

Version 2.16.0.58

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.

## Caveat

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation. Any Changes or modifications not expressly approved by the party responsible for compliance could void the user"s authority to operate the equipment.



## **Customer Information**

- This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the bottom of this equipment is a label that contains, among other information, a product identifier in the format [US: X8PIS00BSTD]. If requested, this number must be provided to the telephone company.
- 2. A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. See installation instructions for details.
- 3. If this equipment [Yeastar MyPBX] causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
- 4. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
- If trouble is experienced with this equipment [US: X8PIS00BSTD], for repair or warranty information, Service can be facilitated through our office at: U.S. Agent Company name:

Commlogik Corporation

Address: 1921 NW 82nd Ave Miami - FL - 33126, USA

Tel: +1 (305) 677-7888

Fax: +1 (305) 677-7889

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

- 6. Please follow instructions for repairing if any (e.g. battery replacement section); otherwise do not alternate or repair any parts of device except specified.
- 7. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
- 8. If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this [US: X8PIS00BSTD] does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.
- 9. If the telephone company requests information on what equipment is connected to their lines, inform them of:

a) The ringer equivalence number [0.0B]



- b) The USOC jack required [RJ11C]
- c) Facility Interface Codes ("FIC") [METALLIC]
- d) Service Order Codes ("SOC") [9.0y]
- e) The FCC Registration Number [US: X8PIS00BSTD]
- 10. The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for this product is part of the product identifier that has the format US: AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point. For this product the FCC Registration number is [US:X8PIS00BSTD] indicates the REN would be 0.0B.

## 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

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

# **1.2 Hardware Specifications**

## 1.2.1 Exterior Appearance

#### 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.

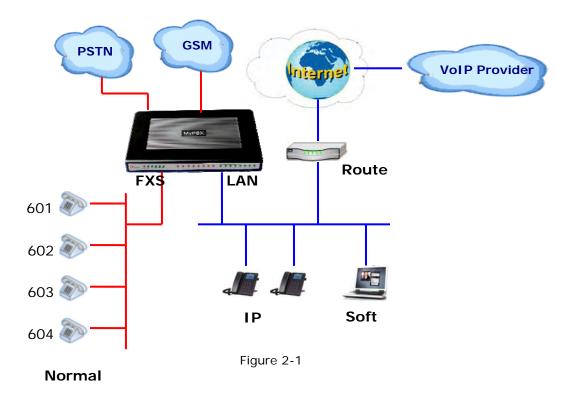


LED Alternating - Red and Green (fast blink): FXO port is in use LED Dual - Green and Red (slow blink): FXS port is ringing LED Dual - Green and Red (fast blink): FXS port is in use

# 2. System set up

# 2.1 Connection Drawing

### A. MyPBX



# 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

a http://192.168.5.186/	Nicrosoft Internet Explorer	
<u>File Edit View</u> Favorites		-
3 Back 🔹 🕥  🖹	🖹 🔣 🔎 Search 👷 Favorites 🥹 🎯 - 😓 🥽	
Address 🛃 http://192.168.5.18	6/	链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	~
	MyPBX Configuration Panel	
Yeastar	User Login	
	User Name:	
	Password;	
	Language: English	
	Login Rését	
		×
👸 Done	🔮 Internet	

Figure 3-1



# 3.2 Status Monitor

### 3.2.1 Line Status

	Embedded H	ybrid IP-PBX 1 Ø	for Sn	nall Business	es					Logou
tatus Monitor		Tree 2	4	Busy	Exte	ensions old	a Unav	ailable	Ringing	
xtensions										
runks utbound Routes		<u>500</u> (SIP)	9	<u>501</u> (SIP)	a	<u>502</u> (SIP)	4	> 503 (SIP)	Ì	<u>504</u> (SIP)
hone Provisioning		505 (SIP)	7	601 (FXS)	7	602 (FXS)	-	> 603 (FXS)	27	604 (FXS)
bound Call Control 🌋 R ueues		<u>605</u> (FXS)	9	<u>606</u> (FXS)						
ustom Prompts					Tr	unks				
ing Groups bound Routes	Status	Signal		Trunk Nar	ne	Туре	User Name	Port/Hostnan	ne/IP Rea	chability
lacklist	Registered			305@192.168	4.138	SIP	305	192.168.4.1	138	OK
ternal Settings	OK (1006 ms)			192.168.5.	138	SP-SIP		192.168.5.1	138 OK (	1006 ms)
ternal Settings 🄇 🍣	Failed			<u>aa</u>		SP-IAX		111111	UNF	KNOWN
usiness Hours	Idle			pstn4		FXO		Port 4		
eature Codes	Disconnected			pstn5		FXO		Port 5		
P Settings	Disconnected			pstn6		FXO		Port 6		
AX Settings	Disconnected			pstn16		FXO		Port 16		

Figure 3.2.1

#### **MyPBX Status Description:**

#### Extensions:

- 1) 🦥 : Extension is unregistered
- 2) 💐 : Extension is idle
- 3) 🗳: Extension is ringing
- 4)  $\overline{\mathbf{v}}$ : Extension is busy in a call
- 5) **3**: 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

Y : No signal.

🏋 : Poor.

**\*** : Average.

Til : Good.

Til: 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).



	<u>I</u> ools <u>H</u> elp							
Back * 🔘 🖹	😰 🏠 🔎 Se	earch 🤺 Favorites 🍕	3 🗟 - 💐					
ress 🗃 http://192.168.5.	186/cgi/WebCGI?1000							🖌 🄁 Go 👹
MyPBX	Embedded H	iybrid IP-PBX for s	Small Busine					Logout
Status Monitor 🛸	Extension	\$						
Line Status	and the second			E	stension List			
asic *	FXS Extension	ons						
Extensions	Port	Extension		ull Name	Caller ID			
Trunks	3	601		601	601	🐕 Edit	🗴 Delete	
Outbound Routes	4	602		602	602	M Edit	🗶 Delete	
Auto Provision	5	603		603	603	Ni Edit,	🗴 Delete	
nbound Call Control 🏾 🋸	VolP Extensi							
IVR Queues	+ Create Ne	ew Extension Add M	Iultiple Extension	is 🛛 🔊 Modify Selected I	Extensions X Delete Selecte	d Extensions		
Queues Custom Prompts		ew Extension Add M	lultiple Extension	is Modify Selected I Full Name	Extensions Delete Selecte Caller ID	d Extensions		
Queues Custom Prompts Ring Groups						d Extensions	X Delete	
Queues Custom Prompts Ring Groups Inbound Routes		Extension	Туре	Full Name	Caller ID		X Delete X Delete	
Queues Custom Prompts Ring Groups Inbound Routes Blacklist		Extension 500	Type SIP	Full Name 500	Caller ID 500	S Edit		
Queues Custom Prompts Ring Groups Inbound Routes		Extension 500 501	Type SIP SIP	Full Name 500 501	Calier ID 500 501	¥ <sup>3</sup> Edit 10 <sup>3</sup> Edit	🗴 Delete	
Queues Custom Prompts Ring Groups Inbound Routes Blacklist Internal Settings 🏾 🏝 Options		Extension 500 501 502	Type SIP SIP SIP	Full Name 500 501 502	Caller ID 500 501 502	Se Edit Se Edit Se Edit	X Delete X Delete	
Queues Custom Prompts Ring Groups Inbound Routes Blacklist <b>Internal Settings</b>		Extension 500 501 502 503	Type SIP SIP SIP SIP	Full Name 500 501 502 503	Caller ID 500 501 502 503	Sedit Sedit Sedit Sedit	X Delete X Delete X Delete	

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.

#### ·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.



#### ·DND

Don't Disturb.

#### ·Ring Out

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

#### 8) Volume Settings

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

#### 9) Mobility Extension

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 3 spy modes available for choice: 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.

Edit Extension - 601	Х
General         Port:         1           Extension <sup>(1)</sup> :         601         Caller ID <sup>(1)</sup> :         601	
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.	
Flash Hook Flash Detection ①: 1000 ms	
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 Vser Web Interface Ring Out : 30	
Volume Settings Rxgain <sup>1</sup> : 40%  Txgain <sup>1</sup> : 40%  Txgain <sup>1</sup> : 40%	
Mobility Extension         Enable Mobility Extension         Mobility Extension	
Spy Settings Allow Being Spied Spy Modes:	

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

#### ∙Туре

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) 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.

#### ·Enable SRTP

Enable extension for SRTP (RTP Encryption).

#### Qualify

Send check alive packets to IP phones



#### ·SIP Transport

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

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

8) 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.

#### 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.

#### 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 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.



Extension - 500
General         Type: SP          Solution         Name(1): 500         Caller ID(1): 500
Voicemail          Image: Second state       Voicemail Access PIN #0 : 500
Mail Setting Enable Send Voicemail Email Address
<b>Note:</b> 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       Image: Voicemail         Follow me:       Image: No answer       Transfer to:       Image: Number         Image: When Busy       Image: Number       Image: Number
Other Options Call Waiting DND OUVIEW User Web Interface Ring Out Spy Settings
Allow Being Spied Spy Modes:
Optional Settings ≈
VolP Settings       NAT●:       □       Enable SRTP●:       □       Qualify:       □       Transport:       UDP       □       DTMF Mode●:       RFC2833
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 addr
Mobility Extension          Enable Mobility Extension       Mobility Extension Number 1:
Save X Cancel

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.



