



For Firmware Version: V3.0

2013-12-11



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# 1. Introduction

# **Overview of the IPPBX**

ATCOM IPPBX is a SIP-based IP voice switch with a small embedded OS and rich GUI (Graphical User Interface), providing a powerful networking and corporate communication function. With it, users can quickly deploy an internal communication system for enterprise, as well as configure conveniently applications and value-added services on IP PBX via its GUI, to fit enterprise's own various demands.

Targeting for SOHO user and SMB market with an easy to use graphical interface, IP01 provides a cost-saving solution on their telecommunication/data needs. With IP01, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

# **Hardware Specifications**

CPU	400MHz Blackfin 532 Chip		
NAND Flash	256 M		
SDRAM	64 M		
Analog Port	1		
Network Interface	WAN		

Measurement and Weight

Inner box	100 * 100 * 28mm			
G.W./unit	0.36KG			
Carton MEAS	456 * 442 * 362 mm			
Units per Carton	21 units/ CTN			
G.W./CTN	17 KG/CTN			

#### **Function Features**

Voicemail	Authentication before call outbound
Voicemail to Email	User WEB portal
Blind/Attended Transfer	Blacklist
Call Forward	Call Detail Records(CDR)
Call Parking	Conference
Do not Disturb (DND)	Ring Group
Group / Directed Pickup	Call Queue
Call Recording	Call Back
Call Waiting	IVR
Call Routing	Intercom/Paging
Caller ID	IP Restriction
BLF Support	Firewalls
Music on Hold	DDNS
Video Call	DHCP Server
PPPoE	VLAN

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#### **IPPBX IP01 User Manual**

Port Forward	VPN client
External Storage	Auto Provisioning
Storage Quota Privilege	Custom Dialplan
Auto call recording	Integrated built-in packet capture tools

# 2. Connection and Change IP Address of IPPBX

# 2.1 Connection

The default IP address of IPPBX is: **WAN: 192.168.1.100/255.255.255.0** 

The network scenario should be like below:



1) Connect IPPBX to your PC directly or through switch.

Make sure IP address of your PC is in network 192.168.1.0/255.255.255.0, if not, you need to appoint an IP address for your PC, for example, 192.168.1.3

2) Login IPPBX as administrator via WEB GUI

## User: admin

### Password: atcom

**NOTE**: Language option will be accomplished in future versions, currently, only English is supported. So it's not selectable now.

### 3) Go to Network Settings ->WAN.

Re-set IP of WAN port.



♦ ③ 192. 168. 1. 100			gle «Ctrl+K>	P 🛛 -	<u> </u>	* -
						^
	ATCOM PBX Login					
	L UserName					
	Password					
	🕇 Language	v				
	Login					
		Contact us				
	©2013 ATCOM All Rights Reserved					

# 2.2 WAN Settings

There are two ways to set an IP address to WAN port: DHCP, Static IP.

# DHCP

IPPBX will obtain an IP address automatically from DHCP server when rebooting. It's not recommended to choose this option unless there is a reserved IP for IPPBX in DHCP server so that IPPBX can keep the same IP all the time.

## Static IP

Set an IP address manually according to the real network environment. If IPPBX is behind a router, the gateway is usually set to the IP of the router.

alcom	System Status	PBX Settings	Network Settings	System Settings	Reports	<b>X</b> Logout
Web Access	WAN Setting					
	Орнср					
> WAN	O Static IP					
Firewall	IP Address :		192.16	8.1.100		
DDNS	Subnet Mask :		255.25	5.255.0		
	Default Gateway :		192.16	8.1.1		
VLAN	Primary DNS :		8.8.8.8	;		
VPN	Secondary DNS :		8.8.4.4			
	Save					

# 2.3 System Status

atcom	System Status PBX Settings Network S		Settings	<b>O</b> System Settings	Reports	8 Logout	
> General	General						
Trunk Status	Product Model :	IP01		Application V	ersion : V3.0		
	Kernel Version :	V3.0		System Up T	ime : 0 days (	0 hours 0 minutes 32 seco	onds
Extension Status	System Current Time : Wed Dec 11 15:23:38 2013						
Network				WAN Primary DNS :	8.8.8.8		
	WAN Connection Type :	STATIC			WAN Secondary DNS :	8.8.4.4	
	WAN Mac Address :	80:82:87:00:D9:CB			WAN Gateway :	192.168.1.254	
	WAN IP Address :	192.168.1.159					
	WAN Subnet Mask :	255.255.255.0					
	Peripheral						
	Port 1 :	unplugged					

# 1) General

#### **Product Model**

Show the model of this IPPBX Firmware Version Show the firmware version System Uptime Show the time how long the system has been running System Current Time Show the current time 2) Network

Show the network setting of IPPBX

# 3) Peripheral

Show what kinds of / how many modules are detected.

# 3. Create local extensions and make interior calls

There are two kinds of extensions in IPPBX: FXS extensions and SIP extensions.



# 3.1 FXS extensions

System Status PBX Setti		PBX Settings	() Network Settings	System Settin	gs Reports Logo		
> Extensions	FXS Extensions :						
Trunks CRefresh							
	Port E	Extension F	Full Name	CallerID	Operation		
Inbound Routes	1 <u>1</u> 6	3101		6101	Edit		

It needs support of FXS module, the module installed in IP01 can be:

#### AX110S: single FXS port

Analog phone is available to make calls once connected to the corresponding FXS port, IPPBX configures FXS extension automatically when FXS module is detected and they can't be deleted.

**NOTE**: FXS extension number range is defined in **PBX Settings** -> **Options** -> **Extension Preference**, changing it can change the FXS extension number.

#### 1) General

# Extension

Extension number, i.e. 6101, it is associated with this particular user / phone.

#### Port

The analog port bound with extension.

#### Name

A character-based name for this extension, i.e. 'Bob Jones'.

#### Caller ID

CID showed in the other's phone during a call, default is Extension.

#### 2) Voice Mail

#### Enable Voice Mail

Check this option to enable voice mail account for the extension. Enabled by default.

#### Voice Mail Access PIN Code

Password for accessing this voice mail account, default is 123456. It's also the password for extension to login his administration web page.

#### 3) Mail Setting

#### **Enable Sending Voice Mail**

Check this option to enable PBX send new voicemail to Email address below as an attachment.

#### **Email Address**

The Email address that new voicemail will be send to when Enable Sending Voice Mail is enabled and PBX settings -> SMTP settings is right set.

#### 4) Volume Settings



#### Rxgain

Adjust the volume sent to FXS extension. There are 3 options: 20%,60%,100%.

### Txgain

Adjust the volume sent out by FXS extension. There are 3 options: 20%,60%,100%.

### 5) Flash

### **Hook Flash Detection Time**

Sets the amount of time, in milliseconds, that the hook-flash must remain depressed in order for IPPBX to consider such an event a valid flash event. The default value of it is 1250 ms and it can be configured in 1 ms increments.

### Sequential Hook Flash Interval

Sets the amount of time, in milliseconds, that must have passed since the last hook-flash event received by IPPBX before it will recognize a second event. If a second event occurs in less time than defined in here, then IPPBX will ignore the event. The default value is 750 ms, and it can be configured in 1 ms increments.

### 6) Follow Me

Follow me is a feature to let an incoming call to a called party to be redirected to a third party, the third party can be a voicemail box, ring group, mobile telephone and so on.

When callee is No Answer / Busy / Unreachable, incoming calls will go to voicemail by default, if voicemail is disabled, call will be hung up.

Edit FXS Extension : 1	⊗
General	^
Extension: 6101 Port: 1	
Name : Caller ID : 6101	
Voice Mail	
Voice Mail 4 Voice Mail Access PIN Code : 123456	
Mail Setting	
Enable Sending Voice Mail 9 Email Address :	=
Volume Settings	
Rxgain :         60%           60%	
Flash	
Hook Flash Duration Time : 1250	
Sequential Hook Flash Interval : 750	
Follow Me	
Call Forward : ① Always 🗹 When no answer 🗹 When busy Forward To : ③ Voice Mail 〇 Number :	_
Other Options	
Pickup Group : 0 Call Waiting <b>1</b> Ring Out : 30 Use Web Interface <b>1</b>	
Storage Quota Privilege : Basic V D DND Call Recording Call Recording	
Submit Cancel	~

### 7) Other Options

#### **Pickup Group**

Allows extension to answer someone else's telephone call by dialing the group call pickup code (defined in **PBX Settings->Feature Codes->General**), the two extensions must be in a same pickup group.

IPPBX supports 10 pickup groups: 0-10, **None** means the extension belongs to none pickup group, extensions in group None can't pick up others' ring call and also can't be picked up by others.

#### **Call Waiting**

Check this option to enable the Call Waiting capability for this extension. Then the extension can answer a new call when it is already on the line. If this Option is checked, the follow me option "When busy" will be unavailable.

#### **Ring Out**

Set the ring timeout for this extension. IPPBX will stop ringing the extension if the time is up and there is still no answer.

#### **Use Web Interface**

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When checked, user can login the administration web page of this extension with extension number and voice mail pin code as username and password.

#### Storage Quota Privilege

Set capacity of disk space for this extension to store voicemail in IPPBX.

Restricted: 1 M Basic: 2 M Regular: 3 M Privileged: 4 M Super: 5 M

**NOTE**: If you set **System Settings** -> **External Storage** and set Move Files Created Before 0 days ago. This storage limitation will be of no use.

#### DND

Do not disturb, once checked, the extension will be unavailable.

#### Call Recording

Check this option to record all interior calls made by this extension, user can check the recordings on the personal web portal and with record account (password is the same with admin) too.

# 3.2 SIP extensions

SIP extension is a SIP Account that allows an IP Phone or an IP SoftPhone client to register on IPPBX. It can be created / modified / deleted one by one or in batch.

#### 3.2.1 Add new extensions

a. Add single extension

Click Create New Extension to add an extension

**NOTE**: SIP extension number range is defined in **PBX Settings** -> **Options** -> **Extension Preference**, changing it can create extensions in others number range.

#### 1) General

#### Туре

SIP

#### Name

A character-based name for this extension, i.e. 'Bob Jones'

#### Caller ID

CID showed in the other's phone during a call, default is Extension.

#### Extension

Extension number, i.e. 6000, it is associated with this particular user / phone.

#### Password

Authentication for SIP phone to register and make calls.



# 2) SIP Setting

### NAT

Try this setting when IPPBX is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports.

#### Can Reinvite

By default, IPPBX will route the media streams from SIP endpoints through itself. Enabling this option causes IPPBX to attempt to negotiate the endpoints to route the media stream directly, bypassing IPPBX. It is not always possible for IPPBX to negotiate endpoint-to-endpoint media routing.

