuon.
	DHCP Address Range:	192.168.1.100 to	192.168.1.149			More	
	Client Lease Time:	0 minutes	(0 means one da	iy)			
	Static DNS 1:	0.0.) . 0				
	Static DNS 2:	0.0.	0.0				
	Static DNS 3:	0.0.	0.0				
	WINS:	0.0.) . 0				
							CISCO SYSTEMS
		_		_			
			Save Settings	Cancel C	hanges		6 min had min has

Figure D-1

Step2. Enable DHCP Server on MyPBX.

Login MyPBX web interface, System Settings \rightarrow DHCP Server \rightarrow Enable DHCP Server.



Status Monitor	▶ DHCP Server ♦
Status Monitor	
Line Status	DHCP Server
Basic 🙁	DHCP is not running
Extensions	
Trunks	(v) Enable
Outbound Routes	Routers : 192.168.5.1
Auto Provision	Subnet Mask : [255,255,0
Inbound Call Control 🔕	Primary DNS : 192.168.5.1
IVR	Secondary DNS :
Queues	Allow IP Address From: 192.168.5.2
Custom Prompts	
Ring Groups Inbound Routes	To: 192.168.5.254
Blacklist	
Diackiist	Save Cancel
Internal Settings 🛛 🖄	
Options	
Business Hours	
Feature Codes	
SIP Settings	
IAX Settings Voicemail Settings	
DISA	
Conferencing	
Paging Groups	
PIN User Settings	
Speed Dial Settings	
Music On Hold Prompts	
Network Settings 🙁	
LAN Settings	
WAN Settings	
Firewall	
DHCP Server	
1/LANLCottings	

Figure D-2

Step3. Configure phones on MyPBX auto-provision page.

1. Login MyPBX web interface, Basic \rightarrow Auto Provision \rightarrow Create New Phone.

🗿 MyPBX - Embedded Hyb	rid IPPBX for Small Busines	ses - Microsoft Internet Explorer		
Ele Edit View Favorites	Tools Help			
🚱 Back 🝷 💮 🕤 💌	🗟 🏠 🔎 Search 🤺	avorites 🤣 🍰 🚽		
Address 🙆 http://192.168.5.18	86/cgi/WebCGI?1000			✓ ラ Go 链接
MyPBX	Embedded Hybrid IP-	PBX for Small Businesses		Logout
Status Monitor 🔹	🕨 Auto Provision 🔅			
Line Status			Auto Provision	
	Phone			
Basic 🔹 Extensions	+ Create New Phone			<u> </u>
Trunks	MAC	Name	Extension	
Outbound Routes Auto Provision	Upload a file			
Inbound Call Control 🙁	#	Name	Options	
IVR	▶ IP phone			_
Queues	+ Scan Yealink Phone			
Custom Prompts	- ocur realment none			
Ring Groups Inbound Routes				
Blacklist		M	ac Address List	
Internal Settings 🔹				
Options				
Business Hours Feature Codes				
SIP Settings				
IAX Settings				
Voicemail Settings				
DISA				
Conferencing				<u> </u>
ê				🔮 Internet

Figure D-3

2. Fill in the phone detail message on the pop-up windows.

Input IP Phone's MAC address, configure Name, Call waiting, Line, Extension, Label, Line active for the phone.



Create New Ph	one		X
General M Key As Se	AC: 00156511189E	Name: Rela	Call Waiting : Enabled
Line			
🔽 Line1	Extension: 500	✓ Label: 500) Line Active: 🔽
🗖 Line2	Extension:	Label:	Line Active: 📁
🗖 Line3	Extension:	🕑 Label:	Line Active: 厂
🗖 Line4	Extension:	🕑 Label:	Line Active: 📕
🗖 Line5	Extension:	🕑 Label:	Line Active: 📕
🗖 Line6	Extension:	👻 Label:	Line Active: 📁
		🖌 Save 🛛 🗙 Cancel	

Figure D-4

Step4. Turn on the power and connect the network cable to IP Phone.

Remark: The factory default setting of DHCP for IP Phone is enable, so you can skip this step to step 5.

If the DHCP is disable, please follow below step to enable it. (e.g.: Yealink's IP Phone).

- 1. Login IP phone's web page.
- 2. Enable DHCP.

Yealink								
Easy vop	Status	Account	Network	Phone	Contacts	Upgrade	Security	
		Internet F	Port (WAN)	PC Port / /	Advanced			
	IP , Sul De Prir Sec C PP Use	Address Address bnet Mask fault Gateway mary DNS condary DNS POE		Cancel		addres server Static Set th Mask, I addres Secon manua PPPol	·	t

Figure D-5

Step5. Finish.



APPENDIX F How Do I Configure Distinctive Ring Tones

Step1: On your IP phone, navigate to the Phone settings web configuration page and find the Distinctive Ring Tone section.

For each custom ring tone, enter the Internal Ringer Text (can be digits or text) to trigger the ring tone. For example, you may enter "Family". e.g.: Yealink's IP phone.

