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Safety Notice

Please read the following safety notices before installing or using this IP PBX. They are crucial for safe and reliable operation of the device. Failure to follow the instructions contained in this document may result in damage to your PBX and void the manufacturer's warranty.

1. Please use the external power supply which is included in the package. Other power supplies may cause damage to the device, affect performance or induce noise.
2. Before using the external power supply in the package, please check your building power voltage. Connecting to inaccurate power voltage may cause fire and damage.
3. Please do not damage the power cord. If the power cord or plug is impaired, do not use it. Connecting a damaged power cord may cause fire or electric shock.
4. Ensure the plug-socket combination is accessible even after the PBX is installed. In order to service the PBX it will need to be disconnected from the power source.
5. Do not drop, knock or shake the device. Rough handling can break internal circuit boards.
6. Do not install the device in places where there is direct sunlight. Also do not place the device on carpets or cushions. Doing so may cause the device to malfunction or cause a fire.
7. Avoid exposing the device to high temperature (above 40°C), low temperature (below -10°C) or high humidity. Doing so could cause damage and will void the manufacturer warranty.
8. Avoid letting the device come in contact with water or any liquid which would damage the device.
9. Do not attempt to open the device. Non-expert handling of the device could cause damage and will immediately void the manufacturer warranty.
10. Consult your authorized dealer for assistance with any issues or questions you may have.
11. Do not use harsh chemicals, cleaning solvents, or strong detergents to clean the device.
12. Wipe the device with a soft cloth that has been slightly dampened in a mild soap and water solution.
13. If you suspect your device has been struck by lightning, do not touch the device, power plug or phone line. Call your authorized dealer for assistance to avoid the possibility of electric shock.
14. Ensure the PBX is installed in a well-ventilated room to avoid overheating and damaging the device.
15. Before you work on any equipment, be aware of any hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents if you are in a situation that could cause bodily injury.

1 Overview

1.1 Brief Introduction of LAVoice Series

LAVoice Series IP Phone System is the most innovative solution for VoIP telecommunication in the SMB (Small and Medium-sized Business) market. They provide not only traditional PBX functionality such as automated attendant and voicemail, but also offer many advanced telephony features, including remote extensions, remote office connection, IVR, call recording, call detail records(CDR). All of these can serve to greatly enhance business operations at reduced operational cost.

In this manual, we will introduce how to install and configure the LAVoice LVX30V2 and LVX100sV2 IPPBX systems.

Each model is introduced below:

LAVoice LVX30V2 has 2 analog ports installed onboard by default. The available options are detailed below:

	FXS	FXO	GSM
LAVoice LVX30	1	1	Optional
	0	2	Optional

LAVoice LVX100s consists of two main components:

- LVX100s Main Case
- Modules

There are 2 slots in the system and modules can be utilized as in the following table.



LVX100s Slot / LVX100s Module	Slot 1	Slot 2
LVX--M4X	✓	✓
LVX-M4O	✓	✓
LVX-M2XM2O	✓	✓

1.2 Main Features

1. Black List
2. BLF(Busy Lamp Field)
3. Caller ID
4. Call Detail Records (20000 records)
5. Call Center Queues (20)/ Callback
6. Call Parking/ Call Forward/ Call Transfer/ Call Waiting
7. Call Record /Ring Group Record/ Call Queue Record
8. Conference Bridge (20 Conferences)
9. DISA (Direct Inward System Access) /Paging and Intercom
10. DID/Smart DID/ DOD
11. Dial by Name
12. DHCP Server
13. Do Not Disturb(DND)
14. DDNS(Dyndns.org /No-ip.com /zoneedit.com/ freedns.afraid.org/
www.oray.com/ 3322.org)
15. Audio Codec: Opus, G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX
16. Video Codec: VP8, H.261/ H.263 / H.263+ / H.264
17. Flexible Dial Plan
18. IP Phone Feature Code
19. IPv4 / IPv6
20. One Number Stations
21. Music On Hold
22. Phonebook/LDAP (5000 contacts)
23. Ring Group
24. Speed Dial
25. Skype for SIP
26. SIP/ IAX Extension Registration
27. SNMPv2
28. Static /DHCP /PPPoE Network Access
29. System Backup
30. T.38 Pass-through
31. USB Mobile Hard Disk Record (Scalable)
32. Video Call
33. Voicemail
34. Virtual Fax
35. VPN Server (L2TP / PPTP / OpenVPN/ IPSec, up to 20 connections for VPN clients)
36. VPN Client (L2TP / PPTP / OpenVPN / N2N/ IPSec)

37. Web-based Administration and configuration
38. Extension User Portal
39. Webdial/ WebRTC
40. IP Phone Provisioning

1.3 Modules

			
2FXS	2FXO	1FXOS	1GSM
			
LVX-M4X	LVX-M4O		
			
LVX-M2X2O	LVX-M1E1		
			
LVX-M2G	LVX-M3G		

Notice:

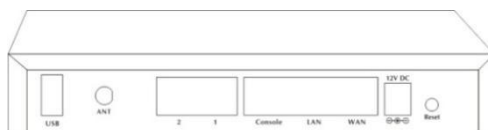
- 1) Lava Module cards will only function in LAVoice IP PBX from Lava;
- 2) Module cards for LAVoice LVX100s are packed separately but contained in the same package as the LAVoice system.

1.4 Mechanical Design

1.4.1 LAVoice LVX30



LAVoice LVX30V2 Front Panel



LAVoice LVX30V2 Rear Panel

- 1 * Reset Button
- 1 * Power Interface (DC 12V 2A)
- 2 * Ethernet Interface (WAN/LAN:10/100Mbps)
- 2 * Analog Ports(FXO/FXS)
- 1 * ANT Port (GSM)
- 1 * USB Interface (for storage)
- 1 * Console Interface

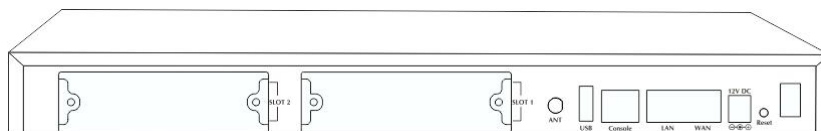
L VX30 LED Indication

LED Label	Function	Status	Indication
PWR	Power Status	On	Power on
		Off	Power off
SYS	System Status	On	System initiating
		Blink	System is functioning
		Off	System failure
WAN	WAN Status	On	Connected but no data transmitting
		Blink	Data transmitting
		Off	Disconnected
LAN	LAN Status	On	Connected but no data transmitting
		Blink	Data transmitting
		Off	Disconnected
1	FXO Status	Red	Channel available
		Blink	Channel ringing
		Off	Channel failure
2	FXS Status	Green	Channel available
		Blink	Channel ringing
		Off	Channel failure

1.4.2 LAVoice LVX100s



LAVoice LVX100sV2 Front Panel



LAVoice LVX100sV2 Rear Panel

1 * Reset Button

1 * Power Port (DC 12V 2A)

2 * Ethernet Interface (WAN/LAN:10/100Mbps)

1 * USB Interface

1 * Console Interface

Applicable module for slot1 &2: Analog/ GSM/ WCDMA Module Cards

L VX100s LED Indications

LED Label	Function	Status	Indication	
PWR	Power Status	On	Power on	
		Off	Power off	
SYS	System Status	On	System initiating	
		Blink	System is functioning	
		Off	System failure	
WAN	WAN Data Status	On	Connected but no data transmitting	
		Blink	Data transmitting	
		Off	Disconnected	
LAN	LAN Data Status	On	Connected but no data transmitting	
		Blink	Data transmitting	
		Off	Disconnected	
1-4 (SLOT1/2)	SLOT 1/2 Status	FXS	Green	Channel available
			Blink	Channel ringing
			Off	Channel failure
		FXO	Red	Channel available
			Blink	Channel ringing
			Off	Channel failure
GSM	Red	Channel available		

			Blink	Channel ringing
			Off	Channel failure
		WCDMA	Red	Channel available
			Blink	Channel ringing
			Off	Channel failure

1.5 Model Comparison Table

Items		LAVoice LVX30	LAVoice LVX100s
System Capacity	Concurrent Calls	15	30
	Extension Users	30	100
	Voicemail and Recording	36,000 mins (.gsm)	36,000 mins (.gsm)
		4000 mins (.wav)	4000 mins (.wav)
	Conference Rooms	20	20
Hardware Capacity	SDRAM	512MB DDR3	1GB DDR3
	Memory (default)	8GB SD card	8GB SD card

1.6 Environmental Requirements

Operating Temperature: 0 °C ~40 °C

Storage Temperature: -20 °C ~ 55 °C

Humidity: 5~95% Non-Condensing

1.7 Package Contents

LAVoice Main Case	1
Power Adaptor	1
Ethernet Cable	1
Quick Installation Guide	1
Warranty Card	1
Rack Mount Ear	2
Screws	10

1.8 Compatible Endpoints

- Any SIP compatible IP Phone (Desktop Phones and Soft Phones for Windows, Linux, iOS and also Android platforms).
- IAX compatible endpoints
- Analog Phones and Fax Machines
- Web Extensions (WebRTC)

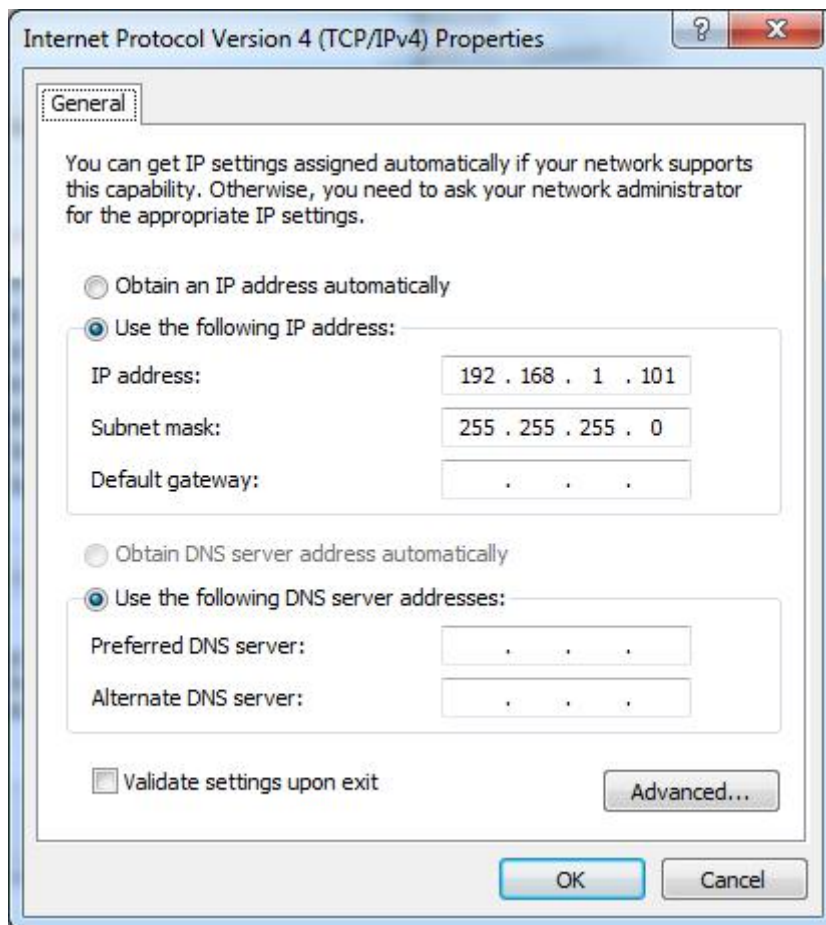
2 Getting Started

2.1 Connect LAVoice IPPBX to your LAN

2.1.1 System Login

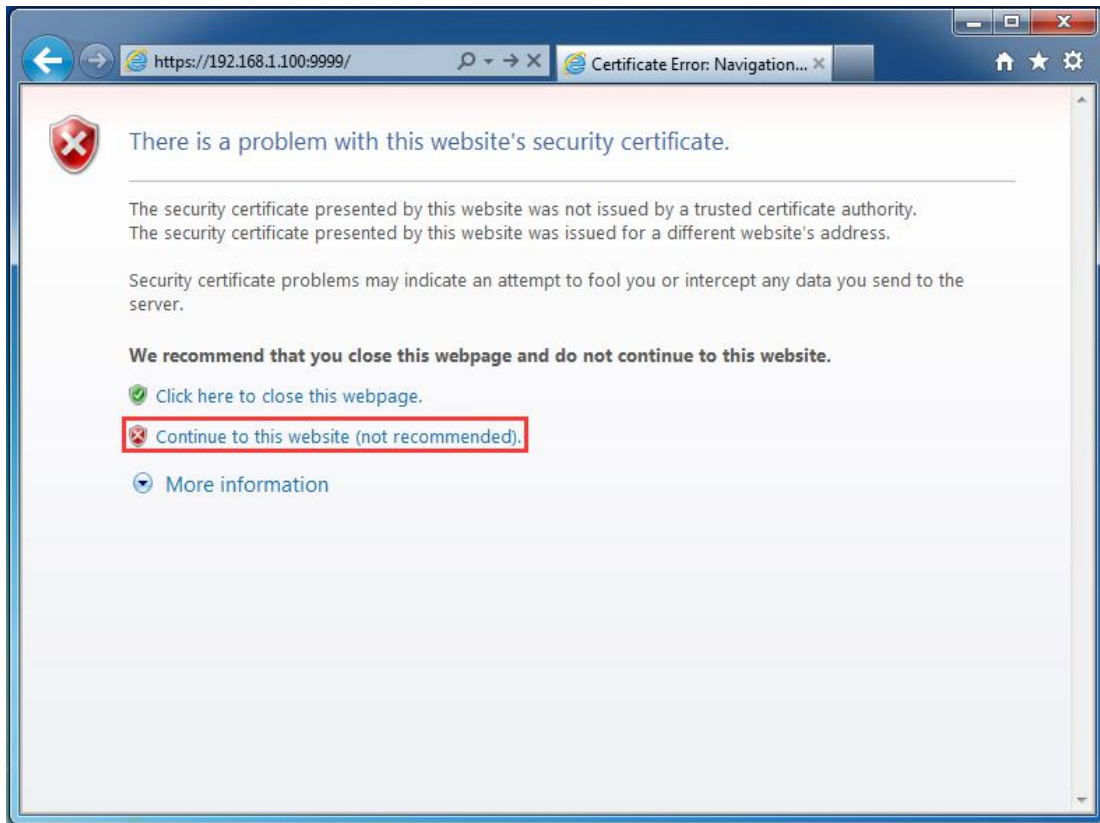
LAVoice series IPPBXs are preconfigured with a static IP address of 192.168.1.100 on the devices WAN port (192.168.10.100 on LAN port). If your network is configured with a different IP range to the LAVoice system default address, then you will need to change the IP address to something more appropriate before connecting to your local LAN.

Please connect your PC directly to the WAN interface of the IPPBX and change the network profile of the PC to an IP address of 192.168.1.101 and Subnet mask of 255.255.255.0.

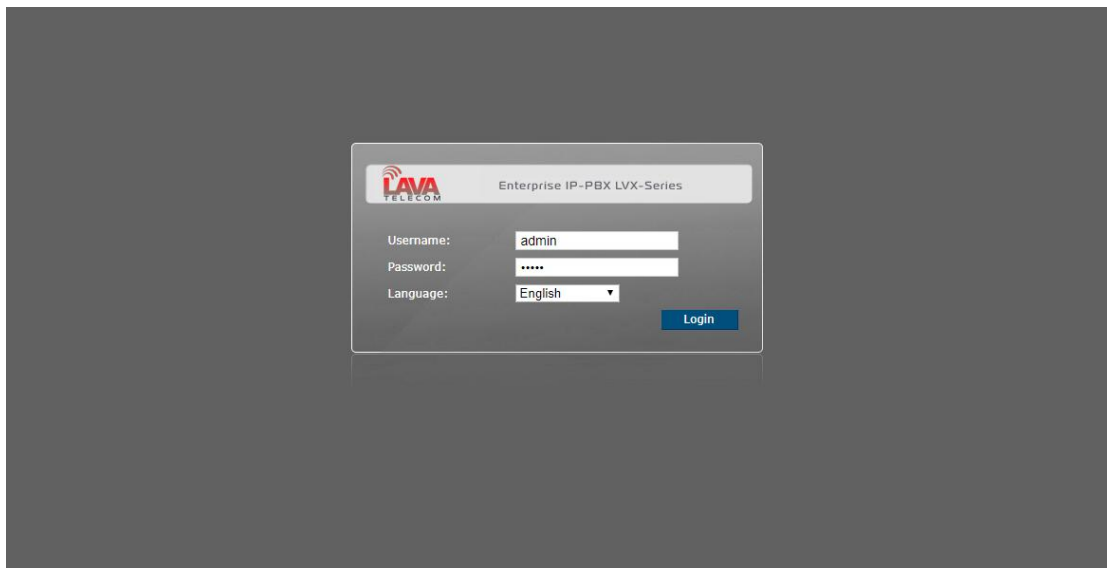


Now you can access the Web interface by inputting <https://192.168.1.100:9999> into your Internet browser address bar and pressing Enter.

You'll now be presented with a Certificate Error notice as below, please click "Continue to this website..." and you will be directed to the login page. Please ensure your IE browser version is at least version 9 or you may not be able to access the web interface.



Login page appears as below:



Type in the default username: admin, and default password: admin to login. After successful login, you will be notified to change the default admin password. Please follow the instructions within the notice to do this. To ensure the device is secure, the admin password must be complex so please set a strong password that uses a combination of letters, numbers and also special characters.

Notice:

1. LAVoice Series IPPBX Web GUI supports the following 11 languages:

English, Chinese, Arabic, Persian, Portuguese, Italian, French, Spanish, Russian, Turkish and Thai.

You can select your native language or if this is not available then the most familiar one to login. We are continuously adding more languages to meet the needs of our customers from all around the world.

2. Extension number can be used to login to the LAVoice IPPBX Web GUI

3. Operator user can login to the LAVoice IPPBX Web GUI to monitor the system status and check call logs and faxes. Login username is operator and password is password.

2.1.2 Configure Network Profiles

Navigate to Web Menu *Network Settings-->Network->IPv4 Settings*.

LAVoice IPPBX WAN interface can be configured to operate in Static, DHCP or PPPoE mode. In the majority of deployment scenario's it is standard practice to configure the unit in Static mode. DHCP and PPPoE will be described later in [chapter 5](#).

To configure your LAVoice system in Static mode, you must assign an available static IP address along with corresponding subnet mask, gateway and DNS to the WAN interface of the LAVoice IPPBX. For example, you could assign an IP address of 192.168.1.254, Subnet Mask: 255.255.255.0, Gateway: 192.168.1.1, DNS: 8.8.8.8.

Network

IPv4 Settings IPv6 Settings VLAN Settings

WAN Port Setup

IP Assign:

IP Address:

Subnet Mask:

Gateway:

Primary DNS:

Alternative DNS:

LAN Port Setup

IP Address: Subnet Mask:

IP AddressV1:

IP AddressV2:

Subnet MaskV1:

Subnet MaskV2:

After modifications are complete, please click the "Save" button to save the configuration. You will now be presented with a dialog box asking you to reboot the system to make the changes effective. Please reboot the system and once complete you can connect the IPPBX to your local LAN switch.

2.2 User extensions

Navigate to web menu *Basic->Extensions*.

This page lists all user extensions on LAVoice V2 system. Here you can add/bulk add, delete/bulk delete user extensions and also edit/bulk edit the user extension properties.

By default, 10 extension numbers within the range of 800 to 809 have been created for you to use.

The screenshot shows the 'Extensions' management page. At the top, there are two tabs: 'Extensions' (selected) and 'Upload/Download Extensions'. Below the tabs is a search bar with the label 'Extension:' and buttons for 'Search' and 'Show All'. Underneath are four action buttons: 'New User', 'Batch Add', 'Edit Selected', and 'Delete Selected'. The main content is a table titled 'Extensions' with the following data:

<input type="checkbox"/>	Name	Extension	Port	Protocol	DialPlan	Outbound CID	Options
<input type="checkbox"/>	1 800	800	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	2 801	801	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	3 802	802	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	4 803	803	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	5 804	804	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	6 805	805	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	7 806	806	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	8 807	807	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	9 808	808	--	SIP	DialPlan1		Edit
<input type="checkbox"/>	10 809	809	--	SIP	DialPlan1		Edit

2.2.1 New Extensions

You can add further extensions one by one by clicking the “New User” button or bulk add extensions by clicking “Batch Add” button and completing the popup shown below.

The 'Batch Add' popup window has the following fields and controls:

- Extension Start:** 810
- Extension End:** 829
- DialPlan:** DialPlan1 (dropdown menu)
- Password:** (Random)
- Buttons:** Save, Cancel

- **Extension Start/Extension End:** These two fields define the new extension range to be generated.
- **DialPlan:** Select a dial plan for the new extensions.
- **Password:** A secure random password consisting of numbers, letters and special characters is the recommended choice and can be selected by selecting the “Random” checkbox. Alternatively, you can specify the same password for all new extensions. If you choose this option then please ensure a secure password is set.

2.2.2 Other Extension Ranges

We have limited the user extension number range in the LAVoice IPPBX to be between 800 and 899. If you require more extensions or you want extensions in other number ranges then you need to change the extension range before you can add new extensions.

Please navigate to web menu *Advanced->Options->General*.

In the “[Extension Preferences](#)” section you can change the user extension range.

Extension Preferences		
User Extensions	<input type="text" value="400"/>	to <input type="text" value="499"/>
Conference Extensions	<input type="text" value="900"/>	to <input type="text" value="909"/>
IVR Extensions	<input type="text" value="610"/>	to <input type="text" value="629"/>
Queue Extensions	<input type="text" value="630"/>	to <input type="text" value="639"/>
Ring Group Extensions	<input type="text" value="640"/>	to <input type="text" value="659"/>
Paging Group Extensions	<input type="text" value="660"/>	to <input type="text" value="679"/>
Web Extensions	<input type="text" value="680"/>	to <input type="text" value="699"/>

In the above example, the user extension range has been changed to be between 400 and 499. If you now go back to the extension page you’ll be able to add new extensions within this range.

2.3 IP Extension Registration

2.3.1 Desktop IP phones

The following example details how to register Lava LV-4SC IP phone on your LAVoice IPPBX system.

Step 1:

Press the softkey “Status” beneath the phone screen, here you can see the IP Address of the IP phone.

Step 2:

Open the IP phone web interface by entering the phone IP address into the web browser address bar.

Step 3:

Default login credentials are username admin and password admin.

Step 4:

After successful login, navigate to the phone web menu VOIP->SIP, and register an extension number as below example.

- **Primary SIP server:** IP address of the LAVoice IPPBX.
- **SIP User ID:** User extension number from the LAVoice IPPBX user extension page.
- **Authenticate ID:** The same as Authentication user
- **Authenticate Password:** The password of the extension.
- **Name:** Name of the extension user.

2.3.2 Softphone on Windows PC

Softphones including 3CX, Bria, Zoiper and many other softphone APPs all work well with LAVoice V2 IPPBX. Below is an example of registering Zoiper to LAVoice IPPBX system as an extension from your Windows PC.

Step 1:

Download Zoiper from <http://www.zoiper.com/>.

Step 2:

Install and run Zoiper on your Windows device.

Step 3:

Click menu “Settings” and select “Create a new account” and select “SIP” protocol and click Next.

Step 4:

Complete the register credentials as in the example below:



Step 5:

Click Next to complete the registration process.

2.3.3 Softphone on Android phone, iPhone or iPad


The majority of softphones detailed previously in this section have mobile editions for both Android and iOS platforms. You can download these APPs and install them from your mobile phone APP Store.

Below is an example of how to register Zoiper softphone to LAVoice IPPBX as an extension from your iPhone:


Step 1:

Run Zoiper on your iPhone and tap  menu.

Step 2:

Tap  Accounts menu.


Step 3:

Tap  to create a new account.

Step 4:

You will be asked "Do you already have an account(username and password)?" tap "Yes" and then tap "Manual configuration" to continue.

Step 5:

Tap  SIP account to configure the account as in the below example:

SIP Account	
Account name:	403
Domain:	192.168.1.254
User name:	403
Password:	••••••
Caller ID:	403

Step 6:

After entering the register credentials, tap “Register” to register to LAVoice IPPBX system as an extension.

2.4 Phone Provisioning

If you plan to deploy a large number of IP phones, phone provisioning is a useful feature as it can reduce the time and effort required to deploy phone extensions. There are 2 methods to auto provision your IP phones, DHCP and PnP.

2.4.1 Phone Provisioning by PnP

Navigate to web menu *Advanced->Phone Provisioning*.

Here on this page you can see the term “PnP”, which refers to Plug and Play. By using this technique you don’t have to undertake any configurations directly on the IP phones, but instead only some minimized configurations on the IPPBX system. After this configuration is complete you can plug the phones to your LAN and once they start up, they are ready for phone calls through the IPPBX system.

Click on “PnP Settings” tab.

Plug and Play(PnP) Settings

Phones Settings | PnP Settings

Plug and Play(PnP) Settings

Enable:

Interface: WAN

Custom URL:

Multicasting Address: 224.0.1.75

Port: 5060

Save Cancel

On this page, tick “Enable” to enable PnP feature.

- **Interface:** Select WAN or LAN depending on which interface you have connected the IPPBX to your local LAN.

- **Custom URL:** Custom URL tells the IP phones where to obtain the configuration files for auto provisioning. You should read your IP phone user manual to determine which kind of files it requires for auto provisioning. Then you create/upload these files to a FTP/TFTP/HTTP server for the phones to download. The URL can be IP address or domain name with subdirectory. For example: [http://192.168.1.2/phones/\\${MAC}.conf](http://192.168.1.2/phones/${MAC}.conf). With “Custom URL” configured, you don’t have to add phones from the “Phone Settings” tab.
- **Multicasting Address:** IP phones which support PnP can use multicast discovery of SIP Registrar. Multicast registrations are addressed to the well-known “all SIP servers” multicast address “sip.mcast.net” (224.0.1.75 for IPv4).
- **Port:** SIP signaling port, default is 5060.