#### **DTMF Mode**

Select DTMF sending mode, there are three modes: **rfc2833**, **inband**, **info**. The DTMF setting in here should be as same as that in SIP phone, otherwise IPPBX will not detect the users' input correctly during a call.

Auto means IPPBX will match anyone of them according to the setting of SIP phone.

#### Transport

The transplant protocol type for VoIP data package, default is UDP. Please make sure TCP is enabled in PBX Settings -> SIP Settings before using TCP.

#### **SIP Quality**

Check this option to force keep alive.

#### 3) Voice Mail

#### **Enable Voice Mail**

Check this option to enable voice mail account for the extension. Enabled by default.

#### Voice Mail Access PIN Code

Password for accessing this voice mail account, default is the extension number. It's also the password for extension to login his administration web page.

#### 4) Mail Setting

#### Enable Sending Voice Mail

Check this option to enable PBX send new voicemail to Email address below as an attachment.

#### **Email Address**

The Email address that new voicemail will be send to when Enable Sending Voice Mail is enabled and PBX settings -> SMTP settings is right set.

#### 5) Follow Me

Follow me is a feature to let an incoming call to a called party to be redirected to a third party, the third party can be a voicemail box, ring group, mobile telephone and so on.

When callee is No Answer / Busy / Unreachable, incoming calls will go to voicemail by default, if voicemail is disabled, call will be hung up.

#### 6) Other Options



#### **Pickup Group**

Allows extension to answer someone else's telephone call by dialing the group call pickup code (defined in **PBX Settings->Feature Codes->General**), the two extensions must be in a same pickup group.

IPPBX supports 10 pickup groups: 0-10, **None** means the extension belongs to none pickup group, extensions in group None can't pick up others' ring call and also can't be picked up by others.

#### **Call Waiting**

Check this option to enable the Call Waiting capability for this extension. Then the extension can answer a new call when it is already on the line. It also needs the call waiting support of IP phone. If this Option is checked, the follow me option "When busy" will be unavailable.

#### **Ring Out**

Set the ring timeout for this extension. IPPBX will stop ringing the extension if the time is up and there is still no answer.

#### **Use Web Interface**

When checked, user can login the administration web page of this extension with extension number and voice mail pin code as username and password.

Edit Voip Extension : 6000	
General	
Type : SP V	Name : 6000 Caller ID : 6000
Extension : 6000	Password : pw6000
Sip Settings	
NAT : 🗖 🕒	Can Reinvite : DTMF Mode : rfc2833 V
Transport : UDP 🗸 🔮	SIP Qualify 🤑
Voice Mail	
☑ Enable Voice Mail 🏮	Voice Mail Access PIN Code : 6000
Mail Setting	
Enable Sending Voice Mail	Email Address :
Follow Me	
Call Forward :      Always      When no	answer 🗹 When busy Forward To : <ul> <li>Voice Mail</li> <li>Number :</li> </ul>
Other Options	
Pickup Group : 0 🗸 🗸 🕽	Call Waiting Ring Out : 30 Use Web Interface
Storage Quota Privilege : Basic 🗸	DND DND Call Recording Call Recording



### Storage Quota Privilege

Set capacity of disk space for this extension to store voicemail.

Restricted: 1 M

Basic: 2 M

Regular: 3 M

Privileged: 4 M

Super: 5 M

**NOTE**: If you have set **System Settings** -> **External Storage** and set Move Files Created Before 0 days ago. This storage limitation will be of no use.

#### DND

Do not disturb, once checked, the extension will be unavailable.

#### **Call Recording**

Check this option to record all interior calls made by this extension, user can check the recordings on the personal web portal and with record account (password is the same with admin) too.

### Advanced Configuration

Preferred Codec		
First : a-law 🗸	Second : u-law	Third : GSM 🗸
Fourth : None 🗸	Fifth : None 🗸	Sixth : None 🗸
IP Restriction		
Enable IP Restriction		
Permitted Rule 1 :	(ip address/subnet mask) 🚺	
Permitted Rule 2 :	(ip address/subnet mask)	
Permitted Rule 3 :	(ip address/subnet mask)	
Permitted Rule 4 :	(ip address/subnet mask)	

#### **Preferred Codec**

Set the allowed codec and priority for SIP phone. The options are below:

Audio: A-law, U-law, GSM, SPEEX, G726, G722, ADPCM, G729

Video: H261, H263, H263P, H264

**NOTE**: There must be at least one same audio/video codec chose in IPPBX extension settings and SIP phone codec settings, otherwise, It's impossible to make audio/video calls between IPPBX and SIP phone.

### **IP** Restriction

Enable it to permit trusted IP/network register to this extension number. This is an useful way to improve the security of IPPBX.



Set trusted IP: xx.xx.xx/255.255.255.255, for example: 192.168.1.160/255.255.255.255 Set trusted network: xx.xx.xx/subnet mask, for example: 192.168.1.0/255.255.255.0

b. Click Add Multiple Extensions to add multiple extensions.

Add Multiple Extension	ons		8
Create 10 💙	Type SIP/IAX 🗸	Extensions starting from	
	🗸 Create 🎽	Cancel	

# 3.2.2 Modify extensions

- a. Click Edit behind an extension to edit the extension.
- b. Choose one or more extensions, click

Modify Selected Extensions to modify them simultaneously.

# 3.3 Register onto IPPBX with your IP phone

🕙 ATCON IP-Phone Configuration System - Nozilla Firefox		_	PX	
Eile Edit Yiew Higtory Bookmarks Tools Help				1
TATCOM PBX Configuration System × C ATCOM IP-Phone Configuration Sys × +				
🔄 🕘 192. 168. 1. 202/user. asp	- <b>-</b> 1	<b>F</b>	*	
			^	•
atcom ATCOMIPPHONE Device Configuration				
Device Configuration				
System Status 🔗 Account / Account 3 adr	lmin			
Network 😣				
Account				
Account 1				
Account 2 Enable : Yes 💌 Extension created in APBX				
Account 3 Display Name : 6000 User ID : 6000				
Phone Settings Authoriticate D : 6000 Paceword : Paceword set in	APBX			
Indata for this extension	i on			
Opdate         V         SIP Server :         192.168.1.100         SIP Port :         5060				
Call Log Vise Outbound Proxy: No Vise Outbound Proxy:				
Outbound Proxy Server : Outbound Proxy Port : 5060				
Register Expires : 30 Subscribe Expires : 3600				
Transport Type : UDP v SIP 100Rel Require : No v				
Must be the same as that in AFBX for this extension				
Codec Configuration				
Prefer Codec : G726 💌 User Prefer Codec Only : No 💌				
DTMF Tx Method : RFC2833 💌				
Call Feature Setting				
Massage Waiting · No · Default Ring · 1			~	,

Login the web administration page of IP phone and set the account information.

After successfully register with 6000 and 6001, you can make interior calls among 6000, 6001, 6101(FXS) now.

# **3.4 Extensions Status**

This page is used to check the extensions status.



Idle: The extension is registered and idle.

Busy: The extension is on the phone.

Ringing: The extension is ringing.

Hold: The extension is on hold

Unavailable: The extension is not registered and unreachable.

**NOTE**: If this page response slowly, please be patient to wait the output before check other pages. Otherwise, other pages cannot be displayed correctly since IPPBX is accessing database while status checking, and database is locked for other pages' request.

atcom	System Status	<b>2</b> + PBX Settings	Network Settings S	<b>X</b> ystem Settings	Reports Logout
General	Idle	Busy	Ringing	Hold	Unavailable
	Status	Extension	Туре	New Messages	Old Messages
Trunk Status	1 Unavailable	6000	SIP	0	0
<ul> <li>Extension Status</li> </ul>	2 Idle	6001	IAX	0	0
	3 Unavailable	6002	SIP/IAX	0	0
	4 Idle	6003	SIP	0	0
	5 Ringing	6006	SIP	0	0
	6 🕞 Idle	6007	SIP	0	0
	7 Unavailable	6008	IAX	0	0
	8 🔲 Unavailable	6009	SIP/IAX	0	0
	9 Ringing	6101	FXS	0	0
	10 Idle	6102	FXS	0	0
	10 🖌 🚺 🖣 Page	1 of 1 🕨 🕅 💭			Displaying 1 to 10 of 10 items
	<b>1</b>				Refresh

# 3.5 Feature Codes

1) General

### **Call Recording**

Record a call while in the call.



Dial Call Recording Code to begin recording and dial it again to stop recording during a call.

#### **Checking Voicemail**

Users can check their Voicemail by dialing this code on their phone.

#### **Attended Transfer**

Routed a call to a third party only if the third party answers the call. The call flow should be like below:

- 1. Phone A call B, B answers the call.
- 2. B presses feature code(\*3) and C's number to transfer the call to C
- 3. If C answers B's call, B can talk to C and A is on hold
- 4. If Phone B hangs up, A will talk to C, transfer is successful.
- 4' If Phone C hangs up, B connects back to A, transfer is failed

### **Blind Transfer**

Blind transfer is when a call is routed to a third party, the original call is ended, and no check is made to determine

whether the transferred call is answered or if the number is busy.

The call flow should be like below:

- 1. Phone A call B, B answers the call.
- 2. B presses feature code(\*03) and C's number to transfer the call to C

#### **Group Call Pickup**

Pick up a ring call for other extensions in the same pickup group.

The call flow should be like below:

- 1. C calls A, phone A is ring, but A is not at his/her seat.
- 2. Extension A and B are in the same pick up group, B can dial Group Call Pickup code to pick up the ring call, and talk to C.

#### **Direct Call Pickup**

Pick up a ring call for an appointed extension. The call flow should be like below:

- 1. C calls A, phone A is ring, but A is not at his/her seat.
- 2. B can dial Direct Call Pickup code + A's extension number to pick up the ring call, and talk to C.

#### Intercom

Connect directly to a specified phone.

The call flow should be like below:

- 1. A dial Intercom code + B's extension nubmer.
- 2. If Phone B supports page/intercom, it will answer the call automatically.

### 2) Call Park

It allows a person to park a call on IPPBX and continue the conversation from any other telephone set. The call flow should be like below:

- 1. A and B are on the conversation.
- 2. A dial call park code (e.g. \*6), PBX will tell A a park extension (e.g. 701) and then hang up the call. B is parked on PBX.
- 3. C dial park extension: 701, PBX will bridge C and B.

### 3) Call Forward

Users can configure their follow me settings via their phones.

#### **Reset to Defaults**

Reset follow me settings by dialing \*70 (default code, can be changed). After dialing in, PBX will prompt a "beep", then the setting is completed and the call will be hung up.