Yealink					
Sta	atus Account	Network	Phone	Contacts Up	ograde Security
Pre	eference Features DSS	S Key EXT Key	Voice Ring	Tones Dial Plan	SMS
1	Internal Ringer Text	Far	nily	0	
	Internal Ringer File	Rin	g1.wav 💌		E HOTE
2	Internal Ringer Text				
	Internal Ringer File	Rin	g2.wav 💌		
3	Internal Ringer Text				
	Internal Ringer File	Rin	g3.wav 💌		
4	Internal Ringer Text				
	Internal Ringer File	Rin	g4.wav 💌		
5	Internal Ringer Text				
	Internal Ringer File	Rin	g5.wav 💌		
6	Internal Ringer Text				
	Internal Ringer File	Rin	g6.wav 💌		
7	-				
	Internal Ringer File	Rin	g7.wav 💌		
8	-				
	Internal Ringer File	Rin	g8.wav 💌		
9	-				
	Internal Ringer File	Rin	g1.wav 💌		
10	-				
	Internal Ringer File	Rin	g1.wav 💌		
	Confirm		Cancel		

Figure E-1

Step2. Configure the 'Distinctive Ringtone' on MyPBX.

MyPBX web interface, Inbound Routes \rightarrow Edit Inbound Route, fill in the Internal Ringer Text on 'Distinctive Ringtone'.



Edit Inbound Route: allin				Х
General				
Route Name	🛈 : allin			
DID Number (i :			
Extension	0:			
Caller ID Number	0:			
Distinctive Ringtone				
Member Trunks			•	
Available Trunks			Selected	
	 >>>	pstn1(FXO) pstn2(FXO) pstn6(FXO) GSM11(GSM BriTrunk7(BF BriTrunk8(BF 8032(SIP) voipprovider	RI) RI)	
During Office Hours				
	End Call	_		
	C Extension		Extension 500	*
	C Voicemail		Voicemail 500	~
Destination:	IVR		IVR welcome	*
	C RingGroup		RingGroup ringgroup_def:	~
	C Conference	Room	Conference Room 640	

Figure E-2

Step3. Finish.



APPENDIX G How to Use Email to SMS

How to use Email to SMS

You need to send an email to the specified email address (you set in Email Settings. In this case, it is lears@yeastar.com).The content of this email will be sent to the number you want as message. The subject (title) of the email will determine the number. Here are some examples of the formats to the subject of the email.

Example:

1. Send message with no PIN code and default GSM port.

Format: phonenumber

if the subject is "12345678", the text of this email("Welcome to Yeastar!") will be sent to number "12345678" through the first available GSM trunk(No pin code should be set by administrator).

To:	lears@yeastar.com 🥖 🗙
Subject:	12345678
Insert:	🛿 Attachments 🛛 🎦 Office docs 🛛 🔄 Photos 🔻 💽 From Bing 🔻 🤓 Emoticons
Tahoma	• 10 • B / U 画画画 注 注 律 律 😣 🚝 📥
Icome to	Yeastar!

Figure F-1

2. Send message with no PIN code and specified GSM port.

Format: port:portnumber-phonenumber

if the subject is "port:9-12345678", the text of this email ("Welcome to Yeastar!") will be sent to the number "12345678" through GSM trunk 9 (No pin code should be set by administrator).



ments 🛛 🚼 Office docs 🛛 🔄 Photos 🔻 🔂 From Bing 🔹 🤓 Emoticons
10 • B I U 副 副 田 田 田 田 田 田 田 田 田 田 田 田 田 田 田 田 田

Figure F-2

3. Send message with PIN code and default GSM port.

Format: 500:pincodenumber-phonenumber

if the subject is "500:987-12345678", the text of this email("Welcome to Yeastar!") will be sent to number "12345678" through the first available GSM trunk("987" is the pin code set by administrator).

To:	lears@ye	astar.cor	m 🥖 X			
Subject:	500:987-1	12345678				
Insert:	🛿 Attachm	ents 🎖	Contraction of Contract of Con	📓 Photos 🔻	🔁 From Bing 🔻	😂 Emoticons
Tahoma	•	10 🔹	B <i>I</i> <u>U</u>		三 三 (年 (年 🤮	
elcome to	Yeastar!					

Figure F-3

4. Send message with PIN code and specified GSM port.

Format: 500:pincodenumber-port:portnumber-phonenumber if the subject is "500:987-port:9-12345678", the text of this email("Welcome to Yeastar!") will be sent to number "12345678" through GSM trunk 9("987" is the pin code set by administrator).

rearseyeastan	.com 🤌 X	
500:987-port:9)-12345678	
Attachments	🔡 Office docs 🔄 Photos 🔻 💽 From Bing 🔻 🤓 Emoticons	
• 10	• B / U 画 画 画 扫 扫 律 律 🛞 🚝 📥	
	Attachments	500:987-port:9-12345678 Attachments $\stackrel{23}{{{{{{{{{{\underset$

Figure F-4



APPENDIX H How to Use DID

Direct inward dialing (DID), also called direct dial-in (DDI) in Europe and Oceania, is a feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk.

