

# DVX-2002F / DVX-2005F

Small and Medium Business

IP-PBX

VERSION 1.00

## USER MANUAL

( for Administrator )



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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 the 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 it. Non-expert handling to 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 it with 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.

## **Chapter 1 Brief Introduction**

### **1.1 Brief Introduction of IPPBX**

The DVX-2002F/DVX-2005F IP PBXs are designed to provide SMEs (small & medium enterprises) with all the standard and advanced features that are normally only available from large, expensive, legacy PBX manufacturers. Aimed at businesses with up to 100 extensions, the DVX-2002F/DVX-2005F Series IP PBXs are based on SIP and OpenSource Asterisk 1.8, with whose innovative modular telephony design, that is easy to expand the PBX to meet the growing needs of your business.

Each model will be introduced in detail below:

DVX-2002F is configured with 2 FXO ports.

DVX-2005F is configured with 4 FXO ports.

### **1.2 Main Features**

1. SIP/ IAX Extension Registration
2. Video Call
3. USB Mobile Hard Disk Record (Scalable)
4. IP Phone Provisioning
5. Call Record /Ring Group Record/ Call Queue Record
6. Web-based Administration and configuration
7. Web-based Extension User Management
8. Voicemail
9. Caller ID
10. Call Parking
11. Call Forward
12. Call Transfer
13. Call Waiting
14. Call Center Queues
15. Black List
16. Phonebook
17. Flexible Dial Plan
18. Virtual Fax (fax to email, and email to fax)
19. DID
20. Dial by Name
21. Speed Dial
22. Do Not Disturb
23. Callback
24. Skype for SIP
25. Ring Group

26. Conference Bridge (Three Conferences)
27. Music On Hold
28. DISA (Direct Inward System Access) /Paging And Intercom
29. Call Detail Record
30. IP Phone Feature Code
31. BLF(Busy Lamp Field)
32. Static /DHCP /PPPoE Network Access
33. DHCP Server
34. System Backup
35. T.38 Pass-through
36. Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX
37. Video Codec: H.261/ H.263 / H.263+ / H.264
38. VPN Server (L2TP / PPTP / OpenVPN, up to 10 connections for VPN clients)
39. VPN Client (L2TP / PPTP / OpenVPN / N2N)
40. SNMPv2
41. IPv4 / IPv6
42. DDNS(Dyndns.org /No-ip.com /zoneedit.com)

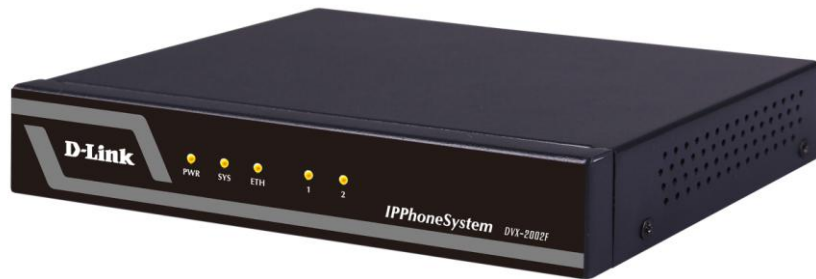
### 1.3 Module



DVX-2020 Module

## 1.4 Hardware Interfaces

### 1.4.1 DVX-2002F



DVX-2002F Front Panel



DVX-2002F Rear Panel

- 1 \* Reset Button
- 1 \* Power Interface (DC 12V 2A)
- 1 \* Ethernet Interface (10/100Mbps)
- 2 \* FXO Ports

DVX-2002F LEDs Indication



Indication	Function	Status	Explanation
PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	Blink	System Works
		Off	System Fails
ETH	WAN or LAN Data Status	Blink	Data Transport
FXO	FXO	Red	Channel Loading Success
		Blink	Channel Ringing
		Off	Channel Loading Failure

**1.4.2 DVX-2005F**



DVX-2005F Front Panel



DVX-2005F Rear Panel

- 1 \* Reset Button
  - 1 \* Power Interface (DC 12V 2A)
  - 1 \* Ethernet Interface (10/100Mbps)
  - 1 \* Console Interface
  - 1 \* USB Interface
- Slot 1 / Slot 2 for DVX-2020 Module Cards

Indication	Function	Status	Explanation
PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	Blink	System Works
		Off	System Fails

ETH	Data Status	Blink		Data Transport
		Off		No Data Transport
USB	U-disk Status	Off		Module not running
		On		Module Works
1-4(SLOT1/2)	SLOT 1/2 Status	FXO	Red	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure

**1.4.3 Model Comparison Table**

Items		DVX-2002F	DVX-2005F
System Capacity	Concurrent Calls	10	20
	Extension Users	30	100
	Voicemail and Recording	21,000 mins (GSM) 3000 mins (wav)	21,000 mins (GSM) 3000 mins (wav)
Hardware Capacity	SDRAM	128MB DDR2	256MB DDR2
	Memory (default)	4GB SD card	4GB SD card
Power Supply	Input	AC 100-240V	AC 100-240V
	Output	DC 12V/1A	DC 12V/2A

**1.4.6 Environmental Requirements**

1. Working Temperature: 0 °C ~40 °C
2. Storage Temperature: -20 °C ~ 55 °C
3. Humidity: 5~95% Non-Condensing

**1.4.7 Packing List**

Host	1 set
Power Supply	1 piece
Ethernet Cable	1 piece
Quick Installation Guide	1 piece
CD	1 piece

## Chapter 2 Getting Started

(Take DVX-2005F as example for the guide)

### 2.1 Before Making a Call

#### 2.1.1 Login IP PBX

##### Getting IP Address

There are three ways to set the IP address: Static, DHCP, PPPoE.

Default IP: [192.168.1.100:9999](http://192.168.1.100:9999)

Notice: you have to add port number 9999 after this IP address.

##### Defaults and Function Key

- |    |                      |                                       |
|----|----------------------|---------------------------------------|
| 1. | Web Panel User name: | admin                                 |
| 2. | Web Panel Password:  | admin                                 |
| 3. | *60                  | Enter Voicemail Box                   |
| 4. | 900/901/902          | Default three conference room numbers |
| 5. | #                    | Blind Transfer                        |
| 6. | *2                   | Attended Transfer                     |
| 7. | *                    | Disconnect Call                       |

##### Administrator Login

After connecting the IP PBX to the local area network and setting your laptop to the 192.168.1.x subnet, launch the web browser and bring up the system login page by entering the following URL: <http://192.168.1.100:9999>. You will see the login interface as below:



Input username and password, press the “Login” button and you will see the configuration interface below.

1. Default username: admin and password: admin

**Notice**

1. Please use IE(7.0 or higher version), Chrome, Firefox web browser.
2. If you do not see the interface above after inputting default IP and port number, please check whether your computer IP address is in the same segment with your IP PBX.
3. For Security reasons, please modify the username and password after login successfully. You can modify these by selecting: **【System】** --- **【Management】**
4. With the default setting, if there is no activity on the page for more than one minute, the system will timeout and automatically log out. To continue making configuration changes, you will need to login again.

The screenshot displays the D-Link web interface. On the left is a navigation menu with options: Home, Operator, Basic, Inbound Control, Advanced, Network Settings, Security, Report, and System. The main content area is titled 'System Info' and includes the following sections:

- Network:** Ethernet IP: 192.168.1.100, MAC: 00:60:6E:72:C2:FC
- Storage:**

Disk	Total:	3.0G	Used:	177.9M
Ext Disk	Total:	N/A	Used:	N/A
- Slot Info:**

SLOT 1				SLOT 2			
1	2	3	4	1	2	3	4
FXO	FXO	FXO	FXO	N/A	N/A	N/A	N/A
- Device Info:**

Model No.:	DVX-2005F	System Version:	***
------------	-----------	-----------------	-----

At the bottom, it shows 'Current Time: 04/14/14 18:15' and 'Run Time: 16 min'.

- |   |             |   |
|---|-------------|---|
| 1 | Network     | WAN IP and MAC will be displayed  |
| 2 | Storage     | Total storage and used storage will be displayed  |
| 3 | Channels    | Channel information will be displayed based on the modules installed                            |
| 4 | Device Info | Model No. And system version will be displayed.<br>System version is subject to the actual one. |

**Commonly Used Buttons**

On the home page, besides system info, there are other function buttons as below:

1. Logout Logout the Web panel
2. Activate Changes Activate the changes for your current configuration

**System Menu**

System Menu includes the following sub menu:

Home	Display device information
Operator	Extension / Trunk / Channel Status
Basic	Basic configuration on extension, trunks, etc.
Inbound Control	Configuration of Inbound Route, IVR and Black List, etc.
Advanced	Configuration of extension's default information, Conference Call, Call Transfer, Function Key, etc.
Network Settings	Configuration of Routing, Network, VPN, DHCP and other related network parameters
Security	Configuration of Firewall, SSH, FTP
Report	Record List, Call Logs and System Logs
System	Time Settings, Management, Back Up and Upgrade, etc.

### 2.2.2 Basic Configuration

#### Extension Configuration

DVX-2002F/DVX-2005F Supports SIP/ IAX2 and analog extensions as well as the ability to "Batch Add Users" by uploading extensions file.

Click **【Basic】** -> **【Extensions】** to configure:

The screenshot shows the D-Link web interface for extension configuration. On the left is a navigation menu with options: Home, Operator, Basic, Extensions, Trunks, Outbound Routes, Inbound Control, Advanced, Network Settings, Security, Report, and System. The main content area is titled "Extensions" and contains buttons for "Extensions" and "Upload/Download Extensions". Below these is a search bar with "Extension:" and "Search Show All" buttons. Further down are buttons for "New User", "Batch Add Users", and "Delete Selected Users". The core of the page is a table with the following columns: Name, Extension, Port, Protocol, DialPlan, Outbound CID, and Options. The table lists 20 extensions, each with an "Edit" link in the Options column.

Name	Extension	Port	Protocol	DialPlan	Outbound CID	Options
1 800	800	--	SIP	DialPlan1		Edit
2 801	801	--	SIP	DialPlan1		Edit
3 802	802	--	SIP	DialPlan1		Edit
4 803	803	--	SIP	DialPlan1		Edit
5 804	804	--	SIP	DialPlan1		Edit
6 805	805	--	SIP	DialPlan1		Edit
7 806	806	--	SIP	DialPlan1		Edit
8 807	807	--	SIP	DialPlan1		Edit
9 808	808	--	SIP	DialPlan1		Edit
10 809	809	--	SIP	DialPlan1		Edit
11 810	810	--	SIP	DialPlan1		Edit
12 811	811	--	SIP	DialPlan1		Edit
13 812	812	--	SIP	DialPlan1		Edit
14 813	813	--	SIP	DialPlan1		Edit
15 814	814	--	SIP	DialPlan1		Edit
16 815	815	--	SIP	DialPlan1		Edit
17 816	816	--	SIP	DialPlan1		Edit
18 817	817	--	SIP	DialPlan1		Edit
19 818	818	--	SIP	DialPlan1		Edit
20 819	819	--	SIP	DialPlan1		Edit

Click **【New User】** to see the extension configuration interface as below:

**New** X

---

**General**

SIP:  IAX2:

Name: 817 Extension: 817

Password: Z\_2Aj3V%BV Outbound CID: \_\_\_\_\_

Dial Plan: DialPlan1 Analog Phone: None

---

**Voicemail**

Voicemail:  VM Password: 1234

Delete VMail:  Email(Fax/Voicemail): \_\_\_\_\_

---

**Other Options**

Web Manager:  Agent:  Call Waiting:

Allow Being Spied:  Pickup Group: 1

Mobility Extension:  Mobility Extension Number: \_\_\_\_\_

---

**VoIP Settings**

NAT:  Transport: UDP SRTP:

DTMF Mode: RFC2833 Permit IP: \_\_\_\_\_

---

**Video Options**

Video Call:

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

**Extension Settings:**

Item	Explanation
SIP/IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g.: Tom.
Extension	Extension Number connected to the phone, e.g.: 888.
Password	Same password as voicemail. (4-16 digits, e.g.:123456)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu "Outbound Routes".
Analog Phone	Please choose the relative FXS port for your analog phone.
Voicemail	Check this option to enable the voicemail account.
VM Password	Set password for Voicemail, for security reasons, do not use the extension number or any easy combination like "1234"
Delete VMail	Check this option to delete voicemail from the PBX after it's sent by email.
Email (FAX/Voicemail)	Extension user's email address to receive email messages with attached fax or voicemail (you need configure the fax to email/voicemail options), e.g.: <a href="mailto:Tom@gmail.com">Tom@gmail.com</a>
Web Manager	Allow this user to login to the Extension Management Panel to manage extension options including voicemail, call recording, and call transfer, etc when you select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting

Allowing Being Spied	Check this option to allow this extension to be monitored (listened to or "spied").
NAT	Check this option if extension user or the phone is located outside the NAT(Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After check this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g.: Outbound Call, Internal Call, and Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS
SRTP	Enable SRTP (Secure Real-time Transport Protocol)
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary..
Video Call	Check to enable video calling for this extension. And select the video codec's you need to use.
Permit IP	Set device ip address or subnet permitted to register this extension with the IP PBX, e.g.:192.168.1.77 or 192.168.10.0/255.255.255.0. Devices with other IP addresses are not allowed to register this extension with the IP PBX.
Audio Codecs	Select what audio codec's you need to use.



**Notice:**

1. There are 30 default extensions which number started with "8"\*; you can add or delete extension by your requirement.
2. Maximum extensions: 100

**Upload/Download Extensions**

Click **【Upload/Download Extensions】** to batch add extensions as below:

Download the extension template from the **【Download Extensions Template】** , open the template using an editor or application like Microsoft Excel and carefully add extension information based on the template format and save.