**NOTE**: Default Follow Me settings are as below:

Always: Disabled 
When no answer: Enabled 
When busy: Enabled

Forward to: Voice Mail

elecon	System Status PBX	Settings	() Network Setti	ings System Sett	ings	Reports	<b>X</b> Logout
Extensions	General						
Trunks	Call Recording *1			Checking Voicemail	*2		
Inbound Routes	Attended Transfer *3			Blind Transfer	*03		
Outbound Routes						0	
> Feature Codes	Group Call Pickup			Direct Call Pickup	*04		
IVR	Intercom *5	9					
Hunt/Ring Group	Call Park						
Conference		*6					
Callback	Call Park						
Paging/Intercom	Extension Range to Park Calls :	701-720					
Time Intervals	Park Time Before Recalled(Second)	: 60					
Queue	Call Forward						
DISA	Reset to Default	*70					
Blacklist			•				
Options	Enable Unconditional Call Forward	*71		Cancel Unconditional	Call Forward	*071	
SIP Settings	Enable Call Forward On Busy	*72		Cancel Call Forward (	On Busy	*072	
IAX Settings	Enable Call Forward On No-Answer	*73		Cancel Call Forward (	On No-Answe	*073	
SMTP Settings	Call Forward to Number	*74		Call Forward to Voice	Mail	*074	
Music On Hold			<b>~</b>				
Custom Prompts	Enable Do Not Disturb	*75		Cancel Do Not Disturt	)	*075	
Custom Dialplan			📃 💾 Sa	ive			

# **Enable/Cancel Unconditional Call Forward**

Enable/Disable call forward Always function.

#### **Enable/Cancel Call Forward On Busy**

Enable/Disable call forward When Busy function.

#### **Enable/Cancel Call Forward On No-Answer**

Enable/Disable call forward When No Answer function.

#### **Call Forward to Number**

Set the destination for call forward to number by dialing \*74 (default code, can be changed), if the number is not set yet, dial \*74+number to set it.

#### **Call Forward to Voice Mail**



Set the destination for call forward to voicemail.

### **Enable/Cancel Do Not Disturb**

Enable/Disable Do Not Disturb function

# 3.6 SMTP Settings

### 1) Voice Mail to Email Setting

#### **Email Address**

The Sender Email Address IPPBX used to send voicemail.

#### Password

The password for above Email Address/Account.

#### **SMTP Server**

SMTP server that above email address/account is located in.

#### Port

Port for SMTP server, for example: Gmail server use port 465 to send / receive email.

#### Use SSL/TLS to send secure message to server

Some servers need to authenticate sender before sending email, then the box should be checked.

#### **Test SMTP Settings**

Check whether the SMTP setup is OK. PBX will send an email to the test email address using above SMTP setting information. If the test failed, please check that information and network connection.

alteon	System Status	PBX Settings	Network S	) Settings	System Settings	Repor	rts Logout
Extensions	SMTP Settings						
Trunks	Email Address :	bty@atcomemail.com		Password :		•••••	
Inbound Routes	SMTP Server :	smtp@gmail.com		Port :		465	
Outbound Routes	Min Messages Time :	5		Max Messa	ges Time :	120	
Feature Codes		100		Say CID :		No	
IVR	-						
Hunt/Ring Group	Say Duration :	No 🗸 🕘		Envelope :		No	× <b>(</b> )
Conference	Review :	Yes 🗸 🗸		Delete Mes	sage After Notification:	No	~ <b>(</b> )
Callback	Ask Caller to Dial 5 🌗						
Paging/Intercom	Use SSL/TLS to send	secure message to server 🅕					
Time Intervals	Save Test	SMTP Settings					

NOTE: After SMTP setting, please set Email address for each extension to achieve Voicemail to Email function.

#### 2) Voice Mail Setting

#### Max Messages

Queue

This limits the number of messages in a voicemail folder. The maximum value is 9999 (hard coded) and the default 100. When a mailbox has more than this number of messages in it, new messages can not be recorded and "voice mail box is full" is played to the caller.

#### Max Messages Time



This defines the maximum amount of time in seconds of an incoming message. Use this when there are many users and disk space is limited. The default value is 120 (2 minutes), 0 means there will be no maximum time limit enforced.

### Min Messages Time

This setting can be used to eliminate messages which are shorter than a given amount of time in seconds. The default value for this setting is 5.

#### Say CID/Duration

Read back caller's telephone number / message duration prior to playing the incoming message when checking it.

#### Envelope

Envelope controls whether or not IPPBX will play the message envelope (date/time) before playing the voicemail message.

#### Review

Let a caller review their message before committing it to a mailbox.

#### **Delete Messages After Notification**

When voicemail to email is set, after voicemail is sent out via email successfully, the voicemail will be deleted from IPPBX. This is intended for use with users who wish to receive their voicemail ONLY by email.

#### Ask Caller to Dial 5

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

# 3.7 Conference

Allows participants dial into a virtual meeting room from their own phone, support up to 20 participants.

atcom		System Status	PBX S	ettings	Network Settings	System Settings	F	<b>Leports</b>	<b>X</b> Logout
Extensions	Conf	erence :							
Trunks	+ A	bb							
Inbound Routes	С	onference Room		PIN #			O	peration	
Outbound Routes	1 6	Edit Conference	e : 6800				Edit	Delete	
Outbound Roales	2 6	81					Edit	Delete	
Feature Codes	3 6	8			<b>0</b>		Edit	Delete	
IVR	4 6	81 Extension :	6800		• PIN : 123		Edit	Delete	
Hunt/Ring Group	5 6	81		Save	A Cancel		Edit	Delete	
	6 6	86					Edit	Delete	
> Conference	7 6	806					Edit	Delete	
Callback	8 6	807					Edit	Delete	-

#### **Conference Room**

Extension number of conference room, participant dial it to get into the room.

#### PIN#

Used for authentication before participants dial into the room, IPPBX will playback MoH for the first participant.



# 3.8 Paging / Intercom

Dial a code and connect directly to a built-in two-way announcement and talkback function on one or more phones, support up to 20 participants.

## **Paging Group Number**

Extension number of paging group, dial it to reach this group.

## Duplex

If checked, caller and callees all can speak and hear. Otherwise, only caller can speak, and callees can hear.

elicom	System Status	PBX Settings	Network Settings	System Settings	Reports	<b>X</b> Logout
Extensions	Paging/Intercom :					
Trunks	+ Add					
Inbound Routes	Add Paging Group					
Outbound Routes						_
Feature Codes	Paging Group Num	ber: 6700	Du Du	plex : 🗖 🌗		
IVR		6002(SIP/IAX)		>> 6001(IAX)		
Hunt/Ring Group	Paging Group Men	6003(SIP) 6006(SIP) 6007(SIP)	=	6000(SIP)		
Conference	5 5 1	6007(SIP) 6008(IAX) 6009(SIP/IAX)	~	← <<		
Callback						
> Paging/Intercom			🔊 Save 🥢 A Cance	el		
Time Intervals						

# 3.9 Options

### 1) General Preference

### **Ring Timeout**

Default Ring Timeout for an extension if Ring Out for it is not set.

# Max Call Duration

This defines the maximum amount of time in seconds for a interior call, 0 means no limit, default is 6000s.

#### **Music On Hold**

This define which Music on hold is used when transfer/call park/on hold/Conference etc.

#### **Tone Region**

This defines how the default dial tone, busy tone, and ring tone look like, please select your country or nearest neighboring country here.



atcom	System Status	PBX Settings		Network S	,	Systen	<b>Ö</b> n Settings	Reports	<b>X</b> Logout
Extensions	General Preference								
Trunks	Ring Timeout : 30	9			Max Call Du	ration :	6000		
Inbound Routes	Music On Hold : defaul	: v 🛛			Tone Regio	n :	United States/Nor	rth An 🗸 🌗	
Outbound Routes	Follow Me Prompt no	~							
Feature Codes									
IVR	Extension Preference								
Hunt/Ring Group	VOIP Extensions :	6000	to	6099					
Conference	FXS Extensions :	6101	to	6108					
Callback	Ring Group Extensions :	6600	to	6609					
Paging/Intercom	IVR :	6680	to	6699					
Time Intervals	Paging Group Extensions :	6700	to	6709					
Queue	Conference Extensions :	6800	to	6809					
DISA			to						
Blacklist	Queue Extensions :	6900	to	6909					
> Options					Save				

#### 2) Extension Preference.

Defines the range for VOIP / FXS / Ring Group / Voice Menu / Paging Group / Conference / Queue Extensions. The extension length must be between 3 and 9 digits. The maximum quantity can be supported for each are as below:

VOIP extension	100
FXS extension	8
Ring Group	9
Voice menu/IVR	16
Paging Group	9
Conference	9
Queue	9

# 4. Create SIP trunk to communicate with VoIP provider

In this page, we can also configure the 'service provider' trunk, which doesn' t need the use name and password for authorization, when you have bought a trunk from provide with IP address only, please choose 'service provider' trunk .

# 4.1 Create SIP trunks

Go to	PRY	Settings	- ~ '	Trunke	Click
90 10	<b>FD</b> A	Settings	->	munks,	CIICK

+ New VolP Trunk to add a new SIP trunk.

Туре	
SIP	
Trunk	Name

A unique label to help you identify the trunk.



#### Provider Hostname/IP

Hostname or IP of your VoIP provider, default port is 5060

#### Account Name

The username that your service provider configured

#### Authuser

The username that your service provider configured for authentication, generally, it's same as Account Name.

#### Password

The password configured for the user in your service provider side.

#### **Enable Outbound Proxy**

Outbound Proxy is a SIP proxy server, it acts, like any proxy server, as a middleman between two communicating agents, serving as a transit point for all SIP traffic. It can be used to solve the SIP one-way-audio issue.

#### **Outbound Caller ID**

The Caller ID used when using outbound proxy.

#### **SIP Transport**

The transplant protocol type for VoIP data package, default is UDP. Please make sure TCP is enabled in PBX Settings -> SIP Settings before using TCP.

#### **Maximum Outbound Calls**

Define the maximum quantity of outbound connections (simultaneous calls) that can be used on this trunk. Inbound calls are not counted in. 0 means no connection limit.

#### **Preferred Code**

Set the allowed codec and priority for this trunk.

#### Call Recording

Check this option to record all exterior calls made by this trunk. User can check the recordings with record

account(password is the same with admin) .