MyPBX support DID, you can configure DID in inbound route. Related settings: **DID Number, Extension, Destination.**

eneral				
Route Name				
DID Number 🛈				
Extension 🛈				
Caller ID Number 🕕				
Distinctive Ringtone 🛈				
Distinctive Kingtone	•			
Enable Callback	No 🗸	Callb	ack Settings	
ember Trunks				
Available Trunks			Selecte	d
5503302 (FXO) 5503305 (FXO)		a l		
5503306 (FXO)	-			
pstn5(FXO)	-	18		
pstn6 (FXO) pstn7 (FXO)	*			



·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.

Extension

Define the extension for DID number, this field only valid when use E1 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.

Destination

If you don't set the extension, you can set the destination of the call here.

Example 1:

Step1: You set the DID number (5503XXX in this example). Step2: You choose the destination (Ring Group in this example).

The configuration of this example means when the incoming call with DID number 5503XXX (7 digits number start with 5503) will go to the destination Ring Group.

If you choose the destination, please leave the Extension form blank.



eneral					
Route Nam	ne 🛈 : E	3RI1			
DID Number	0:5	5503XXX			
Extension	ι 🛈 : 🗍				
Caller ID Number	0				
Distinctive Ringtone	0.				
Enable Call	back : N	No 👻	Callback	Settings	
ember Trunks					
Available Trunks				Selected	
5503301 (FXO)		»»			
5503302 (FXO) 5503305 (FXO)	Â				
5503306 (FXO)	E				
	Contraction of the second s				
pstn6 (FXO)		~			
pstn6 (FXO) pstn7 (FXO)	-				
pstn6 (FXO) pstn7 (FXO)	•	<u>ج</u>			
pstn5 (FXO) pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	•				
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	_	44			
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)		end Call		Fabracian an 500	
pstn6 (FXO) pstn7 (FXO)	0	End Call Extension		Extension 500	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0	End Call Extension Voicemail		Voicemail 500	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	0	End Call Extension			•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0	End Call Extension Voicemail		Voicemail 500	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0 0 0	End Call Extension Voicemail IVR		Voicemail 500 IVR welcome	• • nalSe •
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0 0 0 0	End Call Extension Voicemail IVR RingGroup		Voicemail 500 IVR welcome RingGroup Natio	• • nalSe •
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0 0 0 0	End Call Extension Voicemail IVR RingGroup Conference		Voicemail 500 IVR welcome RingGroup Natio Conference Room	▼ ▼ malSe ▼
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	0 0 0 0 0	End Call Extension Voicemail IVR RingGroup Conference DISA		Voicemail 500 IVR welcome RingGroup Natio Conference Room DISA test	 ✓ ✓ ✓ ✓ ✓ ✓ ✓ ✓ ✓

Figure G-2

Example 2:

Step1: You set the DID number (6001-6099 in this example). Step2: You set the Extension (6001-6099 in this example).

The configuration of this example means when the incoming call with DID number 6001 to 6099 will go to the destination 6001 to 6099(number 6001 to extension 6001, number 6002 to extension 6002).

The destination you set below will be disabled if you set the Extension.



General		
Route Name 🛈 :	BRI1	
DID Number 🛈 :	6001-6099	
Extension 🛈 :	6001-6099	
Caller ID Number 🛈 :		
Distinctive Ringtone 🛈 :		
Enable Callback :	No - Callbac	<u>k Settings</u>
lember Trunks		
Available Trunks		Selected
5503301 (FXO) 5503302 (FXO)	»»	-
	→	
5503305 (FXO) 5503306 (FXO)		
5503306 (FXO) pstn5 (FXO)	*	
5503306 (FXO)	Ŧ	

Figure G-3



APPENDIX I How to Use BLF Key to Choose the PSTN line.

MyPBX allows you to choose the specific PSTN line to make outbound call by pressing the BLF key on the IP Phone.

Follow the steps to do the configuration with your Yealink phone

1. We want to choose pstn1 or pstn2 to call out.

MyPBX F	mbedded Hybrid IP-PBX for Small		Lo
Status Monitor 🛞	Manage Trunks 🌾	Trunk List	
Basic 🔕	Trunk Name	Port/Hostname/IP	
Extensions	pstnl	1	MEdit
<u>Trunks</u> Outbound Routes	pstn2	2	N Edit
Auto Provision	pstn3	3	🔊 Edit
	pstn4	4	K) Edit

Figure H-1

2. Configure the IP Phone:

Memo	ry Key >> 🕜			
Key	Туре	Value	Line	Extension
DSS Key 1	BLF	← pstn1	Line 1	pstn1
DSS Key 2	BLF		Line 1	pstn2

Figure H-2

Test

When you press DSS Key 1/2, the phone will connect to pstn1/pstn2 line. If pstn1/pstn2 is not busy, you will hear the dial tone. You can dial the number you want and use this line to call out then.

<Finish>