Notice:

Phone provisioning only works for IP phones that are in the same LAN where the LAVoice IPPBX is deployed.

After enabling PnP feature, click on “Phone Settings” tab and click “New Phones” to generate the configuration files for the phones to be added to the IPPBX system.

- **Manufacturer:** Manufacturer of the IP phone, currently, LAVoice V2 supports phone provisioning phones from the following manufacturers: SNR, Lava Grandstream, Yealink, Cisco
- **Model:** You must specify the exact model number of the phone, even if the phone is from the same manufacturer. This is because different models require different configuration files.
- **MAC:** LAVoice IPPBX uses the MAC address of the phone to identify it on the local LAN as part of the provisioning process and it essential that you enter the correct MAC address for your IP Phone
- **Extension:** The extension number selected here will be auto configured to the phone with the MAC address given above.
- **Label:** Specify the user name of the phone.

Once you have added your new IP Phone(s) as described above, configuration files will be generated in the background of the IPPBX system. You can now connect the phone(s) to your LAN and once the phone(s) have booted up they will download configuration files from the IPPBX system and complete auto configuration with the extension numbers you provided.

2.4.2 Phone Provisioning by DHCP

If you want to auto provision your IP phones using DHCP, please make sure they support DHCP option 66.

Please navigate to web menu: *Network Settings->DHCP Server* to enable DHCP service for the IP phones first. Please refer to [chapter 5.4](#).

Once DHCP is enabled you can add the phones in the same way as instructed above in Phone Provisioning by PnP section, however, enabling PnP is not required in this scenario.

Notice:

If you are going to enable DHCP service on the LAVoice IPPBX system, please ensure there is no other DHCP server in the same LAN. If possible you can put the IPPBX and IP phones in a separate VLAN.

2.5 Analog Extensions

If your LAVoice IPPBX is equipped with an FXS port then you can configure an analog extension on your IPPBX system. This can be an ordinary analog phone or it can be a fax machine for sending and receiving faxes. The green LED indicates the RJ11 interface is FXS, you should connect the analog phone/fax machine to the FXS port of the IPPBX.

Navigate to web menu: *Basic->Extensions*, click “[New User](#)” button to add an analog extension.

New		X	
General			
SIP:	<input checked="" type="checkbox"/>	IAX2:	<input type="checkbox"/>
Name:	411	Extension:	411
Password:	UdTQ1P#@4g	Outbound CID:	
DialPlan:	DialPlan1	Analog Phone:	Port 2
Voicemail			

In the “[Analog Phone](#)” dropdown list, select an FXS port number for this new extension. This will allow the analog phone/fax machine connected to this port to be assigned with this extension number. The phone can now make and receive phone calls in the same manner SIP/IAX extensions do.

2.6 Extension Status

You can check the status of all extensions configured on your LAVoice IPPBX via the *Operator* page “[Operator](#)” section.

Operator

Extensions

● Idle	● Ringing	● InUse	■ Hold	● UnAvailable
---	---	--	---	---

● 401 401(SIP)	● 402 402(SIP)	● 403 403(SIP)	● 404 404(SIP)	● 405 405(SIP)
● 406 406(SIP)	● 407 407(SIP)	● 408 408(SIP)	● 409 409(SIP)	● 410 410(SIP)

Total:10 Online:3 Current Call(s):0

Here in this section, you can view real-time status of all extensions. Including idle(online), ringing, in use and also on hold.

2.7 Advanced Extension Configurations

2.7.1 Edit Properties of One Extension

On the *Basic->Extension* page, you can click the "Edit" button to edit the properties of one extension number.

Edit X

General

SIP: <input checked="" type="checkbox"/>	IAX2: <input type="checkbox"/>
Name: <input type="text" value="401"/>	Extension: <input type="text" value="401"/>
Password: <input type="password" value="123456"/>	Outbound CID: <input type="text"/>
DialPlan: <input type="text" value="DialPlan1"/>	Analog Phone: <input type="text" value="None"/>

Voicemail

Enable: Password:

Delete VMail: Email(Fax/Voicemail):

Other Options

Web Manager: <input type="checkbox"/>	Agent: <input type="checkbox"/>	Call Waiting: <input checked="" type="checkbox"/>
Allow Being Spied: <input type="checkbox"/>	Pickup Group: <input type="text" value="1"/>	
Mobility Extension: <input type="checkbox"/>	Mobility Extension Number: <input type="text"/>	

VoIP Settings

NAT: <input type="checkbox"/>	Transport: <input type="text" value="UDP"/>	S RTP: <input type="checkbox"/>
Qualify: <input checked="" type="checkbox"/>	Remote Extension: <input type="checkbox"/>	
DTMF Mode: <input type="text" value="RFC2833"/>	Permit IP: <input type="text"/>	

Video Options

Video Call: H.261 H.263 H.263+ H.264 VP8

Audio Codecs

Disallowed

Allowed

Below are the explanations for the configuration options:

General

- **SIP:** Tick the checkbox to activate SIP protocol.

- **IAX2:** Tick the checkbox to activate IAX2 protocol.
- **Name:** Alias of this extension which can be the name of the extension user.
- **Extension:** Number of this user extension.
- **Password:** The password used for the phones to register. This can be set manually or can be generated by the IPPBX system. Auto generated password consists of numbers, letters and special characters. If this is an analog extension then password is of no use as analog phones are not required to register.
- **Outbound CID:** Choose a number to show to the external called party. This feature only works with SIP trunks if the ITSP(Internet Telephony Service Provider) allows this number to be passed.
- **DialPlan:** Defines which type of numbers the extension can dial.
- **Analog Phone:** The FXS port number. An analog phone attached to this port will use this extension number.

Voicemail

- **Enable:** Activate voicemail service for this extension.
- **Password:** Password for extension user to access the voicemail facilities.
- **Delete VMail:** Delete voice messages if the system has sent the message to user via email.
- **Email:** Email address of this extension user.

Other Options

- **Web Manager:** If enabled, users can use their extension number and voicemail password to login to the IPPBX system web GUI.
- **Agent:** If enabled, this user extension can be a call queue agent.
- **Call Waiting:** With this option enabled, busy extensions will hear the call-waiting tone, and can use hook-flash to switch between callers. This option is only for analog extensions, for IP extensions you have to configure this feature directly on the IP phones.
- **Allow Been Spied:** Enable this option to allow other extension users the ability to spy on the phone calls of this extension by using feature codes.
- **Pickup Group:** Define a pickup group for this extension, extensions in the same pickup group can help pickup an incoming call on other ringing extensions in the same pickup group using feature code *8. Available values are from 0 to 63.
- **Mobility Extension:** An external number can be specified here e.g. your mobile phone number. If you now call the IPPBX using the mobile phone specified, you will hear a dial tone and will have full access to the IPPBX system functionalities just as a standard extension user does.
- **Mobility Extension Number:** When “Mobility Extension” as described above is enabled, enter your external phone number here.

VoIP Settings

- **NAT:** Check this option if extension user or the phone is located behind a router.
- **Transport:** Choose UDP, TCP or TLS as the transport protocol for SIP signaling.

- **SRTP:** Secure Real-time Transport Protocol(SRTP) encrypts the RTP traffic to secure your VoIP phone calls. Before enabling this option you need to ensure the end point can also support SRTP.
- **Qualify:** Asterisk sends a SIP OPTIONS command regularly to check that the device is still online.
- **Remote Extension:** By activate this option to enable this extension number to be able to be registered from Internet.
- **DTMF Mode:** Defines how the system detects DTMF tones, the default setting is rfc2833, it can be changed if necessary.
- **Permit IP:** Defines which IP address or network address is allowed to register to this extension number, other addresses will be unable to register. Addresses can be private or public IP Addresses.

Video Options

- **Video Call:** Tick the checkbox to enable video call support. Supported video codecs are H.261, H.263, H.263+, H.264, VP8.

Audio Codecs

LAVoice V2 supported audio codecs are ulaw, alaw, G.722, G.726, G.729, GSM, Opus and Speex. Enable the ones you require by moving the audio codecs to the “Allowed” column.

2.7.2 Search Extension

If there are too many extensions on the extensions page, it is difficult to locate a single extension number to edit its properties, you can search by specifying the extension number and clicking “Search” button.

Extension: [Search](#) [Show All](#)

[New User](#)
[Batch Add](#)
[Edit Selected](#)
[Delete Selected](#)

Extensions

	Name	Extension	Port	Protocol	DialPlan	Outbound CID	Options
<input type="checkbox"/>	1 405	405	--	SIP	DialPlan1		Edit

2.8.3 Edit Properties of Multiple Extensions

Tick the checkboxes of the extensions you wish to edit, and click “Edit Selected” button and you are able to edit the options as below:

Edit Selected X

General

Password: DialPlan:

Voicemail

Enable: Password:

Delete VMail: Email(Fax/Voicemail):

Other Options

Web Manager: Agent:

Pickup Group:

VoIP Settings

NAT: Transport: SRTP:

Qualify:

DTMF Mode: Permit IP:

Video Options

Video Call: H.261 H.263 H.263+ H.264 VP8

Audio Codecs

alaw
ulaw
g722
g729

Disallowed **Allowed**

If configured here, the selected extensions will have the same properties with the exception of the extension numbers.

Notice:

Here you are configuring mutual parameters for the selected extensions, if you provide an IP address here in the "Permit IP" field, then only the unique endpoint with this IP can register to these extensions. Only consider this if these selected extensions are for an individual gateway or a remote office, otherwise please do not configure here or please specify a network address.

2.7.4 Upload/Download Extensions

The upload/download extensions feature can be used to backup or bulk add extensions of the IPPBX system using text files. Supported file formats are CSV and TXT.

Click on the “Upload/Download Extensions” tab on *Basic->Extensions* page and you will see the menu as below:

Upload/Download Extensions

[Extensions](#) [Upload/Download Extensions](#)

Upload Extensions

Please choose file to upload: No file chosen

Download Extensions Template

Extensions Template

Right Click here to Save as Template File (.csv)

Right Click here to Save as Template File (.txt)

Download Extensions(.csv)

[Download Extensions](#)

- **Upload Extensions:** Here you can upload .csv or .txt file to generate extensions.
- **Download Extensions Template:** Here you can download a template file in .csv or .txt format. Inside there are examples which you can follow to add your desired new extensions in the same format. Once complete, the new file can then be used to upload to LAVoice IPPBX system to generate new extensions.
- **Download Extensions(.csv):** Here you can download the existing extensions in the system for backup. The downloaded CSV file can be used for extension list recovery.

3 IPPBX Basic

3.1 Trunks

A trunk on an IPPBX system is essential for extensions to be able to make outbound phone calls. LAVoice LVX30V2 IPPBX system supports FXO, GSM and VoIP trunks for outbound calls. LVX100sV2 support FXO,GSM, WCDMA and VoIP trunks.

If your IPPBX system is equipped with FXO ports then you can attach the PSTN (Public Switched Telephone Network) lines to the FXO ports for the extensions to make phone calls through the local telephone company. If your system is equipped with GSM modules or WCDMA modules then you are able to make phone calls through your mobile carrier network.

3.1.1 VoIP Trunks

Asterisk PBX can register as a SIP user agent to a SIP proxy (provider). If you have subscribed to a VoIP service from an ITSP, then with the account details provided by them you can configure a VoIP trunk on your LAVoice IPPBX system for the user extensions to share and make outbound phone calls.

Navigate to web menu *Basic->Trunks*. Click “[New VoIP Trunk](#)” button and complete the account details provided to setup the trunk as in the example below.

New VoIP Trunk [X]

Description: International

Protocol: SIP ▼

Peer Mode:

Host: gw1.sip.us :5060

Maximum Channels*: 0

Prefix: _____

Outbound CID: _____

Trunk Outbound CID Preferred:

Without Authentication

Username: 525274xxxx

Authuser: 525274xxxx

Password: *****

Advanced Options

From Domain: gw1.sip.us Insecure: port,invite

From User: 525274xxxx Qualify(sec): 2

DID Number: _____ Transport: UDP ▼

DTMF Mode: RFC2833 ▼ NAT: SRTP:

Auto Fax Detection:

Context: Default ▼ Language: Default ▼

Audio Codecs

ulaw alaw G.722 G.729 G.726 GSM Speex opus

Video Codes

H.261 H.263 H.263+ H.264 VP8

- **Description:** A name for this trunk.
- **Protocol:** SIP or IAX2 protocol.

- **Peer Mode:** If enabled then Host blank will be hidden. Peer mode requires only that the authorization matches rather than the IP address.
- **Host:** The SIP server domain or IP address.
- **Maximum Channels:** Maximum calls that can be made through this trunk at the same time, 0 means unlimited.
- **Prefix:** The prefix number you enter here will be added in front of any number you dial via this trunk. This feature is seldom required so please leave this field blank.
- **Caller ID:** The number you want to display to the called party.
- **Without Authentication:** If the service provider doesn't require a username and password for this account to register to their server then you can enable this option.
- **Username:** Username provided by VoIP Provider.
- **Authuser:** The optional authorization user for the SIP server
- **Password:** Password provided by VoIP Provider.

Advanced Options

- **Domain:** Your service provider's domain name.
- **Insecure:** Default value is "port, invite" ; "port"--Allow matching of peer by IP address without matching port number; "invite"-- Do not require authentication of incoming INVITEs.
- **From User:** fromuser=yourusername; Many SIP providers require this.
- **Qualify(sec):** Asterisk sends a SIP OPTIONS command regularly to check that the device is still online. Default value is 2(sec).
- **DID number:** Self defined, and can be used to setup number DID.
- **Transport:** Default transport type for SIP messages.
- **DTMF Mode:** Used to inform the system how to detect the DTMF(Dual Tone Multi Frequency) key press. Choices are inband, rfc2833, or info. By default we use RFC2833.
- **NAT:** With this option enabled, Asterisk may override the address/port information specified in the SIP/SDP messages, and use the information (sender address) supplied by the network stack instead. This feature is often required when there is a firewall located between the PBX and the service provider.
- **Context:** Custom dial plan for this trunk, by default it uses the "default" dial plan. Configure only if this trunk is for branch office integration, so calls coming from the other side can dial out from this IPPBX trunk directly. DO NOT change unless you fully understand how this feature works.
- **Language:** You can choose a desired language of the system voice prompts to play to the incoming calls from this trunk. For example, if the call is not answered or the user is busy the IPPBX system will notify the caller to leave a voice message in the language you set.
- **Audio Codecs:** Select the audio codec/codecs the provider can support.
- **Video Codecs:** If the ITSP supports video calls then you can enable compatible video codecs here for video phone calls.

With the exception of configuration options related to your service provider and your account details, please do not change the trunk advanced parameters if you are not familiar with them. After the SIP trunk is successfully added you can see it listed here on this page.

List of Trunks					New VoIP Trunk
	Provider Name	Type	Hostname/IP	Username	Options
1	International	SIP	gw1.sip.us	525274xxxx	Edit Delete

By clicking “Edit” you can modify the trunk settings and by clicking “Delete” you can remove this trunk from the IPPBX system.

3.1.2 FXO and GSM Trunks

FXO Trunks

On the IPPBX front panel, red LED indicates the RJ11 interface is FXO. You should attach the telephone wire from your telecom socket to the FXO ports. Once connected you should be able to see the connection status on *Operator* page “FXO/FXS/GSM Ports” section.

FXO/FXS/GSM Ports				
Status	Signal Strength	Type	Port	BLF Label
Connected		FXO	1	Channel1
Connected		FXO	2	Channel2
Connected		FXO	3	Channel3
Connected		FXO	4	Channel4
Disconnected		FXO	5	Channel5
Connected		FXO	6	Channel6
Connected		FXO	7	Channel7
Connected		FXO	8	Channel8

To be able to make calls on your FXO interface you will first need to create a trunk(s). To create a trunk you need to navigate to web menu *Basic->Trunks->FXO/GSM Trunks*.

Click “New FXO/GSM Trunk” button and you’ll see the available port numbers that can be used.

New FXO/GSM Trunk X

Description: _____

Lines: **FXO:** 1 2

Prefix: _____

Advanced Options

Call Method: ▼

Busy Detection: ▼ Busy Count:

Busy Pattern: _____ Language: ▼

Input Volume: ▼ Output Volume: ▼

Caller ID Start: ▼ Caller ID Signaling: ▼

Answer on Polarity Switch: ▼ Hangup on Polarity Switch: ▼

Auto Fax Detection:

- **Description:** A name for this FXO trunk.
- **Lines:** Available FXO and GSM ports.
- **Prefix:** The prefix number you enter here will be added in front of any number you dial via this trunk. This feature is seldom required so please leave this field blank.
- **Call Method:** If in this trunk you have more than 1 FXO/GSM ports selected, then this

parameter defines how to use these ports for outbound phone calls.

- **Busy Detection:** Enable busy tone detection, it is also possible to specify how many busy tones to wait for before hanging up.
- **Busy Count:** Specify how many busy tones to wait for before hanging up, configurable only if Busy Detection is enabled.
- **Input Volume:** The volume of the incoming calls from FXO channel/channels.
- **Output Volume:** The volume of the outgoing calls from FXO channel/channels.
- **Busy Pattern:** If busy detection is enabled, it is also possible to specify the cadence of your busy signal.
- **Language:** You can choose a desired language of the system voice prompts to play to the incoming calls from this trunk. For example, if the call is not answered or the user is busy the IPPBX system will notify the caller to leave a voice message in the language you set.
- **Answer on Polarity Switch:** When enabled, FXO (FXS signaled) ports watch for a polarity reversal to mark when an outgoing call is answered by the remote party.
- **Hangup on Polarity Switch:** In certain countries, a polarity reversal is used to signal the disconnection of a phone line. If the "hangup on polarity switch" option is selected, the call will be considered "hung up" on a polarity reversal.

When creating a FXO trunk, if you are not competent with the advanced options then please do not configure or change the default values.

GSM Trunk

If you have ordered GSM or WCDMA modules for your LAVoice LVX100sV2 IPPBX, the user extensions will be able to make and receive phone calls from the mobile network.

You first have to insert SIM cards into the SIM slots of the GSM/WCDMA modules and then install the modules to the LVX100s IPPBX module slots. Antennas should be properly installed and placed in open space for better signal reception. After completing the above, power on the LVX100s IPPBX and you'll be able to configure GSM/WCDMA trunks in exactly the same way as you configure FXO trunks.

GSM Specification

Module	Working Frequencies
LVX-M2G	GSM/GPRS 850/900/1800/1900MHz

3.2 Outbound Routes

Outbound Routes allow you to define a set of dial rules that tell your LAVoice IPPBX which Trunks (phone lines) to use when people dial external telephone numbers. A simple installation will direct LAVoice IPPBX to send all calls to a single trunk. However, a complex setup could have for

example an outbound route for emergency calls, another outbound route for local calls, another for long distance calls, and perhaps even another for international calls.

With all of the above possibilities, you may have to configure several trunks on your LAVoice IPPBX system and therefore you will need to configure several dial rules and maybe also several dial plans.

3.2.1 Dial Rules

Navigate to web menu *Basic->Outbound Routes->DialRules*.



By default there are no existing dial rules configured in the IPPBX system. You need to click “New DialRule” button to add a new dial rule.

New DialRule [X]

Rule Name: Domestic

PIN Set: forDialRule Record in CDR:

Call Duration Limit: _____ seconds

Time Rule:

Place this call through:

Available Trunks: Custom Pattern: 9XXX.

Selected Trunks: fxo(FXO/GSM)

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

Delete 1 digits prefix from the front and auto-add digit _____ before dialing

Save Cancel

- **Rule Name:** A name for this dial rule.
- **PIN Set:** A collection of PIN codes for granting outbound phone calls. See [chapter 4.13](#).
- **Record in CDR:** Record the PIN codes used for outbound phone calls along with the user extension number and the dialed numbers in to the call logs.
- **Call Duration Limit:** Specify the maximum call time using this dial rule.

- **Time Rule:** Set a time condition when this dial rule can be used.
- **Available Trunks:** All existing trunks in the IPPBX system.
- **Selected Trunks:** Trunk/Trunks that can be used by this dial rule.
- **Custom Pattern:** Dial patterns act like a filter for matching numbers dialed with trunks. The various patterns you can enter are similar to Asterisk's definition of them:
 - X — Refers to any digit between 0 and 9
 - N — Refers to any digit between 2 and 9
 - Z — Any digit that is not zero. (E.g. 1 to 9)
 - . — Wildcard. Match any number of anything. Must match **something**.
- **Delete ___ digits prefix from the front and auto-add _____ digit before dialing:** The first blank allows you to strip some digit/digits before dialing out, here if required, you need to complete the number of digits to delete. The second blank is to prepend some digit/digits before dialing out, here you need to fill in the exact number of digits to be added in front of the dialed number. For example a user dialing 912345678 using the dial rule example above, the prefix 9 at the first digit will be removed, and 00 will be added, so eventually the number called will actually be 0012345678.

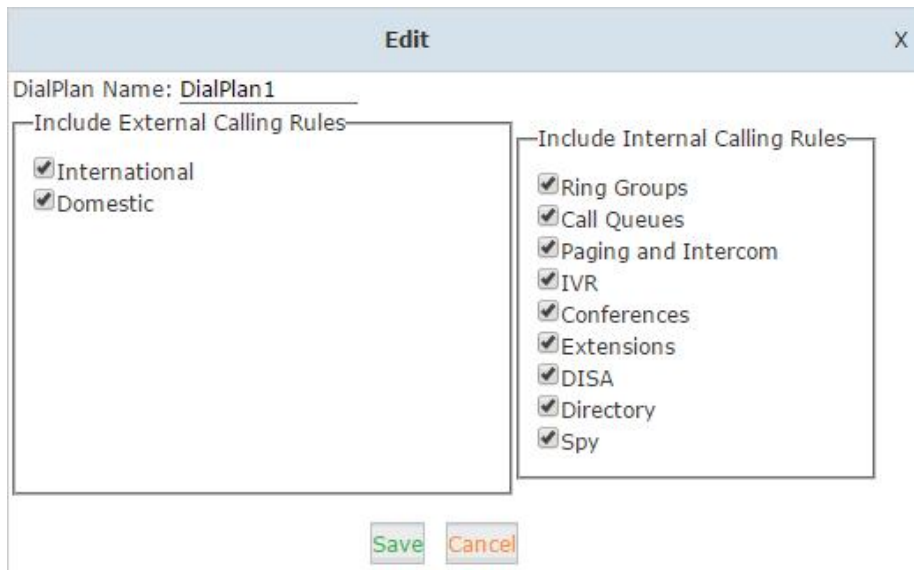
3.2.2 Dial Plans

Navigate to web menu *Basic->Outbound Routes->DialPlans*.

DialPlans

Default		DialPlan Name	Rules	Options
<input checked="" type="checkbox"/>	1	DialPlan1	Ring Groups, Call Queues, Paging and Intercom, IVR, Conferences, Extensions, DISA, Directory, Spy	Edit Delete

A default dial plan already exists in the IPPBX system. For most installations you just have to click “[Edit](#)” button on the default dial plan “[DialPlan1](#)” and tick on all dial rules to enable them, now extension users will be able to call any destinations using the trunk lines of the IPPBX system.



Calling rules in the left column are for external calls and calling rules in the right column are for internal calling. If you want to restrict some users from calling out through specific trunk lines or you don't want them to be able to call certain internal destinations, you can create a new dial plan by clicking the “[New DialPlan](#)” button.



In the new dial plan you should disable the rules you don't want others to use and save. After this, go to the extension configuration page and give the extension a different dial plan which ensures the restrictions you made take effect.

3.3 Inbound Control

The Inbound Control section is where you define how LAVoice IPPBX system handles incoming calls. Typically, you determine the phone number that outside callers have called (DID Number) and then indicate which extension, Ring Group, Voicemail, or other destination to which the call should be directed.

3.3.1 Inbound Destinations

A call destination in LAVoice IPPBX system might be an IVR menu that instructs the callers to press certain digits to route their calls, a queue to wait for specific telephone services, a ring group to call a number of user extensions, or virtually any other type of process to route the call in whatever way is desired. A call may have several destinations throughout its lifespan.

Below is a list of call destinations available in LAVoice IPPBX system:

- Extension
- Voicemail
- IVR
- Ring Group
- Paging Group
- Conference
- Call Queue
- DISA
- Time Rule
- FAX
- Dial By Name
- Hangup

3.3.2 IVR

IVR, or interactive voice response, is responsible for the menus people hear and respond to when they call up a company or business and hear the words for example: "press 1 for sales, press 2 for marketing, press 0 to speak to the operator".

IVR Prompts

To configure an IVR menu on LAVoice IPPBX system you'll first need to record your IVR prompts, these IVR prompts will communicate to the callers the menu options that they have e.g. press one for sales.

Navigate to web menu: *Inbound Control->IVR Prompts*

On this page you can delete the default voice prompts and click "New Voice" button to record a new voice prompts from a designated extension.



The image shows a "New Voice" dialog box with the following fields and values:

File Name:	office_hours
Format:	WAV (16-bit) ▼
Extension used for recording:	402 ▼

Buttons: Record, Cancel

Click "Record" button and the extension will ring, pickup the extension and speak to record your

message. Once recording is complete your voice prompts will be listed on this page.
There is another way to add voice prompts to the system, click "Upload Voice Prompts" tab.

Upload IVR Prompts

[IVR Prompts](#) [Upload IVR Prompts](#)

Upload IVR Prompts

Note: The sound file must be mp3, wav(16bit/8000Hz/Mono), gsm, ulaw or alaw!
The size is limited in 15MB!