Туре:	SIP	~						
Trunk Name :	elastix		0	Provider Hostnam	e/IP: 192.16	8.1.173	: 5060	0
Account Name :	200		0	Authuser :	200			
Password :	200		0	Call Recording	0			
Domain :				From User :				
Enable Outbound Proxy	0							
Outbound Proxy :				Outbound Caller I	D :			
SIP Transport :	UDP	~	0					
Maximum Outbound Calls :	100		0					
Preferred Code :								
First : a-law 🗸 🗸		Second :	u-law	~	Third :	GSM	~	
Fourth : G729 🗸		Fifth :	None	~	Sixth :	None	~	
dvance Configuration								•

# Advance Configuration

DOD(Direct Outward Dialing Number) Setting

Set the Outbound number for different extensions. For example:

lvance Configu	ration		
OOD Setting			
DOD :123456	Associated Extension : 600	0	8
OD Number : 1	23456	Associated Extension : 6000 🗸 🗸	🕂 Add DOD



# 4.2 Check SIP Trunk Status

After creating trunk, go to **System Status** -> **Trunk Status** to check the SIP trunk Status, make sure it's registered.



elicom		System Status	PB	X Settings	() Network Settir	ngs	System Settings	Reports	<b>X</b> Logout
		Туре		Trunk Name		Status		Port/HostName/IP	
General	1	SIP		Elastix		Registered	Ł	192.168.1.158	
	2	SIP-SIP		IP08		Registered	Ł	192.168.1.157	
<ul> <li>Trunk Status</li> </ul>	3	FXO		FXO2		Disconneo	ted	PORT2	
<ul> <li>Trunk Status</li> <li>Extension Status</li> </ul>	3	FXO		FXO2		Disconneo	ted	PORT2	Refres

# 4.3 Make outbound calls

Go to PBX Settings -> Outbound Routes, click

Add to add an outbound route.

#### Outbound Route Name

A unique label to help you identify the outbound route.

#### **Dial Pattern**

A filter for marching numbers you dial, the call will be forwarded out via Selected Trunks only when it matches the dial pattern here. In patterns, some characters have special meanings.

X means Any Digits from 0-9

**Z** means Any Digits from 1-9

N means Any Digits from 2-9

[1234-9] means Any Digits in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

. means one or more digits

! 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.

For example: Once set Dial Pattern: 2XX Strip: 0, that means any calls to 200-299 will be forwarded out. Please do not simply set it to X., otherwise all telephone numbers with 2 or 2+ digits will be matched, this outbound route probably affect your interior calls, unless your local extensions is just a single figure.

**Strip**: The number of digits that will be stripped from the front of the dialing string before the call is placed via Selected Trunks. See example in Chapter 6.

**Prepend these digits:** Allows the user to specify digits that are prepended before the call is placed via the trunk. **Password:** Authentication for Selected Extensions before dialing out.

**Outbound Extension Selection**: Select extensions which can dial out with this outbound route. In my case, only 6000 and 6001 can dial out with this trunk.

Outbound Trunk Selection: Select trunks which calls are forwarded out through.



		8
	Elastex	
l		
2XX 🚺	Strip digits from	front
before d	dialing 😲	
	6102(fxs2)	X
	Selected Trunks	
	2xx   before of   Image: state of the state of th	Image: Strip   Image: Strip   Image: Defore dialing   Image: Defore dialing

# 4.4 Make inbound calls

Go to **PBX Settings** -> **Inbound Routes**, click **H** Add to add an inbound route.

### 1)General

#### Caller ID

Define the Caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info. Special characters described in chapter 4.3 can be used here as same.

#### **DID number**

Define the expected DID number if your trunk passes DID on incoming calls. Leave this blank to match calls with any no DID info. The DID for SIP trunk is usually as same as the account (others can only dial the account number to dial into PBX), then just leave it blank is OK.

## Extension

Define the extension for DID number. This field is usually unavailable for most VoIP trunk, if you VoIP provider support DID, you can set it. 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. Up to 100 DID numbers can be set.

For example: Set DID number: 6000-6010, Extension: 6000, All inbound calls to 6000-6010 will be forwarded to extension 6000. Set DID number: 6000-6010, Extension: 6000-6010, inbound calls to 6000-6010 will be forwarded to corresponding extension.

Add Inbound Route							
General							
Inbound Route Name :	fromElastix	] 🛛	Caller ID	: 2	xx		
DID Number :	200	9	Extension	n:			
Inbound Trunk Selection							
Available Trunks				Selected Trur	iks		
FXO3 FXO4 GSM5 GSM7 sv4 u520	<	>> → ↓ <<	ela	astix			<
Time		- <u> </u>					
Time Interval :				~			
Path							
Destination Type :	Extension	~	Destina	tion :	6000(SIP)	~	
Fax Detection							
Destination : Faxes	~			6001(IAX)	~		
		Save	✦ Cancel				

### 2)Inbound Trunk Selection

Select the trunks for which this inbound route apply.

# 3)Time

Select appropriate time intervals for when this inbound route apply.

### 4)Path

Set the destination for incoming calls. If Extension is set, this option will not take effect.



# 5. Create Service Provider to connect two IP PBXes

Service Provider is used to interconnect two IP PBXes, it must be created in both IP PBXes, after setting service provider and outbound route, extensions of each IP PBX can call the other's directly according to the outbound route.

# 5.1 Create Service Provider

Go to **PBX Settings** -> **Trunks**, Click **I low** Service Provider to add a new service provider.

#### Туре

SIP

### Provider Hostname/IP

Hostname or IP of your VoIP provider, default port is 5060.

#### **Provider Name**

A unique label to help you identify the service provider.

#### Call Recording

Check this option to record all calls made by this trunk. User can check the recordings with record

account(password is the same with admin).

#### **Maximum Outbound Calls**

Define the maximum quantity of outbound connections (simultaneous calls) that can be used on this trunk. Inbound calls are not counted in. 0 means no connection limit.

#### Transport

The transplant protocol type for VoIP data package, default is UDP. Please make sure TCP is enabled in PBX Settings -> SIP Settings before using TCP.

#### Preferred Code

Set the allowed codec and priority for this trunk.

#### Advance Configuration

#### DOD(Direct Outward Dialing Number) Setting

Set the Outbound number for different extensions. Detailed example can be found in Chapter 4.1.

Ne	w Service Provider					8
	Type : Provider Name :	SIP V	Provider Hostname/IP :	192.168.1.78	5060	
	Maximum Channels :	100	Transport :	UDP	v <b>9</b>	_
	Preferred Code :	First : Fourth :		Second : u-law Fifth : None	✓         Third :         GSM         ✓           ✓         Sixth :         None         ✓	-
	Advance Configuration		Save Acance			•



# 5.2 Check Service Provider Status

After setting service provider in the other PBX, the status will become to Registered.

atcom		System Status	PB	X Settings	Network Settir	ngs	System Settings	Reports	Cogout
		Туре		Trunk Name		Status		Port/HostName/IP	
General	1	SIP		Elastix		Registere	d	192.168.1.158	
	2	SIP-SIP		IP08		Registere	d	192.168.1.157	
Trunk Status	3	FXO		FXO2		Disconne	cted	PORT2	
Extension Status									Refresh

# 5.3 Make outbound calls

Go to **PBX Settings** -> **Outbound Routes**, click to add an outbound route.

The setting here is as same as that for SIP trunk, please refer to Chapter 4.3 to set it.

ld Outbound Route					E
General					
Outbound Route Name :			toIP08		
This place will be replace	d				
Dial Pattern :	9XX		Strip	digits from front	]
Prepend these digits :		before diali	ng 🕕		
Password :		9			
Outbound Extension Selec	tion				
Available Extensions				Selected Extensions	]
6002(SIP/IAX) 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX) 6009(SIP/IAX)		>> 1 4 <<		6000(SIP) 6001(IAX) 6101(fxs1)	
Outbound Trunk Selection					
Available Trunks				Selected Trunks	
FXO3 FXO4 GSM5 GSM7		> 1 J		IP08	J

Then 6000/6001 can dial 900-999 to corresponding extension (900-999) in other end directly.

# 5.4 Make inbound calls

# Go to **PBX Settings** -> **Inbound Routes**, click **I** add an inbound route.

Even there is no inbound route set, two IPPBXes can communicate with each other, however we can make it more functional with setting DID to those differ from local extensions.

## Example 1: DID + Extension--Make calls more easier.

DID: 00-03 Extension: 6000-60003

Extensions from other end can dial 6000-6003 extension directly to reach these extensions, besides, they can dial 00-03 to reach them if their outbound route allow. In this case, Path is of no use.

Add Inbound Route							8
General							
Inbound Route Name :	IP08		Caller I	D :			
DID Number :	00-03		Extensi	on :	6000-6003		
Inbound Trunk Selection							
Available Trunks				Selected T	runks		
FXO4 GSM5 GSM7 elastix sv4 u520			1	P08			<
Time							
Time Interval :				~			
Path							
Destination Type :	End Call	~	Destin	ation :		~	
Fax Detection							
Destination : Custom E	mail			6002(SIP/IA)	X) 🗸		
		Save /	✦ Cancel				

### Example 2: DID + Path--Forward calls to other applications.

DID: 6600 Path: Ring Group

Extensions from other end can dial local extensions directly, besides, they can dial 6600 to reach the destination set in Path. In this case, they can dial 6600 to reach the ring group.



Add Inbound Route			8
General			
Inbound Route Name :	IP08	Caller ID :	9
DID Number :	6600	Extension :	0
Inbound Trunk Selection			
Available Trunks		Selected Trunks	
FXO4 GSM5 GSM7 elastix sv4 u520		IP08	
Time			
Time Interval :		~	
Path			
Destination Type :	Ring Group	Destination : 6600	Y
Fax Detection			
Destination : Faxes	~	6001(IAX)	
	Save ,	A Cancel	

# 6. Make outbound/inbound calls to/from PSTN network

# 6.1 Make sure FXO modules are installed

If there are FXO modules installed in your IPPBX, IPPBX configures analog trunk automatically when they are detected.

ateon	System Status	<b>2+</b> PBX Settings	Network Settings	System Settings	Reports	<b>S</b> Logout				
Extensions	Analog Trunk :	Analog Trunk :								
> Trunks	🗘 Refresh	🗘 Refresh								
	Trunk Name		Port		0	peration				
Inbound Routes	1 FX02		2			Edit				

**NOTE**: Before using them, please make sure FXO port is connected with PSTN line (InService). The connection



### status can be checked in System Status -> Trunk Status

elicom		System Status	PB	<b>X</b> X Settings	() Network Settin	ngs	System Settings	Reports	<b>S</b> Logout
		Туре		Trunk Name		Status		Port/HostName/IP	
General	1	SIP		Elastix		Failed		192.168.1.158	
	2	SIP-SIP		IP08		Registere	d	192.168.1.157	
<ul> <li>Trunk Status</li> </ul>	3	FXO		FXO2		InService		PORT2	
Extension Status									Refresh

# 6.2 Make outbound calls

Go to PBX Settings -> Outbound Routes, click

Add to add an outbound route.