Select the extension file to upload from **【Upload Extensions】**

Download current extensions information from **【Download Extensions(.csv)】**

**2.2.3 Time Based Rules**

Create a Time Rule. For example, Business Hours.

Select the starting and ending time, starting and ending days of the week, specific start and end dates and/or start and ending month of the year.

When an inbound call is processed, if the current time of the PBX is within these parameters, then the “if time matches” destination will be used for the call. If the current time of the PBX is outside these parameters, then the “if time does not match” destination will be used for the call.

Please set from this page: **【Time Based Rule】** --- **【New Time Rule】** :

**New Time Rule** X

Rule Name:

**Time & Date Conditions**

Start Time:  :  End Time:  :   
 Start Day:  End Day:   
 Start Date:  End Date:   
 Start Month:  End Month:

**Destination**

if time matches:   
 if time unmatches:

New Time Rule:

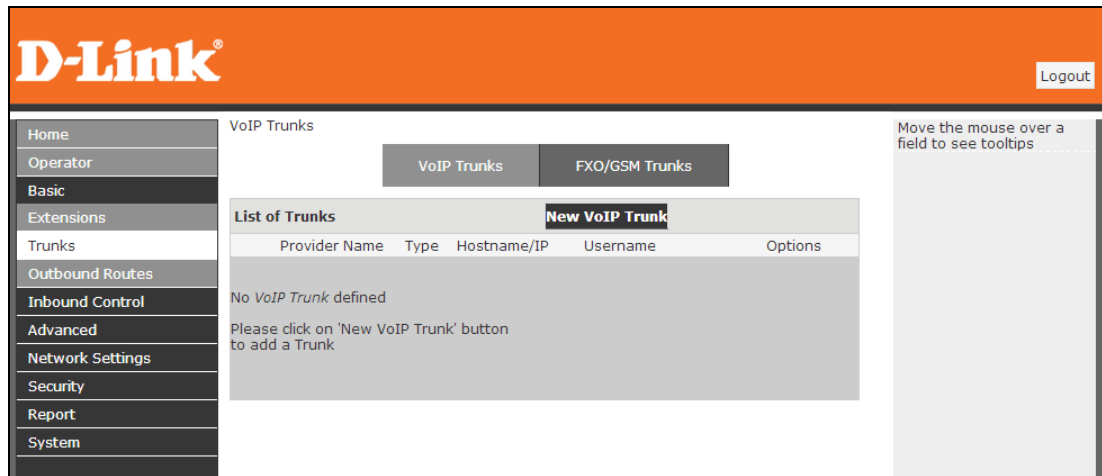
Item	Explanation
Rule Name	Define the name for this Time Rule.
Time & Date Conditions	Set parameters for Time/Day/ Date/ Month.
Destination	Select destination if time matches or does not match the conditions set. For example for Business Hours, “if time matches”, select operator extension during Business Hours. If outside business hours, select “if time does not match” destination of Operator voicemail

**2.3 Outbound Call**

**2.3.1 Trunks**

If you want to set up outbound route connected to PSTN (Public Switch Telephone Network) or VoIP provider, please configure on this page: **【Basic】** -> **【Trunks】**





DVX-2005F supports two kinds of trunks for your choice: VoIP or SIP Trunk and FXO.

**How to add each trunk:**

**VoIP Trunks**

Click **VoIP Trunk** -> **New VoIP Trunk** :

**New VoIP Trunk** X

Description: \_\_\_\_\_

Protocol: SIP

Host: \_\_\_\_\_ :5060

Maximum Channels\*: 0

Prefix: \_\_\_\_\_

Caller ID: \_\_\_\_\_

Without Authentication

Username: \_\_\_\_\_

Authuser: \_\_\_\_\_

Password: \_\_\_\_\_

**Advanced Options**

Domain: \_\_\_\_\_ Insecure: port,invite

From User: \_\_\_\_\_ Qualify(sec):  2

DID Number: \_\_\_\_\_ Transport: UDP

DTMF Mode: RFC2833 NAT:  SRTP:

Auto Fax Detection:

Context: Default Language: Default

**Audio Codecs**

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

**Video Codes**

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

Save
Cancel

**VoIP Trunks Reference:**

Item	Explanation
Description	Description of SIP trunk.
Protocol	Select protocol for outbound route, SIP or IAX2.

Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; "0" = no limitation).
Prefix	The prefix will be added in front of your dialed number automatically when the trunk is in use.
Caller ID	This Caller ID will be displayed when user make outbound call. Note: This function must be supported by local provider.
Without Authentication	If your trunk is static IP based and does not require a registration string when connecting the IP PBX, check this option.
Username	Username provided by VoIP Provider.
Password	Password provided by VoIP Provider.
Advanced Options	Advanced options for this trunk, e.g.: codec's, dial plan, etc.

The outbound trunk will be in the list of VoIP Trunk when the trunk is added successfully.

**FXO Trunk**

Click **【FXO/GSM Trunk】** -> **【New FXO/GSM Trunk】** :

New FXO/GSM Trunk X

Description: \_\_\_\_\_

Lines: **FXO:**  3  4  
**GSM:** \_\_\_\_\_

Prefix: \_\_\_\_\_

**Advanced Options**

Call Method: Order ▾

Busy Detection: Yes ▾      Busy Count: 3

Input Volume: 40% ▾      Output Volume: 40% ▾

Call Progress: No ▾      Progress Zone: US ▾

Busy Pattern: \_\_\_\_\_      Language: Default ▾

Answer on Polarity Switch: No ▾

Hangup on Polarity Switch: No ▾

Auto Fax Detection:

Save
Cancel

**FXO/GSM Trunk Reference:**

Item	Explanation
Description	Description for this trunk.
Lines	Check one or more channels FXO to be included in this trunk group
Prefix	The prefix will be added to the dialed number automatically when this trunk is in use.
Advanced Options	Advanced Options for this trunk, e.g.: Call Method, Busy Detection, etc.

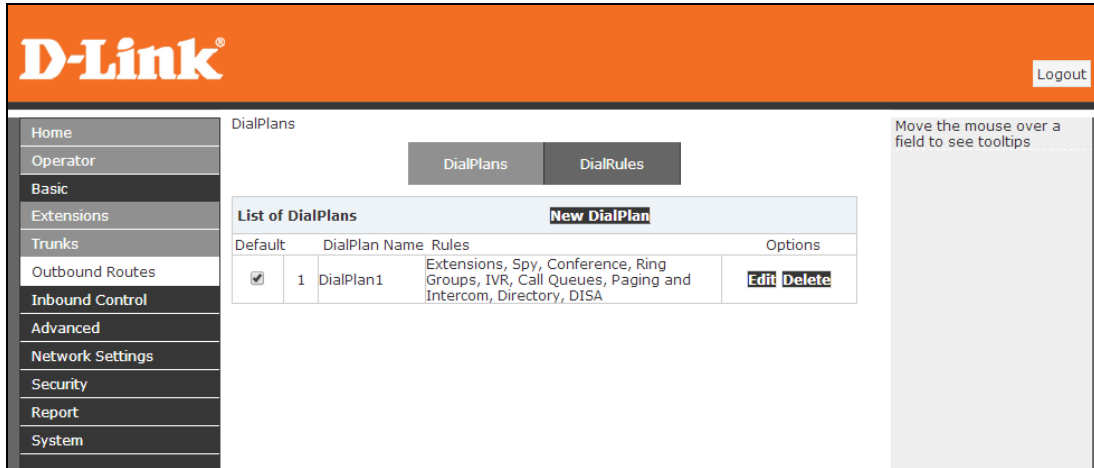
Select one or more of the available channels to be used for this trunk group.

Note: each channel can only be included in one trunk group. If no channels appear then all available channels are already defined.

### 2.3.2 Outbound Routes

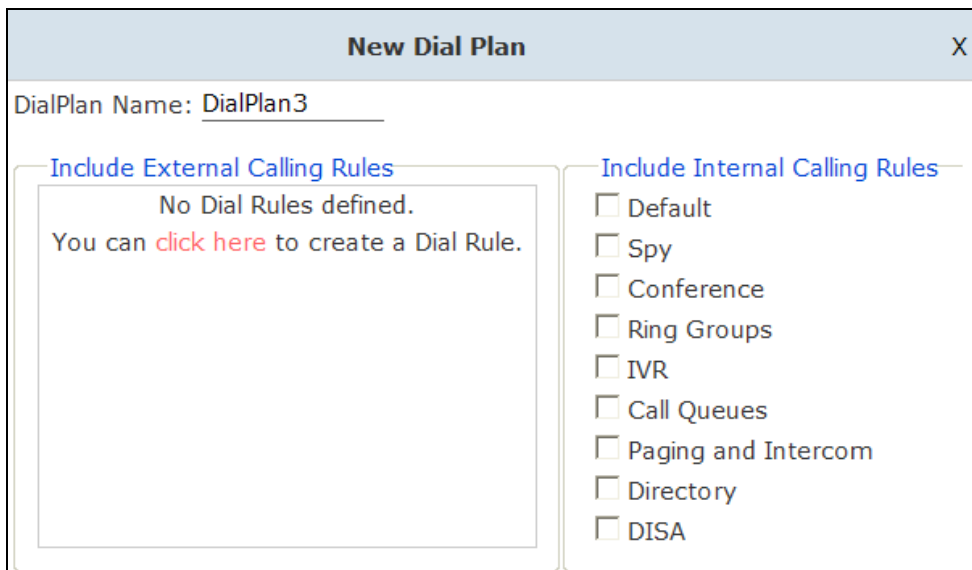
Outbound Routes are used to define which trunk groups are used by a specific extension when placing outbound calls. If you don't allow an extension user to place external calls, please ignore this part.

Please configure on this page: **【Basic】** -> **【Outbound Routes】**



You can configure the basic match pattern of outbound routes and create different dial plan on this page. Create as many different dial plans as you need to determine how you need extensions to be allowed to make calls. For example, create “Internal DialPlan” to include all Internal Calling Rules but do not select any outbound dial rules. Select “Internal DialPlan” for all extension users that do not need the ability to make external calls.

Click **【DialPlans】** -> **【New DialPlan】** :



You can create one or more DialRules for DialPlans from this page:

**New DialRule** X

Rule Name: \_\_\_\_\_

PIN Set:

Call Duration Limit: \_\_\_\_\_ seconds

Time Rule:

Start Time: 09 : 00 End Time: 17 : 30

Start Day: Mon End Day: Fri

Place this call through:

test(FXO/GSM) voip(SIP)	<input type="button" value="»»"/>  <input type="button" value="→"/>  <input type="button" value="←"/>  <input type="button" value="««"/>	
<b>Available Trunks</b>		<b>Selected Trunks</b>

Custom Pattern: \_\_\_\_\_

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

Delete \_\_\_\_\_ digits prefix from the front and auto-add digit \_\_\_\_\_ before dialing

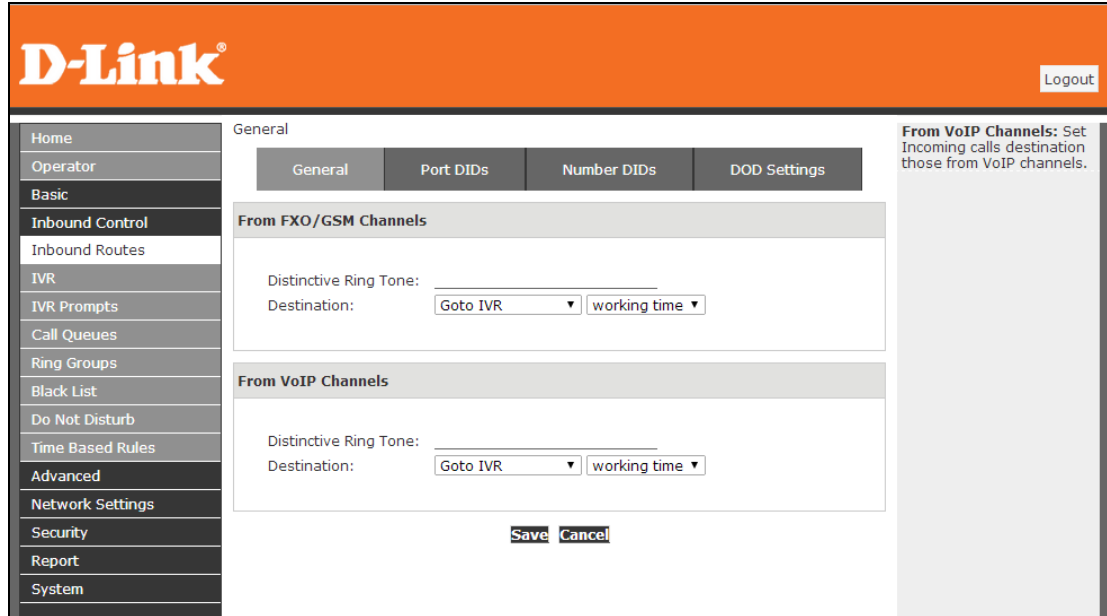
Reference:

Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule.
Call Duration Limit	Set the duration limit for a call, beyond which the call will be auto hung up
Time Rule	Set the time interval for this DialRule, beyond which the call based on this DialRule won't work
Place this call through	Select one of the trunk groups that have been set up to use for this dial rule
Custom Pattern	<b>N</b> any digit from 2 to 9 <b>Z</b> any digit from 1 to 9 <b>X</b> any digit from 0 to 9 . One or more digits
Delete[ ]digits prefix	How many digits will be deleted from what the user dialed to what is actually sent over the trunk. For example, user dialed 94166445775 and you selected to delete 1 digit, then 4166445775 is sent out the trunk.
Auto-add digit[ ]	If add digit "9", when dial 12345, 912345 will be sent.