Edit View Favorites Io	IPPBX for Small Businesses - Micr ols <u>H</u> elp	soore meetinge expe	1101		_
Back • 🔘 💌 💈	Search 👷 Favorites 🕻	A.B.	3		
s 🙆 http://192.168.5.186/cg		a la cara			v 🗗 G
<b>AyPBX</b>	Embedded Hybrid IP-PBX for	Small Businesse			Loo
	Manage Trunks Ø				
tus Monitor 🔹 — e Status			T	runk List	
	Analog Trunk			Tarin Line	
ic 🌲	Trunk Name		Port	Hostname/IP	
insions nks	pstn1	_		1	S Edit
bound Routes	pstn2			2	😼 Edit
Provision	pstn6			6	🖞 Edit
und Call Control 🖈 🕕	GSM Trunk				
	Trunk Name		Port	Hostname/IP	
ues	GSM11			11	S Edit
tom Prompts g Groups	BRI Trunk				
und Routes	Trunk Name		Port	Hostname/IP	
klist	BriTrunk7			7	🔊 Edit
nal Settings 🔹	BriTrunk8			8	🔊 Edit
	VolP Trunk			1	
iness Hours	+ New VolP Trunk				
ture Codes Settings	Provider Name	Туре	Hostname/IP	User Name	
Bettings	8032	SIP	192.168.5.99	8032	N Edit X Delete
email Settings	Service Provider				
A. ferencing	New Service Provider				
ing Groups					
User Settings	Provider Name	Туре		ime/IP	
ed Dial Settings ic On Hold Prompts	192,168.5.190	SIP	192.16	8.5.190	Delete

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 ①:	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
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 🛈 : GSM	13
Volume Setting 🛈 : 40%	<b>•</b>
PIN Code:	
	ou failed to enter your correct PIN on, 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	<ul> <li>Overlap Dial ①:</li> </ul>	no 🔻	
PRI Indication 🛈 :		<ul> <li>Enable Facility<sup>1</sup>:</li> </ul>	Enabled 💌	
Nsf0:	none	<ul> <li>Echo Cancellation<sup>1</sup>:</li> </ul>	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 :	A	ssociated 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

#### ∙Туре

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.

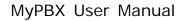
#### ·DTMF mode

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

2) DOD Setting•DODDirect Outward Dialing Number.

#### ·Associated Extension

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





Edit VolP Trunk - test			Х
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			
Maximum Channels 🛈 :	0		
Caller ID 🛈 :			
	Enable Outbound Proxy Server		
Codecs :	First: a-law Second: u-law	▼ Third: GSM ▼	
	Fourth: None <b>•</b> Fifth: None		
Transport:	UDP  Enable SRTP	Qualify: 🔽	
	rfc2833 🔻		
DOD Settings			
DOD:	Associated Extension: 60	1 ▼ ↑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

#### ∙Туре

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•DODDirect Outward Dialing Number.

#### ·Associated Extension

The extension make call out via BRI 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 X 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•DODDirect Outward Dialing Number.

#### Associated Extension

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



Edit Service Provider Trunk-S	PS-192.168.4.138	Х		
Туре:	SIP 🔻			
Provider Name:	192.168.4.138			
Hostname/IP:	192.168.4.138 : 5060			
Maximum Channels 🛈 :	0			
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  Associated Extension : 601				
Save X Cancel				
Figure 3.3.2.5				

## 3.3.3 Outbound Routes

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

MyPBX - Embedded Hyt	orid IPPBX for Sma	ll Businesses - Microsoft I	nternet Explorer			
<u>File E</u> dit <u>V</u> iew Favorites	Tools Help					27
🚱 Back 🔹 🔘  💌	🧃 🏠 🔎 Sea	rch 🔆 Favorites 🙆 👔	3· 3 🖻			
Address 🙆 http://192.168.5.18	86/cgi/WebCGI?1000					💌 🄁 Go 链接
MyPBX	Embedded Hy	/brid IP-PBX for Small	l Businesses			Logout
Status Monitor *	🕨 Manage Outl	oound Routes 💠 👘				
Line Status	New Outpo	und Route	Outbound Routes			
Basic *		Route Name	Dial Pattern			
Extensions	3 ±	pstnout	9.	🔂 Edit	🗴 Delete	
Trunks	Ŧ \$	to190	Х.	🔛 Edit	🗴 Delete	
Outbound Routes Auto Provision						
Inbound Call Control R IVR Queues Custom Prompts Ring Groups Inbound Routes Blacklist						
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings						
🕘 Done					🥥 Inte	ernet

Figure 3.3.3

#### 3.3.3.1 New Outbound Route

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

#### 1) General

#### •Route Name

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

#### ·Dial Pattern

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

- X : Any Digit from 0-9
- **Z** : Any Digit from 1-9
- **N** : Any Digit from 2-9

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

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

Example 1: **NXXXXXX** will match any 7 digits phone number. Example 2: **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 .



w Outbound Route		
Route Name		
Dial Pattern 🧿:	1	
Strip 🛈 :	digits from	front
Prepend these digits 🛈 :		before dialing
Password:		
T.38 Support	No	
Rrmemory Hunt		
Member Extensions		1
Available Extensions		Selected
501(SIP) 502(SIP) 503(SIP)	~	
Member Trunks		Selected
1234(FXO) pstn5(FXO) pstn7(FXO) pstn7(FXO) BriTrunk9(BRI) BriTrunk10(BRI) Mexico(SIP)	84 200	
<u></u>	Save 🔀 Cance	

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)

## News:

When provisioning Yealink and Snom IP phone, MyPBX is not needed to be set as the only DHCP server any more.



Phone Provision	ing 🗘					
		Pł	none Provisioning			
General Settings	for Yealink					
- Phone Book						
Configured Phon						
+ Add Phone	Add Bulk Phones 🔊 Config	ure the Selected Phones X Delete the Selec	ted Phones			
🔲 ID	MAC Address	Manufacturer	Phone Type	Name	Extension	Enabled
Not Configured P	hone					
🔊 Configure the S	elected Phones Refresh					
	ID	MAC Address	Manufacturer		Phon	е Туре
	1	001565113844	Yealink			
	2	<u>001565114094</u>	Yealink			-
Upload a file						
#		Name			Options	

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

**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.



General Settings IVI Leanink 4.	C. Dester Desse Desterior
	Go Back to Phone Provisionin
General Preferences	Toola -
Language 🔮: Web server Type:	
	• Fixed C Prefix
	admin
Time Zone:	+8 China(Beijing)
Primary NTP Server:	cn.pool.ntp.org
Secondary NTP Server:	
Daylight Saving Time:	Disabled 🔹
Time Format:	12 Hour 🗸
Date Format:	WWW MMM DD 🔹
Voicemail:	
	Automatic     Custom
Codecs A	
Audio Codec For T12,T18,T20,T22,T26,T28,T32,T38 Disable Codecs	Enable Codecs
G723_53	PCMU
G723_63 G726-16	PCMA
G726-24 G726-32	G722
G726-40	
	2.2
Audio Codec For VP530 and VP-2009	
Disable Codecs	Enable Codecs
G722	»» PCMU
G723 GSM	PCMA 6729
AACLC iLBC	
	10.0
Video Codec For VP530 and VP-2009	
Disable Codecs	Enable Codecs
	>>> H264 H263
	mp4v-es
	**
Remote Phone Book 奈	
Phonebook UF	RL Phonebook Name
1:	
2:	
3:	
4:	
5:	
Firmware Download Server 🚿	
Enabled:	Disabled •
T12 Firmware Name:	
T18 Firmware Name:	
T20 Firmware Name:	
T22 Firmware Name:	
T26 Firmware Name:	
T28 Firmware Name:	
T32 Firmware Name:	
T38 Firmware Name:	
VP530 Firmware Name:	
VP-2009 Firmware Name:	
Server Type:	
Ocher Type.	ftp (• http
HTTP URL:	C ftp C http

Figure 3.3.4.1



### 3.3.4.2 Phone book

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

•	Phone Book 🗄	
	Create New Contact	
• 0	Contacts	
	No Contact Defined	
• 0	Deny List	
	No Deny List Defined	
	Upload Phonebook	
	Note: All the existing phonebooks of the IP phone would be deleted automatically if the phonebooks are configured in this way.	
	No Phonebook Uploaded	

Figure 3.3.4.2

### 1) Add Contact

#### ∙Туре

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.

	None 💌	Group:		•
Nick Name		Favorite	No -	
organization 0:		Title 1:		
Email <sup>1</sup> :		Birthday 0:		
First Name <sup>()</sup> :		Family Name		
Office Number:		Mobile Number:		
Home Number:				
Sub Number				
Sub Number		Sub Number:		↑Add Sub
Sub Name:		Sub Number:		↑Add Sub
		Sub Number:		↑Add Sub
Sub Name:		Sub Number:		↑Add Sub



Figure 3.3.4.2.1

## 2) Upload Phonebook

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

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

## 3.3.4.3 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

### · 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.

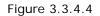
Add Phone				Х
General				
Enabled:	Yes 🔻			
MAC Address:	001565	Name:		
Manufacturer:	Yealink	Phone Type:	T28 🔻	]
Call Waiting:	Enabled 💌	Key As Send:	#	
Auto Redial:	Disabled -	Auto Answer:	Disabled -	
Phone Book:	Enabled -			
Line				
	xtension:	Label:	Line	Active:
Line2 E	xtension:	Label:	Line	e Active:
🗆 Line3 E	xtension:	Label:	Line	e Active:
Line4 E	xtension:	Label:	Line	Active:
Line5 E	xtension:	Label:	Line	Active:
Line6 E	xtension:	Label:	Line	e Active:

Figure 3.3.4.3

2) Audio codec

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

C Custom	
Disable Codecs	Enable Codecs
G723_53	»» PCMA
G723_63	PCMU
G726-16	G729
G726-24 G726-32	G722
G726-32 G726-40	
0720-40	←