#### **Outbound Route Name**

A unique label to help you identify the outbound route.

#### **Dial Pattern**

A filter for marching numbers you dial, the call will be forwarded out via Selected Trunks only when it matches the dial pattern here. In patterns, some characters have special meanings.

X means Any Digits from 0-9

Z means Any Digits from 1-9

**N** means Any Digits from 2-9

**[1234-9]** means Any Digits in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

. means one or more digits

**Strip**: The number of digits that will be stripped from the front of the dialing string before the call is placed via Selected Trunks.

For example: If set Dial Pattern: 9, Strip: 1, Prepend 123, user need to dial 94567 to dial PSTN number 1234567

### Prepend these digits

Allows the user to specify digits that are prepended before the call is placed via the trunk.

#### Password

Authentication for Selected Extensions before dialing out.

#### **Outbound Extension Selection**

Select extensions which can dial out with this outbound route. In my case, only 6000,6001 can prefix 9 to dial out.

#### **Outbound Trunk Selection**

Select trunks which calls are forwarded out through. If there are more than one FXO trunk chose, PBX will try to dial out through next trunk if the previous one failed.



Add Outbound Route						8
General						
Outbound Route Name :						
This place will be replaced						
Dial Pattern :	9.	•	Strip	1	ligits from front	
Prepend these digits :		before dia	aling 🕛			
Password :	123	0				
Outbound Extension Selection Available Extensions 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX) 6009(SIP/IAX) 6102(fxs2)	n	>> 1 ×		Selected Extens 6000(SIP) 6001(IAX) 6101(fxs1)	ions	<
Outbound Trunk Selection						
Available Trunks				Selected Trunks		
FXO4 GSM5 GSM7 IP08 elastix sv4		> 1 4 V		FX03		~
	Save	e	A Cance			

# 6.3 Make inbound calls

Go to **PBX Settings** -> **Inbound Routes**, click **I** to add an inbound route. Just setting Selected Trunks

and Path is OK.

### 1) General

# Caller ID

Define the Caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info. Special characters described in chapter 5.2 can be used here as same.

#### **DID** number

Just leave it blank.

#### Extension

It's unavailable for Analog trunk, leave it blank.

# 2) Inbound Trunk Selection



Select the trunks for which this inbound route apply.

# 3) Time

Select appropriate time intervals for when this inbound route apply.

# 4) Path

Set the destination for incoming calls.

Add Inbound Route							
General							
Inbound Route Name : fromPSTN	Caller ID :						
DID Number : Leave	them blantiktension :						
Inbound Trunk Selection							
Available Trunks	Selected Trunks						
GSM5 GSM7 IP08 elastix sv4 u520	FX03						
Time							
Time Interval :	✓						
Path							
Destination Type : Extension	✓ Destination : 6002(SIP/IAX) ✓						
Fax Detection							
Destination : No Detect	✓						
Sav	ave Acancel						

# 7. Inbound Call Control

# 7.1 Time Interval

Set the Time Interval for inbound route.



atcom

Ac	ld Time Intervals			8
	Time Interval Name :		lunchbreak	
		۲	By day of week	
			Mon 💌 to Fri 💌	
		0	By Days of a Month	
			Date: 1 👻 to 1 😪 Month: January 😪	
	Time:		Entire Day	
			Start Time: 11:45 AM End Time: 01:30 PM	
			Save Alose	

# 7.2 Hunt / Ring Group

This defines a 'virtual' extension that rings a group of phones simultaneously / one by one, stopping until any one of them is picked up.

Ec	lit Hunt/Ring Group : 6600			8
	Ring Group Name :	6600	Ring Group Number :	6600
	Ignore Call Forward 🏮	Enable MOH 🌗	Music On Hold :	×
	Ring Mode :	🔿 Parallel 🚺 💿 Serial 🌗		
	Ring interval :	20		
	Ring group members : 🌗	Available Extensions : 6002(SIP/IAX) 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX)	>> 	Selected Extensions : 6000(SIP) 6001(IAX)
	No answer forward to :	End Call	✓ 🕒	

### **Ignore Call Forward**

By default, if there is call forward enabled in IP phone, IPPBX will call that forwarding number. when this option is checked, IPPBX will ignore that call forward setting.

# Enable MoH


When MoH is choose, caller will hear MoH other than ring tone.

#### Parallel

Ring all the members at the same time.

#### Serial

Ring all the members one by one.

#### No answer forward to

Set the failover destination if there is no answer.

## 7.3 Queue

Usually used in Call Centre to queue customers for the next available operator.

#### 1) General

#### Queue Name

Name of the queue

#### **Queue Number**

Extension number of the queue, dial it to get into the queue

#### Queue Password

Used as authentication for users before being dynamic agent.

#### **Queue Agent Timeout**

Ring timeout in seconds when calling an agent

#### **Queue Max Wait Time**

The maximum time in seconds for a caller can wait in the queue before being pulled out. (0 means unlimited)

#### **Queue Ringing Strategy**

Strategy for IPPBX ring the agents.

**RingAll**: Ring all available agents simultaneously until one answers.

LeastRecent: Ring agents which was least recently called.

FewestCalls: Ring agents with the fewest completed calls.

Random: Ring agents in a random way.

**RRmemory**: Round robin with memory, remembers where it left off in the last ring pass.

#### 2) Agents

Select Static Agent here. there are two kinds of agents:

#### Static Agent: chose here

**Dynamic Agent:** users can dial 'Queue number + \*' to log in as dynamic agent, and 'Queue number + \*\*' to log out.

In this case, users can dial 6900\* to being a dynamic agent (need to enter password 123), and 6900\*\* to log out.

#### 3) Caller Position Announcement

Announce queue position and / or estimated hold time to caller

#### 4) Period Announcement



This allows a message like "Thank you for holding, your call is important to us." to be played at regular intervals while a caller is in the queue

**NOTE**: The key point with announcements is that they are only played within the timeout/retry period set on the queue. For the most part this works OK as when all queue members are busy/unavailable, the timeout/retry period is effectively ignored (i.e. you can consider the queue to always be in this state) and announcements will be played as per your setting of the announce-frequency and periodic-announce-frequency parameters. When a handset is available and the queue is ringing it, the timeout/retry timeouts become critical. For example, if you want announcements every 20 seconds, but the timeout is set to 60 seconds, when a queue member is ringing, you will only ever get announcements every 60 seconds.

#### 5) Event

This allows callers waiting in the queue to dial a key to go to other destination.

#### 6) Failover Destination

This define the failover destination for callers when the max wait time is up.

#### 7) Others

#### Music On Hold

Select Music On Hold Class for this Queue

#### Leave When Empty

This option controls whether calls 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 logged in, or if all logged in agents are unavailable.

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 three options:

Yes: Callers can join a call queue with no agents or only unavailable agents.

No: Callers cannot join a queue with no agents or if all agents are unavailable.

#### 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

How long does IPPBX wait before trying all the members again.

#### Wrap Up Time

After a successful call, how long to wait before sending a potentially free member another call.



it Call Queue : 6900					
General					
Queue Name :	6900		Queue Number :	6900	
Queue Password :	1234		Queue Agent Timeout :	45	
Queue Max Wait Time :	1800		Queue Ringing Strategy		
-			Quoto ranging bautogy		
Agent <del>s</del>					
Available Agents :			Selected Agents :		
6002(SIP/IAX) 6003(SIP)	^	>	> 6000(SIP) 6001(IAX)		<u>~</u>
6006(SIP) 6007(SIP)			→ 6101(fxs1)		
6008(IAX)	<b>v</b>	-			~
Caller Position Announce	ment				
Announce Position :	Yes	~	Announce Holdtime :	Yes	~
Frequency : 15s V					
Period Announcement					
Prompt : he	ello-world 🗸		Frequency :	40s	~
Event					
Кеу: *	Y Action :	End Call	<ul> <li>✓ Destinati</li> </ul>	ion :	~
Failover Destination					
Action : En	nd Call 🗸		Destination :	0	~
Others					
Music On Hold :	default	~ <b>0</b>	Leave When Empty :	No	~ <b>!</b>
Join Empty :	Yes	~ <b>9</b>	Agent Announcement :	hello-world	~ <b>Q</b>
Join Announcement :	hello-world	~ \rm \rm	Retry :	30	
Wrap Up Time :	30				

## 7.4 IVR

Callers are presented with a recorded menu and respond by selecting a digit or, in some cases, by entering an extension number. The automated attendant eliminates the need for a live operator to handle the call.

#### $\otimes$ Edit IVR : 6680 0 0 **IVR Name :** IVR Number : 0 🗸 times 🕕 3 Key Timeout : 3 Repeat Count : hello-world $\sim$ Prompt : Custom Prompt Allow Dialing of Other Extensions I Extension $\sim$ 6001(IAX) $\sim$ Press 0 to trigger : Press 1 to trigger : $\sim$ 6000(SIP) $\sim$ Voice Mail Press 2 to trigger : $\sim$ $\sim$ $\sim$ $\sim$ Press 3 to trigger : $\sim$ Press 4 to trigger : $\sim$ $\sim$ $\sim$ Press 5 to trigger : ¥ $\sim$ Press 6 to trigger : $\sim$ $\sim$ Press 7 to trigger : $\sim$ $\sim$ Press 8 to trigger : $\sim$ $\sim$ Press 9 to trigger : V $\sim$ Press \* to trigger : $\sim$ $\sim$ Press # to trigger : ~ 0 No Entry Forward to : No Action $\sim$ No Action ~ 0 Invalid Forward to : $\sim$ Save

#### Voice Menu Name

Name of the Voice Menu

#### Voice Menu Number

Extension number of the voice menu, dial it to get into the voice menu

#### **Key Timeout**

How long for IPPBX to wait user's input

#### **Repeat Count**

How many times to play prompt

#### **Allow Dialing of Other Extensions**

Allow dialing local extensions

atcom



#### **Key Press Event**

Dial digit to trigger corresponding event

#### No Entry Forward to

The destination for incoming call if there is none input

#### Invalid Forward to

The destination for incoming call if there is invalid input

## 7.5 DISA

DISA (Direct Inward System Access) allows someone calling in from outside to obtain an "internal" system dialtone and dial out as if a local extension.