2.4 Inbound Call

2.4.1 Inbound Routes

Click **【Inbound Control】** -> **【Inbound Routes】**



**General**

Distinctive Ring Tone: Mapping the custom ring tone file, e.g.: Set distinctive ring tone as “External”, the phone will play this ring tone when receiving the call.

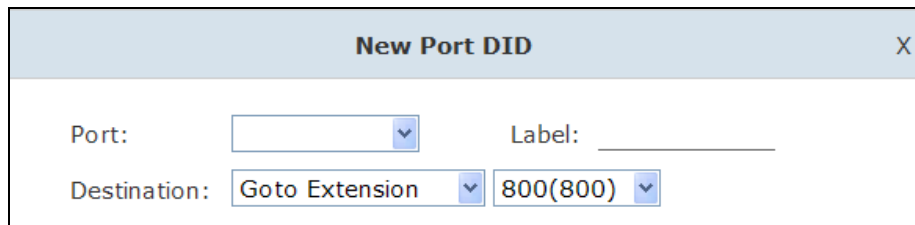
**Note: The phone must support such feature as well.**

Select all calls coming in on a specific port (FXO/ VOIP) and select which destination (Extension User, IVR, Queue, Conference Bridge, IVR, etc) should answer those calls. Setting the label will assign this label to be displayed.

**Port DIDs**

To have incoming calls from a PSTN trunk port (FXO trunk) answered by a specific extension user, call queue, conference bridge, or IVR, please configure here:

Click **【Port DIDs】** -> **【New Port DIDs】** :



1. Port            Select the trunk group port
2. Label         Set a label for this port. Incoming calls from this port will display The specified label.
3. Destination   Incoming calls will be answered by the specified destination (extension user, call queue, conference bridge, or IVR)

**Number DIDs**

If you want to select the destination of inbound calls on VoIP Trunks based on the incoming DNIS (dialed number or DID). You can specify the DID and destination (user extension, queue, conference bridge, or IVR):

Click **【 Number DID 】** -> **【 New Number DID 】** :

**New Number DID** X

DID Number:

Destination: Goto Extension ▼ 800(800) ▼

1. DID Number Set DID Number
2. Destination Select the extension for access directly(Extension User/ Call Queue/ conference/ IVR)

**DOD Settings**

To configure outbound calls from user extensions to answer with specified destinations (user extension, queue, conference bridge, IVR) , please click **【 DOD Settings 】** -> **【 New DOD 】**

**New DOD** X

DOD Number:

Destination: Goto Extension ▼ 800(800) ▼

- 1 DOD Number Set the DOD (direct outbound dial) number, and use it to match the Caller ID.
- 2 Destination Outbound calls will access directly to this destination (user extension, queue, conference bridge, or IVR)

**2.4.2 IVR**

IVR (Interactive Voice Response) or Automated Attendant will allow callers to select from a specific set of options by pressing the selected digit on their telephone dial pad.

Click **【 Inbound Control 】** -> **【 IVR 】** :

Click **【New IVR】** to create a new IVR:

Key	Action
0	Disabled
1	Disabled
2	Disabled
3	Disabled
4	Disabled
5	Disabled
6	Disabled
7	Disabled
8	Disabled
9	Disabled
*	Disabled
#	Disabled
t	Disabled

Item	Explanation
Name	Enter a descriptive name for the IVR
Extension	Enter a unique extension or IVR number. This number is used to access the IVR from an internal extension
Custom	Click "Custom" to choose a DialPlan for IVR
Please Select	Select the IVR prompt that will provide the caller with instructions on what options are

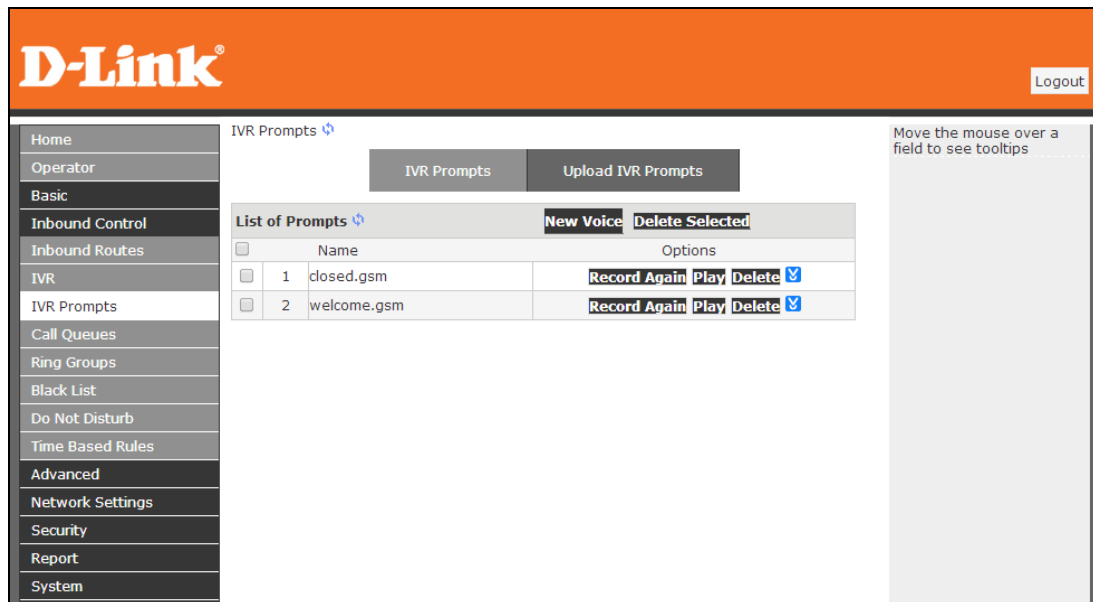
	available. To configure the prompt in this page: <b>【IVR Prompt】</b>
<a href="#">Repeat Loops</a>	Loop times to repeat playing the IVR prompt if the caller does not select an option
<a href="#">Dial Other Extension</a>	Allow user to dial other extensions besides of the listed options
<a href="#">Keypress Event</a>	Select the available options beside the designated digit

### 2.4.3 IVR Prompts

IVR prompts can be recorded by using any extension registered to the PBX or they can be uploaded from the “Upload IVR Prompt” section below.

#### IVR Prompts

**【IVR Prompts】**



Click **【IVR Prompts】** ---- **【New Voice】** to create new IVR prompt:

**New Voice**
X

File Name:

Format: GSM ▼

Extension used for recording: 800 ▼

1. File Name Define a name for this voice file.
2. Format Select the voice format,GSM/WAV(16bit) supported only.
3. Extension used for recording: Select the extension which is used for recording the IVR prompt.

Click **【Record】** , the extension will ring, and the prompt can be recorded after picking up the phone.



To hear the existing recording, please click **【Play】** :

**Play record voice**X

Extension used for playing:  ▼

Select the extension, click **【Play】** , the selected extension will ring, and you will hear the recorded prompt after picking up the phone.

### Upload IVR prompt

**【Upload IVR prompt】**

The screenshot shows the D-Link web interface. At the top left is the D-Link logo, and at the top right is a 'Logout' button. A left sidebar contains a menu with items: Home, Operator, Basic, Inbound Control, Inbound Routes, IVR, IVR Prompts (highlighted), Call Queues, Ring Groups, Black List, Do Not Disturb, Time Based Rules, Advanced, Network Settings, Security, Report, and System. The main content area is titled 'Upload IVR Prompts' and contains two tabs: 'IVR Prompts' and 'Upload IVR Prompts'. Below the tabs is a box with the title 'Upload IVR Prompts' and a red note: 'Note: The sound file must be wav(16bit/8000Hz/Mono), gsm, ulaw or alaw! The size is limited in 15MB!'. Below the note is a text prompt 'Please choose file to upload:' followed by a 'Please choose' button and an 'Upload' button. On the right side of the interface, there is a tooltip that says 'Move the mouse over a field to see tooltips'.




**Notice:**

DVX-2002F/DVX-2005F supports custom audio file with wav,gsm,ulaw,alaw format. Recordings must be smaller than 15MB.

### 2.4.4 Ring Groups

A Ring Group (sometimes called a Hunt Group) is a way to ring a collection of extensions by dialing a single extension number. The methodology used to ring that collection of extensions is called the ring strategy. Once the timeout (number of seconds) is reached, the call will then be directed to the “if not answered” or failover destination.

To configure a Ring Group Click**【Inbound Control】**-> **【Ring Groups】**->**【New Ring Group】**:

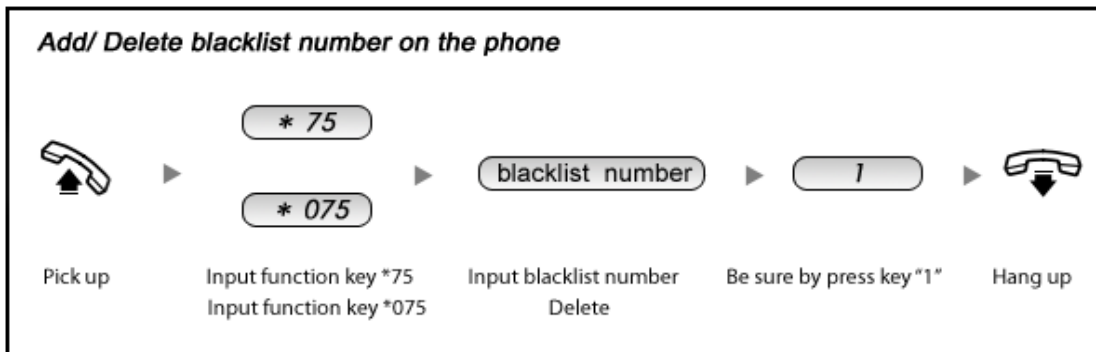
1. Name Define a name for the Ring Group
2. Strategy Select “Ring All” or “Ring in order”
3. Ring Group Members Select the Ring Group Member from “the Available Channels”, click  to add.
4. If not answered You can choose to forward the call to extension, voicemail ring group, IVR or hang up if not answered.

**2.4.5 Blacklist**

The Blacklist feature allows the blocking of specific phone numbers by Caller ID. Click **【Inbound Control】** -> **【Blacklist】** -> **【New Blacklist】**

Input the caller ID in the space provided. Once configured, future calls from this caller ID will be blocked.

To maintain this list of blocked numbers, see the instructions in the following diagram:



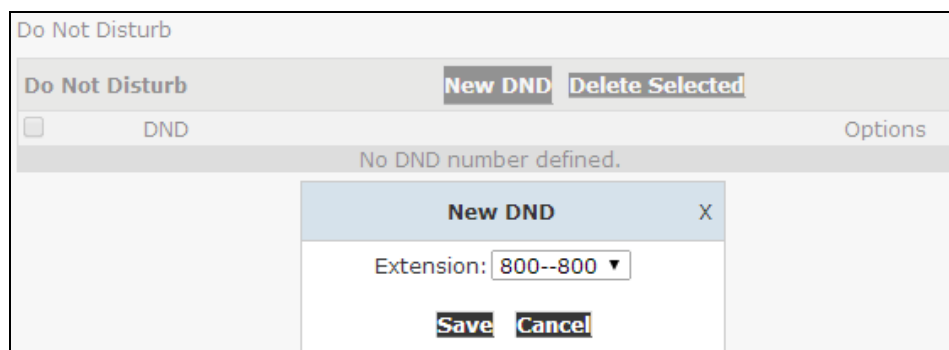
Reference:

Item	Explanation
*75	When the registered extension user inputs *75 + blacklist number, this number will be added in the list of Blacklist Number.
*075	When the registered extension user inputs *075+blacklist number, this number will be deleted in the list of Blacklist Number.

### 2.4.6 Do Not Disturb

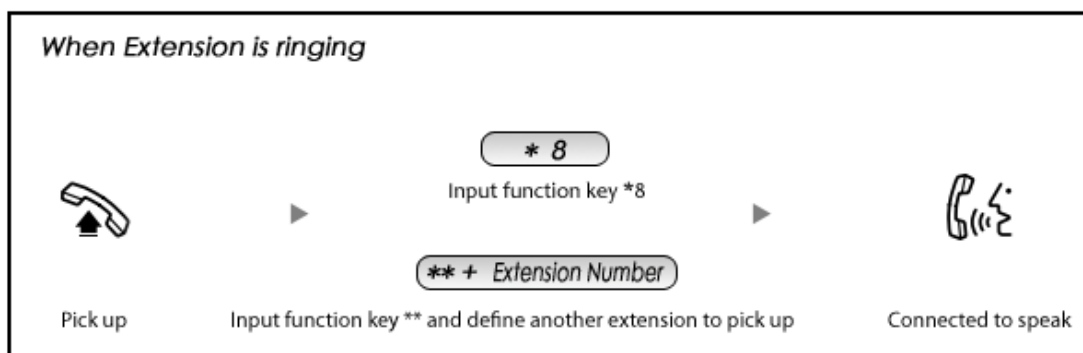
The administrators can config DND for extensions on this page:

Click **【Inbound Control】** -> **【Do Not Disturb】** :



### 2.4.7 Call Pickup

This feature allows users to answer a call that is ringing on another user’s extension by pressing the selected feature code on their own phone as shown in the diagram below.