Please choose file to upload: No file chosen

Here you can select a pre-recorded voice prompts file from your operating system to upload and once complete your file will be listed on *Voice Prompts* page. Now you can use your file to setup your personalized IVR menu.

IVR menu

Navigate to web menu: *Inbound Control*->*IVR*.

Click "New IVR" button to add an IVR menu.

New IVR X

IVR Settings

Name: Extension:

Welcome Message

Please Select: [Custom Prompts](#)

Repeat Loops:

Timeout:

Dial other Extensions: [\(Custom\)](#)

Keypress Events

Key	Action
0	Goto Extension ▼ 401(401) ▼
1	Goto Ring Group ▼ sales ▼
2	Goto Ring Group ▼ marketing ▼
3	Disabled ▼
4	Disabled ▼
5	Disabled ▼
6	Disabled ▼
7	Disabled ▼
8	Disabled ▼
9	Disabled ▼
*	Disabled ▼
#	Disabled ▼
t	Goto Extension ▼ 401(401) ▼
i	Goto Extension ▼ 401(401) ▼

Let's look at the above example where your IVR message says "Press 1 for sales, press 2 for marketing, press 0 for operator". If the caller is on the IVR menu, and after they hear the voice prompts they press 1 then the sales ring group will ring, if 2 is pressed then the Marketing ring group will ring, if 0 is pressed then will the IVR will ring the operator extension.

IVR Settings

- **Name:** Name for this IVR menu.
- **Extension:** Extension number for the IVR, by calling this number you can access the IVR menu.

Welcome Message

- **Please Select:** Select a voice prompts for this IVR menu.
- **Custom Prompts:** Click this button to navigate to *Inbound Control->IVR Prompts* page for new voice prompts.
- **Repeat Loops:** Define how many times to play the IVR menu to the caller.
- **Timeout:** Timeout for key pressing of each IVR loop.
- **Dial other Extensions:** If enabled, the caller can dial extension numbers directly when in the IVR.
- **Custom:** By clicking “Custom” you can set a dial plan for this IVR menu and the callers on the IVR will be able to dial other destinations that the dial plan allows.(Not recommended)
- **Key Press Events:** Define which destination to go by pressing a key on the phone keypad. If undefined keys are pressed then they will be handled by the “i” parameter, “i” which means invalid. And “t” stands for timeout, after all IVR loops are completed without the caller pressing any key then the incoming call will be handled by “t” parameter.

3.3.3 Ring Group

In a ring group, an incoming call will ring the phones of everyone in the group at the same time.

To configure a ring group please navigate to web menu: *Inbound Control->Ring Groups*.

Click “New Ring Group” button to add a ring group.

The screenshot shows a configuration window titled "Edit - sales" with a close button (X) in the top right corner. The window is divided into several sections:

- Name:** sales
- Strategy:** RingAll (dropdown menu)
- Ring Group Members:** A list box containing 403(SIP) 403, 404(SIP) 404, 405(SIP) 405, and 406(SIP) 406.
- Available Channels:** A list box containing 401(SIP) 401, 402(SIP) 402, 407(SIP) 407, 408(SIP) 408, 409(SIP) 409, 410(SIP) 410, 411(SIP) 411, and 412(SIP) 412.
- Navigation:** Four arrow buttons (double left, single left, single right, double right) are positioned between the two list boxes.
- Label:** A text field with the value "Label:".
- Extension for this ring group:** 640
- Ring (each/all) for lasting time(sec):** 20
- If not answered:** A list of radio buttons:
 - Goto Extension
 - Goto Voicemail
 - Goto Ring Group
 - Goto IVR
 - Hangup
- Buttons:** "Save" (green) and "Cancel" (orange) buttons at the bottom center.

The extensions in the “Available Channels” column can be added to the ring group as a ring group member.

- **Name:** Name for this ring group.

- **Strategy:** Defines how to ring the group members; selecting “RingAll” will ring all the member extensions at the same time, selecting “Ring In Order” will ring the member extensions one by one.
- **Ring Group Members:** The extensions selected to be the members of the ring group.
- **Available Channels:** All available extensions/channels can be added to the ring group.
- **Label:** Extensions can be members of multiple ring groups and therefore by giving each ring group a different label, if an incoming call rings a ring group the label will be displayed on the phone screen along with the caller ID. Therefore a ring group member will know which ring group the call is coming from.
- **Extension for this ring group:** Reach the ring group member by calling this extension.
- **Ring (each/all) for lasting time(sec):** Ring duration of the group members.
- **If not answered:** Defines a destination to redirect incoming calls to if no one answers from within the ring group.

3.3.4 Call Queue

A call queue places incoming calls in line to be answered while extension users are busy with other calls. The queued calls are distributed to the next available extension user in the order received. Once a call queue has been created, it can be assigned to specific extensions and configured to feature greetings, messages, and hold music.

To configure a call queue please navigate to web menu *Inbound Control->Call Queue*.

There are 3 existing call queues pre-configured and all you have to do is click “Edit” button to configure them. If you require more call queues then click “New Call Queue” to add more.

New X

Call Queue Reference:

Queue Number: Label:

Ring Strategy: ▼

Agents:

You do not have any users defined as agents!
[click here](#) to manage users.

Queue Options:	Announcements:
Agent TimeOut(sec): <input type="text" value="15"/> Auto Pause: <input type="checkbox"/> Wrap-Up-Time(sec): <input type="text" value="10"/> Max Wait Time(sec): <input type="text"/> Max Callers: <input type="text" value="8"/> Join Empty: <input type="checkbox"/> Leave When Empty: <input type="checkbox"/> Auto Fill: <input checked="" type="checkbox"/> Report Hold Time: <input type="checkbox"/>	<p>Caller Position Announcements</p> Frequency(sec): <input type="text" value="30"/> Announce Hold Time: <input type="text" value="No"/> ▼ <p>Periodic Announcements</p> Repeat Frequency(sec): <input type="text" value="0"/> Announcements Prompt: <input type="text"/> ▼ <p>If not answered</p> Destination: <input type="text" value="Hangup"/> ▼

Here we can see in the “Agents” field there are no available agents to be assigned to the call queues. Click “[click here](#)” and you’ll be redirected to the extension page to determine which extensions will be employed as call queue agents.

Tick the checkbox of the extension numbers which will be employed as call queue agents, then click “[Edit Selected](#)” button and tick the “Agent” option in the “[Other Options](#)” section.

Other Options

Web Manager: Agent:

Pickup Group:

Save and go back to *Inbound Control->Call Queues* page again and now you will be able to configure the existing call queues and add new call queues with available agents.

Edit [X]

Call Queue Reference:

Queue Number: Label:

Ring Strategy:

Agents:

406 407 408 409 410

Queue Options:	Announcements:
Agent TimeOut(sec): <input type="text" value="15"/> Auto Pause: <input type="checkbox"/> Wrap-Up-Time(sec): <input type="text" value="10"/> Max Wait Time(sec): <input type="text"/> Max Callers: <input type="text" value="8"/> Join Empty: <input type="checkbox"/> Leave When Empty: <input type="checkbox"/> Auto Fill: <input type="checkbox"/> Report Hold Time: <input type="checkbox"/>	Caller Position Announcements Frequency(sec): <input type="text" value="30"/> Announce Hold Time: <input type="text" value="Yes"/> Periodic Announcements Repeat Frequency(sec): <input type="text" value="0"/> Announcements Prompt: <input type="text"/> If not answered Destination: <input type="text" value="Hangup"/>

- **Queue Number:** Define an extension number to identify the queue.
- **Label:** Define the label for the queue. A user can be an agent of multiple queues, by giving a label for the call queue, if an incoming call is distributed to an agent the label will be displayed on the phone screen along with the caller ID. So a call queue agent knows which call queue the call is coming from.

Ring Strategy

- **RingAll:** Ring all available agents until one answers(default)
- **RoundRobin:** Starting with the first agent, ring the extension of each agent in turn until the call is answered.
- **LeastRecent:** Ring the extension of the Agent who has least recently received a call
- **FewestCalls:** Ring the extension of the Agent who has taken the fewest number of calls.
- **Random:** Ring the extension of a random Agent.

- **RRmemory**: RoundRobin with Memory, like RoundRobin above, except instead of the next call starting with the first agent, the system remembers which extension was last called and begins the round robin with the next agent .
- **Agent**: Check each agent that you want to be a member of this specific Call Center Queue.
- **Agent TimeOut(sec)**: Specify the number of seconds to ring an agent's extension before sending the call to the next Agent (based on Ring Strategy).
- **Auto Pause**: If an Agent's extension rings and the Agent fails to answer the call, automatically pause that agent to stop them receiving further calls from the queue.
- **Wrap-Up-Time(sec)**: This is the amount of time in seconds that an agent has to complete work on a call after which the call is disconnected. (Default is 0, which means no wrap-up time.)
- **Max Wait Time(sec)**: Calls that have been waiting in the queue for this number of seconds will be sent to the "If not answered" destination.
- **Max Callers**: Max number of callers who are allowed to wait in the queue. (Default is 0, which means unlimited.) when the maximum number of callers in the queue is reached, subsequent callers will be sent to the "If not answered" destination.
- **Join Empty**: Allow callers to enter the Queue when no Agents are available. If this option is not defined, callers will not be able to enter Queues without available agents - callers will be sent to the "If not answered" destination.
- **Leave When Empty**: If this option is selected and calls are still in the queue when the last agent logs out, the remaining callers in the Queue will be transferred to the "If not answered" destination. This option cannot be used with Join Empty simultaneously.
- **Auto Fill**: Callers will be distributed to Agents automatically.
- **Report Hold Time**: Report the hold time of the next caller for Agent when the Agent is answering the call.
- **Frequency(sec)**: Repeat frequency to announce the hold time for callers in the Queue. ("0" means no announcement).
- **Announce Hold Time**: Announce the hold time. Announce (yes), do not announce (no) or announce once (once), There will be no announcement when the hold time is less than 1 minute.
- **Repeat Frequency(sec)**: Interval time to play the voice menu for callers. ("0" means do not play).
- **Announcement Prompt**: Select an IVR prompt to be used as the Announcements Prompt.

3.3.5 Time Based Rules

Many businesses have fixed working hours where they know for example that they are only open Monday to Friday between 9am and 6pm and will be closed for business at all other times. Time conditions in LAVoice IPPBX allow you to control what happens to inbound calls both during and

outside normal business hours.

Navigate to web menu: *Inbound Control*->*Time Based Rules*.

Click on the “*Time Settings*” tab, you may create a new time rule or edit the example one, just specify the business hours during the workdays.

Edit X

Rule Name: office time

Time Settings

Sun:	Start Time: 00 : 00	Add	Delete	
	End Time: 00 : 00			
Mon:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			14:00-18:00
Tue:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			14:00-18:00
Wed:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			14:00-18:00
Thu:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			14:00-18:00
Fri:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			14:00-18:00
Sat:	Start Time: 00 : 00	Add	Delete	09:00-12:00
	End Time: 00 : 00			

Save Cancel

After the business hours have been specified, you may also want to specify the holidays of the company, on which the company will be closed for holidays.

Please click on the “*Holiday Settings*” tab, and click on the “*New Time Rule*” button to specify all the holidays on which the office will be closed.

New Time Rule X

Rule Name: my holidays

Holiday Settings

(YYYY/MM/DD HH:MM - YYYY/MM/DD HH:MM)

2017/04/30 00:00 - 2017/05/03 23:59

Start Date: Apr 30 2017 Start Time: 00 : 00 Add

End Date: May 3 2017 End Time: 23 : 59 Delete

Save Cancel

You may add your holidays one by one by specifying the start date and time and the end date and time of the holidays.

Once the business hours and holidays all have been configured, you need to define the inbound rules according to the time conditions that you have configured. Please click on the “*Time Based*”

Rules” tab. And click on “Edit” button of the existing time rule.

The screenshot shows a dialog box titled "Edit" with a close button (X) in the top right corner. Inside the dialog, the "Rule Name" is "TimeRule". Below this is a section titled "Time Settings" containing a "Time Rule" dropdown menu set to "office time". Underneath is a "Destination:" section with two rows: "If time matches:" with a dropdown set to "IVR -- working time", and "If time does not match:" with a dropdown set to "IVR -- closed time". Below the "Time Settings" is a section titled "Holiday Settings" with "Holidays:" set to "my holidays" and "Destination:" set to "Voicemail 801". At the bottom of the dialog are two buttons: "Save" (green) and "Cancel" (orange).

In the “Time Rule” dropdown list select the time rule you have defined for business hours. And in the “Destination” section specify where to route the inbound calls during and out of the business hours you have defined.

If you also defined the holidays of the company, you may select the holiday set in the “Holidays” dropdown list and set a destination for the inbound calls on holidays.

Once done, you will need to direct all inbound calls to the time rule instead of any other destinations, please go to *Inbound Control->Inbound Routes* page, set the inbound call destination as “Goto Time Rule” and then select the exact time rule that you have defined, then all inbound calls will be routed according to the time conditions you have configured.

The screenshot shows the "General" tab of a configuration page. At the top, there are four tabs: "General" (highlighted in orange), "Port DIDs", "Number DIDs", and "DOD Settings". Below the tabs are two sections: "From FXO/GSM Channels" and "From VoIP Channels". Each section has a "Distinctive Ring Tone:" label followed by a text input field, and a "Destination:" label followed by two dropdown menus. In both sections, the first dropdown is set to "Goto Time Rule" and the second is set to "Time Rule -- TimeRule". At the bottom of the page are two buttons: "Save" (green) and "Cancel" (orange).

3.3.6 Office Closed Timing

Office closed timing is an extending of time based rules, you can manually activate office closed timing by feature code. This feature allows much more flexible time conditions to be temporarily

applied for the offices which may have some unscheduled businesses and activities off the time table of the time based rule/rules.

For example, the office opens in the morning but there's an event in the afternoon and by then nobody will be able to answer phone calls. You can direct the inbound calls to an extension's voicemail or the closed time IVR.

Enable Office Closed Timing	
Enable Office Closed Timing: *81	Disable Office Closed Timing: *081
Destination:	<input type="text" value="Voicemail 800"/>
<input type="button" value="Save"/>	<input type="button" value="Cancel"/>
Status: Enable	

- **Enable Office Closed Timing:** By dialing the feature code on the phone, you can activate office closed timing. (Default is *81)
- **Disable Office Closed Timing:** By dialing the feature code on the phone, you can deactivate office closed timing. (Default is *081)
- **Destination:** The destination of the inbound calls while office closed timing is activated. It needs to be pre-configured before you can use this feature.
- **Save:** Save the settings of office closed timing.
- **Cancel:** Cancel the settings.
- **Status:** Status of office closed timing, "Enabled" or "Disabled".

3.3.7 Inbound Routes

General

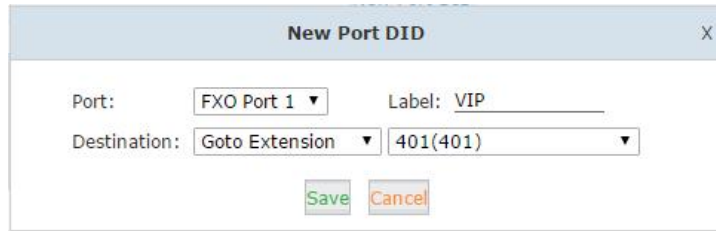
For both FXO channels and VoIP channels, you can define default inbound destinations. If you don't want the inbound calls to always go to an IVR menu, ring group or extension, then you can use a time rule to handle the inbound calls.

From FXO/GSM Channels	
Distinctive Ring Tone:	<input type="text" value=";info=domestic"/>
Destination:	<input type="text" value="Goto Time Rule"/> <input type="text" value="Time Rule -- TimeRule"/>

From VoIP Channels	
Distinctive Ring Tone:	<input type="text" value=";info=international"/>
Destination:	<input type="text" value="Goto Time Rule"/> <input type="text" value="Time Rule -- TimeRule"/>

Port DIDs

If some of the FXO/GSM ports are dedicated to a specific calling service and you want them handling differently to your generic service then you can configure “Port DIDs” here.



New Port DID	
Port:	FXO Port 1
Label:	VIP
Destination:	Goto Extension
	401(401)
Save Cancel	

For the above example, all inbound calls from FXO port 1 will be directed to extension number 401. General inbound control will still work with other ports which have not been configured with port DIDs.

Number DIDs

Number DID is only for inbound control of VoIP channels and not FXO channels. If you have a VoIP trunk for outbound and inbound phone calls, then your service provider will issue you with a DID number with which people can call you on.

Click “Number DIDs” tab and click “New Number DID” button to add a number DID rule:



New Number DID	
DID Number:	51097214
Label:	Inquiry
Destination:	Goto Extension
	410(410)
Save Cancel	

In this example, if the caller calls your DID number 51097214 the call will go directly to extension 410, general inbound control will not work with this DID number. If you experience problems setting inbound DID then please check with your service provider to confirm the exact DID number that the service provider is passing to the LAVoice IPPBX.

DOD Settings

DOD is also known as direct outward dialing, by specifying the number of an external caller in the LAVoice IPPBX system, when this caller calls in, this call can be directed to a destination directly without restriction of time rule or IVR.

Click DOD Settings tab and click New DOD to add a record.



New DOD	
DOD Number:	02885337096
Destination:	Goto Extension
	405(405)
Save Cancel	

For this example, if the caller 02885337096 calls the office number, the call will go directly to extension 405.

4. IPPBX Advanced

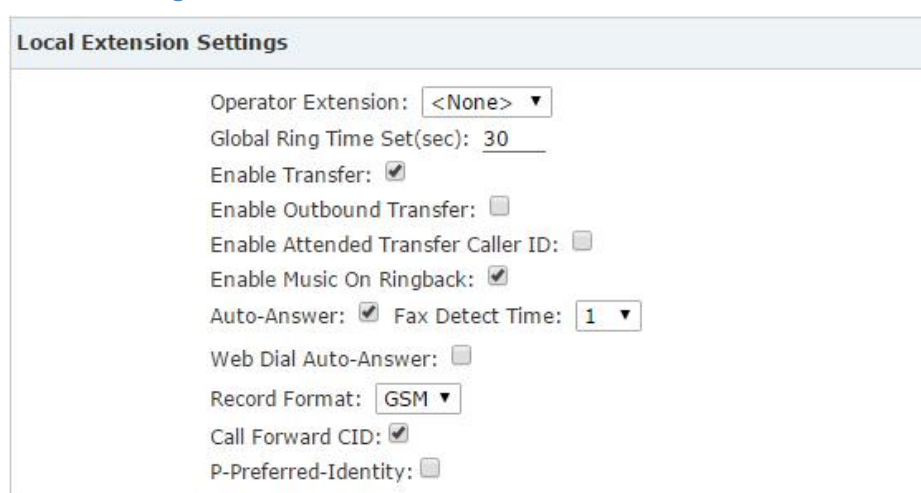
4.1 Global IPPBX Advanced Settings

4.1.1 General

Navigate to web menu *Advanced->Options->General*.

Here on this page you can configure some global options for all user extensions. In the “Local Extension Settings” section you can view the below options that can be configured.

Local Extension Settings



Local Extension Settings	
Operator Extension:	<None> ▼
Global Ring Time Set(sec):	30
Enable Transfer:	<input checked="" type="checkbox"/>
Enable Outbound Transfer:	<input type="checkbox"/>
Enable Attended Transfer Caller ID:	<input type="checkbox"/>
Enable Music On Ringback:	<input checked="" type="checkbox"/>
Auto-Answer:	<input checked="" type="checkbox"/>
Fax Detect Time:	1 ▼
Web Dial Auto-Answer:	<input type="checkbox"/>
Record Format:	GSM ▼
Call Forward CID:	<input checked="" type="checkbox"/>
P-Preferred-Identity:	<input type="checkbox"/>

- **Operator Extension:** Choose an extension to be operator extension. When an incoming call has been directed to voicemail, then by pressing ‘0’ the caller will be put through to the operator extension.
- **Global Ring Time Set(sec):** If not specifically configured, an incoming call will ring the extension for the time given here.
- **Enable Transfer:** If enabled, the extension users will be able to perform call transfers.
- **Enable Outbound Transfer:** If enabled, the outbound calls will be able to be transferred.
- **Enable Attended Transfer Caller ID:** Normally if you use feature code *2(This will be introduced in [chapter 4.19](#)) to transfer a call to another extension, the extension user only sees your extension number as caller ID but not the actual caller ID, by enabling this option the real caller ID will be passed to the user extension.
- **Enable Music On Ringback:** If enabled, callers will hear music instead of ringback tone when calling other extensions.
- **Auto-Answer:** Auto-answer enables the IPPBX to automatically answer the inbound calls from analog ports.
- **Fax Detect Time:** If auto-answer enabled, you are able to configure the fax auto detection time here.

- **Web Dial Auto-Answer:** Enable/disable auto answer of the extension numbers while dialing from Web GUI.
- **Record Format:** Choose GSM or WAV as the call recording format.
- **Call Forward CID:** Allow passing the real caller ID to the forwarded number.
- **P-Preferred-Identity:** The P-Preferred-Identity header is used among trusted SIP entities (typically intermediaries) to carry the identity of the user sending a SIP message as it was verified by authentication.

Default Settings for New User

Default Settings for New User

SIP: IAX2: Web Manager: Call Waiting:
 Agent: Voicemail: Delete VMail: VM Password: 1234
 NAT: Transport: UDP+TCP SRTP:
Audio Codecs
ulaw alaw G.722 G.729 G.726 GSM Speex Opus

In this section, options are defined for the creation of new extensions. If you have one of the options enabled, then so will any newly created extensions.

Extension Preferences

Extension Preferences

User Extensions 401 to 499
 Conference Extensions 900 to 909
 IVR Extensions 610 to 629
 Queue Extensions 630 to 639
 Ring Group Extensions 640 to 659
 Paging Group Extensions 660 to 679
 Web Extensions 680 to 699

The user extension number and system extension number ranges are defined here to avoid any conflicts within the LAVoice V2 IPPBX system. You can modify these number ranges according to your requirements.

4.1.2 Global Analog Settings

Global Analog Settings are used for configuring the LAVoice V2 IPPBX system to seamlessly work with the telephone lines from your telecommunications provider.

Navigate to web menu *Advanced->Options->Global Analog Settings*.

Caller ID Detect

Caller ID Detect

Caller ID Detection:

Caller Name:

Caller ID Signaling: Bell-US ▼

Caller ID Start: Ring ▼

CID Buffer Length: 2500 ▼

Ring Debounce: 64 ▼

DTMF Hits Begin: 2 ▼

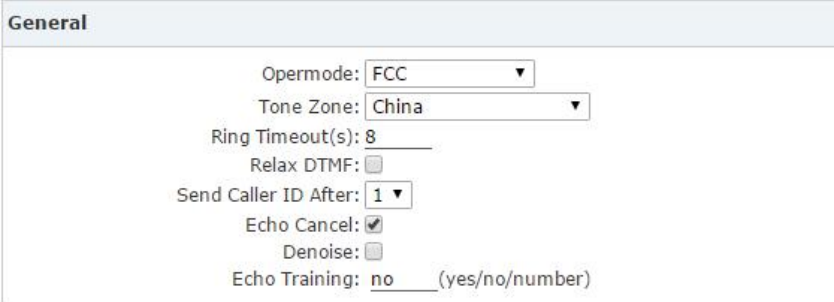
DTMF Misses End: 3 ▼

Detect Caller ID After: 1 ▼

These options are used to teach the LAVoice IPPBX system how to detect caller identity (caller ID) from the PSTN lines on FXO ports.

- **Caller ID Detection:** Enable/Disable Caller ID Detection.
- **Caller Name:** In some countries/regions caller name can be passed through the PSTN lines, by enabling this option the caller name will be received by the IPPBX system along with the caller ID.
- **Caller ID Signaling:** The signaling type applied on the PSTN lines to pass caller ID.
 Bell-US—Also known as Bellcore FSK. Used in the Canada, China, Hong Kong and US.
 DTMF—Dual Tone Multi-Frequency. Used in Denmark, Finland and Sweden.
 V23—Mostly used in UK.
 V23-Japan—Mostly used in Japan.
- **Caller ID Start:** Defines when the caller ID starts.
 Ring—Caller ID starts when a ring is received.
 Polarity—Caller ID starts when polarity reversal starts.
 Polarity(India)—Can be used in India.
 Before Ring—Caller ID starts before a ring received.
- **CID Buffer Length:** The buffer length can be used to store caller ID info.
- **Ring Debounce:** Sets the minimum time in milliseconds to debounce extraneous ring events.
- **DTMF Hits Begin:** Sampling matching value of DTMF caller ID digits, you can choose 1 to 5 digits been matched then to consider it as part of the Caller ID.
- **DTMF Miss End:** Sample matching value of DTMF caller ID digits, you can choose 1 to 5 digits been mismatched then to consider it's not part of the caller ID.
- **Detect Caller ID After:** Sets the IPPBX to detect Caller ID after how many rings been detected.