## 3) Line keys settings

## Configure the DSS keys/Function Keys

emory Key Sett	ings 🛠				
Memory Key					
Key	Туре	)	Value	Line	Extension
DSS Key1	N/A	•		line1 💌	
DSS Key2	N/A	<b>-</b>		line1 👻	
DSS Key3	N/A	<b>•</b>		line1 💌	
DSS Key4	N/A	<b>-</b>		line1 💌	
DSS Key5	N/A	<b>-</b>		line1 🔻	
DSS Key6	N/A	-		line1 💌	
DSS Key7	N/A	-		line1 💌	
DSS Key8	N/A	-		line1 💌	
DSS Key9	N/A	-		line1 💌	
DSS Key10	N/A	-		line1 💌	
ine Keys Setting Line Keys Setting					
Key	Туре	Value	Label	Line	Extension
Line Key 1	N/A	•		Line1	-
Line Key 2	N/A	-		Line2	-
Line Key 3	N/A	•		Line3	-
Line Key 4	N/A	<b>•</b>		Line4	▼
Line Key 5	N/A	<b>•</b>		Line5	▼
Line Key 6	N/A	-		Line6	-

Figure 3.3.4.5

#### 3.3.4.3 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	<ul> <li>Key As Send:</li> </ul>	# ▼	
Auto Redial:	Disabled	Auto Answer:	Disabled <b>•</b>	
Phone Book:	Enabled	•		
Line				
Line1 E	xtension:	- Label:	Line Ac	ctive:
Audio Codec As Genera Custom	I			
I	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 3.3.4.6

## 3.3.4.4 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.

# 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 - Micros	oft Internet Explorer	
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites	<u>T</u> ools <u>H</u> elp			20 A
🕝 Back 🔹 🕥  🖹	🛃 🏠 🔎 Sean	ch 🥎 Favorites		
Address 🖉 http://192.168.5.18	6/cgi/WebCGI?1000			✓ → Go 維接
MyPBX	Embedded Hy	brid IP-PBX for S	mall Businesses	Logout
Status Monitor *	🕨 Manage IVR 4	þ		
Line Status	+ Create New	IVR	IVE	
Basic 🙁	Name	Number	Allow Dialing of Other Extensions	
Extensions	welcome	660	Yes	Se Edit 🗴 Delete
Trunks 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 Valescenal Settings				÷
🕘 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 IVR - 660						х
Number 🛈 :	660					
Name 🛈 :	660					
Prompt <sup>1</sup> :	aa	$\checkmark$	Custom Prom	<u>pts</u>		
Repeat Count 🛈 :	3 🔽					
Key Timeout 🛈 :	3 🔽					
	Allow Di	aling of Other Exten	sions			
🗌 🛈 Keypress E						
Key	/	Action		Destination		
(	)	No Action	~		~	
1	1	Connect to Extension	~	Extension 101		
2	2	Connect to Voicemail		Voicemail 101		
3	3	Connect to RingGroup		Ring Group ringgroup_de	et 🔽	
4	1	Connect to IVR		IVR 660		
E	5	Connect to Conference	e Room 🔽	Conference Room 640		
6	6	Connect to DISA		DISA test		
7	7	Connect to Queues		Queues 680		
8	3	Connect to Faxes	~	Faxes 101	~	
9	9	Dial by Name			*	
ŧ	ŧ	Hangup			*	
	*	No Action			~	
Timeout	)	No Action			~	
Invalid	)	No Action			~	
		V Save	e 🗙 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.

### **•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.



ew Queues	x
Queue Name	692
Queue Number	692
Queue Password 🛈 :	
Queue Agent Timeout 🛈 :	30
Queue Max Wait Time	1800
Queue Ring Strategy 🤨:	ringall
Agents	
Available Agents	Selected
500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP) 506(SIP) 507(SIP)	
Caller Position Announcements Announce Position Announce Hold Time	Yes V
Frequency	: 30 seconds 👻
Periodic Announcements Prompt Frequency	30 seconds and
Events	
Key	: End Call
Destination	
Failover-Destination	End Call
Destination	
Others Music On Hold	calmriver
Leave When Empty ④	
Join Empty 🕖	
Agent Announcement	
Join Announcement	
Retry	
Wrap-up time	: 30
🗸 Sav	/e X 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 💽 to record a new voice prompt Record 🏾 🄀 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.

Upload Music On Hold Prompt	X
WAV format: GSM 6.10 8kHz, Mor	no, 1Kb/sThe file size must not be greater than 1.8MB!
Choose file to upload ①:	浏览
Upload	X: Cancel

Figure 3.4.3.2

## 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.



MyPBX - Embedded Hybr	id IPPBX for S	mall Businesses - Micr	rosoft Internet Explorer	
<u>File Edit View Favorites</u>	Tools Help			27
🕝 Back 🔹 🔘 💌	el 🏠 🔎	Search 🔆 Favorites	🐵 🙆 · 💺 🖻	
Address 🖉 http://192.168.5.186	/cgi/WebCGI?100	)		✓ → Go 链接
MyPBX	Embedded	Hybrid IP-PBX for	Small Businesses	Logout
Status Monitor	🕨 Manage R	ling Groups 🗘		
Line Status	+ New Ri	ng Group	Manage Ring Groups	
Basic *	Number	Name	Members	
Extensions Trunks Outbound Routes Auto Provision	620	ringgroup_default	500(SIP)-501(SIP)-502(SIP)-503(SIP)-504(SIP)-505(S	Sł Edit 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				M
🕘 Done				🥥 Internet

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.

Ring Gr	oup Name 🔍: 625		
Ring Grou	ıp Number 🔍: 625		
	Strategy 💽 Ring all simultaneo	usly 💌	
Seconds to ring each	n member 🔍 : 60		
Ring Group members 🛈			
Available Extensions		Selected	
001(SIP)	*		
002(SIP)			
003(SIP) 004(SIP)			
005(SIP)			
006(SIP)	<u></u>		
006(SIP) 007(SIP)	<u></u>		
:006(SIP) :007(SIP) :008(SIP)			
006(SIP) 1007(SIP) 1008(SIP)			
:006(SIP) :007(SIP) :008(SIP)	End Call		
:006(SIP) :007(SIP) :008(SIP)		Extension 5001	×
5006(SIP) 5007(SIP) 5008(SIP) Destination If No Answer	End Call	Extension 5001 Voicemail 5001	< <
Destination If No Answer Destination If No Answer	<ul> <li>End Call</li> <li>Extension</li> </ul>		<ul> <li>S</li> <li>S</li> </ul>
5006(SIP) 5007(SIP) 5008(SIP) Destination If No Answer	<ul> <li>End Call</li> <li>Extension</li> <li>Voicemail</li> </ul>	Voicemail 5001	*

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 Hyb	rid IPPBX for Small Businesses	- Microsoft Internet Exp	lorer	
Eile Edit View Favorites	Tools Help			4
🕝 Back 🔹 🔘 💌	🧟 🟠 🔎 Search 🐈 Favor	ites 🚱 🔗 🎍	2	
Address 🖉 http://192.168.5.18	6/cgi/WebCGI?1000			💌 🄁 Go 链
MyPBX	Embedded Hybrid IP-PB)	X for Small Busines:	ses	Lagout
Status Monitor *	Manage Inbound Routes	φ		
Line Status	+ New Inbound Route	Manag	e Inbound Routes	
Basic *	Reute Name	DID Number	Caller ID Number	
Extensions Trunks Outbound Routes Auto Provision Inbound Call Control * IVR Queues Outbound Resolution	pstnin			Netete Kanala Ka
Custom Prompts Ring Groups Inbound Routes Blacklist Internal Settings (*) Options Business Hours Feature Codes				
SIP Settings IAX Settings				
🕘 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.

## •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>

How do I configure distinctive ring tones? Please refer to <u>APPENDIX E</u>. Currently distinctive ringtone can be compatible with Yealink and Snom phone.



#### 2) Member Trunks

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

#### 4) Business Days

Define where the calls will be routed during Business Days.

#### ·Office Days

Select one defined business days office days.

#### **·Office Hours Destination**

Configure where to route the incoming calls during office hours.

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

•Extension Route the incoming calls to a specific extension.

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

·IVR Route the incoming calls to a specific IVR.

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

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

•DISA Route the incoming calls to a specific DISA.

### •Queues Route the incoming calls to a specific Queue.

#### ·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.



reate New Inbound Rou	ıte		Х
General			
	Route Name 🛈 :		
	DID Number 🛈 :		
	Extension 🛈 :		
	Caller ID Number 🛈 :		
Dis	tinctive Ringtone 🛈 :		
	Enable Callback : No 🔽	Callback Settings	
Member Trunks	Trunks	Selected	
Available		Scietta	
456(FXO) pstn5(FXO) pstn6(FXO) GSM9(GSM) BriTrunk7(BRI) BriTrunk8(BRI)			
Business Days Office Days :	default 💟		
Destination :	End Call		
Non-office Hours			
Destination :	End Call	V	
<ul> <li>During Holidays</li> </ul>			
Holiday :			
Destination :	End Call		
Fax Detection			
Destination :	No Detect	V	
	🗸 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.



	id IPPBX for Small Businesses - Microsoft Internet Explorer	
<u>File Edit View Favorites</u>		
🕝 Back 🔹 🕥  🖌	🗟 🏠 🔎 Search 🤺 Favorites 🚱 🙆 + 💺 🚍	10 million (1997)
Address 🙆 http://192.168.5.186,	/cgi/WebCGI?1000	💌 🄁 Go 链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesse≤	Lögout
Status Monitor 🔹	Manage Blacklists 🌵	
Line Status	* New Blacklist Manage Blac	klists
Basic *	Biacklist	
Extensions	10000	× Delete
Trunks	10086	× Delate
Outbound Routes	1234567	X Délete
Auto Provision	13850050500	× Delete
Inbound Call Control 😒 IVR Queues Custom Prompts Ring Groups Inbound Routes <u>Blacklist</u>		
Internal Settings Options Business Hours Feature Codes SIP Settings IAX 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	Х
	Blacklist Number :	
	Type : Inbound	
L		

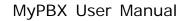
Figure 3.4.6.1

### ·Blacklist number

Enter the number you would like to block.