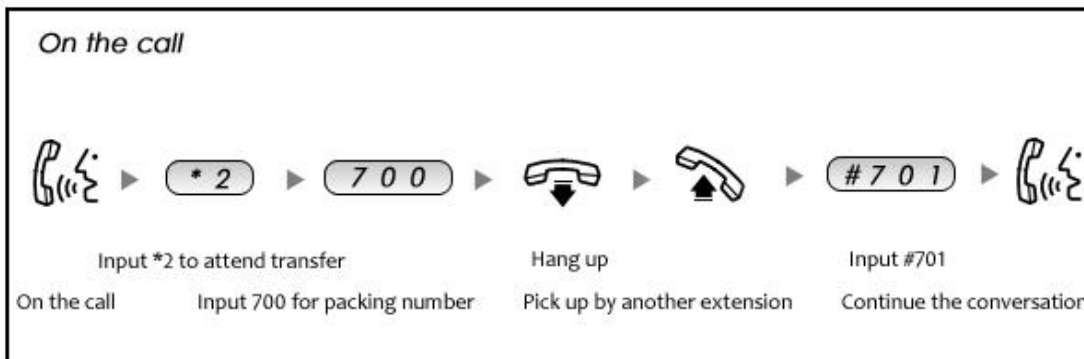
Reference:

Item	Explanation
*8	Input function key *8 to pick up the registered extension which is in the ring at random. This can be defined in <b>【Feature Codes】</b>
**	Input function key ** and define another extension to pick up. This can be defined in <b>【Feature Codes】</b> .

**2.5 During a Call**

**2.5.1 Call Parking**

This feature allows a call to be placed on hold (system will play the parked number, e.g. 701) and then retrieved from any other extension by entering the parked number. After answering the call, to park the call press \*2 700 on the telephone dialpad (to transfer the call to the parking lot 700). This will park the call and the system will play the parking space (e.g. 701). To retrieve the call from the parking lot, anyone can pick up any registered extension and dial the parking space number (e.g. 701) and will be connected with the parked caller. Refer to the diagram below:

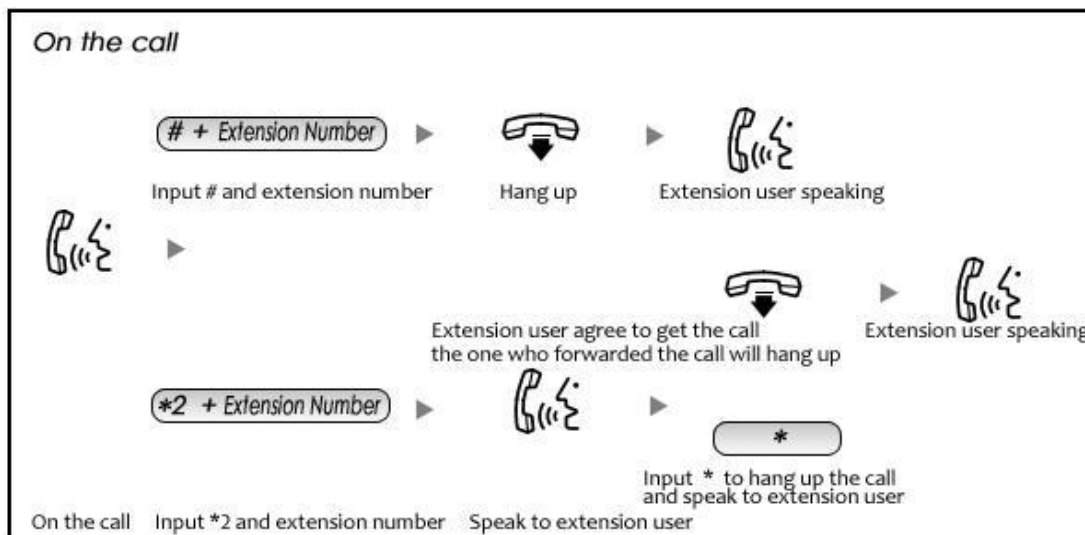


Reference:

Item	Explanation
Extension to Dial for Parking Calls:	Default Number: 700, Define in <b>【Feature Codes】</b>
What Parking space or Extension to park calls on	Default Number : 701 - 720. Define in <b>【Feature Codes】</b>
How many seconds a call can be parked for	Default is 45 seconds. Define in <b>【Feature Codes】</b> .

**2.5.2 Call Transfer**

This feature allows an incoming call that is answered on one extension to be sent to another user’s extension. Refer to the diagram as below:



Reference:

Item	Explanation
Blind Transfer	Default is #. Define in 【Feature Codes】
Attended Transfer	Default is *2. Define in 【Feature Codes】
Complete Attended Transfer	Default is *, it can be used when you use *2. Define in 【Feature Code】
Timeout for answer on attended transfer	Default is 15 seconds. Define in 【Feature Codes】

## 2.6 User Extension Settings

### 2.6.1 Follow Me Settings

This feature allows a call to an extension to be automatically forward to one or more internal extensions or external phone numbers. To allow the user to configure these settings, first the user must be allowed access to the User Web Portal. To do this, select the “Web Manager” box under “Other Options”.

Click **【Basic】** -> **【Extension】** -> **【Edit】** the extension you want to configure.

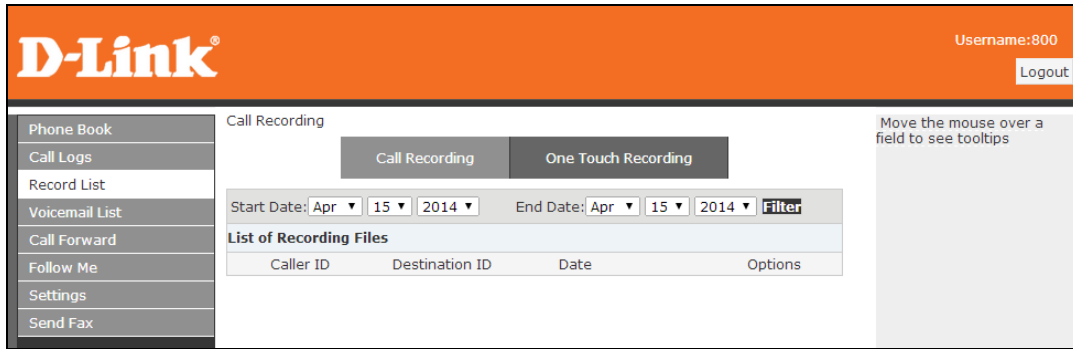
Edit			
<b>General</b>			
SIP:	<input checked="" type="checkbox"/>	IAX2:	<input type="checkbox"/>
Name:	800	Extension:	800
Password:	123456	Outbound CID:	
Dial Plan:	DialPlan1	Analog Phone:	None
<b>Voicemail</b>			
Voicemail:	<input checked="" type="checkbox"/>	VM Password:	1234
Delete VMail:	<input type="checkbox"/>	Email(Fax/Voicemail):	
<b>Other Options</b>			
Web Manager:	<input checked="" type="checkbox"/>	Agent:	<input checked="" type="checkbox"/>
Allow Being Spied:	<input type="checkbox"/>	Pickup Group:	1
Mobility Extension:	<input type="checkbox"/>	Mobility Extension Number:	
<b>VoIP Settings</b>			
NAT:	<input checked="" type="checkbox"/>	Transport:	UDP
DTMF Mode:	RFC2833	S RTP:	<input type="checkbox"/>
<b>Video Options</b>			
Video Call:	<input type="checkbox"/>		
<input type="checkbox"/> H.261 <input type="checkbox"/> H.263 <input type="checkbox"/> H.263+ <input type="checkbox"/> H.264			
<b>Audio Codecs</b>			
<input checked="" type="checkbox"/> alaw <input checked="" type="checkbox"/> ulaw <input type="checkbox"/> G.722 <input checked="" type="checkbox"/> G.729 <input type="checkbox"/> G.726 <input type="checkbox"/> GSM <input type="checkbox"/> Speex			

Check **【Web Manager】** and **【Save】**

Then login the Extension Web Panel:

### 2.6.2 Call Recording

This feature allows users to access calls they have recorded. To configure this setting, please see the diagram below.



### 2.6.3 Call Forward

This feature allows calls to an extension to be automatically forwarded to a specific internal extensions or external phone number. To configure this setting, please see below:

Click **【Call Forward】** :

Forward Settings	
<input type="checkbox"/>	Always _____
<input type="checkbox"/>	Busy _____
<input type="checkbox"/>	No Answer _____

#### Reference

	Item	Explanation
Status	Always	All incoming calls will be forwarded.
	Busy	Forward when extension is busy.
	No Answer	Forward when no answer from extension.

### 2.6.4 Voicemail

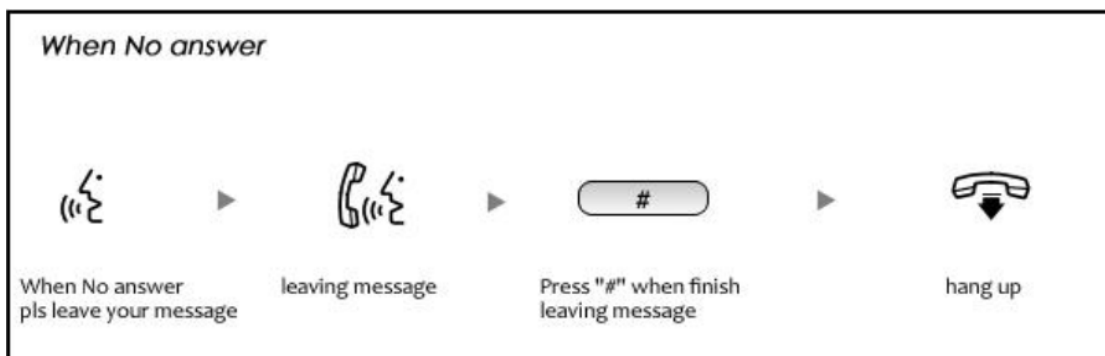
Calls that are not answered have the option to be sent to a voicemail account so the caller can leave a recorded message. Optionally, these recorded messages may be sent to a user's email account.

Click **【Basic】** -> **【Extension】** -> **【Edit】** the extension you want to configure.

Edit <span style="float: right;">X</span>			
<b>General</b>			
SIP:	<input checked="" type="checkbox"/>	IAX2:	<input type="checkbox"/>
Name:	800	Extension:	800
Password:	123456	Outbound CID:	
Dial Plan:	DialPlan1	Analog Phone:	None
<b>Voicemail</b>			
Voicemail:	<input checked="" type="checkbox"/>	VM Password:	1234
Delete VMail:	<input type="checkbox"/>	Email(Fax/Voicemail):	
<b>Other Options</b>			
Web Manager:	<input checked="" type="checkbox"/>	Agent:	<input checked="" type="checkbox"/>
Allow Being Spied:	<input type="checkbox"/>	Call Waiting:	<input type="checkbox"/>
Pickup Group:	1		
Mobility Extension:	<input type="checkbox"/>	Mobility Extension Number:	
<b>VoIP Settings</b>			
NAT:	<input checked="" type="checkbox"/>	Transport:	UDP
DTMF Mode:	RFC2833	SRTTP:	<input type="checkbox"/>
Permit IP:			
<b>Video Options</b>			
Video Call:	<input type="checkbox"/>		
<input type="checkbox"/> H.261	<input type="checkbox"/> H.263	<input type="checkbox"/> H.263+	<input type="checkbox"/> H.264
<b>Audio Codecs</b>			
<input checked="" type="checkbox"/> alaw	<input checked="" type="checkbox"/> ulaw	<input type="checkbox"/> G.722	<input checked="" type="checkbox"/> G.729
<input type="checkbox"/> G.726	<input type="checkbox"/> GSM	<input type="checkbox"/> Speex	

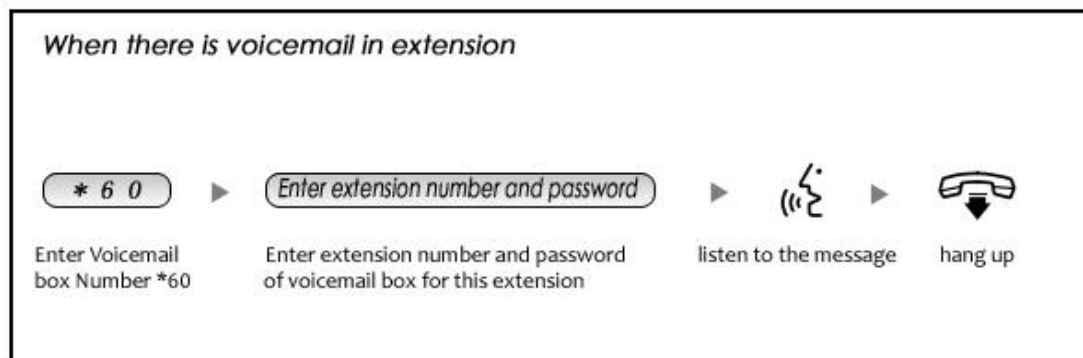
Please enable **Voicemail** before configuration, and configure **VM Password** and **Email** . If incoming calls are not answered, when the default ring time is over, the system will play: "please leave your message and press the "#"key ". Then voicemail will be sent to the specified mailbox by email.

To Leave a Message



To listen to the message using the user's desk phone



**Notice:**

1. Proper Email address is necessary to receive voicemail via email.
2. You must configure the SMTP and Email template. For detail settings, please see the detail configuration guide **【Voicemail】** in Chapter 3.

## 2.7 Call Center (Call Queues)

### Create Agent

To allow a user to be considered an agent in a Call Center queue, please check the “Agent” option for that specific user extension.