General



General

Opermode:

Tone Zone:

Ring Timeout(s):

Relax DTMF:

Send Caller ID After:

Echo Cancel:

Denoise:

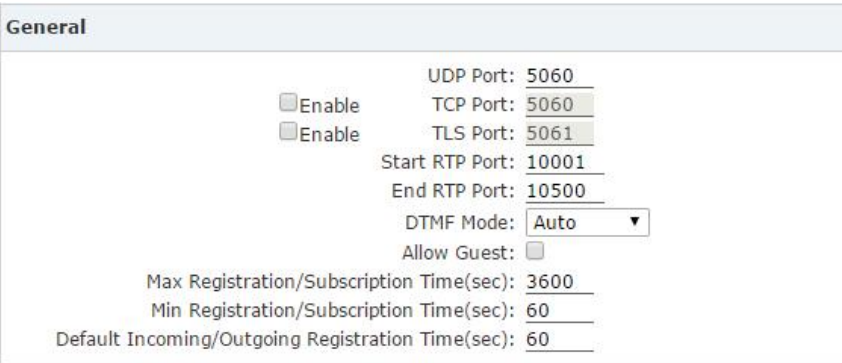
Echo Training: (yes/no/number)

- **Opermode:** Set the Opermode for FXO Ports.
- **ToneZone:** Select the tone zone of your country.
- **Ring Timeout(s):** FXO (FXS signaled) devices must have a timeout to determine if it should hang up before the line is answered. This value can be tweaked to shorten how long it takes before DAHDI considers a non-ringing line to have hung-up.
- **Relax DTMF:** Helps DTMF signal detection.
- **Send Caller ID After:** Certain countries (UK) have ring tones with different ring tones (ring-ring), which means the caller ID needs to be set later on, and not just after the first ring, as per the default (1).
- **Echo Cancel:** Enable/Disable software Echo Cancel algorithm.
- **Denoise:** The denoise parameter will help on noise reduction of the noisy analog lines, especially when gains have been increased on the lines.
- **Echo Training:** Enabling echo training will cause the PBX system to mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo. Value may be "yes", "no", or a number of milliseconds to delay before training (default = 400). This option does not apply to hardware echo cancellers.

4.1.3 Global SIP Settings

Global SIP settings allow you to configure some general and advanced options for the IP-PBX system global SIP preferences. Navigate to web menu *Advanced->Options->SIP Settings*.

General



General

Enable

UDP Port:

Enable

TCP Port:

TLS Port:

Start RTP Port:

End RTP Port:

DTMF Mode:

Allow Guest:

Max Registration/Subscription Time(sec):

Min Registration/Subscription Time(sec):

Default Incoming/Outgoing Registration Time(sec):

- **UDP Port:** SIP over UDP service port. By default Lava IPPBX system uses UDP as SIP transmission protocol. Port number can be changed here if required.
- **TCP Port:** By ticking the “Enable” checkbox you can enable global SIP TCP support. To register a SIP extension over TCP protocol, you’ll have to select TCP transport on the extension configure page, please refer to [chapter 2.8.1](#).
- **TLS Port:** By ticking the “Enable” checkbox you can enable global SIP TLS support. To register a SIP extension over TLS protocol, you’ll have to select TLS transport on the extension configuration page, please refer to [chapter 2.8.1](#).
- **Start RTP Port/End RTP Port:** The UDP ports used by LAVoice IPPBX system to carry RTP voice stream. Do not change the port numbers or you may encounter audio issue with phone calls.
- **DTMF Mode:** The DTMF mode specifies how touch tones will be transmitted to the other side of the call. Possible values for this field are rfc2833, inband, info, and auto.
- **Allow Guest:** This setting determines if anonymous callers are permitted to place calls to the LAVoice IPPBX system. For security precautions please do not enable this option.
- **Max Registration/Subscription Time(sec):** Maximum allowed time of incoming registrations and subscriptions (seconds).
- **Min Registration/Subscription Time(sec):** Minimum length of registrations/subscriptions.
- **Default Incoming/Outgoing Registration Time(sec):** Default length of incoming/outgoing registration.

NAT Support

When the LAVoice IPPBX system is behind a NAT device and needs to communicate to the outside. It needs to know whether it is talking to someone "inside" or "outside" of the NATted network. For example, if you are going to deploy remote extensions you have to tell the LAVoice IPPBX system which network address/addresses are from inside and which are from outside. Below is an example configuration.

NAT Support	
External IP:	<u>117.176.159.157</u>
External Host:	<u>117.176.159.157</u>
External TCP Port:	<u> </u>
External TLS Port:	<u> </u>
External Refresh(sec):	<u>10</u>
Local Network Address:	<u>192.168.1.0/24</u>
Local Network Address:	<u> </u>
Local Network Address:	<u> </u>

- **External IP:** Your static public IP address or domain name.
- **External Host:** This is similar to “External IP” except that the hostname is looked up every "External Refresh" seconds(default 10's).
- **External TCP Port:** Port number of SIP signaling with TCP transport protocol on the public network.
- **External TLS Port:** Port number of SIP signaling with TLS transport protocol on the public

network.

- **External Refresh(sec):** The refresh interval of the “External Host”.
- **Local Network Address:** Your local network address/addresses.

Notice:

If you have one-way audio or no audio issue on the remote extensions then this most probably means that NAT support is not properly configured. Please check your configurations here.

Type of Service

Asterisk supports different QoS settings at the application level for various protocols on both signaling and media. The Type of Service (TOS) byte can be set on outgoing IP packets for various protocols. The TOS byte is used by the network to provide some level of Quality of Service (QoS) even if the network is congested with other traffic.

Type of Service	
TOS for Signaling packets:	CS3 ▼
TOS for RTP audio packets:	ef ▼
TOS for RTP video packets:	AF41 ▼
COS Priority for Signaling packets:	3 ▼
COS Priority for RTP audio packets:	5 ▼
COS Priority for RTP video packets:	4 ▼
DNS SRV Look Up:	<input type="checkbox"/>
Relax DTMF:	<input checked="" type="checkbox"/>
RTP TimeOut(sec):	_____
RTP Hold TimeOut(sec):	_____
Add 'user=phone' to URI:	<input type="checkbox"/>
UserAgent:	VOIP _____

- **TOS for Signaling Packets:** Sets TOS for SIP packets.
- **TOS for RTP audio packets:** Sets TOS for RTP audio packets.
- **TOS for RTP video packets:** Sets TOS for RTP video packets.
- **COS Priority for Signaling packets:** Sets 802.1p priority for SIP packets.
- **COS Priority for RTP audio packets:** Sets 802.1p priority for RTP audio packets.
- **COS Priority for RTP video packets:** Sets 802.1p priority for RTP video packets.
- **DNS SRV Look Up:** Enable DNS SRV lookups on outbound calls.
- **Relax DTMF:** Relax DTMF handling.
- **RTP TimeOut(sec):** Terminate call if there is 60 seconds of no RTP or RTCP activity on the audio channel when we're not on hold. This feature enables the ability to hangup a call in the case of a phone disappearing from the network, for instance if the phone loses power.
- **RTP Hold TimeOut(sec):** Terminate call if 300 seconds of no RTP or RTCP activity on the audio channel when on hold.
- **Add 'user=phone' to URI:** Enable this option if the SIP provider requires ";user=phone" on URI.
- **UserAgent:** Allows you to change the user agent string. The default user agent string also contains the Asterisk version. If you don't want to expose this, change the user agent string

here.

Outbound SIP Registrations

The “Outbound SIP Registrations” configures the register behaviors of LAVoice IPPBX system when registering as a client to the other SIP servers.

Outbound SIP Registrations
Register TimeOut(sec): <input type="text" value="30"/>
Register Attempts: <input type="text" value="10"/>

- **Register TimeOut(sec)**: Retry registration every 30 seconds (default).
- **Register Attempts**: Number of registration attempts before the IPPBX system give up. Default is 10 and 0 means continue forever.

4.1.4 Global IAX Settings

Navigate to web menu *Advanced->Options->IAX2 Settings*.

General
UDP Port: <input type="text" value="4569"/>
Bandwidth: <input type="text" value="low"/>
Max Registration/Subscription Time(sec): <input type="text" value="1200"/>
Min Registration/Subscription Time(sec): <input type="text" value="60"/>

- **UDP Port**: IAX2 signaling and media port, the default is 4569.
- **Bandwidth**: Specify bandwidth of low, medium, or high to control which codecs are used in general.
- **Max Registration/Subscription Time(sec)**: Maximum amount of time that IAX peers can request as a registration expiration interval (in seconds).
- **Min Registration/Subscription Time(sec)**: Minimum amount of time that IAX peers can request as a registration expiration interval (in seconds).

4.2 Virtual Fax

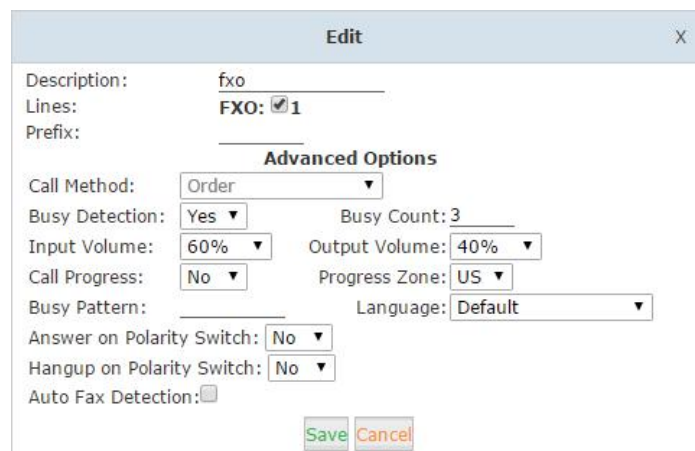
LAVoice IPPBX system has the ability to auto detect incoming faxes and send the received faxes to a user's email box. If you don't wish to send the fax by email then faxes can be saved to a user's extension account.

Notice:

Please enable Virtual Fax services on *Virtual Fax* page first, and then follow the instructions below to configure.

4.2.1 Receive Fax

LAVoice IPPBX system detects incoming faxes from the trunks. To configure LAVoice IPPBX to auto detect incoming faxes please navigate to web menu *Basic->Trunks*.



Click on “[Edit](#)” to edit the trunk(either analog or VoIP trunk) that you want to configure fax auto detection on. Find the “[Auto Fax Detection](#)” option and tick the checkbox. You’ll see a dropdown list from which you can select any extensions to direct the detected faxes to.



If you want the IPPBX system to send the received faxes to an email address(Fax to Email) then please select an extension number starting with “[Virtual Fax](#)”. Then navigate to *Basic->Extension* page to specify the email address in “[Email\(Fax/Voicemail\)](#)”section.

If you require that the received fax is stored in the IPPBX system only then you should select a virtual fax extension without specifying the email address.

Finally, if you want the incoming fax to be handled by a fax machine, please select the extension number assigned to the fax machine.

Notice:

If you are configuring Fax to Email, you also have to configure the SMTP service before it will work. Please refer to [chapter 4.3.3](#).

4.2.2 Send Fax

To send a fax you must first login to the LAVoice IPPBX web interface with an extension number and the voicemail password for this extension. Before doing this please ensure this extension has the “Web Manager” option enabled on the extension configure page.

Other Options
Web Manager:

After login, navigate to the *Send Fax* page.

Send Fax

Send Fax Fax Log

Send Fax

Destination: 02885337096

Send fax must be .tif, .tiff, .txt, .pdf, .jpg or png.

Please choose file to upload: invoice.tif

Enter the fax number and click on “Choose File” to locate the file you are planning to send, upload the file and then send the fax.

There are some optional options for outbound faxes, please navigate to web menu *Advanced->Virtual Fax*.

Virtual Fax

Virtual Fax

Enable:

Country Code: 86

Area Code: 28

Outbound CID: 85337096

Label: Zycoo Co., LTD.

Fax Seat: 4

DialPlan: DialPlan1

- **Enable:** Enable Virtual fax feature for receiving and sending faxes.
- **Country Code:** Enter your country code here.(Optional)
- **Area Code:** Enter your Area Code here.(Optional)
- **Outbound CID:** Only works if the outbound fax is to be sent through VoIP trunks. The other

side receives your fax with this number.

- **Label:** Define custom information to be printed to the header of the fax pages.
- **Fax Seat:** Defines how many users can send fax at the same time.
- **DialPlan:** A dial plan to send faxes.

4.3 VoiceMail

4.3.1 General Voicemail Options

Voice mail allows callers to leave messages for subscribers (user extensions) of the IPPBX system when they are unable to answer the incoming calls.

VoiceMail Reference

VoiceMail Reference	
Max Greeting Time(sec):	30
Dial "0" for Operator:	<input checked="" type="checkbox"/>

- **Max Greeting Time(sec):** Maximum voicemail box greeting message duration.
- **Dial "0" for Operator:** If this option is enabled then callers will be able to dial "0" to transfer out of voicemail to the Operator.

Voice Message Options

Voice Message Options	
Message Format:	WAV (16-bit) ▼
Maximum Messages:	100 ▼
Max Message Time(min):	2 ▼
Min Message Time(sec):	2 ▼

- **Message Format:** The audio file format to be used for the recording.
- **Maximum Messages:** The maximum amount of voice messages for each extension.
- **Max Message Time(min):** The maximum time duration of an individual voicemail message.
- **Min Message Time(sec):** The minimum time duration of an individual voicemail message.

Default minimum duration is 2 seconds, which means voice messages which are less than 2seconds will be ignored by the IPPBX system.

Playback Options

Playback Options	
<input checked="" type="checkbox"/>	Say Message CallerID
<input checked="" type="checkbox"/>	Say Message Duration
<input type="checkbox"/>	Play Envelope
<input type="checkbox"/>	Allow Users to Review

These options are for voicemail message playback.

- **Say Message CallerID:** Announce the Caller ID of the caller who left this message before playing the voice message.
- **Say Message Duration:** Announce the message duration before playing the voice message.

- **Play Envelope:** Announce the date, time and caller ID for the voicemail message.
- **Allow Users to Review:** If enabled, this option will allow users to review the voice message.

4.3.2 Playback Voicemail on the phone

Navigate to web menu *Advanced->Feature Codes*.

On this page, you'll find two feature codes that can be used for checking voicemail.

Voicemail Main Menu: *60

Check Extension Voicemail: *61

Dial *60 and you will enter the main menu of voicemail feature, by specifying the extension number and voicemail password of the required extension then you can check its voicemail and you can do this for any extension by following the system voice guidance.

By dialing *61 from an extension and entering the voicemail password for this extension you can follow the voice guidance to check voicemail of your own extension. Or alternatively, you can configure some advanced options for your voicemail box.

4.3.3 Voicemail to Email

To send received voicemail messages to the user's email box, you need to configure SMTP support, Email format and specify email addresses for the extension users.

Step1:

SMTP Settings

Navigate to web menu: *Advanced->SMTP Settings*.

Define an email account to be used by the system which will send emails with voicemail messages attached to the extension users' email boxes.

SMTP Settings

SMTP Settings:

SMTP Server:

Port:

SSL/TLS:

Enable SMTP Authentication

Username:

Password:

- **SMTP Server:** SMTP server domain, for example: smtp.gmail.com, smtp.tom.com.
- **Port:** Default SMTP service port is 25, but if you are using SSL/TLS then please use port 465.
- **SSL/TLS:** Encrypts a communication channel between the LAVoice IPPBX system and the SMTP server.
- **Enable SMTP Authentication:** If your SMTP server requires authentication then please enable this option and configure the following.
- **Username:** The email account.

- **Password:** The password for this email account.
- **Send Test:** Click “Send Test” to send a test email to see if SMTP is working correctly. If it is working then you’ll receive an email sent by the IPPBX system.

Step 2:

Email Settings

Navigate to web menu: *Advanced->Voicemail->Email Settings.*

On this page you can define the email content that will be sent to the extension users’ email boxes.

Email Settings

General **Email Settings**

Template for Voicemail Emails

Attach voicemail to email

Sender Name IP Phone System

From faxservice@gmail.com

Subject New Voicemail from \${VM_CALLERID}

Message Hello \${VM_NAME}, you received a message lasting \${VM_DUR} at \${VM_DATE} from, (\${VM_CALLERID}).

Template Variables: \${VM_NAME} : Recipient's first name and last name
 \${VM_DUR} : The duration of the voicemail message
 \${VM_MAILBOX} : The recipient's extension
 \${VM_CALLERID} : The Caller ID of the person who left the message
 \${VM_MSGNUM} : The message number in your mailbox
 \${VM_DATE} : The date and time the message was left

- **Attach voicemail to email:** If enabled, the system will send any voice message files received to the extension users’ email box.
- **Sender Name:** Alias for the SMTP email account.
- **From:** The email account from SMTP settings.
- **Subject:** The subject of the email sent by LAVoice IPPBX system.
- **Message:** The content of the email, describes the details of the voicemail message received.
- **Template Variables:** These variables can be used to acquire details of the voicemail messages, which can then be used in the message field to compose the email content.

Step3:

Email Address

Go to the extension details for the user and specify the email address where messages for this user should be sent.

Edit [X]

General

SIP: IAX2:
 Name: 401 Extension: 401
 Password: 123456 Outbound CID:
 DialPlan: DialPlan1 Analog Phone: None

Voicemail

Enable: Password: 1234
 Delete VMail: Email(Fax/Voicemail): example@gmail.com

Once these 3 configuration steps are complete, if user extension 401 receives a new voicemail message then the IPPBX system will send this voicemail message to example@gmail.com.

4.3.4 Playback Voicemail from Web GUI

An extension user can login to the web interface with their extension number and voicemail password if “Web Manager” option is enabled on their extensions.

Navigate to *Voicemail List* page.

Voicemail [↗](#)

Field: New Field: New

List of Voicemail Files				Duration(sec)	Options
<input type="checkbox"/>	Caller ID	Date			
<input type="checkbox"/>	1 "403" <403>	Sat Jan 2 13:47:31 2010	15	<input type="button" value="Play"/>	<input type="button" value="Delete"/> <input type="button" value="⌵"/>

Here on this page you can see all newly received voice messages displayed.

By clicking “Play” button you will be presented with a dialog box that gives you two options to playback this message.

Play [X]

Type 1:

Type 2: Extension used for playing: 401

By clicking button you can playback this message directly from the web interface. By selecting an extension number and clicking on the “Play” button you can playback this message from the selected extension.

4.4 Conference

Conferences allow two or more callers to be joined together so that all parties on the call can hear one another. Conferences are also referred as Conference Bridges or Conference Rooms.

On LAVoice V2 IPPBX system, you can create up to 20 conference rooms. There are 3 default conference rooms preconfigured for you.

4.4.1 Static Conference

Navigate to web menu *Advanced->Conference*. You can click “[New Conference](#)” button to add a new conference room or click “[Edit](#)” button on the existing conference room to change the properties.

Edit		X
Conference Number		
Room Extension:	900	
Conference Password		
Guest Password:	1234	
Administrator Password:	2345	
Conference Options		
Conference DialPlan	Internal	▼
<input type="checkbox"/>	Play hold music for first caller	
<input type="checkbox"/>	Enable caller menu	
<input type="checkbox"/>	Announce callers	
<input type="checkbox"/>	Record conference	
<input type="checkbox"/>	Quiet Mode	
<input type="checkbox"/>	Close the conference when last administrator exits	
<input type="checkbox"/>	Leader Wait	
Save		Cancel

Conference number

- **Room Extension:** Call this extension number to enter the conference room.

Conference Password

- **Guest Password:** If callers use this password to enter the conference then they are ordinary participants.
- **Administrator Password:** If callers use this password to enter the conference then they are administrators and have advanced conference menu features such as inviting people to participate in the conference.

Conference Options

- **Conference DialPlan:** Conference admin can use this dialplan to invite other participants.
- **Play hold music for first caller:** Plays the hold music for the first participant in the conference until another participant enters the conference.

- **Enable caller menu:** Check this option to allow the conference admin to access the conference menu by pressing “*” on the phone.
- **Announce Callers:** Announce all the participants in the room when a new participant enters the conference room.
- **Record Conference:** Record this conference (Recording format is wav). The recorded conference can be searched within *Report->Record List->Conference* page. Please see [chapter 6.3.2](#).
- **Quiet Mode:** If this option is checked then the system will not give any announcement when participants enter or leave the conference.
- **Close the conference when last administrator exits:** If this option is checked then the conference will be terminated when the last administrator exits.
- **Leader Wait:** Wait until the conference leader (administrator) enters the conference before starting the conference.

4.4.2 Dynamic Conference

LAVoice IPPBX system allows you to press a key sequence (feature code) to create a conference during a live call.

Please navigate to web menu *Advanced->Feature Codes*. You can see the feature codes available for conference feature.

Conferences	
Invite Participant:	0
Create Conference:	*0
Return to conference with participant:	**
Return to conference without participant:	*#

- **Invite Participant:** When in a static conference room or a dynamic conference room, if the conference administrator presses 0 they will get a dial tone to invite others to participate in this conference.
- **Create Conference:** During a live call the extension user can press *0 to create a dynamic conference room. The other side will automatically enter the conference as an ordinary participant while the extension user who created this conference will be requested to enter the conference password to enter.
- **Return to conference with participant:** While using the conference menu to invite other people, you can dial ** to return to the conference with invited party.
- **Return to conference without participant:** If the invited party doesn't want to participate in the conference you can press *# to return to the conference without the invited party.

Notice:

After a dynamic conference is created, in reality you have entered a static conference room (by default 900 is the first available conference room). You are able to use conference admin menu to invite others to the conference also others can dial 900 to enter this conference.

4.5 Music Settings

Music Settings, or Music On Hold(MOH) as it is more commonly known on an IPPBX system allows audio files (such as WAV or MP3 files) to be uploaded to the IPPBX system and played back when a caller is placed on hold or is waiting in a queue.

Navigate to web menu *Advanced->Music Settings*.

Music Settings

Music Settings Music Management

Music On Hold Reference

Music: Music 1 ▼

Music On Ringback Reference

Music: Music 2 ▼

Music On Queue Reference

Music: Music 3 ▼

Save Cancel

- **Music On Hold Reference:** Audio files in this selected folder will play to the party which is on hold.
- **Music On Ringback Reference:** Audio files in this folder will be played instead of playing ringback tone to the caller.
- **Music On Queue Reference:** Audio files in this folder will be played when the caller is waiting in a call queue.

There are 10 folders for music files, by default the first 3 folders are preloaded with music files which you may wish to choose. However, if you want to upload your own audio files please click “Music Management” tab.

Music Management

Music Settings Music Management

Music Management

Select Music Directory: Music 1 ▼ Load

Files: ▼ Delete

Upload Music File

Select Music Directory: Music 1 ▼

Note: The sound file must be mp3, wav(16bit/8000Hz/Mono), gsm, ulaw or alaw!
The size is limited in 15MB!

Please choose file to upload: Choose File No file chosen

Upload

In the Music Management section, you can select a music folder and click “Load” button to check which audio files are inside this folder. By clicking “Delete” button you can delete the existing audio files.

In the Upload Music File section, you can select a music folder and browse your PC file system to select your preferred audio file and click “Upload” button to upload the audio file. If there are more than one audio file in the same music folder, they will be played at random.

Notice:

LAVoice IPPBX system can adopt MP3, wav(16bit, 8000Hz, mono), gsm, ulaw and alaw audio file format.

4.6 DISA

Direct inward system access(DISA) allows an outside caller to dial directly into the PBX system and access the system's features and facilities remotely.

This is useful if you want people to be able to for example take advantage of the low rate for international calls that you have available on your system, or to allow outside callers to be able to use the paging or intercom features of the system. Always protect this feature with strong password/passwords, the passwords need to be set on *Advanced->Pin Sets* page which will be introduced in [Chapter 4.13](#).

Navigate to web menu *Advanced->DISA*. Click on “New DISA” button to create a new DISA call target.

The screenshot shows a web-based configuration window titled "New DISA". The window contains several input fields and a checkbox. The "Name" field is set to "international". The "PIN Set" is a dropdown menu currently showing "forDISA". There is a "Without PIN" checkbox. The "Record in CDR" checkbox is checked. The "Response Timeout(sec)" is set to "10". The "Digit Timeout(sec)" is set to "5". The "Extension for this DISA(Optional)" is set to "678". Below these fields is a section titled "Allow Outbound Route" with a "Select DialPlan" dropdown menu set to "DialPlan1". At the bottom of the window are "Save" and "Cancel" buttons.

- **Name:** Alias of the DISA call target.
- **PIN Set:** A set of pin codes used to authorize all callers using the system features and facilities.
- **Without PIN:** If enabled, callers will not be required to enter any pin code to be able to use the system features (Not recommended).
- **Record in CDR:** The pin code that is used will be stored into call logs and can therefore be traced on *Report->Call Logs* page.
- **Response Timeout(sec):**The maximum waiting duration before hanging up if the dialed number is incomplete or invalid. Defaulted 10 seconds
- **Digit Timeout(sec):**The maximum interval time between digits when typing extension number. Defaulted 5 seconds.
- **Extension for this DISA(Optional):** If you want to access DISA by dialing an extension, you can

define an extension number for this DISA.

- **Select DialPlan:** Select a dial plan for this DISA so callers will be able to make outbound phone calls using the trunks on the IPPBX system.

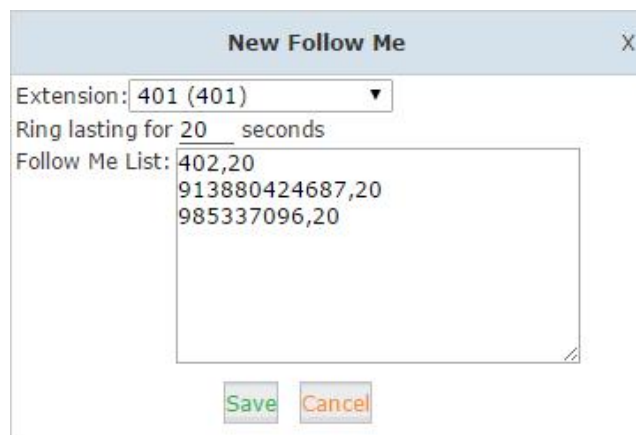
Notice:

After a new DISA is created, it can be included in the inbound control section as a call destination. But this is not recommended as it is not safe because all callers can possibly access DISA functionality. A better option is to configure DOD settings ([Chapter 3.3.7](#)) for the numbers which you want to be able to access DISA.

4.7 Follow Me

The Follow Me feature allows you to set a list of numbers that you may possibly be contacted on. Therefore, if someone calls your extension and you are not available then follow me will work through the list calling each of the numbers in turn until you are contacted or the list is exhausted.

To configure follow me, navigate to web menu *Advanced->Follow Me*. Click on “[New Follow Me](#)” to configure follow me for an extension.