General	
DISA Name : toPSTN PIN # :	
Response Timeout :   10   Digit Timeout :   10	
Outbound Routes	
Outbound : Add	
Outbound : Add	
Save Cancel	

#### 1) General

#### DISA Name

A name for the DISA

#### PIN #

When caller get into the DISA, this password is needed to put before making calls.

#### **Response Timeout**

The maximum time in seconds IPPBX will wait for input from a user.

#### **Digit Timeout**

The maximum time allowed between entry of digits. If exceeded, user input is deemed to have finished.

#### 2) Outbound Trunks

Choose the outbound route that callers can use to dial out.



For example: Both City A and B have a IPPBX, IPPBX-A and IPPBX-B, they are connected with SIP trunk, and IPPBX-A has FXO trunk to connect local PSTN and outbound route for that, DISA can be used as below:

- 1. Create a DISA in IPPBX-A including the FXO trunk.
- 2. Set it as the destination of inbound route for SIP trunk.

After users of IPPBX-B dial into DISA application in IPPBX-A, The DISA application in turn requires the user to enter his passcode, followed by the pound sign (#). If the passcode is correct, the user will hear dialtone on which a outbound call may be placed, so there is no long distance call fees.

## 7.6 Call Back

A callback will hang up on the caller and then call them back, directing them to the selected destination. This is useful for reducing mobile phone charges as well as other applications. Outbound calls will proceed according to the dial patterns in Outbound Routes

dd Callback			
General			
Callback Name :		Callback Number :	
Delay Before Callback :	0	Call Recording	
Path			
Destination Type :	~	Destination :	~
	Save /	A Cancel	

#### **Callback Name**

A name for the Callback

#### **Callback Number**

Enter the number to dial for the callback. Leave this blank to just dial the incoming Caller ID Number.

#### Delay before Callback

Enter the number of seconds the system should wait before calling back.

#### Call Recording

Check this option to record all calls made by this Call back. User can check the recording with record account( password is the same with admin).

#### PATH

Choose the destination which IPPBX will bridge to caller.

#### For Example:

1. Set Callback:

Edit Callback : <mark>pstn</mark>				×
General				
Callback Name :	pstn !	Callback Number :		
Delay Before Callback :	3	Call Recording		
Path				
Destination Type :	Extension V	Destination :	6002(SIP/IAX)	~
	Save Save	← Cancel		

## 2. Set inbound route for FXO trunk:

Add Inbound Route	8
General	
Inbound Route Name : fromPSTN	Caller ID :
DID Number :	Extension :
Inbound Trunk Selection	
Available Trunks	Selected Trunks
FXO4 GSM5 GSM7 IP08 elastix sv4	FX03
Time	
Time Interval :	~
Path	
Destination Type : Callback 🗸	Destination : pstn v
Fax Detection	
Destination : No Detect	~
Save	A Cancel

atcom



3. Dial into IPPBX from PSTN number, and hang up immediately once the call is connected. Then IPPBX dial extension 6002, after 6002 answer the call, IPPBX dial caller and bridge them.

## 7.7 Inbound to Outbound

Outbound Routes can act as destination in Inbound route settings, it let two ends connecting to IPPBX with trunk communicate directly.

#### 7.7.1 FXO trunk to SIP trunk

- 1. Set outbound route for SIP trunk.
- 2. Set Inbound route for FXO trunk.

#### Destination

Choose the proper SIP trunk

#### Number

Set a number on the other end of SIP trunk. it must be set.

3. Dial into IPPBX from PSTN, then IPPBX will dial 202 through SIP trunk.

Edit Inbound Route : fPSTN	8
General	
Inbound Route Name : IPSTN	Caller ID :
DID Number :	Extension :
Inbound Trunk Selection	
Available Trunks	Selected Trunks
	FX04
<	
Time	
Time Interval :	✓
Path	
Destination Type : Outbound Routes V Destination :	toElastix V Number : 202
Fax Detection	
Destination : No Detect ~	~
Save	A Cancel



#### 7.7.2 SIP trunk to FXO trunk

- 1. Set outbound route for FXO trunk.
- 2. Set Inbound route for SIP trunk/Service Provider. Since service provider is more flexible, here we use Service Provider as an example

#### Destination

Choose the proper FXO trunk

#### Number

Set a number on the other end of FXO trunk. it can be blank, then it means the number caller dialed.

3. Dial into IPPBX from SIP trunk, e.g. 83018618, then IPPBX will dial that number through FXO trunk selected in PSTN outbound route.

Edit Inbound Route : fElastix	8
General	
Inbound Route Name : fElastix	Caller ID :
DID Number :	Extension :
Inbound Trunk Selection	
Available Trunks	Selected Trunks
FXO4	Elastix
Time	
Time Interval :	~
Path	
Destination Type : Outbound Routes V Destination :	toPSTN Vumber :
Fax Detection	
Destination : No Detect	✓
Save	A Cancel



## 7.8 Custom Dialplan

Click Click

#### user

This file aimed at defining a "user". It can define a user with an optional SIP phone, Dahdi phone and/or just about any other type of phone.

#### dialplan

This file contains the "dial plan" of Asterisk, the master plan of control or execution flow for all of its operations. It controls how incoming and outgoing calls are handled and routed. This is where you configure the behavior of all connections through IPPBX.

#### dahdi

Asterisk cmd DAHDILookup configuration.

**NOTE**: You must prefix custom\_ to your custom file and context to avoid conflict with the system file,users.conf,extensions.conf and chan\_dahdi.conf.

Ch	noose a file	8
[	Custom Dialplan type :	]
	Choose a file : Browse_ No file selected.	
	🕹 Upload 🛛 🥕 Cancel	

## 7.9 Blacklist

Block incoming calls from specified numbers

Ν	ew Blacklist		8
	Blacklist Number :	202	]
	Save	Acancel 🖉	1
		Canter	



If a number in blacklist dial into IPPBX, caller will hear following prompt: "Then number you have dialed is not in service. Please check the number and try again." Then system will then disconnect the call.

### 7.10 SIP Settings

#### 1) General

#### UDP Port

Set the SIP port (UDP) which IPPBX is listening to.

#### Enable TCP

Enable TCP protocol for SIP.

#### **TCP Port**

Set the SIP port (TCP) which IPPBX is listening to.

#### **Registration / Subscription Time Max**

Maximum duration in seconds of a SIP registration / subscription.

#### **Registration / Subscription Time Min**

Minimum duration in seconds of a SIP registration / subscription.

#### **RTP Port Min / Max**

Set the RTP port range.

## DTMF Mode

Set the default DTMF mode

General			
UDP Port :	5060	Enable TCP 🚺 TCP Port :	5060
Registration/Subscription Time Max :	3600	Registration/Subscription Time Min :	60
RTP Port Max :	20000	RTP Port Min :	10000
DTMF Mode :	rfc2833 🗸 🗸	Video Support :	yes 🗸 🗸

#### 2) NAT

Here provide other two solutions for SIP one-way-audio issue besides outbound proxy. Using any one is OK.

#### a. STUN

Just setting STUN server / port is OK.

There are many public STUN server on Internet: http://www.voip-info.org/wiki/view/STUN



NAT			
Enable STUN : 🗹 🌗			
STUN Server :	stun.sipgate.net	STUN Port :	10000
External IP Address :		External Host :	
External Refresh Interval :		NAT Mode :	v 9
Local Network Identification :		Allow RTP Reinvite :	no 🗸 🕽

#### b. NAT

NAT			
Enable STUN : 🗖 🌗			
STUN Server :		STUN Port :	
External IP Address :		External Host :	atcomtest.f3322.org
External Refresh Interval :	10	NAT Mode :	yes 🗸 🖓
Local Network Identification :	92.168.1.0/255.255.255.0	Allow RTP Reinvite :	nonat 🗸 🗣

The External IP, External Host and Local Network Identification settings are used if you use IPPBX behind a NAT device to communicate with services on the outside.

#### **External IP address**

Address that we're going to put in outbound SIP messages if we're behind a NAT. The externip and localnet is used when registering and communicating with other proxies that we're registered with.

#### **External Host**

Alternatively you can specify an external host, and IPPBX will perform DNS queries periodically. Not recommended for production environments! Use External IP instead.

#### **External Refresh Interval**

How often to refresh External Host if used.

#### NAT Mode

Global NAT settings (Affects all peers and users), is used when Asterisk is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports.

#### Local Network identification

You may add multiple local networks. A reasonable set of defaults are set here.

#### Allow RTP Reinvite

By default, Asterisk tries to re-invite the audio to an optimal path. If there's no reason for IPPBX to stay in the media path, the media will be redirected. This does not really work with in the case where IPPBX is outside and have clients on the inside of a NAT. In that case, you want to set this option to no nat.

## 8. Audios

## 8.1 Music On Hold

Manage audio files for Music On Hold, the format should be .WAV (16 bit, mono 8000 Hz) and .GSM, the size should less than 4 MB.

M	Music On Hold List					
2	🕹 Upload MOH					
	Name	Operation				
1	default	Play Delete				

## 8.2 Custom Prompts

Manage prompts used for Voice Menu. It can be recorded by extensions or uploaded from local PC, the format should be .WAV (16 bit, mono 8000 Hz) and .GSM, the size should less than 4 MB

Cu	Custom Prompts :						
Ŧ	🕂 Record New Prompt 🕹 Upload a Prompt						
	Name	Operation					
1	hello-world	Record Again Play Download Delete					
2	intro	Record Again Play Download Delete					

## 8.3 Language Setting

Set the language of default system prompt audio, English is supported by default. French, Italian, Russian and Spanish need to be download from Internet when chose at the first time. Make sure gateway is right set so that IPPBX can access Internet.

Extensions	Language Setting
Trunks	Language : English 🗸 🕽
Inbound Routes	
Outbound Routes	Save Save

## 9. Network Settings

Description of WAN settings can be found in Chapter 2. All network settings will take effect immediately when you save the change.

## 9.1 Web Access

Choose the web access protocol and port for web server here. HTTP and HTTPs are both supported, default port is 80 and 443 respective.



elicom	System Status	PBX Settings	() Network Settings	System Settings	Reports	S Logout
Web Access	Web Access Setting					
	Web Access Mode :	⊙ HTTP OHTTP	'S			
WAN	HTTP Port :	80	9			
Firewall	HTTPS Port :		]0			
	💾 Save 🛛 Arreset					

## 9.2 Firewall

Firewall is used to prevent unauthorized connections.

#### 1) Enable Firewall

Check it to enable firewall.

#### 2) Common Rule

Accept/Drop the connections from remote hosts.

#### Name

A name for the rule.

#### Description

Simple description for the rule.

#### Protocol

Set the protocol type for connection.