## ∙Туре

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.

#### ·Invalid Phone Number Prompt

Configure the prompt when the dialed phone number is invalid.

#### Busy Line Prompt

Configure the prompt when the dialed phone number is busy.

#### ·Dial Failure Prompt

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

2) Extension Preferences•User ExtensionsThe default value is 500 to 616

•**Ring Group Extensions** The default value is 620 to 629

•Paging Group Extensions The default value is 630 to 639

# ·Conference Extensions

The default value is 640 to 659

•IVR Extensions The default value is 660 to 679

•Queue Extensions The default value is 680 to 689



MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Log
Status Monitor	► General Preferences ♦	
Line Status		
	General Professional	
asić 🛞	Ring Timeout 9: 30 s	1
xtensions	MAX Gall Duration 0 : 6000 s	
unks utbound Routes	Maximum Concurrent Calls 0: 0	
utpound Routes hone Provisioning	Maximum Concurrent Calls 0 0 Music On Hold: cammer *	
none Provisioning		
ound Call Control 🛞		
R	HTTP Bind Port 0: 80	
ueues	FXO Mode 0 FCC +	
ustom Prompts ing Groups	Virtual Ring Back Tone 🛈 - No 🔹	
bound Routes	Distinctive Caller (D <sup>1</sup> ): Yes *	
acklist	Follow Me Prompt: Yes -	
	Invalid Phone Number Prompt	
ernal Settings	Busy Line Prompt 0	
olionu usiness Hours	Dial Failure Prompt	
ature Codes	Extension Proferences	1
P Settings		٩
x Settings	User Extensions - 500 to 616	
icemail Settings	Ring Group Extensions 620 to 629	
IS Settings	Paging Group Extensions - 630 to 639	
SA	Conference Extensions : 640 to 659	
onferences	IVR Extensions _ 660 to 679	
ging Groups IIS Settings		
N User Settings	Queue Extensions : 680 to 689	
allback Settings	Resulto Dofaata	

Figure 3.5.1

## 3.5.2 Business Hours

### 1) General

## •Enable or Disable Business Hours

### 2) Others

## ·Enable Office Closed Timing

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

## ·Disable Office closed timing

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

3) Business DaysYou can setup the business hours here.

4) Holidays

You can setup the holidays here.

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



<b>My</b> PBX	Embedded Hybrid IP-PBX for Small Businesses	^
Status Monitor (*)	Business Hours 💠	
Basic 🔹 Extensions Trunks	CEnable Business Hours CDisable Business Hours Others	E
Outbound Routes Auto Provision	*81 Enable Office Closed Timing *081 Disable Office Closed Timing	
Inbound Call Control I/R Queues Custom Prompts Ring Groups Inbound Routes Blacklist	Business Days           ◆ New Office Hours           Name         Detail           default         Mon 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         Name         Name	
Internal Settings (*) Options Business Hours	Note : Besides business days and office hours, public holiday could be set as well. Those holidays would be treated as non-office time.  New Holiday	
Feature Codes SIP Settings	Name Detail	
IAX Settings Voicemail Settings SMS Settings DISA	Save Cancel	

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 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			
	Cone Touch Record	*1	
	Check Extension Voicemail	*2	
	🔽 Voicemail Main Menu 🛈	*02	
	Attended Transfer	*3	
	Attended Transfer Timeout	15	s
	Blind Transfer	*03	
	Call Pickup	*4	
	Extension Pickup	*04	
	Intercom	*5	
	Normal Spy	*90	
	Vhisper Spy	*91	
	🔽 Barge Spy 🛈	*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 Number	*74 *074	
	Forward to Voicemail	*074	

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.

## Qualify

Send check alive packets to the sip provider.

### ·Remote Party ID

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

## Session -timers

Enable sesstion-timer mode, default: yes

•Sesstion-expires The max refresh interval

•Sesstion-minse The min refresh interval, which mustn't be less than 90s

## Sesstion-refresher

Choose sesstion-refersher, the default is Uas





SIP (Session	Initiation	Protocol)	Configuration	¢

0	
UDP Port	5060
Enable TCP Port 0:	
Enable TLS Port	
RTP Port Start:	
RTP Port End	12000
DTMF Mode	rfc2833 🔻
Max Registration/Subscription Time 🛈	3600
Min Registration/Subscription Time 🔍	60
Default Incoming/Outgoing Registration Time 0	120
Register Attempts 0	8
Register Timeout 0	20
Calling Channel Codec Priority	Yes 🔻
Video Support 🚺	Yes 🔻
Max Bit Rate	384 kb/s
DNS SRV Look Up	No 🔻
User Agent	
NAT	
Note: Configuration of this section is only	required when you use remote extensions.
Enable STUN:	
Enable STUN: STUN Address:	
STUN Port:	
External IP Address	
External Host 🛈 :	
External Refresh Interval	
Local Network Identification	
NAT Mode 🛈 :	ves 🔻
Allow RTP Re-invite	
Allow RTP Re-invite	yes •
Allow RTP Re-invite	Ves  Allowed Codecs
Allow RTP Re-invite Codecs  Codecs  SPEEX G722	Allowed Codecs
Allow RTP Re-invite  Codecs  Available Codecs  SPEEX G722 G726 ADPCM	yes            Allowed Codecs
Allow RTP Re-invite  Codecs  Available Codecs  SPEEX G722 G728 ADPCM G728A G728A	yes            Allowed Codecs
Allow RTP Re-invite Codecs  SPEEX G722 G726 ADPCM G729A MPE04	yes            Allowed Codecs
Allow RTP Re-invite Codecs	yes            Allowed Codecs
Allow RTP Re-invite Codecs  Available Codecs  SPEEX G722 G726 ADPCM G729A MPEG4  G.729 License Key :	yes           Allowed Codecs            u-law
Allow RTP Re-invite	yes           Allowed Codecs            u-law
Allow RTP Re-invite Codecs  Available Codecs  SPEEX G722 G725 G725 G728 G729A G729 G729 G729 G729 G729 G729 G729 G729	Allowed Codecs
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Allow RTP Re-invite ♥:         Codecs         Available Codecs         SPEEX G722 G725 ADPCM G729A MFE04	Allowed Codecs         u-law         a-bay         SM         H281         H283         H283         H284         mer your license key above.
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Allow RTP Re-invite ♥:         Available Codecs         SPEEX       G722         G725       ADPCM         G729       G728         ADPCM       G729A         G.729 License Key :       Image: Colspan="2">Image: Colspan="2" Image: Colspan="2">Image: Colspan="2" Image: Cols	Allowed Codecs     a-law   a-law   a-law   SSM   H283   H283   H284   Inter your license key above.   Cos SIP:   3   Cos SIP:   3   Cos Video:   4   From •
Allow RTP Re-invite ♥:         Available Codecs         SPEEX       G722         G725       ADPCM         G729A       G729A         G729A       G729A         G.729 License Key :       Image: Colspan="2">Note: If you would like to use G.729, please of the colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2"Colspan=	Allowed Codecs     a-law   a-law   a-law   SSM   H283   H283   H283   H284      Inter your license key above.   Cos SIP:   3   Cos Audio:   5   Cos Video:   4   From   INVITE
Allow RTP Re-invite ♥:         Codecs         SPEEX G722 G725 ADPCM G729 ADPCM         G726 ADPCM       G729 G729A         MPE04       □         G.729 License Key :       □         Note: If you would like to use G.729, please of Tos SUP;       C53         Tos SUP;       C53         Tos Video:       AF41         Advanced Settlings        From Field:         To Field:       180 Ringing:	Allowed Codecs         u-law         a-law         SS         H283         H283         H284         ee
Allow RTP Re-invite ♥:         Codecs         Available Codecs         SPEEX       G728         G728       G729         G729A       G729A         M#E04       G         G. 729 License Key :       Image: Colspan="2">Note: If you would like to use G.729, please of the colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Tos SIP:         Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2">Colspan="2"Co	Allowed Codecs     a-law   a-law   a-law   SSM   H283   H283   H284      Inter your license key above.   Cos SIP: 3   Cos Audio: 5   Cos Video: 4   From •   INVITE •   Send • trust
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Figure 3.5.4

## 3.5.5 IAX Settings

## 1) General

#### ·Bind 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 Time

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

#### •Max Registration 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.



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Address 🛃 http://192.168.5.10	86/cgi/webCG721000	链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	<
Status Monitor 🔹	LAX Settings the set of the set o	
Basic Extensions Trunks Outbound Routes Auto Provision MR Gueues Custom Prompts Ring Groups Inbound Routes Blacklist	Bind Port: \$559 Bandwidth: Low V Minimum Registration Time: 60 Maximum Registration Time: 1200 Codecs Allowed Codecs: V u-law V a-law V GSM SPEEX G726 ADPCM G729A H261 H263 H263P H264	
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings DISA	Save K Cancel	K
ej	🖉 Internet	-

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.