The screenshot shows a web-based configuration window titled "New Follow Me". It features a dropdown menu for "Extension:" with the value "401 (401)". Below this, it indicates "Ring lasting for 20 seconds". The "Follow Me List:" is a text area containing three entries: "402,20", "913880424687,20", and "985337096,20". At the bottom of the window, there are "Save" and "Cancel" buttons.

- **Extension:** Select the extension number which will be configured with follow me.
- **Ring lasting for 20 seconds:** Define how long to ring the extension before the call is forwarded out. By default, this is 20 seconds.
- **Follow Me List:** The list of numbers to forward the calls to. Each line is written with the format “number,time”, “number” is one of the number to forward the calls to, “time” defines how long to ring this number. They are separated with a comma without space. Numbers are called in sequence.

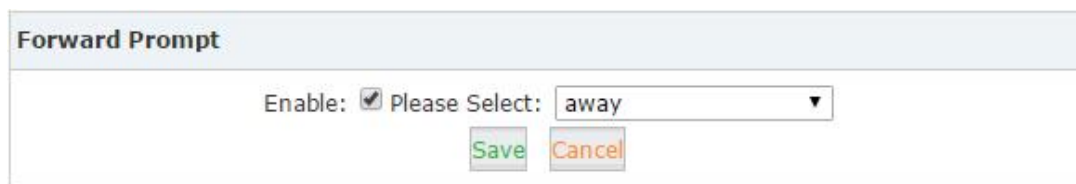
4.8 Call Forward

4.8.1 Configure from the Web

This feature allows calls to an extension to be automatically forwarded to a specific internal extension or external phone number.

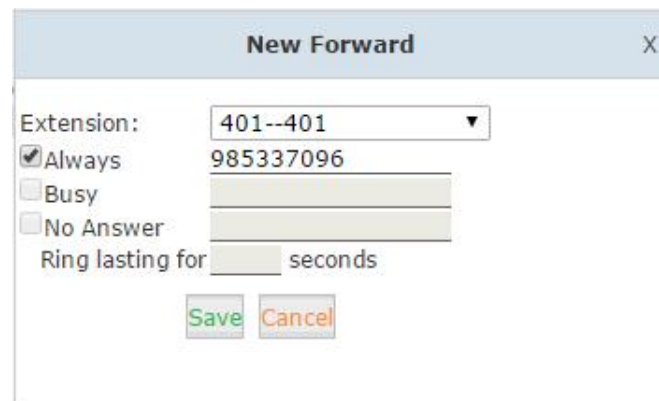
Before configuring call forward you can enable the IPPBX system to play a voice prompt before the call is forwarded. This voice prompts can be recorded or uploaded from the *Inbound Control->IVR Prompts* page.

Once the voice prompt file is ready you can navigate to web menu *Advanced->Call Forward* and enable the system to play back the voice prompt before the incoming call is forwarded.



The screenshot shows a web form titled "Forward Prompt". It contains an "Enable:" checkbox which is checked. To its right is a "Please Select:" dropdown menu with "away" selected. Below these are two buttons: "Save" (green) and "Cancel" (orange).

After the voice prompt is set, click "New Forward" button to set call forward for an extension.



The screenshot shows a modal window titled "New Forward" with a close button (X) in the top right corner. It contains the following fields and options:

- Extension: 401--401 (dropdown menu)
- Always: 985337096
- Busy: [empty text field]
- No Answer: [empty text field]
- Ring lasting for [empty text field] seconds

At the bottom are "Save" (green) and "Cancel" (orange) buttons.

- **Always:** Unconditionally forward the incoming calls.
- **Busy:** Forward the incoming calls only if the extension is busy.
- **No Answer:** Forward the incoming call only if the extension didn't answer.
- **Ring lasting for ____ seconds:** Only configurable for "No Answer" option. It defines how long to ring the extension before forwarding if the extension didn't answer.

Notice:

1. If you are forwarding a call to an external phone number then please ensure that you add a prefix in front of the number if your system requires a prefix to dial out.
2. The forward condition "Always" is mutually exclusive to "Busy" and "No Answer".

4.8.2 Configure from the Phone

Navigate to web menu *Advanced->Feature Codes*.

You'll see feature codes for call forward as follows:

Call Forward

Enable Forward All Calls:	<u>*71</u>
Disable Forward All Calls:	<u>*071</u>
Enable Forward on Busy:	<u>*72</u>
Disable Forward on Busy:	<u>*072</u>
Enable Forward on No Answer:	<u>*73</u>
Disable Forward on No Answer:	<u>*073</u>

With these feature codes, you can activate or deactivate call forward directly from your phones without configuration on the Web GUI.

For example, a LAVoice IPPBX requires prefix 9 to call outbound, and the number you want to forward the calls to is 85337096.

- Activate always call forward: Dial *71985337096, press 1 to confirm.
- Deactivate always call forward: Dial *071.
- Activate call forward on busy: Dial *72985337096, press 1 to confirm.
- Deactivate call forward on busy: Dial *072.
- Activate call forward no answer: Dial *73985337096, press 1 to confirm.
- Deactivate call forward no answer: Dial *073.

4.9 Call Transfer

Call Transfer is used to transfer a call in progress to some other destination. There are two types of call transfer.

- **Attended call transfer** - Where the call is placed on hold, a call is placed to another party, and a conversation can take place privately before the caller on hold is connected to the new destination. It is also referred to as "Supervised Call Transfer".
- **Blind call transfer** - Where the call is transferred to the other destinations without intervention (the other destination could ring out and may not be answered for instance).

Navigate to web menu *Advanced->Feature Codes*. You'll see the feature code for call transfer as below:

```
Transfer
Blind Transfer: #
Blind Transfer Callback: 
Attended Transfer: *2
Disconnect Call: *
Timeout for answer on attended transfer(sec): 15
```

- **Blind Transfer**: In a live call, an extension user can press # key and the IPPBX system prompts "Transfer", you then enter the number to transfer to, this call will be transferred instantly and the user can hangup. If the transferred number doesn't answer this call then it will ring back to the extension user.
- **Blind Transfer Callback**: Determines whether the transferred call should call back to the user who transferred it or not. If enabled and the transferred call was unanswered it will call back to the user who transferred it, if disabled and the transferred call was unanswered it will go to voicemail box of the transferred extension.
- **Attended Transfer**: In a live call, extension user can press *2 and the IPPBX system prompts "Transfer", you then enter the number to transfer to, after someone answers your call, you can introduce this call and hangup at which point the call is transferred.
- **Disconnect Call**: In an attended transfer if the other side doesn't want to take the call to be transferred, you can press * to disconnect with them and get back to the caller.
- **Timeout for answer on attended transfer(sec)**: In an attended transfer, if the third party rang for 15 seconds without answering, the extension user will go back to the caller and the transfer is terminated.

4.10 One Number Stations

One Number Stations is an innovative IPPBX feature unique to Lava IPPBX. With one number stations feature, you can have the same extension number in several different locations.

One number stations feature can put several extension numbers in the same “group”, a main number can be selected from the members and when an incoming call is made to the main number, it will ring all the member extensions including the main number. Any extension with the group calling other extensions will display only the main number.

Navigate to web menu *Advanced->One Number Stations*. Click “[New One Number Stations](#)” button to create a one number stations group.

The screenshot shows a web interface for creating a "New One Number Stations" group. It features two columns of extension numbers. The left column, labeled "ONS Group Members", contains 407, 408, and 409. The right column, labeled "Extensions", contains 403, 404, 405, 406, 410, 411, 412, and 413. Between the columns are two arrow buttons for moving items. Below the columns, there is a "Main Extension" dropdown menu set to 407 and a "Ring lasting for" field set to 20. At the bottom are "Save" and "Cancel" buttons.

Select the extensions from the “[Extensions](#)” column to the “[ONS Group Members](#)” column. In the “[Main Extension](#)” dropdown list select an extension to be the main extension number. Next click “[Save](#)” and you’ll have a new one number stations group.

In this example, no matter whether 407, 408 or 409 makes a call, other extensions only see the calling number as extension 407, while any calls made to 407 will result in all 3 extensions ringing.

As you can see on this page there’s a feature code Switch Station available.

The screenshot shows a configuration field for the "Switch Station" feature code. It consists of a text input field containing the value "*1", followed by "Save" and "Cancel" buttons.

This feature code is used to switch extension during a phone call. For example, if an inbound call called extension 407 and the one number stations member 408 answered this call, you can press *1 from extension 407 or 409 to switch this live call to 407 or 409, then 408 will be disconnected.

4.11 Paging and Intercom

The Paging and Intercom feature allows you to use your phone system as an intercom system, provided that your endpoints (phone devices) support this functionality. The Paging and Intercom feature allows you to define a number (just like an extension or Ring Group number) that will simultaneously page a group of devices. For example, in a small office, you might define a paging group that allows any user to dial 699, allowing them to page the entire office. You can also use the feature code *50/*51 to page/intercom a single extension, by dialing *50/*51 followed by the extension number.

Navigate to web menu *Advanced->Paging and Intercom*. Click “[New Paging and Intercom](#)” button to add a new paging group.

The screenshot shows a 'New' dialog box with the following fields and controls:

- Paging Extension:** 660
- Description:** managers
- Paging Group Members:** 401(SIP) 401, 402(SIP) 402, 403(SIP) 402, 404(SIP) 404, 405(SIP) 405
- Device List:** 406(SIP) 406, 407(SIP) 407, 408(SIP) 408, 409(SIP) 409, 410(SIP) 410, 411(SIP) 411, 412(SIP) 412, 412(IAX2) 412
- Duplex:**
- Buttons:** Save, Cancel

- **Paging Extension:** The extension number for this paging group, by calling this extension number you can reach the group members.
- **Description:** Description of this paging group.
- **Duplex:** If enabled, the group members can talk back to the caller.

By calling the paging extension number, all group member phones will auto answer in speaker mode (requires that the IP phones support auto answer feature), the caller can now make a brief announcement to the group members.

4.12 Web Extensions

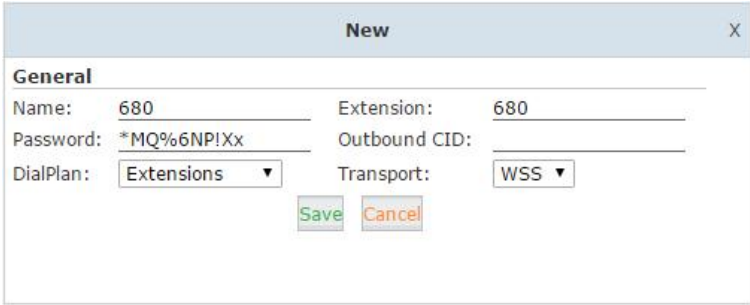
Web Extension is a new feature that makes use of WebRTC technology. You can use any web browser that supports WebRTC to register an extension number to your LAVoice V2 IPPBX system without any plugins.

To register the first Web extensions please follow the steps below:

Step 1:

Create a Web Extension

To create a web extension, navigate to web menu *Advanced->Web Extensions*. Click on “New User” button to add a new web extension.



New	
General	
Name: 680	Extension: 680
Password: *MQ%6NP!Xx	Outbound CID:
DialPlan: Extensions	Transport: WSS
Save Cancel	

- **Name:** Username of this web extension.
- **Extension:** Extension number of this web extension.
- **Password:** Password for registration of this web extension.
- **Outbound CID:** Only works if the call was placed out through VoIP trunks.
- **DialPlan:** Defines which type of numbers the web extension can dial.
- **Transport:** WS or WSS.
- **WS:** WS (WebSocket) Protocol which is an independent TCP-based protocol providing full-duplex communication channels over a single TCP connection. The WebSocket protocol was standardized by the IETF as RFC 6455 in 2011, and the WebSocket API in Web IDL is being standardized by the W3C.
- **WSS:** WSS (WebSockets over SSL/TLS), like HTTPS, WSS is encrypted and we strongly recommend the secure wss:// protocol over the insecure ws:// transport. A variety of attacks against WebSockets are almost impossible if the transport is secured.

Step 2:

Upgrade Web extension patch

As you can see, web extensions use different protocols for signaling and media (WS/WSS) and they are not ordinary SIP/IAX2 extension that can use IP phones or softphones to register so must be treated differently.

Step 3:

Register a Web Extension

After completing the upgrade process (see [chapter 8.6](#)) you can access the WebRTC extension

register interface. Open your web browser and enter URL <https://192.168.1.254:9999/webrtc> (192.168.1.254 should be your IPPBX IP address) you will see the web extension register interface. Please complete the register credentials as below:



Webphone

Name i.e. Homer Simpson
John Doe

SIP URI i.e. sip:homer@your-domain.com
680@192.168.1.254

SIP password
.....

WS URI i.e. wss://your-domain.com:8089/ws
wss://192.168.1.254:8089/ws

[advanced settings](#)

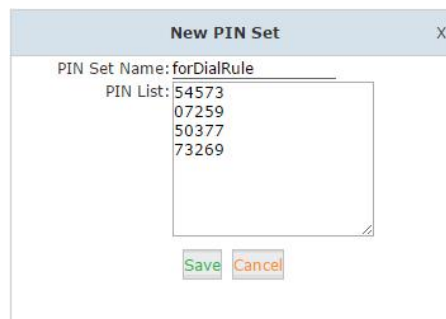
Next, press Enter and the web extension will be registered and is ready for phone calls just like any other standard extension.

WebRTC can even be adapted to the enterprise website which can help an enterprise serve their customers with direct voice communication via their website. For more advanced WebRTC settings please refer to the WebRTC manual.

4.13 Pin Sets

Pin sets can be used to secure your IPPBX system phone services and in particular for outbound dial rules and DISA.

Navigate to web menu *Advanced->PIN Sets*. Click on “**New PIN Set**” button to create a collection of PIN codes.



New PIN Set

PIN Set Name: forDialRule

PIN List: 54573
07259
50377
73269

Save Cancel

Each line is a PIN code, press Enter to add the next PIN code without any symbols.

4.14 Call Recording

LAVoice IPPBX system has built-in ability to record calls. No additional software is required for

recording calls. When LAVoice IPPBX system records a call, both sides of the call are recorded and written out to a file for playback on a computer. Call recording can be used to ensure call quality, or to keep calls for later review. LAVoice IPPBX provides the ability to record all of the calls, or to selectively record calls.

4.14.1 Record All Calls

Navigate to web menu *Advanced->Call Recording*. Click “[New Call Recording](#)” to activate call recording for the extensions you want calls to be recorded.

New Call Recording X

Extension:

401 (401) 402 (402) 403 (403) 404 (404) 405 (405) 406 (406) 407 (407) 408 (408) 409 (409) 410 (410) 411 (411)

Call Recording Time

Always Recording:

Start Time: [] : [] End Time: [] : []

Start Day: [] End Day: []

Call Recording Settings

Inbound Record: Outbound Record:

- **Extension:** Select the extensions which you want their calls to be recorded.
- **Always Recording:** If enabled, all calls from the above selected extensions will be recorded regardless when the calls were made and received.
- **Start Time, End Time, Start Day, End Day:** If Always Recording is unnecessary then you can specify which time durations in a week to record all calls from the above selected extensions.
- **Inbound Record:** Enable to record all inbound calls.
- **Outbound Record:** Enable to record all outbound calls.

The recordings can be searched on *Report->Record List->Call Recording* page. Please see [chapter 6.3.1](#).

4.14.2 One Touch Recording

One Touch Recording is also known as Record on Demand. It allows users to record phone calls selectively.

Navigate to web menu *Advanced->Feature Code*. Here on this page you can see the one touch recording feature code as below:

One Touch Recording
One Touch Recording: *1_____

In a live call conversation, an extension user can use feature code *1 to record this call. With this feature, you don't have to configure recording all calls for the extensions which may cause heavy system resource use if some call recordings are not required.

The one touch recordings can be searched from *Report->Record List->One Touch Recording* page. Please see [chapter 6.3.3](#).

4.15 Smart DID

LAVoice IPPBX system has the ability to route an inbound call directly to an extension if the extension had previously tried to call the number but the call was unanswered. It is convenient for the called party to make a call back and be directly routed to the extension that called them without going through the IVR menu or reception desk.

Navigate to web menu *Advanced->Smart DID*. Tick the "Enable" checkbox to enable Smart DID functionality.

Smart DID

Smart DID

Enable:

[Save](#) [Cancel](#)

Smart DID Rules List [New Smart DID Rule](#)

	Pattern	Strip	Prepend	Options
1	X.			Edit Delete

There is a default Smart DID rule which enables all outbound calls to be monitored by the Smart DID feature. If the call is not answered by the called party, then the called number will be stored into the Asterisk database with the extension number which made this call. If the called party does make a callback to the IPPBX system then the call can automatically be directed to the extension number.

If you don't want all outbound calls monitored by Smart DID, you can modify the existing rule or click "[New Smart DID Rule](#)" to add your custom rule/rules. An example of this is detailed below:

New Smart DID Rule X

Pattern: 17951X.

Strip: 5 digits before dialing

Prepend: +86 before dialing

[Save](#) [Cancel](#)

- **Pattern:** Defines the number format which would be dialed.
- **Strip:** Remove some digits from the front of the dialed number.
- **Prepend:** Prepend some digits in front of the dialed number after manipulated by the "Strip"

option.

The numbers to be dialed will start with prefix 17951 and if they call back, the expected numbers will have +86 in front of them instead of the 5-digit prefix 17951. In such a situation, the outbound and inbound numbers are not the same, you'll need the "Strip" and "Prepend" options to manipulate the dialed numbers to make sure it can match the "same" number when it calls back. If the numbers to be called and the numbers to be received are the same, then you don't have to configure these 2 options. Alternatively, you can configure only one of these 2 options, it will all depend on your actual requirements.

For example, the extension user 401 wishes to call 85337096, and the carrier requires a prefix 17951 to ensure the rate is much cheaper. The user will dial 1795185337096 to place this call. If the called party misses this call then the IPPBX system will store this number +8685337096 with extension number 401 into its database. Later on, if the called party tries to call back, the IPPBX system gets +8685337096 as the caller ID and matches this from its database, once successfully matched, this call will be automatically directed to extension 401.

Notice:

1. The records for Smart DID functionality in the system database will be erased every day at midnight. This means this is a dynamic effective feature and is only designed to handle callbacks made within the same day as the original call.
2. In the "Pattern" field, patterns can be used in the same way as the patterns used to manipulate dialed number in the dial rules. Please refer to [chapter 3.2.1](#).
3. Due to the mechanism of how Asterisk works, at this time Smart DID only works with VoIP trunks and does not work with FXO or GSM/ WCDMA trunks.

4.16 Callback

Callback is to allow a company employee who needs to make a call from their personal phone to call the IPPBX, the IPPBX calls them back and the cost of any future outbound calls are at the companies expense.

Navigate to web menu *Advanced->Call Back*.

Callback Number Settings

Callback Number Settings	
Enable:	<input checked="" type="checkbox"/>
Strip:	2 digits before dialing
Prepend:	0 before dialing
DialPlan:	DialPlan1
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

- **Enable:** Check the checkbox to enable call back feature.
- **Strip:** The received caller ID might have some additional digits in front of it and it will not be possible for you to call back directly, you can specify here to remove some digits before calling back.

- **Prepend:** After the number has been manipulated by the “Strip” option, you can use this option to add some extra digits in front before calling back.
- **DialPlan:** Choose an appropriate dial plan to make sure the IPPBX system has the permissions for outbound calling.

Click “New Callback Number” to add a call back number.

- **Callback Number:** The number which will be used to call into the IPPBX system and will be handled by the Callback feature.
- **Destination:** An extension or another call destination which will be used to call the callback number.

In the above example, if the caller 13880424687 called the IPPBX system, IPPBX will disconnect this call and make a call back to this number using extension 410.

In the call back destination field you can even set the destination to a conference, call queue or DISA, so the callers can access these functionalities all at the companies expense.

4.17 Phone Book

The phone book on the LAVoice IPPBX system is similar to a contact list on a cellular phone. You can add the contacts to the IPPBX system from *Advanced->Phone Book* page. To do this Click “New Contact” to create a new contact record.

- **Name:** Contact name.
- **Phone Number:** Phone number of the contact.
- **Speed Dial:** Speed dial number which can be used to call this contact from another extension.

After contacts have been created they will be listed here on this page.

Phone Book

Phone Book					Import	Export	Delete All	Sync LDAP	
The prefix of speed dial: *99					Save	Cancel			
Field: Name			Filter	Create Contact	Delete Selected				
<input type="checkbox"/>	Name	Phone Number	Speed Dial	Options					
<input type="checkbox"/>	1 John Doe	73459203	01	Call	Edit	Delete			

Here on this page you also have some additional advanced options for the phone book and LDAP configurations.

- **Import:** You can import a contact list from .txt or .csv files.
- **Export:** Export the current contact list as .csv file.
- **Delete All:** Delete all contacts.
- **Sync LDAP:** Synchronize the contacts to an LDAP server.
- **The prefix for speed dial:** Using this feature code with the speed dial code of a contact you can call the contact without knowing their exact number.
- **Filter:** Search contacts by contact name, phone number or speed dial code.
- **Create Contact:** Create a new contact record.
- **Delete Selected:** Delete the selected contacts.
- **Call:** Assign an extension to call this contact.
- **Edit:** Edit the information of this contact.
- **Delete:** Delete this contact.

4.18 LDAP Server

4.18.1 LDAP Server Settings

LDAP(Lightweight Directory Access Protocol) is an open, vendor-neutral, industry standard application protocol for accessing and maintaining distributed directory information services over an IP network. An LDAP server has been embedded into LAVoice IP PBX which is mainly used to centralized and manage the phonebook. LDAP server has generated the phonebook based on created extensions by default.

Navigate to web menu *Advanced->LDAP Server*.

LDAP Server

LDAP Server

Enable:

Username:

Password:

Domain:

Organization:

Port:

- **Enable:** Enable/Disable LDAP Service.
- **Username:** Define the username of the server administrator (e.g.: manager). This setting will be used on the IP Phone.
- **Password:** Define the password of the server administrator. This setting will be used on the IP Phone.
- **Domain:** Define a domain for the LDAP server (e.g.: ldapdomain.com). This setting will be used on the IP Phone.
- **Organization:** Define an organization to describe the members recorded by LDAP (e.g.: zycoo.ltd). This setting will be used on the IP Phone.
- **Port:** LDAP service port, the default port number is 389.

4.18.2 Synchronize Contacts with LDAP Server

Navigate to web menu *Advanced->Phone Book*. Click on the “Sync LDAP” button to synchronize contacts with LDAP server.

Phone Book

Phone Book

The prefix of speed dial:

4.18.3 LDAP Client Settings

After enabling the LDAP server, you need configure a client. For example: Lava LV-2SB IP Phone. Open the web interface of the IP Phone on your browser, navigate to web menu *Directory-LDAP*.

LDAP Name Filter	(cn=%)	?
LDAP Number Filter	((telephoneNumber=%)	?
Server Address	192.168.0.9	?
Port	389	?
Base	dc=pbx,dc=com	?
User Name	cn=admin,dc=pbx,dc=com	?
Password	*****	?
Max.Hits(1~32000)	32000	?
LDAP Display Name	cn	?
Search Delay(0~2000ms)		?
Protocol	<input checked="" type="radio"/> Version2 <input type="radio"/> Version3	?
LDAP Lookup For Incoming Call	<input checked="" type="radio"/> On <input type="radio"/> Off	?
LDAP Sorting Results	<input checked="" type="radio"/> On <input type="radio"/> Off	?

- Filled the LDAP Name Filter:

This parameter specifies the name attributes for LDAP searching. The “%” symbol in the filter stands for the entering string used as the prefix of the filter condition. For example (cn=%), when the name prefix of the cn of the contact record matches the search criteria, the record will be displayed on the IP PHONE LCD.

- Filled LDAP Number Filter:

This This parameter specifies the number attributes for LDAP searching.

- Filled Server Address: Fill the domain name or IP address of the LDAP Server.

For example: 192.168.0.124

- Port(the port of the LDAP Serve) Base, User Name, Password

- Max.Hits: the maximum number of the search results to be returned by the LDAP server.

- LDAP Display Name: the display name of the contact record displayed on the LCD screen.

- Filled the relative value and then click save button the save the settings.

Following is the example screenshot for the configuration.

4.19 Feature Codes

Feature codes allow you to set the special codes that users can dial to access various features.

Navigate to web menu *Advanced->Feature Codes*.

Call Parking

Call Parking
Extension to Dial for Parking Calls: 700
Extension Range to Park Calls: 701-720
Call Parking Time(sec): 45
Enable Call Park BLF notification:

A Parking Lot allows anyone who has received a call to park the call on an extension, allowing any other user to access the parked call. Typically, you receive the call, transfer it to extension 700, and then listen as the system tells you where you can pick up the call (usually extension 701).

Anyone else on your LAVoice IPPBX system can now dial 701 to pick-up the parked call.

A call can be parked for a maximum of 45 seconds as per the definition of “[Call Parking Time](#)”, if nobody picks this call up then it will go back to the extension which parked it.

The “[Enable Call Park BLF Notification](#)” enables the parked extensions 701-720 to be monitored by BLF keys, so if there’s a call that is parked, the extension user will be able to see it from the BLF panel.

Pickup Call

Pickup Call
Pickup Extension: *8
Pickup Specified Extension: **

Pickup call option allows users to pick up calls that are not directed to them by dialing a feature code *8 or **.

“[Pickup Extension: *8](#)” has already been introduced in [chapter 2.8.1](#), as it’s related to the pickup group option of the extension settings.

While “[Pickup Specified Extension: **](#)” can help pickup a call on any ringing extension. Dial ** followed by the extension number and you can pickup a call on a ringing extension if it is in the same pickup group as your extension or not.

Transfer

Please see [chapter 4.9](#).

One Touch Recording

Please see [chapter 6.3.3](#).

Call Forward

Please see [chapter 4.8.2](#).