Click **【Basic】** -> **【Extension】** -> **【Edit】** the extension you want to configure:

Step1: Check **【Agent】** and **【Save】**

Edit			
<b>General</b>			
SIP:	<input checked="" type="checkbox"/>	IAX2:	<input type="checkbox"/>
Name:	800	Extension:	800
Password:	123456	Outbound CID:	
Dial Plan:	DialPlan1	Analog Phone:	None
<b>Voicemail</b>			
Voicemail:	<input checked="" type="checkbox"/>	VM Password:	1234
Delete VMail:	<input type="checkbox"/>	Email(Fax/Voicemail):	
<b>Other Options</b>			
Web Manager:	<input checked="" type="checkbox"/>	Agent:	<input checked="" type="checkbox"/>
Allow Being Spied:	<input type="checkbox"/>	Pickup Group:	1
Mobility Extension:	<input type="checkbox"/>	Mobility Extension Number:	
<b>VoIP Settings</b>			
NAT:	<input checked="" type="checkbox"/>	Transport:	UDP
DTMF Mode:	RFC2833	SRTP:	<input type="checkbox"/>
<b>Video Options</b>			
Video Call:	<input type="checkbox"/>		
<input type="checkbox"/> H.261 <input type="checkbox"/> H.263 <input type="checkbox"/> H.263+ <input type="checkbox"/> H.264			
<b>Audio Codecs</b>			
<input checked="" type="checkbox"/> alaw <input checked="" type="checkbox"/> ulaw <input type="checkbox"/> G.722 <input checked="" type="checkbox"/> G.729 <input type="checkbox"/> G.726 <input type="checkbox"/> GSM <input type="checkbox"/> Speex			

Step2: Click **【Inbound Control】** -> **【Call Queues】**

D-Link®		Logout																		
<ul style="list-style-type: none"><li>Home</li><li>Operator</li><li>Basic</li><li><b>Inbound Control</b></li><li>Inbound Routes</li><li>IVR</li><li>IVR Prompts</li><li>Call Queues</li><li>Ring Groups</li><li>Black List</li><li>Do Not Disturb</li><li>Time Based Rules</li><li>Advanced</li><li>Network Settings</li><li>Security</li><li>Report</li><li>System</li></ul>	<p>Call Queues 1</p> <p>Call Queues 1   Call Queues 2   Call Queues 3</p> <p><b>Call Queue Reference:</b></p> <p>Queue Number: 630   Label: _____</p> <p>Ring Strategy: Random</p> <p><b>Agents:</b></p> <p>You do not have any users defined as agents! <a href="#">click here</a> to manage users.</p> <table border="1"><thead><tr><th>Queue Options:</th><th>Announcements:</th></tr></thead><tbody><tr><td>Agent TimeOut(sec): 15 <input type="checkbox"/> Auto Pause</td><td><b>Caller Position Announcements</b> Frequency(sec): 30 Announce Hold Time: yes</td></tr><tr><td>Wrap-Up-Time(sec): 10</td><td><b>Periodic Announcements</b> Repeat Frequency(sec): 0</td></tr><tr><td>Max Wait Time(sec): _____</td><td>Announcements Prompt: _____</td></tr><tr><td>Max Callers: 8</td><td><b>If not answered</b> Destination: Hangup</td></tr><tr><td><input type="checkbox"/> Join Empty</td><td></td></tr><tr><td><input type="checkbox"/> Leave When Empty</td><td></td></tr><tr><td><input type="checkbox"/> Auto Fill</td><td></td></tr><tr><td><input type="checkbox"/> Report Hold Time</td><td></td></tr></tbody></table>	Queue Options:	Announcements:	Agent TimeOut(sec): 15 <input type="checkbox"/> Auto Pause	<b>Caller Position Announcements</b> Frequency(sec): 30 Announce Hold Time: yes	Wrap-Up-Time(sec): 10	<b>Periodic Announcements</b> Repeat Frequency(sec): 0	Max Wait Time(sec): _____	Announcements Prompt: _____	Max Callers: 8	<b>If not answered</b> Destination: Hangup	<input type="checkbox"/> Join Empty		<input type="checkbox"/> Leave When Empty		<input type="checkbox"/> Auto Fill		<input type="checkbox"/> Report Hold Time		<p><b>Announce Hold Time:</b> Should we include estimated hold time in position announcements? Either yes,no,or only once;hold time will not be announced if &lt;1 minute.</p>
Queue Options:	Announcements:																			
Agent TimeOut(sec): 15 <input type="checkbox"/> Auto Pause	<b>Caller Position Announcements</b> Frequency(sec): 30 Announce Hold Time: yes																			
Wrap-Up-Time(sec): 10	<b>Periodic Announcements</b> Repeat Frequency(sec): 0																			
Max Wait Time(sec): _____	Announcements Prompt: _____																			
Max Callers: 8	<b>If not answered</b> Destination: Hangup																			
<input type="checkbox"/> Join Empty																				
<input type="checkbox"/> Leave When Empty																				
<input type="checkbox"/> Auto Fill																				
<input type="checkbox"/> Report Hold Time																				
<p>Save   Cancel</p>																				

Reference

Item	Explanation
Queue Number	Define an extension number to identify the queue.
Label	Define the label for the queue.
Ring Strategy	<p>RingAll--Ring all available agents until one answers( default)</p> <p>RoundRobin – Starting with the first agent, ring the extension of each agent in turn until the call is answered.</p> <p>LeastRecent – ring the extension of the Agent who has least recently received a call</p> <p>FewestCalls – ring the extension of the Agent who has taken the fewest number of calls.</p> <p>Random – ring the extension of a random Agent.</p> <p>RRmemory -- RoundRobin with Memory, like RoundRobin above, except instead of the next call starting with the first agent, the system remembers which extension was called last and begins the round robin with the next agent .</p>
Agent	Check each agent that is to be a member of this specific Call Center Queue.

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

Reference:

Item	Explanation
Agent TimeOut(sec)	Specify the number of seconds to rin 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 so the stop receiving calls from the queue.
Wrap-Up-Time(sec)	<p>This is the amount of time in seconds that an agent has to complete work on a call after the call is disconnected.</p> <p>(Default is 0, which means no wrap-up time.)</p>
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 the callers who are allowed to wait in the queue. (Default is 0, which means no limitation.). With this number of callers in the queue already, 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 with no 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 "If not answered" destination. This option cannot be used with Join Empty simultaneously.
Auto Fill	Callers will be distributed to Agent 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), it will not be announced when the hold time is less than 1 minute.
Repeat Frequency(sec)	Interval time to play the voice menu for callers. ("0" mean not to play).
Announcement Prompt	Select a prompt as the Announcements Prompt from the IVR Prompts.

### 2.8 Conference Bridge

A conference bridge is a virtual meeting room that allows multiple callers to hear and speak to each other. The conference bridge can be protected with a password so only callers with the password can access the conference. The software supports up to three conference rooms. To configure a conference bridge, go to **【Advanced】** -> **【Conference】** :

Conference Default

Conference Default
Conference 2
Conference 3

**Conference Number**

Room Extension:

---

**Conference Password**

Guest Password:

Administrator Password:

---

**Conference Options**

Conference DialPlan

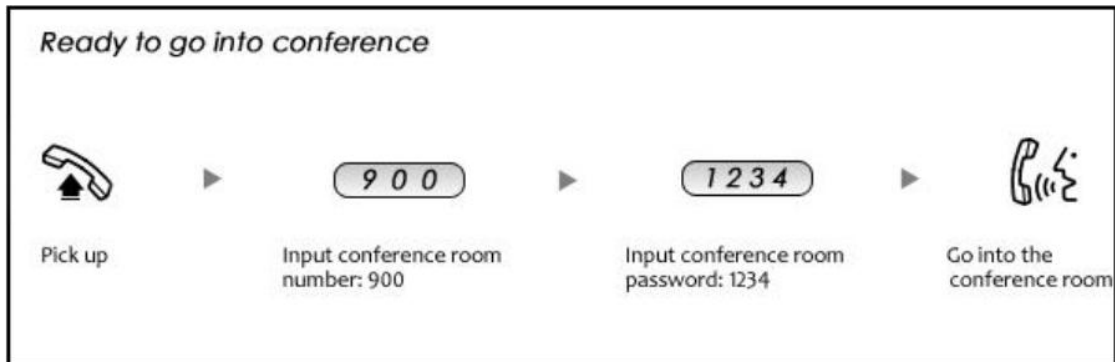
- Play hold music for first caller
- Enable caller menu
- Announce callers
- Record conference
- Quiet Mode
- Close the conference when last administrator exits
- Leader Wait

Reference:

Item	Explanation
Conference Number	The number that internal callers use to access the conference room, the default

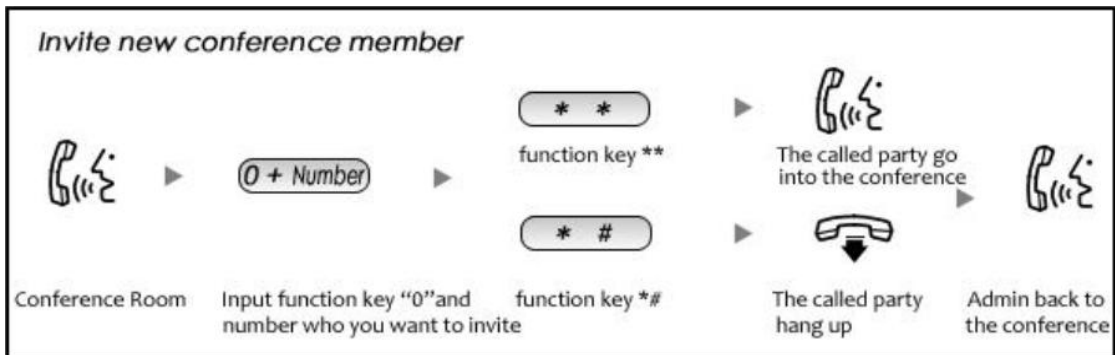
	number is "900".
Conference Password	Password for users to access the conference, e.g.:"1234".
Administrator Password	Password for administrator to access the conference.
Conference DialPlan	Use this dialplan to invite other participants.
Play hold music for the first participant	Check this option to play the hold music for the first participant in the conference until another participant enters in this conference.
Enable caller menu	Check this option to allow the participant to access the Conference Bridge menu by pressing "*" on the dialpad.
Announce callers	Check this option to announce to all Bridge participants that new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If check this option, all the participants in the conference can hear only, but it is not allowed to speak.
Leader Wait	Wait until the conference leader(administrator) entering the conference before starting the conference.

To join a conference, refer to the diagram as below:



While in a conference, the administrator can invite new guest (extension user or external number) into the conference. (Default password for admin is 2345)

As an administrator, to invite a new guest to the conference, refer to the diagram as below:



Chapter 3 Advanced

3.1 Options

General

Default settings for local extension and new extension.

Click **【Advanced】** -> **【Options】** -> **【General】** :

General

General
Global Analog Settings
Global SIP Settings

---

**Local Extension Settings**

Operator Extension:

Global Ring Time Set(sec):

Enable Transfer:

Enable Music On Ringback:

Auto-Answer:

Record Format:

---

**Default Settings for New User**

SIP:  IAX2:  Web Manager:  Call Waiting:

Agent:  Voicemail:  Delete VMail:  VM Password:

NAT:  Transport:  SRTP:

**Audio Codecs**

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

---

**Extension Preferences**

User Extensions  to

Conference Extensions  to

IVR Extensions  to

Queue Extensions  to

Ring Group Extensions  to

**Save** **Cancel**

Reference

Item	Explanation
Operator Extension	Set extension number for Operator.
Global RingTime Set	Set RingTime for every extension.
Enable Transfer	Check to enable Transfer.
Enable Music On Ringback	Check to enable Music On Ringback.
Record Format	Set the format for recording files. (GSM/WAV only)

Default Setting for New User	Check to enable the default settings.
Extension Preferences	Set the rule for extensions.

**Global Analog Settings**

Click **【Advance】** -> **【Options】** -> **【Global Analog Settings】**:

Global Analog Settings

General
Global Analog Settings
Global SIP Settings

**Caller ID Detect**

Caller ID Detection:   
 Caller ID Signaling: Bell-US ▾  
 Caller ID Start: Ring ▾  
 CID Buffer Length: 2500 ▾

---

**General**

Opermode: FCC ▾  
 Tone Zone: China ▾  
 Relax DTMF:   
 Send Caller ID After: 1 ▾  
 Echo Cancel:   
 Echo Training: no (yes/no/number)  
 Busy Detection:   
 Busy Count: 3

Save
Cancel

**Reference:**

Item	Explanation
Caller ID Detection	Enable/Disable Caller ID Detection
Caller ID Signaling	Select the mode of Caller ID Signaling.
Caller ID Start	Ring--Caller ID start before ring. Polarity--Caller ID start when polarity reversal starts.
CID Buffer Length	Default CID Buffer Length
Opermode	Set the Opermode for FXO/GSM Ports.
ToneZone	Select the ToneZone in your country.
Relax DTMF	Enable/Disable Relax DTMF inspection.
Echo Cancel	Enable/Disable Echo Cancel
Echo Training	Set Echo Training (default unit: ms)
Busy Detection	Enable/Disable Busy Detection.
Busy Count	Count the Busy Detection. It will be active when enable Busy Detection.

**Global SIP Settings**

【Global SIP Settings】 is appropriate for advanced administrators. Please contact our technical support department before modifying anything in this section.