#### •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.

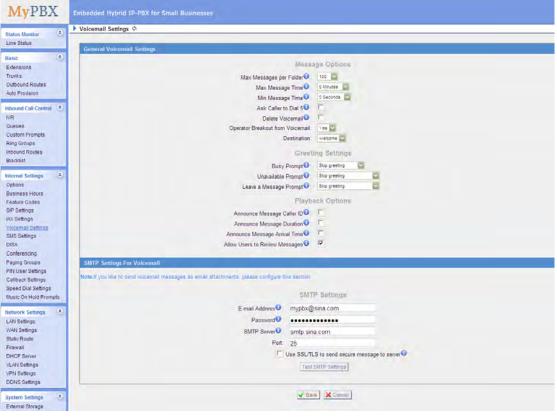


Figure 3.5.6

## 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.

ears@yeastar.com		8
Email Address:	† Add Emai	

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 Se	ettings	
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			🖞 Edit 🛛 🗶 Delete
Enable Email To SMS				
Email To SMS				
		Enable Country Code:		
		Country Code:		
		Receive mails every	60	
		Access Code		
Email Settings				
Email Settings lote: . If you want to use "SMS . If you configure the POP			ail server regularly. Once downloaded, the emails	will be deleted from the mail
Email Settings ote: If you want to use "SMS If you configure the POP			ail server regularly. Once downloaded, the emails chengeng530	will be deleted from the mail
Email Settings ote: If you want to use "SMS If you configure the POP		ownload emails from the ma		will be deleted from the mail
Email Settings ote: If you want to use "SMS If you configure the POP		ownload emails from the ma	chengeng530	will be deleted from the mail
Email Settings lote: . If you want to use "SMS" . If you configure the POP		Email Address Password	chengeng530 •••••• smtp.gmail.com	will be deleted from the mail
Email Settings lote: . If you want to use "SMS" . If you configure the POP		Email Address®: Password®: SMTP Server (SMTP):	chengeng530 •••••• smtp.gmail.com	will be deleted from the mail
Email Settings lote: . If you want to use "SMS" . If you configure the POP		Email Address : Password : SMTP Server (SMTP): SMTP Server Port: Receive Server (POP3):	chengeng530 •••••• smtp.gmail.com 587	will be deleted from the mail
imail Settings Email Settings Vote: I. If you want to use "SMS I 2. If you configure the POP server.		Email Address Email Address Password SMTP Server (SMTP): SMTP Server Port: Receive Server (POP3): Receive Server Port:	chengeng530 ••••••• smtp.gmail.com 587 pop.gmail.com	will be deleted from the mail
Email Settings lote: . If you want to use "SMS" . If you configure the POP		Email Address Email Address Password SMTP Server (SMTP): SMTP Server Port: Receive Server (POP3): Receive Server Port:	chengeng530 smtp.gmail.com 587 pop.gmail.com 995 Use SSL/TLS for security on this server(SMTP)	will be deleted from the mail

Figure 3.5.7.2



## 3.5.8 DI SA

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.

🗿 MyPBX - Embedded Hyb	rid IPPBX for Small Businesses - Microsoft Internet Explorer	
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🕝 Back 🔹 🔘  🖹	🖹 🚮 🔎 Search 😾 Favorites 🤣 🎯 - 😓 🧮	
Address 🙆 http://192.168.5.18	6/cgi/WebCGI?1000	💙 🄁 Go 链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Logaut
Status Monitor *	🕨 Manage DISA 💠	
Line Status	+ New DISA Manage DISA	
Basic 🔅 Extensions Trunks Outbound Routes Auto Provision	No DISA Defined	
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		2
Done		🔮 Internet

Figure 3.5.8.1

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



New DISA		X				
General						
Nan	ne 🛈 :					
PIN	#0:					
Response Timeo	Response Timeout 🛈 : 10					
Digit Timeo	ut 🛈 : 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	x
Extension	640
Admin 🛈 :	
PIN #	
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 Hy	brid IPPBX for Small Businesses - Microsoft Internet Explorer 🗧 🔲 🔀
File Edit View Favorites	
🔾 Back 🔹 🔘  🖹	📓 🏠 🔎 Search 👷 Favorites 🛛 😥 + 😓 📄
Address 🛃 http://192.168.5.	86/cg/WebCG171000 🔽 🄁 Go 🗱
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 Inbound Routes Blacklist	No Paging Groups Defined
Internal Settings A Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings DISA	
ě)	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.

ew Paging Group	х.
Paging Group Number 🛈 :	630
Duplex 🛈 :	
Paging Group members	
Available Extensions	Selected
500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP) 506(SIP) 507(SIP)	**  +- **
Sav	ve Cantel

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

NIS Ø							
		DNIS					
lote:							
1.PSTN trunk and GSM/UMTS trur	k do not need to set DID number.						
2.If you do not set the DID number,	all calls through this trunk will sho	w the DNIS name as Caller Name	9.				
Add DNIS							
Trunk Name	Enable DNIS	DNIS Name	DID Number				
pstn4(FXO)	off		-	🖌 Edit 🗴 Delete			
pstn5(FXO)	off		-	🖌 Edit 🗴 Delete			
pstn6(FXO)	off	-	-	🖌 Edit 🗴 Delete			
BriTrunk8(BRI)	off	-	-	🖌 Edit 🗴 Delete			
BriTrunk7(BRI)	off			🖌 Edit 🔰 Delete			

Figure 3.5.11

## 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.

- 1) Options
- ·Access Code

.Dial this code to access PIN.

## •Prompt for Entry

Prompt caller enter the PIN Number.

## ·Prompt for Entry Failure

Prompt the caller when an invalid PIN is entered.

2) PIN User

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 Hybr	id IPPBX for Small Businesses - Microso	ft Internet Explorer			
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🌀 Back 🔹 🔘  🖹	💈 🚮 🔎 Search 🤸 Favorites 🥝	🔗 🕹 🗟			
Address 🛃 http://192.168.5.186	i/cgi/WebCGI?1000				✓ → Go 链接
MyPBX	Embedded Hybrid IP-PBX for Sm	all Businesses			Logout
Status Monitor *	🕨 PIN User Settings 🕸				
Line Status	+ New PIN User	PIN User Se	ittings		
Basic *	Name	PIN List			
Trunks	PINNumber	0592-0596-0599		🐒 Edit 🗴 Delete	
Outbound Routes Auto Provision	Options	Aurora Codes	1.2.		
Inbound Call Control		Access Code: Prompt For Entry:			
IVR		Prompt For Entry Failure:	· · · · · · · · · · · · · · · · · · ·		
Queues					
Custom Prompts Ring Groups		Save	Cancel		
Inbound Routes					
Blacklist					
Internal Settings 🔹					
Options					
Business Hours Feature Codes					
SIP Settings					
IAX Settings					
Voicemail Settings DISA					-
Elone					M 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 Inbound Route	X
General	
Route Name 1 :	
DID Number 🛈 :	
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(FXO)	
Strip	
Prepend 🛈 : 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.



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🕒 Back 🔹 🔘  📓	👔 🏠 🔎 Search 🤺 Favorites 🤗 🎯 💐		
Address 🛃 http://192.168.5.18	6/cgi/WebCGI?1000		🖌 🌛 Go  链接
MyPBX	Embedded Hybrid IP-PBX for Small Busine		Logaut
Status Monitor	🕨 Speed Dial Settings 🌵		
Line Status	+ New Speed Dial	Speed Dial Settings	
Basic *	Source Number	Destination Number	
Trunks	123	5503305	X Delete
Outbound Routes Auto Provision	Öptions		
Inbound Call Control	Th	e prefix of speed dial 🔍 : *99	
INDURING CALCONING CO IVR Cueues Custom Prompts Ring Groups Inbound Routes Blacklist		Save KCancel	
Internal Settings  Options Business 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, The file size must not be greater than 1.8MB.

Upload Music On Hold Prompt	x
WAV format: GSM 6.10 8kHz, Mono, 1Kb/sThe file size must not be greater than 1.8MB!	
Choose file to upload ①: 浏览	
Upload 🔀 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 Hybrid IPPBX for Small Business	es - Microsoft Internet Explorer		
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Address 🛃 http://192.168.5.186/cgi/WebCGI?1000			🖌 🔁 Go 縦接
MyPBX Embedded Hybrid IP-F	BX for Small Businesses		Logaut
Status Monitor *			
Line Status			
Basic *	C.D.A.		
Extensions		No 💌	
Trunks	Enable SSH:	Yes 💙 Port: 8022	
Outbound Routes	Hostname:	MyPBX	
Auto Provision	IP Address:	192.168.5.186	
Inbound Call Control	Subnet Mask :	255.255.255.0	
IVR	Gateway :	192.168.5.1	
Queues Custom Prompts	Primary DNS :	192 168 5 1	
Ring Groups	Secondary DNS :		
Inbound Routes	IP Address2:		
Blacklist	Subnet Mask2:		
Internal Settings *	Subliti Maskz.		
Options	🖌 Savi	a 🗱 Cancel	
Business Hours			
Feature Codes SIP Settings			
IAX Settings			
Voicemail Settings			
DISA			

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).

WAN Settings I WAN Settings
-----------------------------

WAN Settings	
	Use WAN:
	CDHCP
	Static IP Address
	IP Address:
	Subnet Mask:
	Default Gateway:
	Primary DNS:
	Secondary DNS:
	C PPPoE
	User Name:
	Password:
	J Save M Cancel

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.

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 🔅					
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 Ro	oute Rules			
Destination 0 : Subnet Mask: Gateway: Metric 0 : 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.



		all. a common rule with the 'action' is 'accept',	Firewall has started successfully
Common Rules			
Add Rule			
		No Common Rules Defined	
Auto Defense			
+ Add Rule			
		No Auto Defense Rules Defined	
IP Blacklist			
Add Rule			IP Blacklist Manac
Port	Protocol	Rate	ii Didenisi indite
5060	UDP	120/60s	📢 Edit 🗴 Delete
5060	UDP	40/2s	🔊 Edit 🗴 Delete
8022	TCP	5/60s	🔊 Edit 🗴 Delete
Other Settings			
Disable Ping			

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.