Do Not Disturb

Do Not Disturb

Enable Do Not Disturb: *74
Disable Do Not Disturb: *074

With the Do Not Disturb(DND) feature enabled, an extension can make outbound phone calls but inbound calls to the extension cannot be made.

If an extension user of the LAVoice IPPBX system dials *74 from their phone, the system will play a beep sound to indicate DND has been activated.

To disable DND, simply dial *074, another beep sound will play and DND has been deactivated.

Spy

Spy

Normal Spy: *90
Whisper Spy: *91
Barge Spy: *92

Call Spy allows users to dial the spy feature codes followed by an extension number to listen to the call conversation in real-time.

- **Normal Spy:** For example, extension 410 is talking to someone on the phone, you can dial *90410 to listen to their conversation, however, neither speaker will be able to hear you.
- **Whisper Spy:** Whisper spy is also known as coaching. For example, a new employee is talking to the customer on the phone, their supervisor can dial *91 followed by the employee's extension number to listen to their conversation. The supervisor can talk to the new employee only without the customer hearing the conversation.
- **Barge Spy:** Barge spy is similar to an instant 3-way conference call. While an extension user is talking to someone else on the phone, you can dial *92 followed by their extension number to talk to both of the speakers.

Notice:

Before call spy can be used, you have to make sure the extensions to be spied, have the "Allow Being Spied" option enabled on extension settings page.

Black List

Black List

Blacklist a number: *75
Remove a number from the blacklist: *075

Black list feature allows you to create a list of numbers that are not allowed to call in to the LAVoice IPPBX system.

Any extension user can dial *75 and follow the voice prompts to add the numbers to the LAVoice IPPBX system black list.

To remove numbers from black list, you can dial *075.

Voicemail

Please see [chapter 4.3.2](#).

Conference

Please see [chapter 4.4.2](#).

Call Queues

Call Queues

Pause Queue Member Extension: *95
Unpause Queue Member Extension: *095

Call queue agents can dial *95 to suspend their extension temporarily, new calls will not be distributed to their extensions, until they dial *095 to resume.

Wakeup

Wakeup

Wakeup Advance: *55
Wakeup Add: *55*
Wakeup Delete: *055

- **Wakeup Advanced:** Advanced wakeup call menu for adding, viewing and canceling wakeup calls.
- **Wakeup Add:** Add a wakeup call directly by dialing this feature code followed by a specific date and time in 8-digit number format, for example, feature code is *55*, you can dial *55*08010730 to add a wakeup call of 7:30am on August 1st.
- **Wakeup Delete:** By dialing this code to cancel all requested wakeup calls.

Others

Others

Intercom: *50
Paging: *51
Directory: *3
Check WAN Port IP: **11
Check LAN Port IP: **12
Listen to Account Number: **13

- **Intercom:** The intercom feature code allows you to intercom one extension only. You don't have to create a "Paging and Intercom" group for only one extension if you intend to intercom with only that extension.
- **Paging:** The paging feature code allows you to page one extension only. It's the same as the intercom feature code, the only difference between paging feature code and intercom feature code is by using intercom feature code both sides can talk to each other but using paging feature code, only the caller can talk to the called party.
- **Directory:** Directory is also known as dial by name. Extension users can dial *3 and follow the voice prompts to enter the first 3 letters of another extension user's first or last name and then make a call to an extension number without knowing its extension number.
- **Check WAN Port IP:** By dialing this code you'll hear the system announce the IP address of the LAVoice V2 IPPBX WAN interface. It can be dialed on a registered IP phone or an analog phone connected to the FXS port, whether the analog phone has been assigned with

extension number or not.

- **Check LAN Port IP:** By dialing this code you'll hear the system announce the IP address of the LAVoice V2 IPPBX LAN interface. It can be dialed on a registered IP phone or an analog phone connected to the FXS port, whether the analog phone has been assigned with extension number or not.
- **Listen to Account Number:** By dialing this code you can check the extension number of your phone, either it's an IP phone or analog phone.

5. Network Settings

5.1 Network Basic

5.1.1 IPv4 Settings

LAVoice V2 IPPBX system supports static IP, DHCP and PPPoE for WAN connection, while on LAN port only static IP is supported. If you are configuring your WAN connection as static IP or DHCP, ensure WAN and LAN IP addresses are not in the same network.

Static

Navigate to web menu *Network Settings->Network->IPv4 Setting*.

Network

IPv4 Settings	IPv6 Settings	VLAN Settings
---------------	---------------	---------------

WAN Port Setup	
IP Assign:	Static ▾
IP Address:	192.168.1.254
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	8.8.8.8
Alternative DNS:	4.4.4.4

LAN Port Setup			
IP Address:	192.168.10.254	Subnet Mask:	255.255.255.0
<input checked="" type="checkbox"/> IP AddressV1:	192.168.5.254	Subnet MaskV1:	255.255.255.0
<input type="checkbox"/> IP AddressV2:		Subnet MaskV2:	

Save Cancel

By default, LAVoice IPPBX has been preconfigured with a static IP address of 192.168.1.100 and 192.168.10.100 on WAN and LAN interfaces respectively. If you want to use a static IP then configure required address here and include the address, netmask, gateway and DNS given by your ISP or network administrator.

For the LAN interface, you can specify 2 additional virtual IP addresses. These can be used to access other networks from the LAN port.

DHCP

If your Internet connection automatically provides you with a usable IP address, you can select “DHCP” on the WAN interface.

Network

IPv4 Settings	IPv6 Settings	VLAN Settings
WAN Port Setup		
IP Assign: <input type="text" value="DHCP"/>		
IP Address: <input type="text" value="192.168.1.100"/>		
Subnet Mask: <input type="text" value="255.255.255.0"/>		
Gateway: <input type="text" value="192.168.1.1"/>		
Primary DNS: <input type="text" value="8.8.8.8"/>		
Alternative DNS: <input type="text" value="4.4.4.4"/>		
LAN Port Setup		
IP Address: <input type="text" value="192.168.10.100"/>		
Subnet Mask: <input type="text" value="255.255.255.0"/>		
<input type="checkbox"/> IP AddressV1:	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> IP AddressV2:	<input type="text"/>	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

If DHCP is selected then the WAN interface will not be configurable as it obtains all its network parameters from the DHCP server. DHCP should be used cautiously as all IP extensions register to the IPPBX system through the WAN interface and as DHCP addresses can change and IP extensions need to know the address of the IPPBX at all times. It is best practice to configure WAN address with a Static IP.

PPPoE

LAVoice IPPBX can be connected to the network via ADSL modem by means of Point-to-Point Protocol over Ethernet (PPPoE)dial-up. In such a situation, extensions will subscribe to the IPPBX system through the LAN port, while WAN port can be used for remote extensions.

Network

IPv4 Settings	IPv6 Settings	VLAN Settings
WAN Port Setup		
IP Assign: <input type="text" value="PPPoE"/>		
Username: <input type="text" value="CD85335361"/>		
Password: <input type="text" value="*****"/>		
IP Address: <input type="text" value="192.168.1.100"/>		
Subnet Mask: <input type="text" value="255.255.255.0"/>		
Gateway: <input type="text" value="192.168.1.1"/>		
Primary DNS: <input type="text" value="8.8.8.8"/>		
Alternative DNS: <input type="text" value="4.4.4.4"/>		
LAN Port Setup		
IP Address: <input type="text" value="192.168.10.100"/>		
Subnet Mask: <input type="text" value="255.255.255.0"/>		
<input type="checkbox"/> IP AddressV1:	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> IP AddressV2:	<input type="text"/>	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

If PPPoE is set, you have to specify the username and password given by your ISP and the IPPBX system will dial-up to the ISP and once successfully connected, you will have Internet access on the WAN interface.

LAN port connects to your local network for internal IP extensions to register. If necessary, you can change LAN IP to suit your local network.

5.1.2 IPv6 Settings

IPv6(Internet Protocol Version 6) has been in development for nearly two decades. Now the next-generation protocol is ready to replace IPv4 and assume its place as the back of the Internet.

Today, major Internet service providers (ISPs), home networking equipment manufacturers, and web companies around the world are permanently enabling IPv6 for their products and services. Many organizations, institutions and universities have deployed their own networks on IPv6.

To be able to deliver VoIP calls over IPv6(SIP over IPv6), you can configure LAVoice IPPBX system with IPv6 addresses to be able to deploy it in your IPv6 network infrastructure.

To do this, navigate to web menu *Network Settings->Network->IPv6 Settings*.

Network

IPv4 Settings IPv6 Settings VLAN Settings

WAN Port Setup

Enable:

IPv6 Address:

Prefix Length:

Gateway:

Primary DNS:

Alternative DNS:

Save Cancel

Specify your IPv6 network profile here and you will be able to connect LAVoice IPPBX to your IPv6 network infrastructure.

5.1.3 VLAN Settings

With a layer-3 switch you can configure VLAN on LAVoice IPPBX system to divide the VoIP and data traffic. Voice VLAN can ensure that phones remain working even when the data network is congested.

To set VLAN, navigate to web menu *Network Settings->Network->VLAN*. As you can see here on this page, you are able to configure 4 VLANs, 2 each for WAN or LAN port.

Network

IPv4 Settings	IPv6 Settings	VLAN Settings
WAN VLAN 1		
Enable: <input checked="" type="checkbox"/>		
VLAN ID: <u>2</u>		
VLAN IP Address: <u>172.16.10.2</u>		
Subnet Mask: <u>255.255.255.0</u>		
WAN VLAN 2		
Enable: <input checked="" type="checkbox"/>		
VLAN ID: <u>3</u>		
VLAN IP Address: <u>172.16.20.2</u>		
Subnet Mask: <u>255.255.255.0</u>		
LAN VLAN 1		
Enable: <input checked="" type="checkbox"/>		
VLAN ID: <u>4</u>		
VLAN IP Address: <u>172.16.30.2</u>		
Subnet Mask: <u>255.255.255.0</u>		
LAN VLAN 2		
Enable: <input checked="" type="checkbox"/>		
VLAN ID: <u>5</u>		
VLAN IP Address: <u>172.16.40.2</u>		
Subnet Mask: <u>255.255.255.0</u>		

Ensure VLAN IPs for VLAN1 and VLAN2 of WAN and LAN interfaces are in several different network segments.

5.2 Static Routing

Static Routing is a form of routing that occurs when a router uses a manually-configured routing entry, rather than information from a dynamic routing protocol to forward traffic.

Navigate to web menu *Network Settings->Static Routing*. Click “New Static Routing” to add a new routing record to the system.

New Static Routing		X
Destination Network:	<u>222.209.4.1</u>	
Subnet Mask:	<u>255.255.255.255</u>	
Gateway:	<u>192.168.10.1</u>	

- **Destination:** Set the IP address of destination host or network address. E.g.222.209.4.1, 192.168.10.0.
- **Gateway:** Set the gateway address.

After the new record has been manually created you can see it listed here on this page.

List of Static Routing			New Static Routing	
	Destination Network	Subnet Mask	Gateway	Options
1	222.209.4.1	255.255.255.255	192.168.10.1	Edit Delete

You can click “Edit” button to edit one of the items, or you can delete the item by clicking the “Delete” button.

Click the “Routing Table” tab and you’ll see a detailed list of all the system routing rules, including default and custom ones.

Routing Table

Static Routing Routing Table

Routing Table:

```
Kernel IP routing table
Destination      Gateway         Genmask         Flags Metric Ref    Use Iface
0.0.0.0          192.168.10.1   0.0.0.0         UG    0     0     0 WAN
192.168.1.0     0.0.0.0        255.255.255.0   U     0     0     0 LAN
192.168.7.0     0.0.0.0        255.255.255.0   U     0     0     0 LAN
192.168.10.0    0.0.0.0        255.255.255.0   U     0     0     0 WAN
222.209.4.1     192.168.10.1   255.255.255.255 UGH   0     0     0 WAN
```

5.3 VPN

VPN(Virtual Private Network) is mainly used for setting up long-distance and/or secured network connections. When used on LAVoice IPPBX, all phone calls made and received are encrypted so it secures your remote offices/extensions' phone services. Built-in VPN Server on LAVoice series is an easy way to set up a secured connection between other LAVoice series IPPBXs or IP phones. You don't need to build a dedicated VPN server or buy a VPN router. This is also a workaround to avoid firewall issues when configuring remote VoIP client such as SIP protocol which is notoriously difficult to pass through a firewall due to its random port numbers to establish connection.

LAVoice IP PBX supports four varieties of VPN, they are L2TP, PPTP, OpenVPN and IPSec.

5.3.1 L2TP VPN

L2TP VPN Server

Layer 2 Tunneling Protocol (L2TP) is a tunneling protocol. It does not provide any encryption or confidentiality by itself. Rather, it relies on an encryption protocol that it passes within the tunnel to provide privacy. Here on the LAVoice IPPBX system we use IPSec to do the encryption.

To configure your L2TP server, navigate to web menu *Network Settings->VPN Server*. Check the radio button of L2TP to configure L2TP VPN server.

- **Enable:** Tick the checkbox to enable L2TP VPN server.
- **Remote Start IP, Remote End IP:** L2TP VPN remote network IP range, between start IP and end IP there must be less than 10 available IP addresses.
- **Local IP:** L2TP VPN local server IP address.
- **Primary DNS:** Primary DNS for VPN connection.
- **Alternate DNS:** Alternative DNS for VPN connection.
- **Authentication Method** : Select the authentication method: chap or pap.
pap: Password Authenticate Protocol, PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system.
chap: Challenge Handshake Authentication Protocol
 CHAP takes a more sophisticated and secure approach to authentication by creating a unique challenge phrase (a randomly generated string) for each authentication.
- **Debug:** Tick to enable debug for L2TP VPN connection, debug info will be written into system logs.
- **IPSec:** Enable IPSec encryption for L2TP VPN server.
- **IPSec Local IP:** LAVoice WAN IP which can access Internet.
- **IPSec Password:** Define a password for IPSec VPN client to authenticate.

Notice:

If the LAVoice IPPBX system is behind NAT, you need to open ports 500, 4500 and 1701 on the router/firewall.

For the VPN client to connect you'll need to create a VPN user account.

Click "[VPN User Management](#)" tab and click "[New VPN User](#)" button to add a VPN user account.

New VPN User [X]

Username:

Password:

Availability:

Now the L2TP VPN client can connect to the L2TP VPN server.

L2TP VPN Client

For example, in the branch office you are going to connect another IPPBX system to the head office using L2TP VPN.

Navigate to the web menu *Network Settings->VPN Client*. Check the radio button of L2TP to configure L2TP VPN client.

VPN Client

VPN Client

L2TP PPTP OpenVPN N2N IPsec

Enable:

Server Address:

Username:

Password:

IPsec:

IPsec Local IP:

IPsec Password:

Default Gateway:

- **Enable:** Tick to enable L2TP VPN client.
- **Server Address:** L2TP server public IP.
- **Username:** L2TP VPN user name given by the VPN server.
- **Password:** L2TP VPN user password given by the VPN server.
- **IPsec:** Enable IPsec support.
- **IPsec Local IP:** LAVoice IPPBX WAN IP Address that can access the Internet.
- **IPsec Password:** Set according to the password specified on the server.
- **Default Gateway:** All traffic goes through the L2TP VPN connection.

Notice:

If connection is successfully established, the system will display as follows:

Status: L2TP client VPN remote IP address 172.16.0.1

L2TP client VPN local IP address 172.16.0.x (An IP address between 172.16.0.2 and 172.16.0.9)

5.3.2 PPTP VPN

The Point-to-Point Tunneling Protocol (PPTP) uses a control channel over TCP and a GRE tunnel operating to encapsulate PPP packets. The intended use of this protocol is to provide security levels and remote access levels comparable with typical VPN products.

PPTP VPN Server

To configure your PPTP Server, navigate to web menu *Network Settings->VPN Server*. Check the radio button of PPTP to configure PPTP VPN server.

The screenshot shows the 'VPN Server' configuration page. At the top, there are radio buttons for 'L2TP', 'PPTP', 'OpenVPN', and 'IPSec', with 'PPTP' selected. Below this, there are several configuration fields: 'Enable' (checked), 'Remote IP' (172.16.0.2 - 11), 'Local IP' (172.16.0.1), 'Primary DNS' (8.8.8.8), 'Alternative DNS' (4.4.4.4), 'Timeout(sec)' (20), 'Authentication Method' (checkboxes for chap, pap, mschap, mschap-v2, with mschap and mschap-v2 checked), 'Enable mppe128' (checked), and 'Debug' (checked). At the bottom, there are 'Save' and 'Cancel' buttons.

- **Enable:** Tick the checkbox to enable PPTP VPN server.
- **Remote IP:** PPTP VPN remote network IP range, there must be 10 or less available IP addresses between start IP and end IP.
- **Local IP:** PPTP VPN local server IP address.
- **Primary DNS:** Primary DNS for VPN connection.
- **Alternative DNS:** Secondary DNS for VPN connection.
- **Timeout(sec):** Session timeout for PPTP tunnels.
- **Authentication Method:** Choose method/methods for the authentication of the VPN clients.
 - chap:** Challenge Handshake Authentication Protocol
CHAP takes a more sophisticated and secure approach to authentication by creating a unique challenge phrase (a randomly generated string) for each authentication.
 - pap:** Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system.
 - mschap:** MS-CHAP is the Microsoft version of the Challenge-Handshake Authentication Protocol.
 - mschap-v2:** Microsoft Challenge Handshake Authentication Protocol version 2 (MS-CHAP v2), this provides stronger security for remote access connections.
- **Enable mppe128:** Microsoft Point-to-Point Encryption (MPPE) encrypts data in Point-to-Point Protocol (PPP)-based dial-up connections or Point-to-Point Tunneling Protocol (PPTP) virtual private network (VPN) connections with 128-bit key.
- **Debug:** Tick to enable debug for PPTP VPN connection, debug information will be written into system logs.

You will need to create a VPN user account for a VPN client to be able to connect to the VPN Server.

To create an account, click “[VPN User Management](#)” tab and click “[New VPN User](#)” button to add a VPN user account.

New VPN User X

Username: BranchC
 Password: Dm&2iQE5
 Availability: Yes ▾

Save Cancel

Now the PPTP VPN client will be able to connect to the PPTP VPN server.

Notice:

If the LAVoice IPPBX system is behind NAT, you will need to open ports 1723 on the router/firewall.

PPTP VPN Client

To create your VPN client at the branch office site, open the LAVoice IPPBX web GUI and navigate to web menu *Network Settings->VPN Client*. Check the radio button of PPTP to configure PPTP VPN client.

VPN Client

VPN Client

L2TP PPTP OpenVPN N2N IPSec

Enable:
 Enable 40/128-bit encryption for MPPE:
 Server Address: 117.176.159.163
 Username: BranchC
 Password: ●●●●●●
 Default Gateway:

Save Cancel

- **Enable:** Tick to enable PPTP VPN client.
- **Enable 40/148-bit encryption for MPPE:** Tick to enable 40-bit key (standard) or 128-bit key (strong) MPPE encryption schemes.
- **Server Address:** PPTP VPN server public IP.
- **Username:** PPTP VPN user name given by the VPN server.
- **Password:** PPTP VPN user password given by the VPN server.
- **Default Gateway:** All traffic goes through the L2TP VPN connection.

Notice:

If connection is successfully established the system will display:

Status: Local IP address 172.16.0.x (An IP address between 172.16.0.2 and 172.16.0.9)

Remote IP address 172.16.0.1

5.3.3 OpenVPN

OpenVPN is an open-source software application that implements virtual private network (VPN) techniques for creating secure point-to-point or site-to-site connections in routed or bridged configurations and remote access facilities. It uses a custom security protocol that utilizes SSL/TLS for key exchange. It is capable of traversing network address translators (NATs) and firewalls. It was written by James Yonan and is published under the GNU General Public License (GPL).

OpenVPN allows peers to authenticate each other using a pre-shared secret key, certificates, or username/password. When used in a multiclient-server configuration, it allows the server to release an authentication certificate for every client, using signature and Certificate authority. It uses the OpenSSL encryption library extensively, as well as the SSLv3/TLSv1 protocol, and contains many security and control features.

OpenVPN Server

To create your OpenVPN Server, navigate to web menu *Network Settings->VPN Server*. Check the radio button of OpenVPN to configure your OpenVPN server.

The screenshot shows the 'VPN Server' configuration window. At the top, there are four radio buttons: L2TP, PPTP, OpenVPN (which is selected), and IPsec. Below this, the following settings are visible:

- Enable:
- Stealth:
- Certificate: Done (with 'Create' and 'Delete' buttons)
- Port: 1194
- Stealth Port: 443
- Protocol: TCP (dropdown menu)
- Device Node: TUN (dropdown menu)
- Cipher: Default (dropdown menu)
- Compress Lzo:
- TLS-Server:
- Remote Network: 172.16.0.0 / 255.255.255.0
- Route: 172.16.0.0 / 255.255.255.0
- Client-to-Client:

At the bottom of the window, there are 'Save' and 'Cancel' buttons.

- **Enable:** Tick to enable OpenVPN server.
- **Stealth:** Certain deep packet inspection firewalls might not allow OpenVPN traffic, stealth SSL tunneling can disguise your OpenVPN traffic under the HTTPS traffic which is often seen as HTTPS traffic by the DPI.
- **Certificate:** Certificate is one of the client authentication methods available in OpenVPN.
- **Port:** OpenVPN service port, the default is 1194.
- **Stealth Port:** Stealth service port, the default is 443.
- **Protocol:** You can choose either UDP or TCP. Stealth requires TCP only so if you have stealth enabled then this option is not configurable and the Server will use TCP by default.
- **Device Node:** TUN or TAP; A TAP device is a virtual Ethernet adapter, while a TUN device is a virtual point-to-point IP link.
- **Cipher:** Cipher (or cypher) is an algorithm for performing encryption or decryption.
- **Compress Lzo:** LZ0 is an efficient data compression library which is suitable for data

de-compression in real-time.

- **TLS-Server:** TLS is an excellent choice for authentication and key exchange mechanism of OpenVPN.
- **Remote Network:** OpenVPN remote network.
- **Route:** The route entries adjust the local routing table, telling it which network to route over the VPN.
- **Client-to-Client:** Client-to-Client can enable intercommunication between clients.

5.3.4 IPsec VPN

Internet Protocol Security (IPsec) is a protocol suite for secure Internet Protocol (IP) communications by authenticating and encrypting each IP packet of a communication session.

IPsec can be configured to operate in two different modes, Tunnel and Transport mode. Use of each mode depends on the requirements and implementation of IPsec.

IPsec VPN Server (Tunnel mode)

Tunnel mode is used to encrypt all traffic between secure IPsec Gateways, for example if you have two LAVoice IPPBX's and each acts as an IPsec Gateway for the hosts/IP phones behind it. The WAN ports will be used to connect both LAVoice systems to establish IPsec VPN connection, now all PCs or IP phones on the LAN ports can communicate with each other on both sides via a secure IPsec tunnel.

Navigate to web menu *Network Settings->VPN Server*. Check the IPsec radio button to configure IPsec VPN server.

VPN Server

VPN Server OpenVPN Certificate Download

VPN Server

L2TP PPTP OpenVPN IPsec

Enable:

Type: Tunnel ▾

IPsec Local IP: 117.176.159.163 ▾

IPsec Password: hPC2he@Q

IPsec Remote IP 1: 192.168.1.252

IPsec Remote Network 1: 192.168.200.0 / 255.255.255.0

IPsec Remote IP 2: _____

IPsec Remote Network 2: _____ / _____

IPsec Remote IP 3: _____

IPsec Remote Network 3: _____ / _____

Save Cancel

- **Enable:** Tick the checkbox to enable IPsec VPN server.
- **Type:** Defaults to Tunnel mode.

- **IPSec Local IP:** LAVoice WAN IP, which can be used to connect to the client network.
- **IPSec Password:** Define a password for authentication of the IPSec client.
- **IPSec Remote IP:** IPSec VPN client IP. The client uses this IP to connect to IPSec server.
- **IPSec Remote Network:** Specify the IPSec VPN client LAN network address.

Notice:

1. If the LAVoice IPPBX is behind NAT, port 500 and 4500 must be open on the router/firewall.
2. If the LAVoice IPPBX is connected to the Internet via PPPoE, then IPSec Local IP needs to be the IP address assigned by PPPoE.
3. IPSec VPN server can connect 3 IPSec clients.

IPSec VPN Client (Tunnel mode)

On the remote site, open the web GUI of another LAVoice IPPBX system and navigate to web menu to configure the VPN Client *Network Settings->VPN Client*.

On the VPN Client page choose IPSec and tick “Enable” option to enable IPSec client.

- **Enable:** Tick the checkbox to enable IPSec client.
- **Type:** Ensure this is the same as the IPSec server.
- **IPSec Local IP:** WAN port IP which can connect to the IPSec server.
- **Server Address:** Specify the IPSec server IP.
- **IPSec Password:** Specify the IPSec VPN password defined previously on the server.
- **IPSec Remote Network:** The IPSec VPN server LAN network address.

Notice:

1. After saving the configuration, the client will try to connect to the server using the details provided.
2. If connection is successfully established then the system will display “Status: 1 tunnel has been established!!!”
3. If connection fails then the system will display “Status: There’s no tunnel! Reconnecting...”

IPSec VPN server (Transport mode)

IPSec Transport mode is used for end-to-end communications, NAT traversal is not supported with the transport mode. So if two LAVoice IPPBX's are connected via IPSec transport mode, IPSec only encrypts the communication service ports, unlike Tunnel mode which encrypts the whole LAN subnet.

Navigate to web menu *Network Settings->VPN Server*. Check the IPSec radio button.

VPN Server

VPN Server

L2TP PPTP OpenVPN IPSec

Enable:

Type: Transport

IPSec Local IP: 117.176.159.163

IPSec Password: hPC2he@Q

Save Cancel

- **Enable:** Tick the checkbox to enable IPSec VPN server.
- **Type:** Select Transport mode.
- **IPSec Local IP:** LAVoice IPPBX WAN IP.(This is the same as configuring in Tunnel mode)
- **IPSec Password:** Define a password for authentication of the IPSec client.