**3.2 Voicemail**

Click 【Advanced】 -> 【Voicemail】 -> 【General】:

General

General
Email Settings

**VoiceMail Reference**

Max Greeting Time(sec):   
 Dial "0" for Operator:

---

**Voice Message Options**

Message Format:   
 Maximum Messages:   
 Max Message Time(min):   
 Min Message Time(sec):

---

**Playback Options**

Say Message CallerID  
 Say Message Duration  
 Play Envelope  
 Allow Users to Review

**Reference**

Item	Explanation
MaxGreeting Time(sec)	Maximum recording length for voicemail greetings
Dial "0" for Operator	Select this option to allow callers to press Dial "0" to transfer out of voicemail to the Operator.
Message Format	Save the voice message as this format, WAV(16-bit) or Raw GSM.
Maximum Messages	Maximum voicemail messages to be allowed to leave.
Max Message Time(min)	Maximum Time for each message to be allowed to leave.
Min Message Time(sec)	MinimumTime for each message. The message will be deleted automatically if the time is less than the min message time.
Say Message CallerID	Play the Caller ID of the caller before playing the voice message.
Say Message Duration	Play the message duration before playing the voice message.
Play Envelope	Play the date, time and caller ID for the voicemail message.
Allow Users to Review	Check this option to allow users to review the voice message.



Click **【Advance】** -> **【Voicemail】** -> **【Email Settings】**

Email Settings

General
Email Settings

**Template for Voicemail Emails**

Attach voicemail to email

Sender Name IP Phone System

From test@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}).

Save
Cancel

**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

Reference:

Item	Explanation
Attach voicemail to Email	The voicemail will be sent as attachment to the user's Email.
Sender Name	The sender's name will be displayed when you receive the Email.
From	Mailbox to send email
Subject	Subject of the Email.
Message	Input the Email template.

### 3.3 SMTP Settings

To allow email messages to be sent to users with attached voicemail and faxmail messages, the SMTP settings need to be configured.

Click **【Advance】** -> **【SMTP Settings】**:

SMTP Settings

---

**SMTP Settings:**

SMTP Server: \_\_\_\_\_

Port: 25

SSL/TLS:

Enable SMTP Authentication

Username: \_\_\_\_\_

Password: \_\_\_\_\_

**Send Test**

**Save    Cancel**

Reference

Item	Explanation
SMTP Server	You must set SMTP Server address or domain connected to the DVX-2002F/DVX-2005F, which is used for sending the voice message to Email.
Port	Port number for SMTP server. Default is 25, and it will be changed to 465 when you enable SSL/TLS.
SSL/TSL	Enable SSL/TLS.
Enable SMTP Authentication	If your SMTP server needs authentication, please enable this option, and configure the following.
Username	Input username of your Email.
Password	Input password of your Email.

Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.

**Send Test** X

Email Address: \_\_\_\_\_

Specify the email address and click 【Send】 -to send the test email. Verify that email was successfully sent or not. If no email was received, please modify the SMTP settings and retry.

**3.4 Email to Fax**

Users can send fax by Email. Please configure as below.

Click 【Advanced】 -> 【Email to Fax】

Email to Fax	
Enable:	<input type="checkbox"/>
Username:	<input type="text"/>
Password:	<input type="text"/>
IMAP Server:	<input type="text"/>
SSL/TLS:	<input type="checkbox"/>
Access Code:	<input type="text"/>
Dial Plan:	<input type="text" value=""/> ▼

Check "Enable", input username, password and IMAP Server(server format: imap.XX.com), select the DialPlan, then "Save" and "Activate".

**Practical Case:**

To Send a fax to telephone number 85337096: In DialPlan 1, there is prefix "9" before the telephone number; you need input the **【Access Code】**: 985337096 and make this the subject when sending Email. Then the fax will be sent by Email as attachment.

If you need dial the extension when sending fax, e.g.: fax number: 85337096 ext.800, you need use the **【Access Code】** : 985337096-800 as subject.

**3.5 Music Settings**

Management of Music on Hold, Music on Ringback, Music on Queue.

**【Music Settings】** :

Music Settings	
<b>Music Settings</b>	<b>Music Management</b>
<b>Music On Hold Reference</b>	
Music:	<input type="text" value="Music 1"/> ▼
<b>Music On Ringback Reference</b>	
Music:	<input type="text" value="Music 2"/> ▼
<b>Music On Queue Reference</b>	
Music:	<input type="text" value="Music 3"/> ▼
<b>Save</b> <b>Cancel</b>	

Select the different music file for different Music.

**【Music Management】**

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 wav(16bit/8000Hz/Mono), gsm, ulaw or alaw!  
The size is limited in 15MB!

Please choose file to upload: Please choose

Upload

Item	Explanation
Select Music Directory	Select which Music Directory you wish to load.
File	Display music name under the music file, you can delete it.
Select Music Directory	Select the file where you want to save your uploaded music.
Please choose file to upload	Select the music you want to upload. Note: music file must be WAV(16bit/8000Hz/Single), GSM, ulaw or alaw, and less than 15MB.

### 3.6 DISA

This feature allows an authorized user to call into the PBX and then place an outbound call using another trunk. For example, an employee working out of the office who needs to make an international call using trunks connected to the PBX. By calling the DISA number, after PIN authentication, the caller hears dial tone and can dial the call.

Please configure as below.

Click **【Advance】** -> **【DISA】** -- **【New DISA】**

**New DISA** X

Name: \_\_\_\_\_

PIN Set:   Without PIN

Record in CDR:

Response Timeout(sec): 5 \_\_\_\_\_

Digit Timeout(sec): 3 \_\_\_\_\_

Extension for this DISA(Optional): \_\_\_\_\_

**Allow Outbound Route**

Select DialPlan

**Reference**

Item	Explanation
Name	Define a name for DISA.
PIN Set	User will be prompted to input this number when PIN Authentication is needed.
Record in CDR	Check to record.
Response Timeout(sec)	The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds
Digit Timeout(sec)	The maximum interval time between digits when typing extension number. Default is 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 the DialPlan for this DISA.

**3.7 Follow Me**

This feature allows callers to automatically be forwarded to one or more internal extensions and/or one or more external phone numbers when the call is not answered at the primary extension.

Please configure as below:

Click **【Advanced】** -> **【Follow Me】** -> **【New Follow Me】** :

**New Follow Me** X

Extension:  ▼

Ring lasting for 20 seconds

Follow Me List: 

▲
▼

Select an extension, set the ring duration, and add the numbers in the Follow Me List; **【Save】** and **【Activate】** .

List Format: Extension Number, Ring Duration

E.g.: 806,30

808,20

806 rings, after 30 seconds, the call is going to 808

**【Follow Me Options】**

**Follow Me Options**

Playback the incoming status message prior to starting the follow-me step(sec).

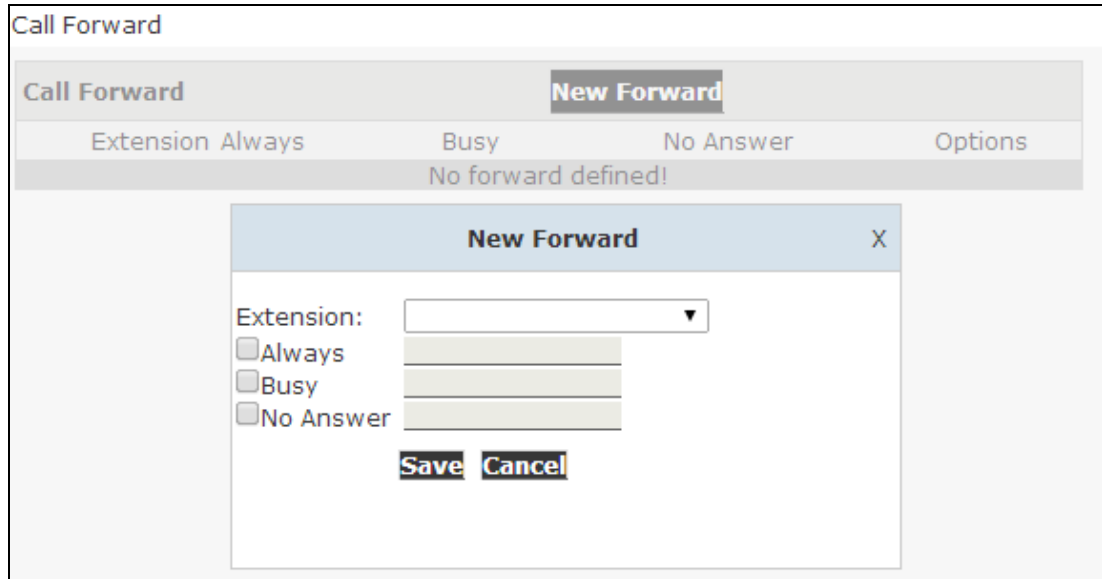
Record the caller's name so it can be announced to the callee on each step.

Playback the unreachable status message if we've run out of all steps or the callee was set not to be reachable.

**3.8 Call Forward**

The administrator can config the Call Forward on this page:

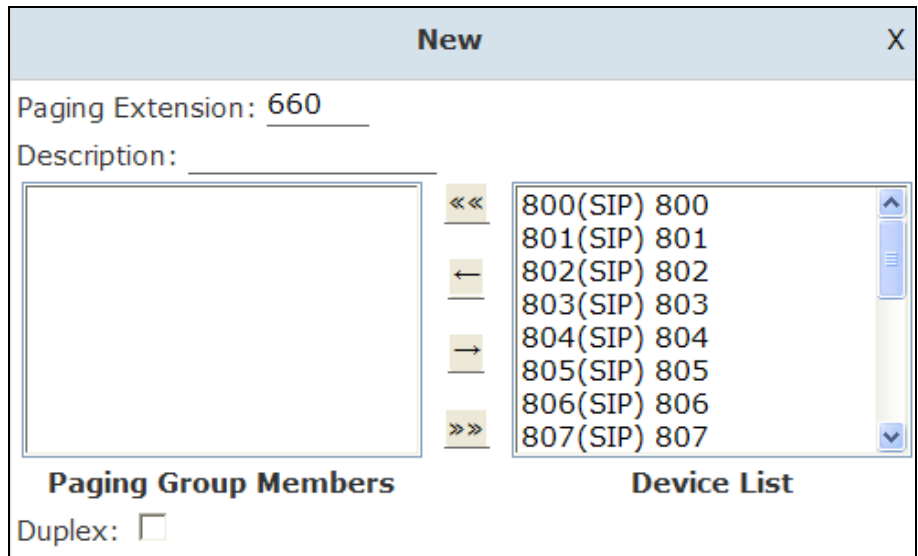
Click **【Advanced】** -> **【Call Forward】** :



### 3.9 Paging and Intercom

This feature allows setting up a Paging group so when the Paging extension is dialed, the listed extensions allow the caller to speak through the speaker phone. The extensions in the Paging group must use phones that support this feature. If the Duplex option is selected, and the listed extensions use phones that support Duplex, then all the phones in the paging group will be able to have two-way conversations.

Click **【Advanced】** -> **【Paging and Intercom】** -> **【New Paging Group】** :



Reference:

Item	Explanation
Paging Extension	Define an extension for this Paging Group.
Description	Define a name for this Paging Group.
Paging Group Members	Selected devices in this Paging Group.
Device List	Select device(s) here to Paging Group.

<p>Duplex</p>	<p>Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it look like an "instant conference".</p>
---------------	--

### 3.10 PIN Sets

This feature allows an administrator to specify a list of PIN codes in a PIN Set. An Outbound Route can be specified that a valid PIN code from a selected PIN Set must be used in order to have access to a give Outbound route (e.g. for long distance or international calling).

Please configure as below.

Click **【Advanced】** -> **【PIN Sets】** -> **【New PIN Set】** :

1. PIN Set Name Define the name for this PIN Set.
2. PIN List Define PIN codes in this list.

### 3.11 Call Recording

This feature allows an administrator to enable Call Recording to record incoming and/or outgoing calls related to the specified extension.

Please configure as below:

Click **【Advanced】** -> **【Call Recording】** -> **【New Call Recording】** :



**New Call Recording** X

Extension:

**Call Recording Time**

Always Recording:

Start Time:  :  End Time:  :

Start Day:  End Day:

**Call Recording Settings**

Inbound Record:                       Outbound Record:

Reference:

Item	Explanation
Extension	Define an extension for recording.
Call Recording Time	Set the time to record.
Inbound Record	Check to record inbound calls.
Outbound Record	Check to record outbound calls.

### 3.12 Speed Dial

This feature allows setting up system wide speed dial numbers that translate a feature code (\*99) plus a two-digit code (00-99) into an external phone number.

Please configure as below.

Click **【Advanced】** -> **【Speed Dial】** -> **【New Speed Dial】** :

**New Speed Dial** X

Notice: Don't forget to add the outbound dial prefix if you would like to dial an outside number

Source Number: \_\_\_\_\_

Destination Number: \_\_\_\_\_

E.g.: prefix is \*99 , speed number is 00, destination telephone number is 85337096.

When dial \*9900, the call is going to 85337096 automatically.

### 3.13 Smart DID

Smart DID: After extension user makes an outbound call, the call is ringing back to DVX-2002F/DVX-2005F, and directed to the extension who made the last call. Please configure as below.

Click **【Advanced】** -> **【Smart DID】** :

Smart DID			
Enable: <input type="checkbox"/>			
<b>Save</b> <b>Cancel</b>			
Smart DID Rules List			New Smart DID Rule
Pattern	Strip	Prepend	Options
1 X.			<b>Edit</b> <b>Delete</b>

Check “Enable” and “Save” to make this function activate.

Click **【New Smart DID Rule】** to display the following diagram:

**New Smart DID Rule**

Pattern: \_\_\_\_\_

Strip: \_\_\_ digits before dialing

Prepend: \_\_\_ before dialing

Input the pattern and define how many digits need to be stripped or prepend, then click “Save”--“Activate”.