#### ٠IP

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	×
Name 0 :	
Description •	
Protocol Port	
IPU:	1
MAC Address : Action :	Drop 🖌

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
Port0:		
Protocol Rate		
Rale .		
	Save X Cancel	

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•PortAuto 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 <sup>1</sup> :	
Protocol 🛈 : UDP 🔽	
IP Packets	
Time Interval 🛈 : 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.

MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Lagont
Status Monitor (A) Line Status	DHCP Server      DHCP Server	
Rasic & Extensions Trunks Outbound Routles Auto Provision	DHCP is not running F Enable Routers 192.168.5.1 Subnot Mask 255.255.255.0	
Inbound Call Control (2) NR Queues Custom Prompts Ring Groups Inbound Routes Blackist	Primary DNS 192.168.5.1 Secondary DNS : Allow IP Address From 192.168.5.2 To: 192.168.5.254 NTP Server	
Internal Settings Options Peature Codes SIP Settings Volcemail Settings USA Contreaming Paging Orkups PRI User Settings Calilack Settings Settings	<u>√</u> Sws <u>¥Curce</u>	

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.

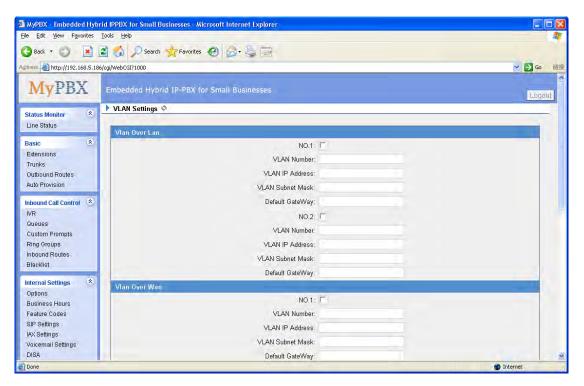


Figure 3.6.6

## 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 Xancel	

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 Φ	
P DDN3 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 of	dynamically.
You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com	
DDNS is not running	
Enable DDNS:	
DDNS Server: dyndns.org	
User Name:	
Password:	
Host Name:	
Save Cancel	
V Save A Califer	

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.

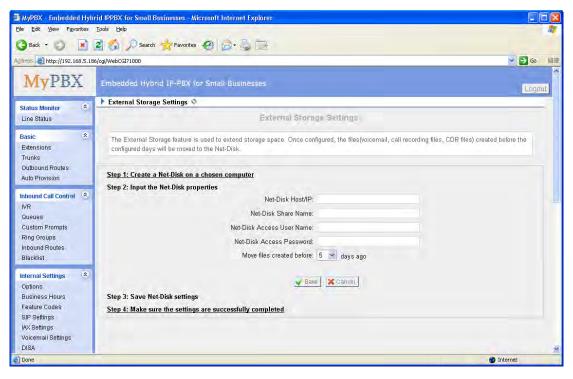


Figure 3.7.1

## 3.7.2 Password Settings

The default password is '**password**'. To change the password, enter the new password and click update. The system will then prompt you re-login using your new password.



MyPBX - Embedded Hyb	rid IPPBX for Small Businesses - Microsoft Internet Explorer	
<u>File Edit View Favorites</u>	Tools Help	A1
🔇 Back 🔹 🔘 🖹	🗈 🏠 🔎 Search 🧙 Favorites 😧 🎯 😼 🔄	
Address a http://192.168.5.18	6/cgi/WebCGI?1000	✓ → Go 链接
MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Logaut
Status Monitor 🏾 🖄 Line Status	Change Password Ø	
Basic *	Change Password Enter New Password: Retype New Password:	
Auto Provision Inbound Call Control & IVR Queues	Save Save	
Custom Prompts Ring Groups Inbound Routes Blacklist		
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings DISA		
Diax Done		S Internet

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.



MyPBX	Embedded Hybrid IP-PBX for Small Businesses
Status Monitor (R) Line Status	ት System Prompts Settings ፡፡ System Prompts Download
Basic (*) Extensions Trunks Outbound Routes Phone Provisioning	Prompts Download Note: Auto-detection is highly recommended, Brit if you prefer to download via HTTP or TFTP server, please contact the local dealer for the prompts.
Inbound Call Control (*) NR Oueues Custom Prompts Ring Groups Inbound Routes Blackist	Local Prompts: English Download Mode: Auto Detection Prompts: English Countral Countral
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings SMS Settings DISA Conferences	

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: 10 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 v: AM v

Figure 3.7.4



## 3.7.5 Backup and Restore

🗿 MyPBX - Embedded Hyb	Hybrid IPPBX for Small Businesses - Microsoft Internet Explorer	
Eile Edit View Pavorites	tes Iools Help	<b>A</b>
G Back + 🔘 🖹	🖹 📓 🏠 🔎 Search 👷 Favorites 🤪 🎯 - چ 🥃	
Address 🛃 http://192.168.5.18	.5.186/cgi/WebCGI?1000	✓ → Go 链接
MyPBX		Logout
Status Monitor *	🔊 🕑 Backup/Restore Configuration Information 🔅	
Line Status	Manage Configuration Backups	
Basic *	Create a New Backup Upload a Backup	a de la companya de la
Extensions Trunks	List Of Previous Configuration Backups:	
Outbound Routes	# Name Date Options	
Auto Provision	1 backup_2011mar28_1617352011mar28.tar Mon Mar 28 16:26:43 2011 Download from System Restore Previous Configu	Ination 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	Â	X
🛃 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.



	rid IPPBX for Small Businesses - Microsoft Internet Explorer
File Edit View Favorites	
🕝 Back 🔹 🔘  📓	🖹 🟠 🔎 Search 🨾 Favorites 🛛 🔗 + 🍃 📄
Address 🛃 http://192.168.5.18	performance of the system leave and report of the system will remain the system of
MyPBX	Embedded Hybrid IP-PBX for Small Businesses
Status Monitor 🔹	🕨 Reset and Reboot Options 🕸
Line Status	Reboot System
Basic *	Reboot System
Trunks Outbound Routes Auto Provision	
Inbound Call Control	
Queues Custom Prompts Ring Groups Inbound Routes Blacklist	Reset to Factory Defaults Warning:A factory reset will arase fail configuration detaion there yetern
Internal Settings Options Business Hours Feature Codes SIP Settings IAX Settings Voicemail Settings DISA	Hamaga in the system
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 <a href="http://www.yeastar.com/download/MyPBX/">http://www.yeastar.com/download/MyPBX/</a>

#### 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.



Ede       Set       Yew       Favorites       Yew       Yew       Set       Yew		d IPPBX for Small Businesses - Microsoft Internet Explorer	BX - Embedded Hyb	🗿 мург
Address @ http://192.166.5.186/cg/WebCdF1000         MyPBX       Embedded Hybrid 1P-PBX for Small Businesses         Status Monitor <ul> <li>Ine Blatus</li> <li>Plate System Firmware Φ</li> <li>Firmware Download Source:</li> <li>Firmware Download Source:</li> <li>Firmware Download Source:</li> <li>Firmware Download Source:</li> <li>HTTP URL  TFTP Server</li> <li>HTTP URL  TFTP Server</li> <li>HTTP URL:</li> <li>Reset Configuration to Factory  Defaults:</li> <li>Status</li> <li>Status</li> <li>Indound Routes</li> <li>Ring Groups</li> <li>Indound Routes</li> <li>Blackist</li> <li>Internal Settings  </li> <li>Options</li> <li>Business Hours</li> </ul>	1			1000
MyPBX       Embedded Hybrid IP-PBX for Small Businesses         Status Monitor       Ine Status         Line Status       Image: Configuration to Factory Configu		🛿 😚 🔎 Search 👷 Favorites 🧐 🎯 - 💺 🧱	ck • 🔘 🖹	G Bac
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Status Monifor   Line Status     Basic   Extensions   Trunks   Outbound Routes   Auto Provision     Inbound Call Control   MR   Queues   Custom Prompts   Ring Groups   Inbound Routes   Blacklist     Internal Settings   Options   Business Hours	Logaut	Embedded Hybrid IP-PBX for Small Businesses	<b>I</b> yPBX	M
Extensions Trunks Outbound Routes Auto Provision Inhound Call Control WR Oueves Custom Prompts Ring Oroups Inhound Routes Blacklist Internal Settings Options Busines Hours		) Update System Firmware לי		
Options Business Hours		HTTP URL     TFTP Server HTTP URL  Reset Configuration to Factory Defaults:	nsions ks ound Routes Provision and Call Control (*) ues om Prompts Groups und Routes	Exten Trunk Outbo Auto F Inbour IVR Queu Custo Ring Inbou
Feature Codes SIP Settings IAX Settings Voicemail Settings DISA	×		nns ness Hours ure Codes Settings iettings email Settings	Option Busin Featu SIP S IAX Se Voice

Figure 3.7.7

## 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.

tatus Monitor	Call Logs 🔅										
tatus Monitor	Search Condition										
	Start Date	11 Jan 1	1970	End Date	11 Jan 1970	Caller/Ca	llee:	Tru	nk: 🛤 🔛		
asic (2) Extensions	Duration			Billing Duration		Sta	itus: Al 🔛	Communication Ty	pe 🗚 📓 Start Se	arch	
Frunks Outbound Routes	Download the records	×0.000	e the record					Show 1-7 Total 7		View: 25	
auto Provision	Timé	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Type	1000	
bound Call Control	1970-01-11 02:02:59	500	640			14	14	ANSWERED	Internal	× Delete	
VR	1970-01-11 01:52:00	500	504			11	9	ANSWERED	Internal	× Detete	
Queues	1970-01-11 01:51:30	500	504			6	6	ANSWERED	Internal	X Delete	
Custom Prompts	1970-01-11 01:50:40	500	504			21	21	ANSWERED	Internal	X Detete	
Ring Groups	1970-01-11 01:50:30	500	504			6	6	ANSWERED	Internal	× Deleta	
nbound Routes Blacklist	1970-01-11 01:27:59	500	6604E		bizpbx	21	0	NO ANSWER	Outbound	× Detete	
iternal Settings (2)	1970-01-11 01:26:45	500	660			3	3	ANSWERED	Internal	X Delete	
Options Business Hours Feature Codes BIP Settings AX Settings Joicemail Settings BMS Settings									< <previous next="">&gt; P</previous>	age 1 /1 G	
DISA Conferencing Páging Gröups PIN User Settings											
Caliback Settings Speed Dial Settings											

Figure 3.8.1



## 3.8.2 System Logs

You can download and delete the system logs of MyPBX.