IPSec VPN Client(Transport mode)

On the remote site, open the client IPPBX web GUI and navigate to web menu *Network Settings->VPN Client*. Check the radio button of IPSec.

VPN Client

VPN Client

L2TP PPTP OpenVPN N2N IPSec

Enable:

Type: Transport

IPSec Local IP: 192.168.1.252

Server Address: 117.176.159.163

IPSec Password: hPC2he@Q

Save Cancel

- **Enable:** Tick the checkbox to enable IPSec VPN client.
- **Type:** Ensure this is the same as the IPSec VPN server.
- **IPSec Local IP:** LAVoice IPPBX WAN IP which can connect to the IPSec server.
- **Server Address:** IPSec VPN server IP.
- **IPSec Password:** Specify the IPSec VPN password defined previously on the server.

Notice:

If a successful connection is established, then the system will display "Status: 2 tunnels have been established!!!". Because the LAVoice IPPBX system encrypts all service ports over UDP and TCP protocols, this means there will be 2 tunnels established.

5.3.5 N2N VPN Client

N2N is an open source Layer 2 over Layer 3 VPN application which utilizes a peer-to-peer architecture for network membership and routing.

On LAVoice IPPBX system we support N2N VPN client, to configure the N2N VPN client, please navigate to web menu *Network Settings->VPN Client*. Check the radio button of N2N VPN and configure the client info.

VPN Client

VPN Client

L2TP PPTP OpenVPN N2N IPSec

Enable:

Server Address:

Port:

Local IP:

Subnet Mask:

Local Port:

Username:

Password:

- **Enable:** Tick this checkbox to enable N2N VPN client.
- **Server Address:** N2N server(supernode) IP address.
- **Port:** N2N service port number. This is 82 by default.
- **Local IP:** VPN local IP.
- **Subnet Mask:** Netmask of the VPN network.
- **Local Port:** N2N local service port.
- **Username/Password:** Used for the N2N server to authorize the connection.

5.4 DHCP Server

DHCP(Dynamic Host Configuration Protocol)is a standardized network protocol used on Internet Protocol (IP) networks for dynamically distributing network configuration parameters, such as IP addresses for interfaces and services.

With DHCP, computers/IP phones request IP addresses and networking parameters automatically from LAVoice IPPBX WAN/LAN port which saves administrators a lot of time when compared with having to configure these settings manually.

5.4.1 DHCP Service

Navigate to web menu *Network Settings->DHCP Server*.

DHCP Server

DHCP Server
DHCP Client List
Static MAC

DHCP Server Settings

Enable:

Interface: WAN ▼

Start IP: 192.168.1.101

End IP: 192.168.1.199

Subnet Mask: 255.255.255.0

Gateway: 192.168.1.1

Primary DNS: 192.168.1.1

Lease Time(min): 1440

TFTP Server:

Save
Cancel

- **Enable:** Enable DHCP service.
- **Interface:** Choose the network port to implement DHCP service.
- **Start IP, End IP:** Specify the DHCP IP address pool.
- **Subnet Mask:** Netmask to be assigned to client devices.
- **Gateway:** Gateway address to be assigned to client devices.
- **Primary DNS:** DNS to be assigned to client devices.
- **Lease Time(min):** Duration for DHCP server to lease an address to a new device. When the lease expires, the DHCP server might assign the IP address to a different device. Default value is 1440 minutes.
- **TFTP Server:** Input the TFTP server address if required which may be used to auto provision your IP phones.

5.4.2 DHCP Client List

Navigate to *Network Settings->DHCP Server->DHCP Client List* and you will see a list of all devices receiving their IP address from the LAVoice IPPBX system.

DHCP Client List

DHCP Server
DHCP Client List
Static MAC

DHCP Client List:

Mac Address	IP Address	Host Name	Expires in
00:0b:82:71:b3:17	192.168.1.157		23:08:17

5.4.3 Static Mac

Static MAC is a useful feature which ensures the DHCP service on LAVoice IPPBX always assigns the same IP address to a specific computer or IP phone on your LAN. To be more specific, the DHCP service assigns this static IP to a unique MAC address assigned to each NIC on your LAN. To create a static Mac, navigate to web menu *Network Settings->DHCP Server->Static MAC*. Click “**New Static MAC**” to add a record to the LAVoice IPPBX system.

The screenshot shows a dialog box titled "New Static MAC" with a close button (X) in the top right corner. Inside the dialog, there are two input fields: "MAC Address:" containing the value "192.168.1.123" and "IP Address:" containing the value "6e72c3d4e5f6". Below these fields are two buttons: a green "Save" button and an orange "Cancel" button.

In this example, the IP address 192.168.1.123 will always be assigned to the device with MAC address 6E:72:C3:D4:E5:F6, lease time will not apply to this IP Address.

5.5 DDNS

Unlike DNS that only works with static IP addresses, DDNS (Dynamic Domain Name Server) is designed to also support dynamic IP addresses, such as those assigned by a DHCP server.

Built-in DDNS feature on LAVoice IPPBX system only requires you to sign up with a Dynamic DNS provider, then with the domain name they provide which maps your IP address on the Internet, you can access LAVoice IPPBX and also other services within your LAN via the domain name without needing to know your Dynamic public IP Address.

After setting DDNS, LAVoice IP PBX phone services can be accessed from remote site via the domain name which your DDNS provider supplied you. Also remote management is possible, even without a static public IP.

LAVoice IPPBX system supports the following DDNS service providers:

- <http://dyn.com/>
- <http://www.noip.com/>
- <http://www.zoneedit.com/>
- <http://www.oray.com/>
- <http://www.3322.net>
- <http://freedns.afraid.org/>

Sign up to one of these DDNS service providers' website and subscribe a dynamic domain name.

Once you have your account details, navigate to web menu *Network Settings* -> *DDNS Settings*.

The screenshot shows the "DDNS Settings" configuration page. At the top, it says "DDNS Settings" in a light blue header. Below the header, there are several configuration options: "Enable:" with a checked checkbox, "DDNS Server:" with a dropdown menu, "Username:" with a text input field, "Password:" with a text input field, and "Domain:" with a text input field. At the bottom of the form are two buttons: a green "Save" button and an orange "Cancel" button.

- **Enable:** Tick to enable DDNS service.
- **DDNS Server:** Select the DDNS service provider which you have subscribed to.

- **Username:** Username you subscribed to the service provider.
- **Password:** Password you used to sign up to the service provider.
- **Domain:** Your domain name.

After completing the above, please configure port forwarding on your router/firewall, then you'll be able to remote access LAvoice IPPBX services from the internet using this dynamic domain. For example, you can port forward port number 9999 and then you can access the LAvoice IPPBX web interface using the URL: [http://your domen name:9999](http://your domain name:9999).

5.6 SNMPv2

Simple Network Management Protocol (SNMP) is an Internet-standard protocol that is widely used in network management systems to monitor network-attached devices for conditions (Alerts) that warrant administrative attention.

SNMPv2 Settings

Read Only	
Enable:	<input checked="" type="checkbox"/>
RO Community:	public
RO Network:	192.168.10.0 / 24
Read and Write	
Enable:	<input checked="" type="checkbox"/>
RW Community:	private
RW Network:	192.168.1.0 / 24
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

With the above configurations, the network 192.168.1.0 can read and write(modify) the system configurations through the web interface, while the network 192.168.10.0 can only read but cannot modify anything.

5.7 TR069

TR069 (Technical Report 069) is a Broadband Forum (formerly known as DSL Forum) technical specification entitled CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.

To configure TR069, navigate to web menu *Network Settings->TR069*.

TR069 Settings	
Enable:	<input checked="" type="checkbox"/>
CPE to ACS URL:	<input type="text" value="http://192.168.1.69/acs"/>
ACS Authentication Mode:	<input type="text" value="BASIC"/>
ACS Username:	<input type="text" value="user"/>
ACS Password:	<input type="text" value="123456"/>
CPE Inform Interval(sec):	<input type="text" value="42200"/>
ACS to CPE URL:	<input type="text" value="http://192.168.1.78:7547"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

- **Enable:** Enable TR069 service
- **CPE to ACS URL:** Input URL to visit ACS, which is used by PBX to connect ACS via CPE WAN management protocol (CWMP)
- **ACS Authentication Mode:** Select ACS Authentication Mode: NONE/ BASIC/ DIGEST
- **ACS Username:** When the PBX sends a request to ACS, ACS will provide username to the authorized PBX.
- **ACS Password:** When the PBX sends a request to ACS, ACS will provide password to the authorized PBX.
- **CPE Inform Interval (sec):** Interval for CPE to connect ACS.
- **ACS to CPE URL:** Input URL to visit CPE. Format: http://IP:port(7547).

5.8 Troubleshooting

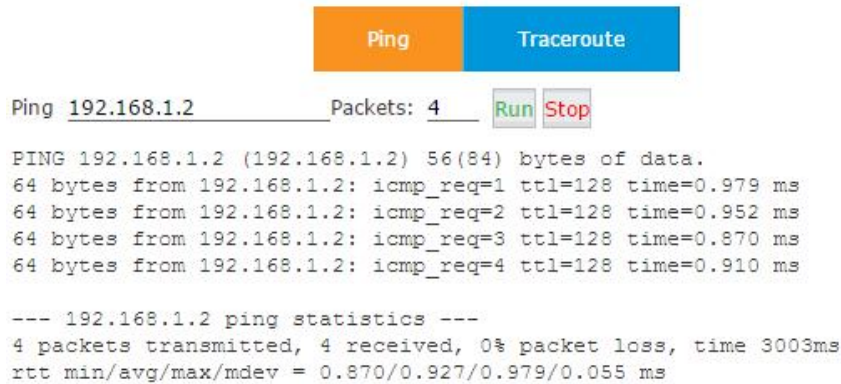
We have included two tools for troubleshooting network problems and they allow you to check the network reachability, ping and traceroute. With these tools you'll get an outside view of your network response time and network topology, which allows you to track down possible errors more easily.

5.8.1 Ping

The ping command is a very common method for troubleshooting the accessibility of devices. It uses a series of Internet Control Message Protocol (ICMP) Echo messages to determine:

- Whether a remote host is active or inactive.
- The round-trip delay in communicating with the host.
- Packet loss.

Troubleshooting



```
Ping 192.168.1.2 Packets: 4 Run Stop

PING 192.168.1.2 (192.168.1.2) 56(84) bytes of data.
64 bytes from 192.168.1.2: icmp_req=1 ttl=128 time=0.979 ms
64 bytes from 192.168.1.2: icmp_req=2 ttl=128 time=0.952 ms
64 bytes from 192.168.1.2: icmp_req=3 ttl=128 time=0.870 ms
64 bytes from 192.168.1.2: icmp_req=4 ttl=128 time=0.910 ms

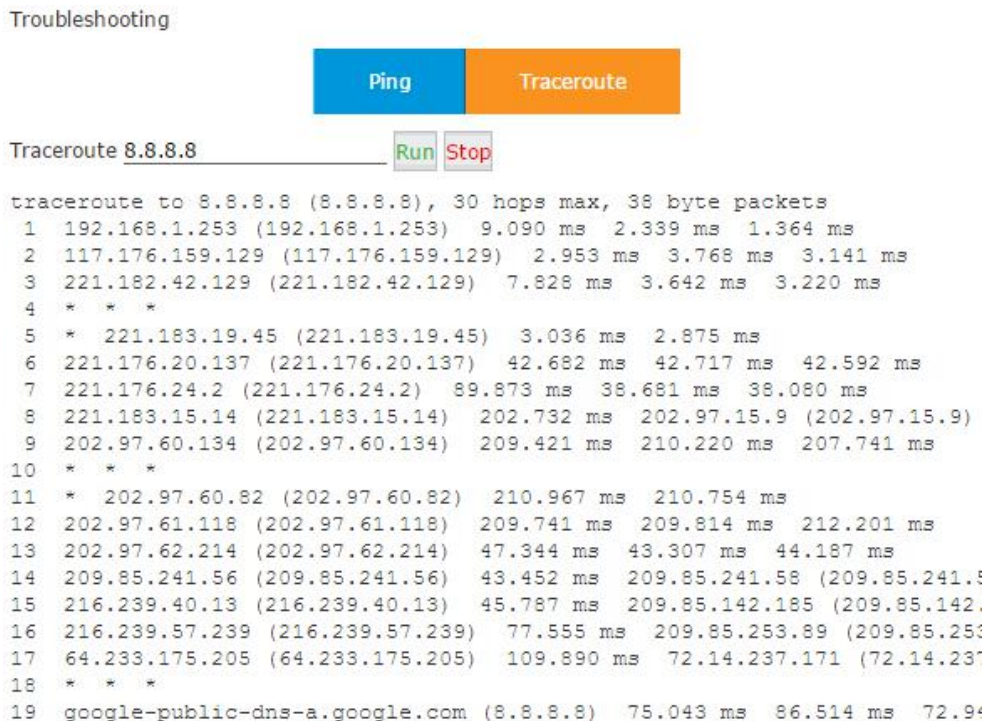
--- 192.168.1.2 ping statistics ---
4 packets transmitted, 4 received, 0% packet loss, time 3003ms
rtt min/avg/max/mdev = 0.870/0.927/0.979/0.055 ms
```

First specify the domain or IP of the host you want to contact and then define how many packets are to be sent, finally click the “Run” button and the command begins to process. You will receive results output from the system indicating the reachability of the destination.

5.8.2 Traceroute

The traceroute command is used to discover the routes that packets actually take when traveling to their destination.

Click “Traceroute” tab and specify the domain or IP address that you want to lookup and then click the “Run” button to start the process.



```
Troubleshooting

Ping Traceroute

Traceroute 8.8.8.8 Run Stop

traceroute to 8.8.8.8 (8.8.8.8), 30 hops max, 38 byte packets
 1 192.168.1.253 (192.168.1.253)  9.090 ms  2.339 ms  1.364 ms
 2 117.176.159.129 (117.176.159.129)  2.953 ms  3.768 ms  3.141 ms
 3 221.182.42.129 (221.182.42.129)  7.828 ms  3.642 ms  3.220 ms
 4 * * *
 5 * 221.183.19.45 (221.183.19.45)  3.036 ms  2.875 ms
 6 221.176.20.137 (221.176.20.137)  42.682 ms  42.717 ms  42.592 ms
 7 221.176.24.2 (221.176.24.2)  89.873 ms  38.681 ms  38.080 ms
 8 221.183.15.14 (221.183.15.14)  202.732 ms  202.97.15.9 (202.97.15.9)
 9 202.97.60.134 (202.97.60.134)  209.421 ms  210.220 ms  207.741 ms
10 * * *
11 * 202.97.60.82 (202.97.60.82)  210.967 ms  210.754 ms
12 202.97.61.118 (202.97.61.118)  209.741 ms  209.814 ms  212.201 ms
13 202.97.62.214 (202.97.62.214)  47.344 ms  43.307 ms  44.187 ms
14 209.85.241.56 (209.85.241.56)  43.452 ms  209.85.241.58 (209.85.241.58)
15 216.239.40.13 (216.239.40.13)  45.787 ms  209.85.142.185 (209.85.142.185)
16 216.239.57.239 (216.239.57.239)  77.555 ms  209.85.253.89 (209.85.253.89)
17 64.233.175.205 (64.233.175.205)  109.890 ms  72.14.237.171 (72.14.237.171)
18 * * *
19 google-public-dns-a.google.com (8.8.8.8)  75.043 ms  86.514 ms  72.941 ms
```

After the process has completed the system will notify you that “Trace Complete” and you can view which routes the packets have taken before reaching their final destination.

5.8.3 TCPDUMP

TCPDUMP is a common packet analyzer that allows users to capture TCP/IP and other packets being transmitted or received over a network to which the LAvoice IPPBX is attached. The captured packets can be downloaded from the IPPBX system and analyzed on your Windows PC to display the SIP traffic details. It can be used to debug a VoIP call problem.

On *System->Troubleshooting->TCPDUMP* page you can do a capture on one of the LAvoice IPPBX Ethernet interface.

Tcpdump

Ping Traceroute **Tcpdump** Channel Monitor

Tcpdump

Capture Trace on Adapter: WAN

Duration(seconds): 20 (1-300)

Start

List of Files Delete Selected

<input type="checkbox"/>	Name	Options
<input type="checkbox"/>	1 20160506033404.pcap	Delete Download

Select an interface and specify the duration of this capture then click on “Start”, the process will begin and now you can make a call to recur the problem.

Once time is up the captured packets will be displayed in the “List of Files” section. You can download it to analyze the SIP packets for troubleshooting purpose.

5.8.4 Channel Monitor

Channel Monitor, technically DAHDI Monitor allows you to monitor signal level on analog channel and record the output to a file. Recorded audio files are by default raw signed linear PCM. You can play it to the speaker to listen to the phone call signaling on the analog channel. Or you can use a sounds editor to visual display the audio level at both the Rx (audio Received by Asterisk) and Tx (audio Transmitted by Asterisk).

Usually Channel Monitor can be used to capture the caller ID signaling of an FXO channel. If you are experiencing caller ID problem you can perform channel monitor on the FXO port and then analyze the captured packets. If needed, you can send this file to Lava [support](#) for help.

Channel Monitor

Ping	Traceroute	Tcpdump	Channel Monitor
------	------------	---------	-----------------

Channel Monitor	
Monitor on channel:	FXS Port 3 ▼
Duration(seconds):	20 (1-300)
<input type="button" value="Start"/>	

List of Files ↻		Delete Selected
<input type="checkbox"/>	Name	Options
No Files		

In the "Monitor on channel" field you should select a channel to be monitored. And then you have to specify the duration to monitor. Then click on "Start" the capture will begin. Now you should make a call in from this channel (port).

After the capture is done you'll get the file listed in the "List of Files" section.


6. Reports

6.1 Register Status

On the register status page you are able to check the extension and SIP/IAX2 trunk status intuitively. You can view from which IP an extension is registered and you can also see the connection state, for example how much delay there is between the IPPBX system and the end point.

6.1.1 SIP User Status

Navigate to web menu *Report->Register Status->SIP User Status*.

Register Status 

SIP Users Status						
Name	Extension	IP	NAT	ACL	Port	Status
401	401	N/A	No	No	N/A	Unregistered
402	402	192.168.7.32	No	No	5060	Registered (4 ms)
403	403	192.168.7.147	No	No	45290	Registered (5 ms)
404	404	N/A	No	No	N/A	Unregistered
405	405	N/A	No	No	N/A	Unregistered
406	406	N/A	No	No	N/A	Unregistered
407	407	N/A	No	No	N/A	Unregistered
408	408	N/A	No	No	N/A	Unregistered
409	409	192.168.7.147	No	No	39480	Registered (4 ms)
410	410	N/A	No	No	N/A	Unregistered
411	411	N/A	No	No	N/A	Unregistered
John Doe	682	N/A	Yes	No	N/A	Unregistered

Here on this page you can see the SIP/IAX2 extensions, web extensions and also the register status of trunk users. Only a trunk that is configured as peer mode will be listed here.

Status and Description

- **Registered:** Registration success.
- **Unregistered:** Registration failure or unapplied.
- **Unreachable:** Network issue.
- **Timeout:** Register request timeout.

6.1.2 IAX2 User Status

To view IAX2 user status, navigate to web menu *Report->Register Status->IAX2 Users Status*.

Register Status [🔗](#)

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status	
IAX2 Users Status				
Name	Extension	IP	Port	Reachability
412	412	192.168.7.32	4569	Registered (2 ms)
413	413	N/A	N/A	Unregistered

Status and Description

- **Registered:** Registration success.
- **Unregistered:** Registration failure or unapplied.
- **Unreachable:** Network issue.
- **Timeout:** Register request timeout.

6.1.3 SIP Trunk Status

To view SIP trunk status, navigate to web menu *Report->Register Status->SIP Trunk Status*.

Register Status [🔗](#)

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status
SIP Trunks Status			
Username	Hostname/IP	Status	
5252742452	gw1.sip.us:5060	Registered	
61921248	183.62.205.209:5060	Registered	

Here you can see all your outbound SIP trunks' status.

Status and Description

- **Registered:** Successfully registered to the service provider and ready for phone calls.
- **Request Sent:** In this status, it's most probable that the network is totally unreachable to the SIP server. Please make sure network setting on the IPPBX system are correct.
- **Waiting for Authentication:** If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to incorrect credentials. Please double check your credentials.
- **Failed:** After trying to register within a certain time period without success, you get "Failed" on the trunk status.

6.1.4 IAX2 Trunk Status

To view IAX2 trunk status, navigate to web menu *Report->Register Status->IAX2 Trunk Status*.

Register Status [🔗](#)

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status
IAX2 Trunks Status			
Username	Hostname/IP	Status	
asterisk	192.168.7.146:4569	Registered	

Here you can see all of your outbound IAX2 trunks' status.

Status and Description

- **Registered:** Successfully registered to the service provider and ready for phone calls.
- **Request Sent:** If in this status, it's most probable that the network is totally unreachable to the service provider. Please make sure network setting on the IPPBX system are correct.
- **Waiting for Authentication:** If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to incorrect credentials. Please double check the credentials again.
- **Failed:** After unsuccessfully trying to register within a certain time period, you will see "Failed" on the trunk status.


6.2 FAX List

Navigate to web menu *Report->FAX List*. You can search any fax received by the IPPBX system.

Fax List

Start Date:	Nov ▼	12 ▼	2015 ▼	Field:	Caller ID ▼	<input type="text"/>	<input type="button" value="Filter"/>
End Date:	Dec ▼	12 ▼	2015 ▼				
Caller ID	Destination	Date	File Name		Status		
02037085791	800	12/04/15 13:15	fax000000007.tif	<input checked="" type="checkbox"/>	Done		
01085790903	800	11/24/15 20:37	fax000000006.tif	<input checked="" type="checkbox"/>	Done		
01085790903	800	11/20/15 16:26	fax000000005.tif	<input checked="" type="checkbox"/>	Done		
02082303466	800	11/18/15 16:06	fax000000004.tif	<input checked="" type="checkbox"/>	Done		
051786244043	800	11/12/15 09:52	fax000000002.tif	<input checked="" type="checkbox"/>	Done		

In the "Start Date" and "End Date" fields specify a time duration, and click "Filter" and you'll be able to view all faxes received during this time period. If you specified a "Caller ID" or "Destination ID" in the "Field" blank then you can get the fax sent/receive by a specific number in this time period.

Faxes can be downloaded to your PC hard drive by clicking the  button.

6.3 Record List

6.3.1 Call Recording

On the web page *Report ->Record List*. You are able to search all recorded call conversations if you have configured the extension to be always recorded.

Call Recording
Conferences
One Touch Recording

Extension: 402 Delete Field: Caller ID

Start Date: Dec 21 2015 End Date: Dec 21 2015 Filter

List of Recording Files Delete Selected

<input type="checkbox"/>	Caller ID	Destination ID	Date	Duration(sec)	Options
<input type="checkbox"/>	1 402	403	2015/12/21 12:00:21	40	Play Delete ↓
<input type="checkbox"/>	2 402	402	2015/12/21 11:46:40	7	Play Delete ↓
<input type="checkbox"/>	3 402	403	2015/12/21 11:46:33	2	Play Delete ↓

- **Extension:** Select an extension number to search the recordings of this extension.
- **Delete:** Delete all recordings from the selected extension number.
- **Field:** Filter the recordings by specifying caller ID or destination ID. For example, if you select “Caller ID” and specify number 401, you will get back the recordings of the calls made by extension 401; if you select “Destination ID” and specify number 401, you get back the recordings of the calls which called extension 401.
- **Start Date/End Date:** Search recordings made during this time period.
- **Delete Selected:** Delete the select recording items.
- **Caller ID:** Caller ID of this recorded call.
- **Destination ID:** The number that was called.
- **Date:** Exact time when this call recording began.
- **Duration(sec):** Duration of the recording.
- **Options:** Playback, delete and download options for the recorded files.
- **Play:** You can playback the recordings directly on the web page or playback on a specific phone.

6.3.2 Conference

All recorded conferences can be found here on *Report->Record List->Conference* page.

Call Recording
Conferences
One Touch Recording

Start Date: Dec 21 2015 End Date: Dec 21 2015 Filter

List of Conference Record Files Delete Selected Delete All

<input type="checkbox"/>	Conference Room	Date	Options
<input type="checkbox"/>	1 900	2015/12/21 14:52:39	Play Delete ↓

- **Start Date/End Date:** Specify a time duration to search the recorded conferences.
- **Delete Selected:** Delete the selected searched results.
- **Delete All:** Delete all searched results.
- **Conference Room:** The number of the recorded conference.
- **Date:** Exact time when the conference began.
- **Options:** Playback, delete or download the recording file.
- **Play:** Playback the recordings directly on the web page or playback on a specific phone.
- **Delete:** Delete the recorded audio file.

6.3.3 One Touch Recording

Call recordings recorded by one touch recording feature code *1 and can be found on *Report->Record List->One Touch Recording* page.

One Touch Recording

Call Recording Conferences One Touch Recording

Extension: 402

Start Date: Dec 21 2015 End Date: Dec 21 2015

List of Recording Files

<input type="checkbox"/>	Caller ID	Destination ID	Date	Options
<input type="checkbox"/>	1 402	403	2015/12/21 14:49:15	<input type="button" value="Play"/> <input type="button" value="Delete"/> <input type="button" value="Download"/>

- **Extension:** Extensions that used one touch recording to record calls will be listed here.
- **Delete:** Delete all recordings for the selected extension number.
- **Start Date/End Date:** Search the recordings during this time period.
- **Delete Selected:** Delete the select recording items.
- **Caller ID:** Caller ID of this recorded call.
- **Destination ID:** The number the caller called.
- **Date:** The exact time when this call began.
- **Play:** Playback, delete and download options of the recording files.
- **Delete:** Delete the recorded audio file.