### 3.14 Callback

This feature allows an external caller to place an inbound call to the Coovpx IP PBX. The inbound call will be disconnected and subsequently the PBX will place an outbound call back to this number and forwarded to defined destination after the call is connected.

Please configure as below.

Click **【Advanced】** -> **【Callback】** :

Callback Number Settings

**Callback Number Settings**

Enable:

Strip: \_\_\_ digits before dialing

Prepend: \_\_\_ before dialing

DialPlan:

**Save** **Cancel**

---

**List of Callback Number** **New Callback Number**

Callback Number	Destination	Options
No Callback Number defined!		

Enable this function; select DialPlan, and define the callback rule (strip digits or prepend prefix).

Click **【New Callback Number】** to add callback number.

**New Callback Number** X

Callback Number:

Destination:

Input callback number and define the destination.

### 3.15 Phone Book

When incoming call Caller ID matches the number in the phone book, the name of matched number will be displayed. Please configure as below.

Click **【Advanced】** -> **【Phone Book】** :

Phone Book

**Phone Book** **Import** **Export** **Delete All**

The prefix of speed dial:  **Save** **Cancel**

Field:   **Filter** **Create Contact** **Delete Selected**

Name	Phone Number	Speed Dial	Options
No Contact defined!!			

Reference:

Item	Explanation
<a href="#">Import &amp; Export</a>	Import & Export a list, make sure it's UTF-8 coded if it's not in English
<a href="#">Delete All</a>	Delete all the contacts from the phone book
<a href="#">The prefix of speed dial</a>	Set the prefix of speed dial
<a href="#">Filter</a>	Search contacts, by name, phone number or speed dial
<a href="#">Create Contact</a>	Create a contact

Delete Contact	Delete a selected contact
Call	Click to call the number directly

Click **【Create Contact】** to see the following diagram:

**Create Contact** X

Name: \_\_\_\_\_

Phone Number: \_\_\_\_\_

Speed Dial: \_\_\_\_\_

**Save** **Cancel**

### 3.16 Feature Codes

Click **【Advanced】** -> **【Feature Codes】** to see the following diagram, and you can define the code for each feature.

**Feature Codes Management**

**Call Parking**

Extension to Dial for Parking Calls: 700

Extension Range to Park Calls: 701-720

Call Parking Time(sec): 45

Parking Hints:

**Pickup Call**

Pickup Extension: \*8

Pickup Specified Extension: \*\*

**Transfer**

Blind Transfer: #

Attended Transfer: \*2

Disconnect Call: \*

Timeout for answer on attended transfer(sec): 15

**One Touch Recording**

One Touch Recording: \*1

**Call Forward**

Enable Forward All Calls: \*71

Disable Forward All Calls: \*071

Enable Forward on Busy: \*72

Disable Forward on Busy: \*072

Enable Forward on No Answer: \*73

Disable Forward on No Answer: \*073

Reference:

Item	Explanation
Extension to Dial for Parking Calls	Define an extension for parking calls.
Extension Range to Park Calls	Define the extension range for parking calls. (e.g.: 701-720)

Call Parking Time(sec)	Define the time for parking calls. IP PBX will return the call to the extension after this time limit has expired.
Pickup Extension	This feature code will pick up a call given that the callers extension and the ringing extension are in the same pickup group and call group.
Pickup Specified Extension	This feature code allows a caller to Pickup a call ringing on the specified extension. Default: Dial**+extension number to pickup the specified extension.
Blind Transfer	To Allow unattended or blind transfer while on a call based on the following steps: <ol style="list-style-type: none"> <li>1. While on a call with caller "A", the user dials the blind transfer key sequence (in this case "#"). The system places the original call with "A" on hold, says "Transfer" then gives a dial tone.</li> <li>2. dial the transferee extension or phone number you wish to transfer the call to "B" and hangup the phone.</li> <li>3. The original caller "A" is transferred immediately to the transferee "B" and "B" sees the callerid of "A".</li> </ol>
Attended Transfer	To Allow attended or supervised transfer while on a call based on the following steps: <ol style="list-style-type: none"> <li>1. While on a call with caller "A", the user dials the supervised transfer key sequence (in this case "*2"). The system places the original call with "A" on hold, says "Transfer" then gives a dial tone.</li> <li>2. dial the transferee extension or phone number you wish to transfer the call to "B" and wait for "B" to answer the phone and talk to "B" to introduce the call.</li> <li>1. If "B" does not wish to take the call, "B" can hang up the call and you are returned to your call with "A".</li> <li>2. If "B" wishes to accept the call, you hang up the phone and caller "A" is transferred to the transferee "B".</li> <li>3. If the call goes to voicemail or you wish to abort the transfer, simply press the "disconnect call" key sequence (in this case "**") and the transfer will be aborted and you will be back on the call with the original caller "A".</li> </ol>
Disconnect Call	Disconnect the current transfer call (for Attended transfer).
Timeout for answer on attended transfer (sec)	Set the timeout value
One Touch Recording	Configure the function key for One Touch Recording
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable "Do Not Disturb"
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklist number.
Voicemail	Configure the function keys for entering voicemail and check extension voicemail.
Invite Participant	In conference, the administrator can invite people into the conference by dialing "0". After pressing "0", you will get dialtone, and you can dial to invite people. After the call is connected, please press ** to direct the people into

	the conference, or *# to hang up the current call and return to the conference.
Create Conference	During the call, you can dial *0 to forward to the conference with the callee.
Return to conference with participant	In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dialtone, and you can dial to invite the participant; when the call is connected, dial "***" to return to the conference with invited participant.
Return to conference without participant	In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dialtone, and you can dial to invite the participant. When the call is connected, you can dial "*#" to hang up and return the conference yourself.
Pause Queue Member Extension	Pause the agent, and the agent cannot receive the call.
Unpause Queue Member Extension	Unpause the agent, and the agent can receive the call.
Others	Function key for Intercom/ Paging/ Directory

**3.17 IP Phone Provisioning**

When many IP Phones are needed, please record the MAC, extension number, and username of each phone according to the format (please take reference of the auto provision script file model for details) , then import the format file, once the phone is connected to the local network, it will get the extension number and password automatically. There are two operation methods to fulfill this function, please see details as below.

**Enable DHCP service**

Click **【System】** -> **【Network Advanced】** -> **【Enable】** DHCP Server in the following diagram:

**DHCP Server Settings**

Enable:

Start IP: 192.168.1.101

End IP: 192.168.1.200

Subnet Mask: 255.255.255.0

Gateway: 192.168.1.1

Primary DNS: \_\_\_\_\_

Lease Time(min): 1440

TFTP Server: \_\_\_\_\_

**Save** **Cancel**

Then Click **【Advanced】** -> **【Phone Provisioning】** -> **【New Phone】** :

New Phone		X
<b>General</b>		
Enable:	<input checked="" type="checkbox"/>	
Manufacturer:	<input type="text"/>	Type: <input type="text"/>
MAC:	<input type="text"/>	
<b>Line</b>		
Line1	Extension: <input type="text"/>	Label: <input type="text"/>
<b>Save</b> <b>Cancel</b>		

Enable Phone Provisioning in **【Basic】** , select the IP Phone manufacture, input MAC of the phone, and select the extension for provisioning.

**Notice**

Supports D-Link Series IP Phone.

---

**Chapter 4 Network Settings**

**4.1 Network**

You can configure the WAN Port, and define the Virtual Interface.

Click **【Network Settings】** -> **【Network】** -> **【IPv4 Settings】** :

Ethernet Port Setup	
IP Assign:	Static ▾
IP Address:	192.168.1.100
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	8.8.8.8
Alternate DNS:	
Virtual Interface	
<input type="checkbox"/> IP AddressV1:	<input type="text"/>
<input type="checkbox"/> IP AddressV2:	<input type="text"/>
Subnet MaskV1:	<input type="text"/>
Subnet MaskV2:	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

**Reference**

Item	Explanation
IP Assign	Static/ DHCP/PPPoE supported.
Virtual Interface	Define the virtual interface for WAN Port.

Click **【Network Settings】** -> **【Network】** -> **【IPv6 Settings】**

Ethernet Port Setup	
Enable:	<input type="checkbox"/>
IPv6 Address:	<input type="text"/>
Prefix Length:	<input type="text"/>
Gateway:	<input type="text"/>
Primary DNS:	<input type="text"/>
Alternate DNS:	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

**IPv6 Reference:**

Item	Explanation
Enable	Enable IPv6, define the IPv6 address, gateway, and DNS.

Click **【Network Settings】** -> **【Network】** -> **【VLAN Settings】** :



<b>VLAN 1</b>	
Enable:	<input type="checkbox"/>
VLAN ID:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
Subnet Mask:	<input type="text"/>
<b>VLAN 2</b>	
Enable:	<input type="checkbox"/>
VLAN ID:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
Subnet Mask:	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

VLAN Reference:

Item	Explanation
Enable	Enable VLAN, define the VLAN address and VLAN ID.

### 4.2 Static Routing

Click **【Network Settings】** -> **【Static Routing】** :

<b>New Static Routing</b>		X
Destination Network:	<input type="text"/>	
Subnet Mask:	<input type="text"/>	
Gateway:	<input type="text"/>	

Item	Explanation
Destination	Set destination network for static routing.
Subnet Mask	Set subnet mask of the destination network.
Gateway	Define the gateway accessing the destination network.

Click **【Network Settings】**->**【Static Routing】**->**【Routing Table】**, the current routing information will be displayed as below:

<b>Routing Table:</b>						
Kernel IP routing table						
Destination	Gateway	Genmask	Flags	Metric	Ref	Use Iface
0.0.0.0	192.168.1.1	0.0.0.0	UG	0	0	0 eth0
192.168.1.0	0.0.0.0	255.255.255.0	U	0	0	0 eth0

### 4.3 VPN Server

DVX-2002F/DVX-2005F supports three kinds of VPN servers: L2TP/PPTP/OpenVPN.

Click **【Network Settings】** -> **【VPN Server】**:

VPN Server	
<input type="radio"/> L2TP <input checked="" type="radio"/> PPTP <input type="radio"/> OpenVPN	
Enable:	<input checked="" type="checkbox"/>
Remote IP:	<u>192.168.11.1</u> - <u>12</u>
Local IP:	<u>192.168.11.90</u>
Primary DNS:	<u>61.139.2.69</u>
Alternate DNS:	<u>8.8.8.8</u>
Timeout(sec):	<u>120</u>
Authentication Method:	<input type="checkbox"/> chap <input type="checkbox"/> pap <input checked="" type="checkbox"/> mschap <input checked="" type="checkbox"/> mschap-v2
Enable mppe128:	<input checked="" type="checkbox"/>
Debug:	<input type="checkbox"/>

Reference:

Item	Explanation
VPN Server Mode	Three kinds of VPN servers L2TP/PPTP/OpenVPN supported (Only one mode can be enabled simultaneously)
Enable	Enable/Disable VPN Server

When the mode is L2TP or PPTP VPN server, click **【Network Settings】** -> **【VPN Server】** -> **【VPN Users Management】**:

List of VPN Users		New VPN User	
Username	Availability	Options	
1 Test	yes	<b>Edit</b>	<b>Delete</b>

This page is used for management of VPN username and password.

When the mode is OpenVPN server, click **【Network Settings】** -> **【VPN Server】** -> **【OpenVPN Certificate Download】**:

List of OpenVPN Certificate		New Certificate	Delete Selected
<input type="checkbox"/>	Certificate Name	Options	
No OpenVPN Certificate defined! Please click on 'Create New Certificate' button to create a new OpenVPN Certificate			

This page is used for management of OpenVPN certificate file.

#### 4.4 VPN Client

DVX-2002F/DVX-2005F supports four kinds of VPN Clients: L2TP /PPTP /OpenVPN /N2N  
 Click **【Network Settings】** -> **【VPN Client】**:

**VPN Client**

L2TP
  PPTP
  OpenVPN
  N2N

Enable:

Enable 40/128-bit encryption for MPPE:

Server Address: \_\_\_\_\_

Username: \_\_\_\_\_

Password: \_\_\_\_\_

Default Gateway:

Reference:

Item	Explanation
VPN Client	Four kinds of VPN Clients supported: L2TP/PPTP/OpenVPN/N2N (Only one mode can be enabled simultaneously)
Enable	Enable/Disable VPN Client

#### 4.5 DHCP Server

Click **【Network Settings】** -> **【DHCP Server】**:

**DHCP Server Settings**

Enable:

Start IP: 192.168.1.101

End IP: 192.168.1.200

Subnet Mask: 255.255.255.0

Gateway: 192.168.1.1

Primary DNS: 61.139.2.69

Lease Time(min): 1440

TFTP Server: \_\_\_\_\_

Click **【Network Settings】** -> **【DHCP Server】** -> **【DHCP Client List】** :

**DHCP Client List:**

Mac Address	IP Address	Host Name	Expires in
6c:3e:6d:e0:f2:00	192.168.1.101	iPhone	expired
00:03:58:45:87:9a	192.168.1.102		expired
0c:74:c2:47:71:6d	192.168.1.103	hnteki-iPhone	expired
20:c9:d0:85:3b:fb	192.168.1.104		expired
08:ed:b9:e7:c5:7f	192.168.1.105	DPVYE1J0WCAAC7I	expired
78:e4:00:8e:c3:99	192.168.1.106	LBSZLACHCIC	22:10:25
68:a3:c4:ef:5d:8b	192.168.1.107	HBWang	1 days 00:00:00
0c:72:2c:5a:39:41	192.168.1.108	MW150R	00:00:57

This page is used to display DHCP Client address and related information.