#### Port

Set the destination port range for connection. The main protocols and default ports IPPBX uses for each application are list below:

НТТР	TCP:80
HTTPS	TCP:443
SIP	UDP:5060
SIP	TCP:5060
RTP	UDP:10000-20000

IP

Set source IP of connection.Format of IP: IP/mask

For example:

192.168.1.156/255.255.255.255 for IP 192.168.1.156

216.207.245.47/255.255.255.255 for IP 216.207.245.47

192.168.1.156/255.255.255.0 for network 192.168.1.0/24

#### Mac Address

Set source Mac of connection. Either IP or Mac Address must be set.

#### Action

Accept: Accept the access from remote hosts.



Drop: Drop the access from remote hosts.

Ne	ew Firewall Comm	inon Rule	8
			^
	Name :	SIPlocal	
	Description :	accept sip package from local network	
	Protocol :	UDP 🗸 🌓	
	Port :	5060 : 5063	
	IP :	192.168.1.0 / 255.255.255.0	
	Mac Address :		
	Action :	ACCEPT V	
		Save Acancel	
			~

#### 3) Auto Defense

Limit connections from remote hosts.

#### Port

Set the destination port range for connection.

#### Protocol

Set the protocol type for connection.

#### Rate

The maximum packets or connections can be handled per second

~
^
=
~

#### 4) SIP Defense:

Limit connections to SIP port from remote hosts.

#### Port

Set the destination port range for connection.

#### Protocol

Set the protocol type for connection.

#### SIP Packets

The maximum packets can be handled per time interval.

#### Time Interval

Time unit which IPPBX uses to manage IP packets received.

N	ew SIP Defense	Rule			8
	Port :	5060	Protocol :	UDP	~
	SIP Packets :	200	Time Interval :	1	seconds
		📃 🖳 Subm	nit 🥕 Ancel		

#### 5) Other Options

#### Disable Ping

Check this to drop ping packets from remote hosts.

#### **Drop All**

Check this to drop all packets or connection from other hosts if there are no other rules defined.

#### 6) Firewall Setting Example :





**Firewall setting** 

🗹 Enable Firewall

Co	Common Rule											
Ŧ	+ New Rule											
	Action	Name	Protocol	IP		Mac Ad	Mac Address Po		ort Operation		ration	
1	ACCEPT	SIPlocal	UDP	192.168.	1.0/255.255.255.0			50	60:5063	Edit	Delete	
2	ACCEPT	SIPprovider	UDP	216.207.	245.47/255.255.255.	.255		50	60:5063	Edit	Delete	
3	DROP	dropothers	UDP					50	60:5063	Edit	Delete	
Au	Auto Defense											
÷	New Rule											
	Port		F	Protocol		Rate			Operation			
1	80		Т	ГСР		50			Ec	Edit Delete		
SI	P Defense											
÷	New Rule											
	SIP Packets				Time in Seconds				Opera	tion		
1	1 200 1			1				Edit	Delete			
Ot	Other Options											
	Disable Ping											

## 9.3 DDNS

Dynamic Domain Name Service (DDNS) is a service used to map a domain name to the dynamic IP address of a network device. IPPBX support 4 DDNS servers below, please go to the website of below servers and apply a domain name then fill related information here.

dyndns.org

dyndns.com

qdns

www.zoneedit.com

DDNS Setting		
Enable DDNS :	Yes	
DDNS Server :	dyndns.org	User Name : voipadmin
Password :	•••••	Hostname : atcomtest.f3322.org
💾 Save		

## 9.4 VLAN

A VLAN (Virtual LAN) is a logical local area network (LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations. Both WAN and LAN support 2 VLANs.

elicom	System Statu	s PBX Settings	Network S		System Settings	Reports	<b>X</b> Logout
Web Access	VLAN over WAN						
	No. 1 :			No. 2 :			
WAN	VLAN ID :	100		VLAN ID :	200		
Firewall	VLAN IP :	192.168.100.100		VLAN IP :	192.168.200.1	00	
	VLAN Subnet Mask :	255.255.255.0		VLAN Sub	onet Mask : 255.255.255.0		
DDNS	Default Gateway :	192.168.100.1		Default G	ateway : 192.168.200.1		
> VLAN	Submit						
VPN							

## 9.5 VPN

A virtual private network (VPN) extends a private network across a public network, such as the Internet. It enables a computer to send and receive data across shared or public networks as if it were directly connected to the private network, while benefitting from the functionality, security and management policies of the private network. This is done by establishing a virtual point-to-point connection through the use of dedicated connections, encryption, or a combination of the two. Currently, only PPTP is supported.

VPN Setting	VPN Setting				
🗹 Enable VPN					
VPN Type :	PPTP V				
VPN Server :	bijou.tenacy-free.com				
User Name :	tenacy				
Password :	91706				
💾 Save					

## 10. System Settings

## 10.1 Change Password

Change the password for admin login, it will take effect immediately.



etcom	System Status	<b>2+</b> PBX Settings	() Network Settings	System Settings	Reports	<b>X</b> Logout
<ul> <li>Change Password</li> </ul>	Change Password					
	New Password :					
Auto Provisioning	Retype New Password :					
Date && Time	Save Save					

## **10.2** Auto Provisioning

IP01 can only configure ATCOM AT8XX serials IP phone via SIP Subscribe + Notify way. Make sure your AT8XX IP phone is using the latest firmware.

#### 10.2.1 Auto Provisioning

#### 1) Phonebook

You can add your contacts here and when you use phone provisioning, IP phone will download the phone book.

#### Add

Name: Input the name of this contact.

Type: These are two types: Directory and Blacklist.

Number: Input the number here.

A	dd Phone Book Rule		⊗
	Name :		
	Туре :	~	
	Number :		
		Save Acancel	

#### 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.



#### 2) Ring Tone

You can upload a Ring Tone before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones.

**NOTE**: Ring tone must be wav file, 8k sampling rate, 8 Bit u-law compression. File size should < 200Kbytes.

#### 3) Firmware

You can upload a firmware before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones.

#### NOTE: the file format must be .fw

10.2.2 Configured Phones

Create New Phone

#### Enable

Enable auto provisioning for this phone

#### Phone Type

Choose the Phone Type

#### **MAC Address**

Input the MAC address for the phone

#### **New Version**

Make sure your phone is with the newest version. If yes, IPPBX will generate the configuration file for IP phone.

#### Manufacturer

Choose the manufacturer.

#### DND

Enable or Disable DND.

#### **Call Waiting**

Enable or Disable Call Waiting.

#### Auto Answer

Enable or Disable Auto Answer.

#### Phone book

Enable or Disable Phone book update .If this option is checked, you must add or upload phone book first.

#### **Ring Tone**

Enable or Disable Ring Tone. If this option is checked, you must upload phone book first.

#### Firmware

Enable or Disable Firmware update. If this option is checked, you must upload firmware first.

#### **Firmware version**

If Firmware option is enable, you must input the correct firmware version here.

**NOTE:** If the firmware version is different from the one you're going to load in IP phone, it will cause problems in IP phone.

#### Line



You can configure the line key settings for this IP phone.

New Configured Phone				E	3
General					
Enable :	Yes 🗸		Phone Type :	AT820 V	
MAC Address :	80:82:87:01:04:D2		New Version :	Yes 🗸	
Manufacturer :	ATCOM 🗸		DND :	Disabled V	
Call Waiting :	Enabled V		Auto Answer :	Disabled V	
Phone Book :	Enabled 🗸		Ring Tone :	Enabled V	
Firmware :	Disabled 🗸		Firmware Version :		
Line					
line1 Extension :	×	Display	/Name :	Line Active :	
☑ line2 Extension :	6007 🗸	Display	Name : Robin	Line Active :	
	S:	ave	A Cancel		

#### 10.2.3 Unconfigured Phones

In this section, IPPBX will scan all the supported IP phones and display here, you can click the 'Edit' button of IP phone and input the corresponding information in the pop-up window.



🕻 Scan 🛛 🥜 Modify Selected Ph	iones		
MAC Address	Manufacturer	Operat	ion
80:82:87:00:C4:48	ATCOM	Edit	
80:82:87:01:04:D6	ATCOM	Edit	
80:82:87:01:09:A8	ATCOM	Edit	
w Configured Phone			
General			
Enable :	Yes 🗸	Phone Type :	AT840 🗸
Enable : MAC Address :	Yes	Phone Type : New Version :	AT840 V
MAC Address :	80:82:87:00:C4:48	New Version :	No
MAC Address : Manufacturer :	80:82:87:00:C4:48	New Version : DND :	No V

#### 10.2.4 Upload Phone Configure

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

**NOTE**: the file format must be .xml.

## 10.3 Date && Time

Set the date and time for IPPBX. The settings will take effect immediately.

General	
GMT TimeZone :	+8 China(Beijing)
<ul> <li>Automatically Synchronize with</li> </ul>	n Internet Time Server :
NTP Server :	pool.ntp.org
◯ Set Date&Time Manually :	
Daylight Saving Time	
Daylight Saving Time :	Disabled
Daylight Saving Time Rule :	
🕒 Save 🛛 🥕 Cancel	

#### 1) General

There are two ways to set Date/Time for IPPBX:

a. NTP server



Make sure the connection between IPPBX and NTP server is OK, if the NTP server is located on Internet, the gateway of WAN should be right set so that IPPBX can access Internet.

GMT TimeZone is also an important arguments for time setting in this way. Please choose the right Time Zone.

b. Manually

#### 2) Daylight Saving Time

There are two ways to set DST:

a. Automatic

Just making sure GMT Time Zone is right set is OK. There have already DST setting in each Time Zone.

b. Manually

However, the DST in some countries is changing every year. If the DST setting in Time Zone is not exact. Please set it manually, the format should be: **start**=*start\_time*;**end**=*end\_time*;**save**=*offset* 

The rule for start / end time is: month/mday/wday/hour:min:second

1<= month <=12 , 0< mday <=31 , 0<= wday <7

month/mday/wday means the first wday coming after month/mday

for example:

start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1 means IPPBX time from the first Sunday coming after April 1th to the first Sunday coming after October 31th will be one hour early.

## **10.4 External Storage**

The External Storage feature is used to extend storage space. Once configured, the files (voicemail/Call recording (>60M) / CDR(>5M) ) created before the configured days will be moved to the Net-Disk.

**NOTE**: The shared folder must be based on Windows operation system.

- 1. Choose a window-based computer that is always in service
- 2. Create a folder
- 3. Share this folder

# atcom



#### 4. Set External Storage

#### **Net-Disk Host/IP**

IP of the PC

#### Net-Disk Share Name

The name of the share folder

#### Net-Disk Access User Name

Account in that PC

#### Net-Disk Access Password

Password for the account, if there is no password required, leave it blank.