6.3.4 Call Recording Playback

On LAVoice IPPBX system, there are two ways to playback recordings.

- Playback on the web interface
- Playback on a specific phone

By clicking the “Play” button on a call recording file you’ll see a dialog box like below:

Play X

Type 1:

Type 2:
Extension used for playing: 802

With “Type 1”, you can click the button you can playback the recording directly on the web interface.

With “Type 2”, you can specify an extension number and click on “Play” and then the extension will ring and you can pickup the call and the recording will play on the phone.

6.4 Call Logs

Call logs are also known as CDR(Call Detailed Records), on the call logs page you can check records for any call that went through the IPPBX system.

Navigate to web menu *Report->Call Logs* and by specifying the time duration and/or Caller ID/Destination ID/Account you can find the call records that you require.

Call Logs

Call Start	Caller ID	Destination ID	Account Code	Duration(sec)	Disposition
2015-12-21 16:35:56	402 <402>	013880424687	50377	240	Answered
2015-12-21 16:33:41	402 <402>	013880424687		43	Answered
2015-12-21 16:08:49	402 <402>	785756211		252	Answered
2015-12-21 16:06:00	85337096 <85337096>	402		55	Answered
2015-12-21 14:52:35	404 <404>	conference		2031	Answered
2015-12-21 14:52:49	402 <402>	conference		2014	Answered
2015-12-21 14:53:07	402 <402>	conference		56	Answered
2015-12-21 14:50:02	404 <404>	conference		42	Answered
2015-12-21 14:50:28	402 <402>	conference		12	Answered
2015-12-21 14:49:15	402 <402>	402		22	Answered
2015-12-21 12:00:21	402 <402>	402		40	Answered
2015-12-21 11:46:40	402 <402>	402		7	Answered
2015-12-21 11:46:33	402 <402>	403		2	Answered

Total:13 35 ▼ Per Page Pages:<< 1 ▼ >>

- **Start Date/End Date:** Define the searching time period by “Start Date” and “End Date”.
- **Field:** Search criteria.
 - Caller ID:** Search by the caller number.
 - Destination ID:** Search by the called number.
 - Account Code:** Search within the pin code which was used for outbound dialing.
- **Download:** Download the search results.
- **Delete:** Delete the search results.
- **Call Start:** The exact time when this call began.
- **Caller ID:** The number of the caller.(By clicking on the number you can add this number to the IPPBX system phone book.)
- **Destination ID:** The number which has been called.(By clicking on the number you can add this number to the IPPBX system phone book.)
- **Account Code:** The pin code that was used for outbound dialing.
- **Duration:** The duration of this phone call.
- **Disposition:** How the calls have been handled. Either answered, no answer or failed.

6.5 System Logs

These logs are IPPBX journals which store all system activities. They can be used for debug purpose if the system is running into exception. Please do not enable these logs if the system is functioning properly as debug information creates large log files which consume space and also utilize system resources.

In the LAVoice IPPBX system, there are 4 kinds of log files.

- **System Log:** System Logs store all system events.
- **PBX Log:** PBX Logs store all Asterisk events.
- **PBX Debug Log:** Asterisk debug logs.
- **Access Log:** Web and SSH access logs.

To enable these logs for the IPPBX system, please navigate to web menu *Report->System Logs*.

And enable the logs by ticking the corresponding checkboxes.

System Logs

System Logs

Enable System Log: Enable PBX Log:

Enable PBX Debug Log: Enable Access Log:

After checking the checkboxes please click “Save” and the log files will be generated.

List of Logs		<input type="button" value="Download Selected"/> <input type="button" value="Delete Selected"/>	
<input type="checkbox"/>	Name	Type	Options
<input type="checkbox"/>	1 debug20151221.log	Debug Log	<input type="button" value="Delete"/> <input type="button" value="Download"/>
<input type="checkbox"/>	2 login201512.log	Login Log	<input type="button" value="Delete"/> <input type="button" value="Download"/>
<input type="checkbox"/>	3 pbx20151221.log	PBX Log	<input type="button" value="Delete"/> <input type="button" value="Download"/>
<input type="checkbox"/>	4 sys20151221.log	System Log	<input type="button" value="Delete"/> <input type="button" value="Download"/>

Each day there will be a new log file generated for each of the log types. Enable them only if you are familiar with these logs for troubleshooting purposes.

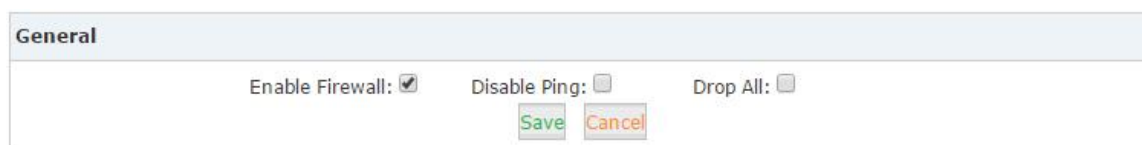
7. Security

7.1 Firewall

LAVoice IPPBX system has been preconfigured with a built-in firewall which prevents your IP phone system from unauthorized access, phone calls and certain other attacks.

To manage the firewall, navigate to web menu *Security->Firewall*.

General

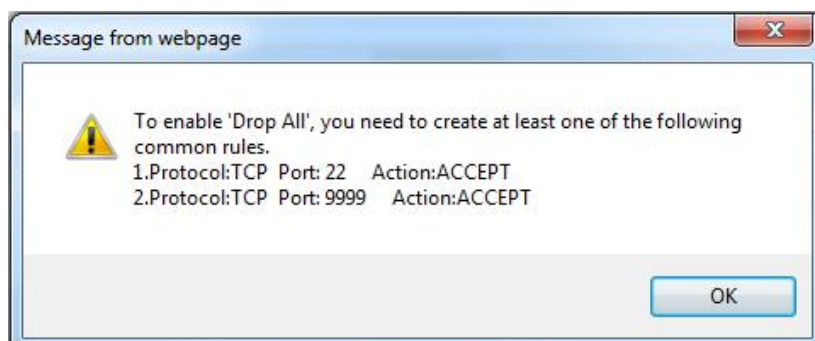


General

Enable Firewall: Disable Ping: Drop All:

Save Cancel

- **Enable Firewall:** By default, the firewall is enabled. You may disable the built-in firewall by unchecking “Enable Firewall” checkbox. Only consider disabling your firewall if your LAVoice IPPBX is behind a router/firewall without any port forwarding from the Internet.
- **Disable Ping:** Ignore ping request. If enabled, you cannot ping the IPPBX system.
- **Drop All:** Drop all packets sent to the IPPBX system, this will cause LAVoice IPPBX system to block all communication with the outside world. Because of this, the system will prompt to add at least one grant rule on port 22(SSH) or 9999(Web) to make sure the IPPBX system is not totally unreachable.



The rule/rules can be created in the “Common Rules” section.

Common Rules

In Common Rules section, you can configure the firewall to grant or deny an IP address or a network from communicating with the IPPBX system. Even the service port number can be specified so it can grant or deny a specific IP or network to access a specific service.

By clicking “Add Rule” button you can add a custom rule for rejecting or accepting an IP address or network address.

- **Name:** A name for this rule.
- **Description:** Optional, you may describe why this rule has been created.
- **Protocol:** Transmission protocol, UDP, TCP or UDP with TCP.
- **Port:** Service port number.
- **IP:** Can be an IP address or a network address.
- **MAC:** Action to be taken according to the Mac address of a device instead of its IP Address. This only works with devices within the same local network because Mac address are not routable.
- **Action:** Select “Drop” to block and “Accept” to grant.

Auto Defense

LAVoice IPPBX system uses Fail2Ban to perform intrusion detection, iptables is used for blocking any attack attempts.

Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IPPBX system, and uses iptables profiles to block brute-force attempts.

In the Auto Defense section you can define some custom rules to help the IPPBX system determine brute-force attempts.

Auto Defense		Add Rule	
Port	Protocol	Rate	Options
5060	UDP	40/2s	Edit Delete
5061	TCP	80/2s	Edit Delete

Click “Add Rule” button to add a new custom rule.

In this example, it will block an IP Address that sends more than 10 packets to the port 9999 within 30 seconds, this rule will prevent brute-force attempts on the web GUI login.

Rejected IP

Any IP address that is banned will be shown in the table of “Rejected IP”. The table will show the IP address of the banned host, as well as what kind of service intrusion was detected.

Rejected IP		
Type	IP	Options
VOIP	212.83.154.178	Delete
VOIP	173.249.158.227	Delete
VOIP	5.189.154.148	Delete

If a host appears incorrectly in the list of rejected IP, you can click on the "Delete" button to remove it from the list.

7.2 Service

Navigate to web menu *Security->Service*.

As we can see here on this page, you are able to configure the SSH and HTTPS services.

- **Enable SSH:** With this option you can enable or disable SSH access to the IPPBX system. It's disabled (unchecked) by default.
- **Port:** By default, SSH service port number is 22, you can change it to any other available port number.
- **Remote SSH Administration:** If this option is checked, remote SSH access will be enabled.
- **HTTPS Port:** Web GUI service port number, the default is 9999 and you can change this to any other port number if required.
- **Remote HTTPS Administration:** If this option is checked then remote web access will be enabled.

Notice:

If you want to remote access to SSH and web GUI of the IPPBX system, you can forward the corresponding ports on your router. Before doing this please ensure you have set strong passwords for root user and web admin user.

7.3 Fail2Ban

Allowed address allows you to add IP addresses and network addresses to the IPPBX system as a whitelist. The IPs in the whitelist will always be treated as trusted IP's and will not be filtered by the firewall rules.

Navigate to web menu *Security->Allowed Address*. Click "Add New IP" button and you can add a trusted IP or network to the system IP whitelist.



Add Allowed IP X

Description:

Protocol: SIP IAX2 HTTPS SSH

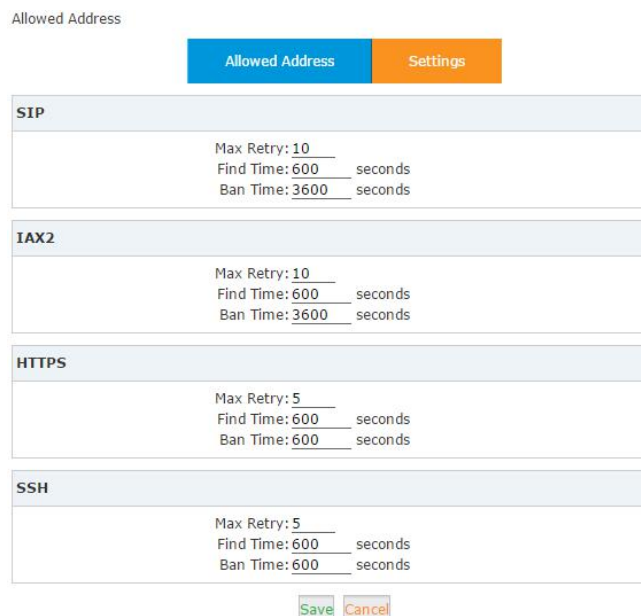
Allowed IP:

Subnet Mask:

Availability: ▾

- **Description:** A name for this entry.
- **Protocol:** Select protocols this IP/network can access.
- **Allowed IP:** IP address or network to be trusted.
- **Subnet Mask:** Netmask for this IP or network.
- **Availability:** Choose "Yes" to activate this entry, choose "No" to deactivate.

Settings



Allowed Address

SIP

Max Retry:
Find Time: seconds
Ban Time: seconds

IAX2

Max Retry:
Find Time: seconds
Ban Time: seconds

HTTPS

Max Retry:
Find Time: seconds
Ban Time: seconds

SSH

Max Retry:
Find Time: seconds
Ban Time: seconds

These options are actually for Fail2Ban, the "Max Retry" limits the authentication attempts. "Find Time" defines the time duration from the first attempt to the last attempt which reaches the "Max Retry" limitation. "Ban Time" is the time in seconds the IPPBX system will block the IP which exceeds max retry.

These settings don't take effect on any allowed addresses.

8. System Advanced

8.1 Time Settings

System time is very important for the IPPBX system, especially if the LAVoice IPPBX system handles inbound phone calls using time rules, then only if the system time is correct will calls be handled properly. Also, call logs and debug logs recorded to the system events use system time. LAVoice IPPBX system supports NTP (Network Time Protocol) and manual time set.

8.1.1 NTP

Navigate to web menu *System->Time Settings*.

By default, LAVoice IPPBX system use NTP to obtain time from Internet time servers. To configure, simply inform the IPPBX system where to find the server by specifying its domain or IP address. Also, please remember to select the correct time zone.

Time Settings

Time Settings

NTP Manual Time Set

NTP Server:

Time Zone: ▼

Once complete, click “Sync” button and the IPPBX system will attempt to synchronize the current time from the Internet. It might take a while depending on your network conditions.

After the process is complete, you’ll receive a notice saying either “Sync Failed!” or “Sync Success!”. If failed, then please check if the IPPBX can access the Internet or please change to another NTP server and try again.

8.1.2 Manual Time Set

If you want to manually set the time for the IPPBX system or for some special reason the IPPBX cannot access the Internet. You can choose to manually set the system time by checking “Manual Time Set” radio button.

Time Settings

Time Settings

NTP Manual Time Set

Year: _____ (YYYY, eg: 2010)

Month: _____ (MM, eg: 05)

Day: _____ (DD, eg: 08)

Hour: _____ (HH, eg: 09)

Minute: _____ (MM, eg: 30)

Synchronize with current PC time

There are two ways to manually set a time on the system.

1. Manually input the time and date info and click “Save”.
2. Synchronize the IPPBX system time with your PC time by clicking “Sync” button and then click on “Save” button.


Once “Save” is clicked the time is manually written or synchronized from the PC and will be stored into the hardware clock chip on the IPPBX motherboard.

8.2 Data Storage

Data storage allows you to upload your recording files, log files and voicemail messages to an FTP server through the Ethernet. If a USB drive is attached to the USB interface then the call recordings will be saved automatically to the USB drive instead of internal storage of the IPPBX system.

8.2.1 USB Data Storage

Plug the USB disk to the USB interface of LAVoice (LVX30V2 and LVX100sV2) IPPBX. Navigate to *Home* page. You’ll see the USB storage info like below snapshot.

Home 

System Info

Network

WAN IP: 192.168.1.7 MAC: 68:69:2E:04:18:1A
 LAN IP: 192.168.10.100 MAC: 68:69:2E:FF:18:1A

Storage

Disk	Total:	5.3G	Used:	2.1G
Ext Disk	Total:	7.5G	Used:	129M

Slot Info

SLOT 1

1	2	3	4
FXO	FXO	FXS	FXS

SLOT 2

1	2	3	4
GSM	GSM	N/A	N/A

Call recordings will be saved into USB drive without additional configurations. And the files can be accessed from IPPBX system Web GUI.

Important Notice:

1. If you are using a mobile HDD please use external power supply to power the mobile HDD.
2. You can plug in the USB drive with IPPBX system in production, but please DO NOT unplug it without cut off the power.

8.2.2 FTP Data Storage

Utilizing your existing FTP server, you can configure the LAVoice IPPBX to upload call recordings, voicemails and call log files to your FTP server. If you don't have one you can even use your Windows PC to setup an FTP server for the IPPBX system to connect to. You must however ensure that your PC is always turned on or at least available at the times when your IPPBX is going to upload files.

Data Storage

Data Storage
Data Storage Log

Data Storage

Enable:

Server Address: 192.168.1.149

Username: U50

Password: ●●●●●●●●

Directory: /uploading

Automatically upload frequency(day): 7

Time of automatically upload: 13 : 01

Forcibly upload when the flash storage is over: 60%

Call Recording: Voicemail: Call Logs:

Save
Cancel

Status: Disabled
Upload Now

After each upload you'll have a new folder created on your FTP server directory named by the date and time of this upload.

Name	Date modified	Type
cdr-custom	12/22/2015 1:30 PM	File folder
monitor	12/22/2015 1:30 PM	File folder
voicemail	12/22/2015 1:30 PM	File folder

Notice:

After each upload, with the exception of call logs(Master.csv inside cdr-custom folder) all other files will be removed from the IPPBX system, including call recordings(files inside monitor folder) and voice messages(files inside voicemail folder). So after each upload you will only have newly generated audio files.

8.3 Management

Navigate to web menu *System->Management*.

8.3.1 User Management

In the “[Change Password](#)” section, you are able to change admin password, also admin username can be changed by adding extra letters following name string “admin”.

Operator user had been disabled by default, if you want to activate Operator user a random password will be generate here. You can use this password or you can change it, but please do make the password strong enough.

Management

Change Password

Administrator:
Username: _____
Password: _____
New Username: admin _____
New Password: _____
Retype New Password: _____

Operator:
Enable:
Username: operator
Password: _____

SSH:
Password: _____
New Password: _____
Retype New Password: _____

Once complete changing admin user credentials, click “[Apply](#)” and you’ll be automatically logged out and redirected to the login page, now you are able to login with the new username and password.

And after Operator user been activated you can login with user name operator and the operator password. Operator user password also can be changed on Operator user portal you can change the password on *Change Password* page.

Root user password also can be changed from this page, it’s recommended not to activate SSH access to the LAVoice LVX30V2 or LVX100sV2 system. If deep troubleshooting of the IPPBX system from command line prompts is required you may activate SSH access from *Security->Service* page. Please deactivate SSH access once done troubleshooting.

8.3.2 Set System Voice Prompts

What are system voice prompts?

System voice prompts guide callers on for example how to place a call or how to use the IPPBX system functionality. One example is while checking voicemail the system voice prompts informs the user to enter voicemail password and in another example if you call someone and they don't answer then the system voice will ask that you should leave a message.

In the "Set Language" section you can decide in which language the system uses for the callers.



Set Language

Set Voice Language: English * [Download] [Delete]

[Save]

At this time, LAVoice IPPBX system(firmware version 2.1.2) supports 22 different languages as the system voice prompts. They are English, English (Australia), Chinese, French, French (Canada), Spanish, Spanish (Mexico), Portuguese, Portuguese (Brazil), Italian, Persian, Arabic, Turkish, Thai, Russian, Polish, Dutch, Korea, Hungary, Vietnamese, Hebrew, Greek and Germany.

The items with * means these languages already exist on the system while others can be downloaded here by clicking the "Download" button.

8.4 Backup

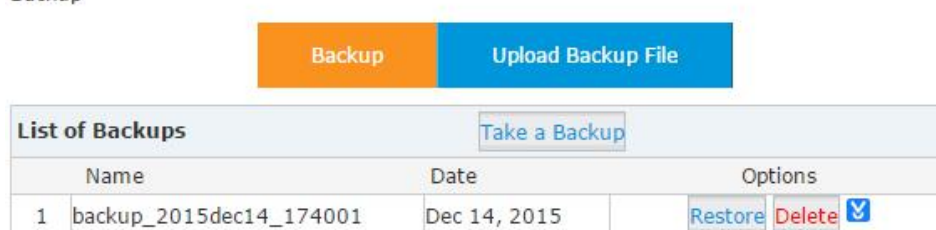
8.4.1 Take a Backup

Taking a backup on LAVoice IPPBX system is the same as when you create a recovery point on your Windows system. By restoring the backup you can recover the LAVoice IPPBX system configurations to the time point when it was still functioning well.

Normally the first backup should be taken when you have finished configuring the IPPBX to work for the very first time. Also, when you have applied new changes to your configuration is always a good time to take another backup.

Navigate to web menu *System->Backup*. Click "Take a Backup" button to create a backup file which will contain all current system configurations.


Backup



[Backup] [Upload Backup File]

List of Backups [Take a Backup]

	Name	Date	Options
1	backup_2015dec14_174001	Dec 14, 2015	[Restore] [Delete] [v]

Once complete, you will see the backup file listed on this page. The file is stored in the file system. At any time, by clicking “Restore” button you can restore your configurations. By clicking “Delete” button you can delete this backup. You can also download the backup to your computer hard disk drive by clicking the  button.

Notice:

If you are downloading the backup to your computer hard drive, please keep this file confidential, because this file contains web admin password, user extension password and many other sensitive information which may compromise your IPPBX system.

8.4.2 Upload Backup File

Click on “Upload Backup File” tab and you are able to upload a backup file from your computer hard drive.

Upload Backup File

[Backup](#) [Upload Backup File](#)

Upload Backup File

Note: Don't change the backup file name.

Please choose file to upload: backup_2015d...015dec22.tar

Notice:

If you are uploading a backup from another IPPBX system, please ensure they have the same hardware configurations. It is not recommended to upload backup files to different IPPBX systems, unless you are pretty comprehensive with Lava IPPBX systems.

8.5 Reset & Reboot

Navigate to web menu *System->Reset & Reboot*.

Reset & Reboot

Factory Defaults

Warning:All the configuration data will be lost when the system is reset to factory default. Please confirm that you have already backed up the configuration before reset.

Keep the current network settings

Reboot

Warning: Rebooting the system will terminate all active calls!

As you can see here on this page, you are able to reset and reboot the IPPBX system directly via web GUI.

8.5.1 Reset

By clicking “[Factory Defaults](#)” button you can reset all configurations for the IPPBX system. In addition to the configurations to be reset, recording files, voicemail messages and call logs will also be erased. So please ensure you have backed up the files you need before resetting.

The whole resetting process will be completed in 2 minutes. If you have chosen to reset network settings also, then you need to login with the default URL <https://192.168.1.100:9999>. Username and password will all be reset to admin.

8.5.2 Reboot

By clicking “[Reboot](#)” you can restart the IPPBX system, the whole process will be completed in 2 minutes.

8.6 Upgrade

Lava will update the IPPBX firmware at regular intervals for new features and bug fixes. You can visit our official [website](#) to check the updates for your IPPBX system.

IMPORTANTLY!

The must download the appropriate file front upgrade software system the Lava IPPBX.

In chapter *Home-Device Info* check *System Version*:

The screenshot shows the web interface for an Enterprise IP-PBX LVX-Series device. The left sidebar contains a navigation menu with items: Home, Operator, Basic, Inbound Control, Advanced, Network Settings, Security, Report, and System. The main content area is titled 'System Info' and includes sections for Network (WAN and LAN IP and MAC addresses), Storage (Disk and Ext Disk usage), and Slot Info (SLOT 1 and SLOT 2). Below this is the 'Device Info' section, which is highlighted with a red box and contains 'Model No.: LVX-100S' and 'System Version: 2.1.6'. At the bottom, it shows 'Current Time: 01/31/18 13:17' and 'Run Time: 1 day, 19:48'.

If version system - 2.1.6, then the required file for the update - `ulmage-md5.ippbx.LVX-100S_v216`

File Name	Upload Date	File Size
<code>lava_v216.zip</code>	29-Jan-2018 03:38	568124

If version system - 1.2.x, then the required file for the update - `ulmage-md5.u50.lava-v120`

File Name	Upload Date	File Size
<code>uimage-md5.u50.lava-v111</code>	04-Dec-2015 09:55	4970045
<code>uImage-md5.u50.lava-v120</code>	29-Apr-2016 09:08	7953645

The downloaded firmware package should be in .rar or .zip format, please extract the package first and upgrade with the `ulmage-md5.xxx` file to upgrade your IPPBX system.

Navigate to web menu *System->Upgrade*. You can see there are two methods you can upgrade the IPPBX firmware, they are web upgrade and TFTP upgrade.

8.6.1 Web Upgrade

Upgrade

Upgrade System Package

WEB Upgrade TFTP Upgrade

Restore Default Set:

Please choose file to upload: uImage-md5.u50v2.v212

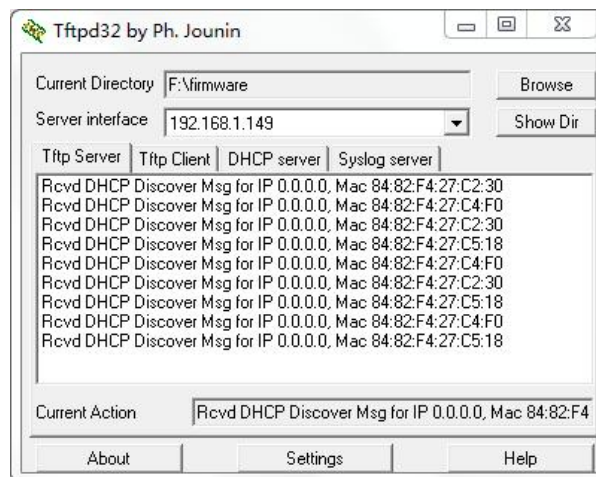
Check “WEB Upgrade” radio button and click “Browse” button to locate the new firmware in your PC hard drive. Click “Upload” and you will be asked to confirm a restart of the IPPBX system to complete the upgrade process. You can click “Yes” to continue upgrading.

Notice:

The “Restore Default Set” option is used to reset the IPPBX system configurations while upgrading, You don’t have to enable this option to reset the IPPBX system and only do so if you do wish to reset to default settings as it will reset all system configurations including the network profiles.

8.6.2 TFTP Upgrade

If you don’t have a TFTP server, you can Google tftpd32 and download this application to setup a lightweight TFTP server on your Windows.



Please click “Browse” on the TFTP application window to locate the new firmware. In the “Server Interface” dropdown list is a list of your PC network interfaces. Please select a correct interface (in the same network) which can access the IPPBX system.

On the IPPBX web GUI please check the “TFTP Upgrade” radio button, and specify the exact firmware file name in the “Enter The Package Name” blank, and in the “TFTP Server IP address” blank please specify the IP address displayed on the TFTP application window.

Upgrade

Upgrade System Package

WEB Upgrade TFTP Upgrade

Restore Default Set:

Enter The Package Name: uImage-md5.u20v2

TFTP Server IP address: 192.168.1.149

Please double check the file name and TFTP server IP address then click “Apply” you will be able to upgrade the firmware just like web upgrade.