#### 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)

Download Selected I	Logs X Delete Selected	1	
		Logs	
	Name	Туре	
Γ	web.log	Web	
	Hardware Log 🛈 Debug Log 🕄	Enable Normal Log	Image: Finally with a state of the state

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 .





-	Statement and a local division of the local	the statement of the st	- 0 -
(a) * http://192.168	84139/cg/WebCGI/1000		n * ¤
MyPBX	Embedded Hybrid IP-PBX for Small Busin		Logout
Status Monitor (8) Line Status	System Information      General      Hardware Version:		
Basic (R) Extensions Trunks Outbound Routes Phone Provisioning	NyPBX-Scandard V4 Firmware Version: 2.16.0.56 Uptime:		
Inbound Call Control (2) IVR Queues Custom Prompts Ring Groups Inbound Routes Blacklist	5:38:59 up 3:17, load average: 1.00, LAN & Hostname: MyPEX MAC Address: 00:15:65:35:11:ae	, 1.00, 1.00	
Internal Settings (A) Options Business Hours Feature Codes SIP Settings Vocemail Settings SMS Settings OtiSA Conferences Paging Groups DNIS Settings	IP Addross: 192.168.4.139 Subnet Mask: 255.255.234.0 Gateway: 192.168.5.1 Primary DNS: 192.160.5.1 Sacondar DNS:		

Figure 3.8.3



# 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         Name       : 501         Caller ID       : 501							
Voicemail          Voicemail       Voicemail Access PIN #0: 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: Always       Image: Voicemail         Follow me:       Image: No answer       Transfer to:       Image: Number         Image: When Busy       Image: When Busy       Image: Number       Image: Number							
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 Explore	r	
Eile Edit View Favorites Iools Help		17
🔇 Back 🔹 🜍 💌 😰 🐔 🔎 Search 🤸	Favorites 🥝 🍰 🔜	
Address 🕘 http://192.168.5.186/		💌 芛 Go - 链接
MyPBX Embedded Hybrid IP	-PBX for Small Businesses	4
MyPBX Configuration	Panel	
7 Yeastar	User Login	
	User Name: 501	
	Password: ••• Language: English	
	Language. Engesti	
	Login Reset	
	Copyright @ 2011 Yeastar Technology, Co., Ltd. All Rights Reserved.	
E 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 🗢					
Voicemail Call Recordings	Folder: New  Viewing 1-1	X Delete Move to Folder N	ew 💌			
Voicemail Settings Settings		Caller ID	Date	Duration	Options	
Customer Feedback						

Copyright © 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	-PBX for Small Businesses			Lo
	Call Recordings O				
bicemail all Recordings bicemail Settings	× Delete	Caller ID	Time	Duration	Options
tlings	-				
Customer Feedback					
		Convright @ 20	11 Yeastar Technology Co., Ltd. A	I Rights Reserved	

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.



MyPBX	Embedded Hybrid IP-PBX for Small Businesses	Logout
	🕨 Volcemall Settings 🕸	
Voicemail Cail Recordings Voicemail Settings Settings	Voicemail Settings Voicemail Password Enter again to confirm Email Address 0: Enable Voicemail 0: P Enable Send Voicemail 0: C @ Save @ Cance	
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.

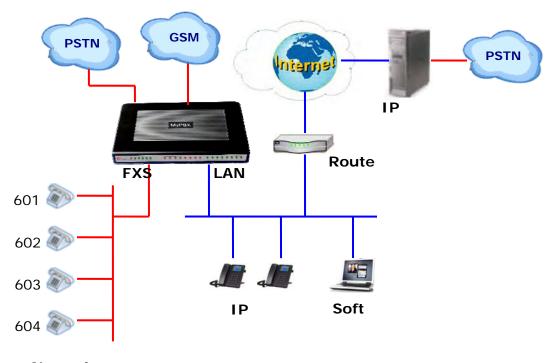
Image: Control of the second seco	
	Follow Me P No Answer Transfer To: Number When Busy Call Waiting Call Waiting Mobile Extension Number Cut

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Figure 4-6



# 5. Use MyPBX



Normal 5.1 אימוגע טענטטund 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.

•VolP 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.



### 5.1.1 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			x
Route Name 🛈 :	pstnout		
Dial Pattern 🛈 :	9.		
Strip 🛈 :	1 digits from	front	
Prepend these digits		before dialing	
Password:			
Member Extensions			
Available Extensions		Selected	
		00(SIP) 01(SIP) 02(SIP) 03(SIP) 04(SIP) 05(SIP)	
Member Trunks			
Available Trunks		Selected	
8032(SIP) 192.168.5.190(SPS) pstn6(FXO) GSM11(GSM) BriTrunk8(BRI) BriTrunk7(BRI)		istn1(FXO) istn2(FXO)	
	Save X Cance	al l	

Figure 5-1

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.



### 5.1.2 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		Х
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	
	Enable Outbound Proxy Server	
	UDP 🗹 Enable SRTP 🛈 : 🗖	
Caller ID 🛈 :		
DOD Setting		
DOD:	Associated Extension: 500 💌	∱Add DOD
	Save X Cancel	

Figure 5-2

### 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 Name0:	vopiout
Dial Pattern 🛈 :	0.
Strip 🛈 :	1 digits from front
Prepend these digits	before dialing
Password:	
Member Extensions	
Available Extensions	Selected
	<pre>&gt;&gt;&gt; 500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) €- 505(SIP) ««</pre>
Member Trunks Available Trunks	Selected
pstn1(FX0)           pstn2(FX0)           pstn6(FX0)           GSM11(GSM)           BrTrunk7(BRI)           BrTrunk8(BRI)           192.168.5.190(SPS)	>>>> voipprovider(SIP) 8032(SIP)
	Save X Cancel

Figure 5-3

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.

### 5.2 Incoming call

### 5.2.1 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→ Create New IVR

Edit IVR welcome					X
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	•		V	
3	No Action	•		V	
4	No Action	•		V	
5	No Action	•		V	
6	No Action	•		V	
7	No Action	•		V	
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 5-4

### 2. Add Inbound Routes

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





ate New Inbound Route			
General			
Route Nar	ne🛈 : allin		
DID Numb	er 🛈 :		
Extensio	on 🛈 :		
Caller ID Numb	er 🛈 :		
Distinctive Ringtor	ne 🛈 :		
Member Trunks			
Available Trunks		Selected	
<b></b>	»» ostn1(FX)		
	pstn2(FX0	o)	-
	→ pstn6(FX0 GSM11(G	SM)	=
	← BriTrunk7 BriTrunk8		
	8032(SIP) voipprovid		~
Ľ.	««		
During Office Hours			
	C End Call		
	C Extension	Extension 500	*
	C Voicemail	Voicemail 500	*
Destination:	IVR	IVR welcome	~
	C RingGroup	RingGroup ringgroup_defi	*
	C Conference Room	Conference Room 640	~
	C DISA	DISA	~
	C Queues	Queues	*
	C Faxes	Faxes 500	~
	Outbound Routes	Route Name pstnout	~
Outside Office Hours	C End Call		
		Extension 500	• •
	C Extension		*
Destination		Voicemail 500	<b>*</b>
Destination:	• IVR	IVR welcome	*
	C RingGroup	RingGroup ringgroup_defi	
	C Conference Room	Conference Room 640	*
	C DISA	DISA	*
	C Queues	Queues	*
	C Faxes	Faxes 500	*
	Outbound Routes	Route Name pstnout	~



Figure 5-5

# 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 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

share 🧊	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 <u>sharing and security</u> .
	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

The External Storage feature is used to extend storage space. files) created before the configured days will be moved to the N	
Step 1: Create a Net-Disk on a chosen computer	
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
V Save	Cancel
Step 3: Save Net-Disk settings	

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.

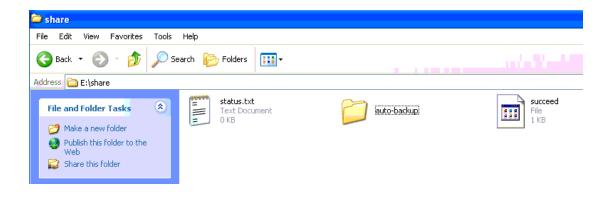




Figure B-3 Autobackup setting succeed

# APPENDIX C 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

and the second
required when using remote extensions.
Г
yeastar.3322.org
20
192.168.5.0/255.25
yes 🐱
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 Y 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

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 D 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.