#### **Backup Period**

How often PBX move its voicemail/call recording/CDR data to PC

#### Move Files Created Before days ago

Choose what files should be move to PC according to their creation time. 0 means immediately. Note that even 0 is set, the size of call recording must be more than 60 M, that of CDR must be 10M. Then IPPBX will move them to PC.



Change Password	Net Disk Settings		
	Net-Disk Host/Ip :	192.168.1.156	
Auto Provisioning	Net-Disk Share Name :	Share	
Date && Time	Net-Disk Access User Name :	user	
	Net-Disk Access Password :	•••••	
<ul> <li>External Storage</li> </ul>	Backup Period :	30 v minutes	
Firmware Upgrade	Move Files Created Before :	30 v days ago	
	Save		

## 10.5 Firmware Upgrade

Firmware for IPPBX There are two ways to upload Kernel / Application for IPPBX:

#### 1) HTTP

Upload them from local PC. It will reset IPPBX to factory default settings by default.

### 2) TFTP

Upload them from TFTP server, the Kernel / Application must be located in base directory of TFTP server.

Firmware Upgrade				
Note: If you select the http mode when upgrading kernel, the config will be erased automatically!				
Image Type :	<ul> <li>Application</li> </ul>	OKernel		
Upgrade Method :	Онттр	<ul><li>● TFTP</li></ul>	Reset Config :	
Server :			File :	

Upgrade

Choose Reset Config will reset the configuration.

## **10.6 Backup and Restore**

#### 1) Backup

Create Backup for configuration / System audio prompt / Voice Mail. The backup can be downloaded to local PC.

New Backup	
🗖 Backup Config 🗖 Backup Voice 🗖 Backup Voice Mail	
File : backup_20130816_371	
🔛 Backup 🦳 🥕 Cancel	



#### 2) Restore

Click Restore to restore corresponding backup, backup file can be uploaded from local PC. Backup will be used after IPPBX reboot. It can't be used for different product models.

Ba	Backup and Resotre				
+	Create a Backup 🛛 🕹	Upload a Backup			
	Name	Date	Operation		
1	backup_20130816_3	2013-08-16 17:32:38	Download From System Restore Delete		

## 10.7 Reboot && Reset

Reboot or Reset IPPBX.

Reboot System
Warning : Rebooting system will terminate all active calls!
Reboot
Reset to Factory Default
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!
Reset to Factory Default

You need to choose the product model and extension length/prefix the first time to login IPPBX after it is reset to factory default settings.

( 192.168.1.100/admin/express_setup.html	⊽ C Soogle ⟨Ctrl+K⟩	🔎 🕄 💌 💌 🕈 🤗 🖛
		_
Extension Format		
Product Model : IP02	~	
IP01 IP02	_	
1F02 1P04	_	
IP08		
IP204A		
IP4G		





		▼ C Google ≪trl+K>	🔎 🖾 💌 🖛 🐐
	Extension Format		
	Extension Length : 4 Extension Prefix : 6 Ne	× ×	
3 192.168.1.100/admin/express_setup.html		V C Scogle (Ctrl+K)	🔎 🔀 💌 💌 🕈 🧚
	Extension Assignment		
	Extension Assignment Extension For SIP User : 6000	- 6099	
		- <u>6099</u> - <u>6108</u>	
	Extension For SIP User : 6000		
	Extension For SIP User : 6000 Extension For FXS Port : 6101	- 6108	
	Extension For SIP User : 6000 Extension For FXS Port : 6101 Extension For Ring Group : 6600	- 6108 - 6609	
	Extension For SIP User : 6000 Extension For FXS Port : 6101 Extension For Ring Group : 6600 Extension For Paging Group : 6700	- 6108 - 6609 - 6709	
	Extension For SIP User :       6000         Extension For FXS Port :       6101         Extension For Ring Group :       6600         Extension For Paging Group :       6700         Extension For Conference :       6800	- 6108 - 6609 - 6709 - 6809	
	Extension For SIP User :       6000         Extension For FXS Port :       6101         Extension For Ring Group :       6600         Extension For Ring Group :       6700         Extension For Conference :       6800         Extension For Conference :       6800         Extension For Queue :       6900	- 6108 - 6609 - 6709 - 6809 - 6809	

## 11. Reports

## 11.1 Call Detail Records

Display the Call Detail Records, the operation for it can be search, delete and download.

#### 1) Search

Users can search the records they needs according to Source, Destination, and / or Time.

#### 2) Delete

IPPBX supports two delete operations: delete selected CDR and delete all CDR.

#### 3) Download

It can be downloaded to local PC



Syslog	From : 🛗 To : 🛗 Source : Destination : 🔍 🔍 Search									
		🖆 Download All CDR 🍵 Delete Selected CDR 🍵 Delete All CDR								
		Source	Destination	Start Time	End Time	Duration	Billable Duration	Disposition		
	11		s	2013-08-21 08:18:41	2013-08-21 08:18:47	6	0	ANSWERE		
	12	02196802000	s	2013-08-21 08:18:36	2013-08-21 08:18:38	2	0	ANSWERE		
	13		s	2013-08-20 19:25:15	2013-08-20 19:25:21	6	0	ANSWERE		
	14		s	2013-08-20 19:25:06	2013-08-20 19:25:12	6	0	ANSWERE		
	15		s	2013-08-20 19:24:56	2013-08-20 19:25:02	6	0	ANSWERE		
	16		s	2013-08-20 19:24:51	2013-08-20 19:24:52	1	0	ANSWERE		
	17	900	6003	2013-08-20 18:48:14	2013-08-20 19:00:25	731	729	ANSWERE		
	18	900	6003	2013-08-20 18:48:11	2013-08-20 18:48:12	1	1	ANSWERE		
	19	900	6003	2013-08-20 18:48:03	2013-08-20 18:48:09	6	6	ANSWERE		
	20	900	6003	2013-08-20 18:47:10	2013-08-20 18:47:17	7	5	ANSWERE		

## 11.2 Syslog

## 1) Syslog

Set the Syslog level and download it.

Call Detail Records	Syslog				
	The Log Level : warning				
> Syslog	Save 🗠 Downlaod Log				
	🔚 Save 🔄 🖆 Downlaod Log				

#### 2) Packet Capture Tool

This feature always used to capture packets for technician. Users could specify the destination IP address and port to get the packets.

#### IP

Specify the destination IP address to get the packets.

#### Port

Specify the destination port to get the packets.

Packet Capture Tool						
IP :						
Port :						
Start	Stop Download					

## 12. Web Interface for extension

PBX allows users to check their voicemail /Call Recording/CDR, and set personal settings.



		^
ATCOM PBX Login		
£ 6003		
â ••••		
🛣 Language	~	
Login		
	<ul> <li>▲ 6003</li> <li>▲ ●●●●●</li> <li>★ Language</li> <li>Login</li> <li>▲ Website ● Products ≈ Support</li> </ul>	<ul> <li>▲ 6003</li> <li>▲ 6005</li> <li>★ Language</li> <li>✓</li> </ul>

- Check Use Web Interface option in PBX Settings -> Extensions management settings to allow this extension to login its own web interface.
- 2. Enter the IP of PBX in the browser.
- 3. Login with extension number / Voice Mail Access PIN Code as username / password

## 1) Voice Mail Checking

Users can listen / download / delete / move voice mail here.

Voice Mail	Voice Mail Logout							
Voice Wildin								
Call Recordings	Caller ID	Time	Duration	Operation				
	1 '900' <900>	2013-08-20 18:45:57	18	Download Play Delete				
Call Detail Records	2 '900' <900>	2007-01-01 02:53:23	16	Download Play Delete				
Dereanel Settinge	5 💌 🚺 🖣 Page 1	of 1 🕨 🔰 💭		Displaying 1 to 2 of 2 items				
Personal Settings	Old Voice Mail							
	Caller ID	Time	Duration	Operation				
	5 V I4 4 Page 1 of 1 V I 5 Displayin							
	Sorry, no data exist!							
	Urgent Voice Mail							
	Caller ID	Time	Duration	Operation				
	5 💌 🚺 🖣 Page 1	of 1 🕨 🔰 💭		Displaying 1 to 0 of 0 items				
	Sorry, no data exist!							

## 2) Call Recording Checking



Voice Mail	Call Recordings Logo							
	Caller ID	Time	Duration	Operation				
Call Recordings	1 900	2013-08-20 18:48:18	738	Download Play Delete				
	<			<u>&lt;</u>				
Call Detail Records	10 🕶 🚺 🖣 Pa	ge 1 of 1 🕨 🔰 📢		Displaying 1 to 1 of 1 items				
Personal Settings								

## 3) CDR Checking

#### Users can check their CDR here.

					:		:	
		Source	Destination	Start Time	End Time	Duration	Billable Duration	Disposition
Call Recordings	1	900	6003	2013-08-20 18:48:14	2013-08-20 19:00:25	731	729	ANSWERED
	2	900	6003	2013-08-20 18:48:11	2013-08-20 18:48:12	1	1	ANSWERED
	3	900	6003	2013-08-20 18:48:03	2013-08-20 18:48:09	6	6	ANSWERED
Call Detail Records	4	900	6003	2013-08-20 18:47:10	2013-08-20 18:47:17	7	5	ANSWERED
	5	900	6003	2013-08-20 18:45:43	2013-08-20 18:46:24	41	39	ANSWERED
Personal Settings	6	900	6003	2013-08-20 18:44:26	2013-08-20 18:45:40	74	67	ANSWERED
	7	900	6003	2007-01-01 02:53:11	2007-01-01 02:53:57	46	46	ANSWERED
	8		6003	2007-01-01 01:54:13	2007-01-01 01:54:22	9	3	ANSWERED
	9		6003	2007-01-01 01:52:20	2007-01-01 01:52:48	28	15	ANSWERED
	10		6003	2007-01-01 01:50:58	2007-01-01 01:51:45	47	44	ANSWERED

## 4) Personal Settings

Users can set voice mail / voice mail to email / follow me / ring timeout here.

Voice Mail	Personal Settings						
voice maii							
Call Recordings	General						
Call Detail Records	Name: 6003						
	Voice Mail						
Personal Settings	Voice Mail Enable : Yes Voice Mail Access PIN Code : 6003						
	Mail Setting						
	Enable Sending Voice Mail to Email Address :						
	Follow Me						
	Call Forward : Always 🗹 When no answer 🗹 When busy Forward To : 📀 Voice Mail 🛇 Number :						
	Other Options						
	Ring Out: 30 DND Save Reset						

--Finish--