	DHCP Server @
Status Monitor 🔹	
	DHCP Server
Basic 2	DHCP is not running
Extensions	₩ Enable
Outbound Routes	Routes 192 106 5.1
Auto Provision	
	Subnet Mask . 255 255 0
Inbound Call Control	Primary DNS 192 168 5 1
IVR Queues	Secondary DNS .
Custom Prompts	Allow IP Address From: 192168.5.2
Ring Groups	To: 192 168 5 254
Inbound Routes Blacklist	The second advector by
Constant of the local distance of the local	V Save X Cancel
Internal Settings 🙁	
Options	
Business Hours Feature Codes	
SIP Settings	
IAX Settings	
Volcemail Settings	
DISA	
Conferencing Paging Groups	
PIN User Settings	
Speed Dial Settings	
Music On Hold Prompts	
Network Settings	
LAN Settings	
WAN Settings	
Firewall DHCP Server	
DHCP Server	

Figure D-2

**Step3.** Configure phones on MyPBX auto-provision page.

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

	rid IPPBX for Small Businesses - Microsof	t Internet Explorer		
Ele Edit Yew Favoritès	Iools Help	A. R -		<i></i>
ddress @ http://192.168.5.18		0.32		✓ → Go 報
MyPBX	Embedded Hybrid IP-PBX for Sm	all Businesses		Lagout
Status Monitor	Auto Provision 💠			
Line Status		Auto	a Provision	
Basic (*) Extensions	Phone     Create New Phone			
Trunks Outbound Routes	MAC	Name	Extension	
Auta Provision	Upload a file			
Inbound Call Control	#	Name	Options	
IVR Queues Custom Prompts	IP phone     Scan Yeallink Phone			
Ring Groups Inbound Routes Blacklist		Mac Addre	oss List	
Internal Settings *				
Business Hours Feature Codes				
SIP Settings				
IAX Settings Voicemail Settings				
DISA				
Conferencing				o 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 Phon	e		x
General MAC Key As Send	: <b>00156511189E</b> : <b>#                                   </b>	Name: Rela Ca	II Waiting : Enabled
_ Line			
🔽 Line1	Extension: 500 💌	Label: 500	Line Active: 💌
🗖 Line2	Extension: 📉 👻	Label:	Line Active: 📁
🗖 Line3	Extension:	Label:	Line Active: 📁
🗖 Line4	Extension:	Label:	Line Active: 📕
🗖 Line5	Extension:	Label:	Line Active: 📕
🗆 Line6	Extension:	Label:	Line Active:
	V	Save X 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							
	Status	Account	Network	Phone	Contacts	Upgrade	Security
	IP Su De Pri Se Us	ICP atic IP Address Address bnet Mask fault Gateway mary DNS condary DNS	Port (WAN)	PC Port	Advanced	addres server Set th Mask, addres Secon manua PPPol	evice will acquire its IP is from the DHCP automatically. : IP Address IP Address, Subnet Default Router IP is, Primary DNS, dary DNS fields ally.

Figure D-5

Step5. Finish.



# APPENDIX E 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.

ink						<u> </u>	-
	Status	Account	Network Pl	none	Contacts	Upgrade	Security
	Preferer	nce   Features   DSS	Key   EXT Key   Voice	Ring	Tones   Dial P	lan   SMS	
	1	Internal Ringer Text	Family		0		
		Internal Ringer File	Ring1.wav	*	]	NOTE	
	2	Internal Ringer Text					
		Internal Ringer File	Ring2.wav	*	]		
	з	Internal Ringer Text					
		Internal Ringer File	Ring3.wav	*	]		
	4	Internal Ringer Text					
		Internal Ringer File	Ring4.wav	×			
	5	Internal Ringer Text					
		Internal Ringer File	Ring5.wav	*	]		
	6	Internal Ringer Text					
		Internal Ringer File	Ring6.wav	*	]		
	7	Internal Ringer Text					
		Internal Ringer File	Ring7.wav	¥	]		
	8	Internal Ringer Text			]		
		Internal Ringer File	Ring8.wav	*	]		
	9	Internal Ringer Text					
		Internal Ringer File	Ring1.wav	*	]		
	10	Internal Ringer Text					
		Internal Ringer File	Ring1.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 🛈 :			
Extension 🛈 :			
Caller ID Number 🛈 :			
Distinctive Ringtone 🛈 :	Family		
Member Trunks			
Available Trunks		Selected	
	>>> pstn1(FX0 pstn2(FX0) pstn6(FX0) GSM11(GS BriTrunk7(E BriTrunk8(E 8032(SIP) voipprovide	) ) M) BRI) BRI)	
During Office Hours	End Call		
	Extension	Extension 500	~
	Voicemail	Voicemail 500	~
Destination:	IVR	IVR welcome	*
C	RingGroup	RingGroup ringgroup_def:	~
	Conference Room	Conference Room 640	

Figure E-2

Step3. Finish.



# APPENDIX F 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).

Subject: 1	12345678
Insert: 🔒 🗸	Attachments 🛛 🎦 Office docs 🛛 🔄 Photos 🔻 🔁 From Bing 🔻 🤓 Emoticons
Tahoma	• 10 • B / U 墨 書 書 註 註 律 律 🚷 🚝 🔺

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).



Subject:	port:9-123456	78			
Insert: 🕻	Attachments	Sa Office	docs	🔄 Photos 🔻 🔁 From Bing 🔻 😁 Emoticons	
ahoma	• 10	• B	<u>U</u>	■ ≡ ≡ != != 律律 🤮 🚝 🔺	
ome to Y	eastar!				
Joine to 1	eastari				

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).

Subject:	500:987-	12345	578						
Insert:	Attachm	ents	Sa Offi	ce docs	E Pho	otos 🔹 🚺	From B	ing 🔹	😂 Emoticons
Tahoma	•	10	• B	ΙU	EE	<b>≡ j</b> Ξ	i≡ ∰ ₹	E 🔒 :	A= A
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).

Insert: 🛙 Attachments 🛛 🚼 Office docs 📰 Photos 🔻 🗔 From Bing 🔻 😅 Emoticon	
Insert: Attachments of Onice docs Photos (Onice docs Photos)	5
ahoma 🔹 10 🔹 🖪 🛛 💆 📰 🚍 🗮 🗮 🛱 🛃 📥	
come to Yeastar!	

Figure F-4



# **APPENDIX G 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 N		_	
Koute N:	ame 🔍 :		
DID Numbe	ər 🛈 :		
Extensi	on 🛈 :		
Caller ID Numbe	er 🔍 :		
Distinctive Rington	ne 🛈 :		
Enable Cal	lback : No 🗸	Callback Setting	s
ember Trunks🛈			
Available Trunks		Sel	ected
5503301 (FXO)	>>>>		
5503302 (FXO)			^
5503305 (FXO)	E →		
5503306 (FXO) pstn5 (FXO)			
	+		
pstn6 (FXO)			
pstn6 (FXO) pstn7 (FXO)			

Figure G-1

#### ·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.



Route Name	🛈 : E	3RI1		
DID Number	1 : 5	5503XXX		
Extension	<b>0</b> :			
Caller ID Number	<b>0</b> :			
Distinctive Ringtone	<b>0</b> :			
Enable Callba		No 🗸 Callback	Settings	
ember Trunks				
Available Trunks			Selected	
5503301 (FX0) 5503302 (FX0) 5503305 (FX0) 5503306 (FX0) pstn5 (FX0)	* E	>>>> >>>		
pstn6 (FXO) pstn7 (FXO)	-	+		-
pstn6 (FXO) pstn7 (FXO)	•	€		+
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)		88		-
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)				-
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	0	88	Extension 500	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	0	End Call	Extension 500 Voicemail 500	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO)	0000	End Call Extension		•
pstn6(FXO) pstn7(FXO) pstn8(FXO) uring Office Hours	00000	End Call Extension Voicemail	Voicemail 500	•
pstn6(FXO) pstn7(FXO) pstn8(FXO) uring Office Hours	000000000000000000000000000000000000000	End Call Extension Voicemail IVR	Voicemail 500 IVR welcome RingGroup NationalSe	· · · · · · · · · · · · · · · · · · ·
pstn6(FXO) pstn7(FXO) pstn8(FXO) uring Office Hours	000000000000000000000000000000000000000	End Call Extension Voicemail IVR RingGroup	Voicemail 500 IVR welcome RingGroup NationalSe	
pstn6(FXO) pstn7(FXO) pstn8(FXO) uring Office Hours		End Call Extension Voicemail IVR RingGroup Conference Room	Voicemail 500 IVR welcome RingGroup NationalSe Conference Room 640	•
pstn6 (FXO) pstn7 (FXO) pstn8 (FXO) uring Office Hours	000000000000000000000000000000000000000	End Call Extension Voicemail IVR RingGroup Conference Room DISA	Voicemail 500 IVR welcome RingGroup NationalSe Conference Room 640 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	x : No - Call	lback Settings	
Member Trunks Available Trunks		Selected	
5503301 (FXO)	>>>>		
5503302 (FXO)	and the second se		
5503302 (FXO) 5503305 (FXO) 5503306 (FXO)	≡ →		
5503305 (FX0) 5503306 (FX0) pstn5 (FX0)			
5503305 (FXO) 5503306 (FXO)	≡ →		

Figure G-3



# APPENDIX H 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 E	mbedded Hybrid IP-PBX for Small	Businesses	Log		
Status Monitor 🙁 👤	Manage Trunks $\phi$				
Line Status	Frunk List				
Basic 🛞	Irunk Name	Port/Hostname/IP			
Extensions	pstnl	1	S Edit		
Trunks Outbound Routes	pstn2	2	S Edit		
Auto Provision	pstn3	3	S Edit		
Contraction of the Contraction of the Contraction	pstn4	4	N Edit		

Figure H-1

2. Configure the IP Phone:

Memor	ry Key >> 🕜			
Кеу	Туре	Value	Line	Extension
DSS Key 1	BLF 👻	pstn1	Line 1	pstn1
DSS Key 2	BLF -	pstn2	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>