When DHCP Server distributes address, the Client's MAC address is associated with the IP address, and then the device will get the same IP address every time.

Click **【Network Settings】** -> **【DHCP Server】** -> **【Static MAC】** -> **【New Static MAC】** :

**New Static MAC** X

MAC Address: \_\_\_\_\_

IP Address: \_\_\_\_\_

#### 4.6 DDNS Settings

After setting DDNS (Dynamic Domain Network Server), DVX-2002F/ DVX-2005F IP PBX settings will be visited remotely. Click **【Network Settings】** -> **【DDNS Settings】**:

**DDNS Settings**

Enable:

DDNS Server:

Username: \_\_\_\_\_

Password: \_\_\_\_\_

Domain: \_\_\_\_\_

DVX-2002F/ DVX-2005F support DDNS provided by Dyndns.org / No-ip.com / zoneedit.com.

#### 4.7 SNMPv2 Settings

SNMP(Simple Network Management Protocol): Used for remote management.

Click **【Network Settings】** -> **【SNMPv2 Settings】**:

SNMPv2 Settings

**Read Only**

Enable:

RO Community:

RO Network: \_\_\_\_\_ / \_\_\_\_

---

**Read and Write**

Enable:

RW Community:

RW Network: \_\_\_\_\_ / \_\_\_\_

#### Reference

Item	Explanation
Enable	Enable "Read Only" of SNMP
RO Community	Define the name of RO Community of SNMP
RO Network	Define network of RO

## 4.8 Trouble Shooting

You can ping other network device through DVX-2002F/ DVX-2005F IP PBX and track network routing by command "Traceroute".

Click **【Network Settings】** -> **【Trouble Shooting】** :

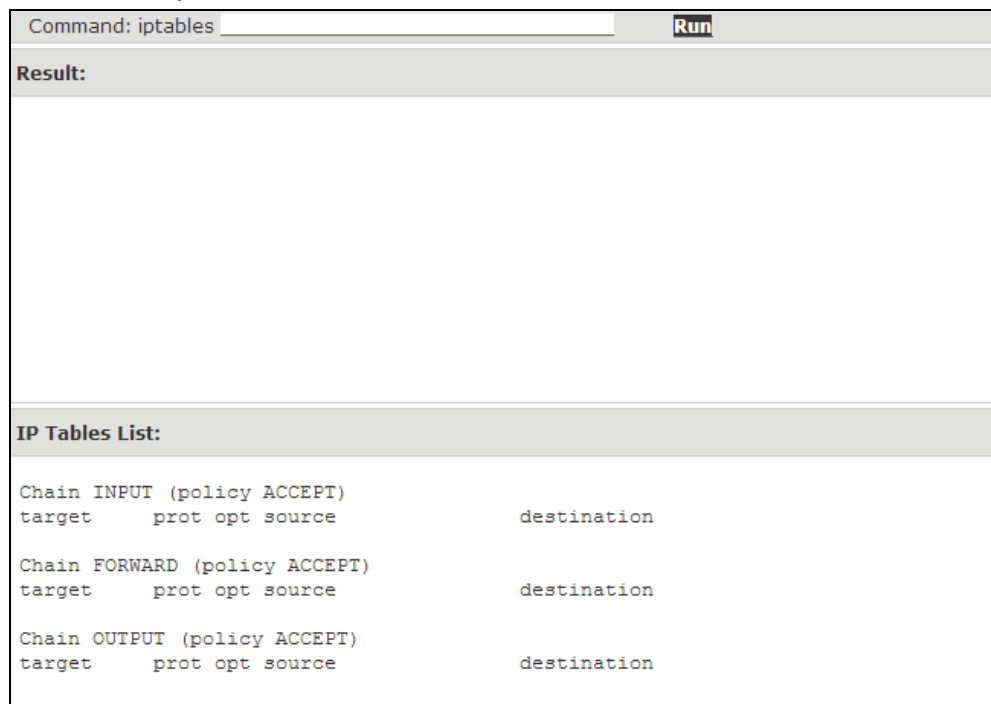
```
Ping 192.168.1.1      Packets: 4      Run Stop
PING 192.168.1.1 (192.168.1.1): 56 data bytes
64 bytes from 192.168.1.1: seq=0 ttl=64 time=1.677 ms
64 bytes from 192.168.1.1: seq=1 ttl=64 time=0.964 ms
64 bytes from 192.168.1.1: seq=2 ttl=64 time=1.057 ms
64 bytes from 192.168.1.1: seq=3 ttl=64 time=0.950 ms

--- 192.168.1.1 ping statistics ---
4 packets transmitted, 4 packets received, 0% packet loss
round-trip min/avg/max = 0.950/1.162/1.677 ms
```

**Chapter 5 Security**

**5.1 Firewall**

Click **【Security】** -> **【Firewall】**



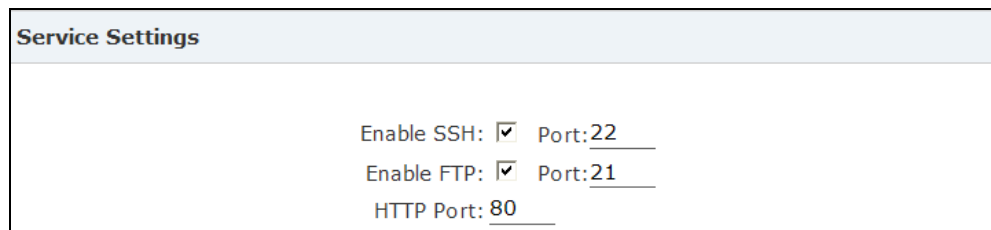
Iptables Command:

- Check iptables list iptables -L -n
- Clear iptables list iptables -F
- Deny an IP (192.168.0.3) iptables -A INPUT -s 192.168.0.3 -j DROP
- Deny every IP to access 80 port iptables -A INPUT -p tcp --dport 80 -j DROP
- Deny IP (192.168.0.3) to access 80 port iptables -A INPUT -s 192.168.0.3 -p tcp --dport 80-j DROP

**5.2 Service**

**【Service】** : settings of SSH/FTP and HTTP Port.

Click **【Security】** -> **【Service】** :



Enable SSH to login background management system through SSH.

Enable FTP to allow uploading files to system through FTP.

### 5.3 SIP Allowed Address

Define an allowed address, from which every SIP request will never be filtered or refused.

Click **【Security】** -> **【SIP Allowed Address】** :

SIP Allowed Address

List of SIP Allowed IP Address **Add Allowed IP**

Allowed IP	Options
No SIP Allowed Address defined!	

**Add Allowed IP** X  
Allowed IP: \_\_\_\_\_  
Subnet Mask: \_\_\_\_\_  
**Save** **Cancel**

Chapter 6 Report

6.1 Record List

Check recordings of specified extension or conference here, or delete the recording file.

【Record List】 :

Extension:	<input type="text"/>	<b>Delete</b>
Start Date:	Apr ▾ 15 ▾ 2014 ▾	End Date: Apr ▾ 15 ▾ 2014 ▾ <b>Filter</b>
<b>List of Recording Files</b>		<b>Delete Selected</b>
<input type="checkbox"/>	Caller ID	Destination ID Date Options

【Conference】 :

Start Date:	Apr ▾ 15 ▾ 2014 ▾	End Date: Apr ▾ 15 ▾ 2014 ▾ <b>Filter</b>
<b>List of Conference Record Files</b>		<b>Delete Selected</b> <b>Delete All</b>
<input type="checkbox"/>	Conference Room	Date Options

【One Touch Recording】

Extension:	<input type="text"/>	<b>Delete</b>
Start Date:	Apr ▾ 15 ▾ 2014 ▾	End Date: Apr ▾ 15 ▾ 2014 ▾ <b>Filter</b>
<b>List of Recording Files</b>		<b>Delete Selected</b>
<input type="checkbox"/>	Caller ID	Destination ID Date Options

6.2 Call Logs

Check call logs by caller ID or callee ID.

Click 【Report】 -> 【Call Logs】 :

Call Logs						
Start Date:	Apr ▾ 15 ▾ 2014 ▾	Field:	Caller ID ▾	<b>Filter</b>		
End Date:	Apr ▾ 15 ▾ 2014 ▾			<b>Download</b>	<b>Delete</b>	
Call Start	Caller ID	Destination ID	Account Code	Duration(sec)	Disposition	



**Notice**

Duration in the call logs is not real charged duration. If you need billing, PSTN must support polarity reversal function, and meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for the DVX-2002F/ DVX-2005F IP PBX.



The number in the call logs can be added in the phone book directly:

Call Logs

Start Date: Feb 1 2014      Field: Caller ID      Filter

End Date: Mar 6 2014      Download      Delete

Call Start	Caller ID	Destination ID	Account Code	Duration(sec)	Disposition
2014-02-28 14:24:38	18380217610	805		0	NO ANSWER
2014-02-28 14:24:31	<18380217610>				
2014-02-28 14:24:00	806 <806>	315828035910		9	ANSWERED
2014-02-28 14:23:10	806 <806>	315828035910		0	NO ANSWER
2014-02-28 14:22:49	ReDial 315828035910	315828035910		0	ANSWERED
2014-02-28 14:22:57	<315828035910>				
2014-02-28 14:19:56	805 <805>	218380217610		14	ANSWERED
2014-02-28 14:20:06	<315828035910>	callback		3	ANSWERED
2014-02-28 14:19:29				19	ANSWERED
2014-02-28 14:04:46				3	ANSWERED
2014-02-28 13:45:22				16	ANSWERED
2014-02-28 13:45:56				7	ANSWERED
2014-02-28 13:45:33				4	ANSWERED
2014-02-28 13:44:05				0	ANSWERED
2014-02-28 13:44:43				0	ANSWERED
2014-02-28 13:44:16				6	ANSWERED
2014-02-28 13:42:34				0	ANSWERED
2014-02-28 13:43:01	805 <805>	conference		30	ANSWERED
2014-02-28 13:42:48	805 <805>	812		0	ANSWERED
2014-02-28 13:41:34	805 <805>	806		1	ANSWERED
2014-02-28 13:42:06	805 <805>	conference		5	ANSWERED
2014-02-28 13:41:50	805 <805>	812		0	ANSWERED
2014-02-28 13:41:16	805 <805>	806		1	ANSWERED
2014-02-28 13:41:16	805 <805>	900		16	ANSWERED

**Create Contact** X

Name: \_\_\_\_\_

Phone Number: **218380217610**

Save Cancel

### 6.3 System Logs

Click **Report** -> **System Logs** , you can download/ delete the system logs.

System Logs

**System Logs**

Enable System Log:       Enable PBX Log:

Enable PBX Debug Log:       Enable Access Log:

**Save** **Cancel**

**List of Logs** **Download Selected** **Delete Selected**

<input type="checkbox"/>	Name	Type	Options
No log file found!			

Chapter 7 System

7.1 Time Settings

Time settings for DVX-2002F/ DVX-2005F IPPBX. The system supports either NTP or Manual Time Set.

【NTP】 :

**Time Settings**

NTP       Manual Time Set

NTP Server:

Time Zone:

Reference:

Item	Explanation
NTP Server	Define the NTP Server. You can input the IP address or domain of this server, whether it's local or remote. Default server is pool.ntp.org. Be aware that the DVX-2002F/ DVX-2005F IP PBX needs to be able to connect to an NTP server to properly function.
Time Zone	Select your time zone so that the system will set time based on the time zone.

【Manual Time Set】 :

**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 **Sync**

After entering Year/ Month/ Day/ Hour/ Minute, then save and activate.

Or, you can click 【Sync】 to synchronize with current PC time.

**7.2 Data Storage**

When you need mass storage of recording files, voicemails, call logs, etc, you can upload these files to FTP server through FTP Data Storage based on the specified time frequency  
Click **【System】** -> **【Data Storage】** :

**FTP Data Storage**

Enable:

Server Address:

Username:

Password:

Directory:

Automatically upload frequency(day):

Time of automatically upload:  :

Forcibly upload when the flash storage is over:

**Reference**

Item	Explanation
Enable	Enable FTP Data Storage.
Server Address	Set FTP server address (IP address or domain).
Username	Username for login FTP.
Password	Password for login FTP.
Directory	Define a directory used for storage on FTP server.
Automatically upload frequency (day)	Define frequency by days to upload the data.
Time of automatically upload	Define the time to upload the data.
Forcibly upload when the flash storage is over	Forcibly upload data when flash storage is over the percentage value.

Check from **【Data Storage Log】** :

Data Storage

Data Storage Log

Data Storage Log

Refresh

Clear

Click **【Refresh】** to refresh data storage log.

Click **【clear】** to clear data storage log.

**7.3 Management**

**【Management】** is used for modify password of DVX-2002F/ DVX-2005F IPPBX, and the settings of system voice.

Click **【System】** -> **【Management】** :

**Change Password**

Password: \_\_\_\_\_  
 New Password: \_\_\_\_\_  
 Retype New Password: \_\_\_\_\_

**Apply**

---

**Set Language**

Set Voice Language: English ▼

**Save**

### 7.4 Backup

Click **【System】** -> **【Backup】**

Backup
Upload Backup File

**List of Backups**

**Take a Backup**

	Name	Date	Options
1	backup_2014apr15_181017	Apr 15, 2014	<span style="background-color: #cccccc; padding: 2px 5px;">Restore</span> <span style="background-color: #cccccc; padding: 2px 5px;">Delete</span> <span style="font-size: 1.2em; vertical-align: middle;">▼</span>

Reference:

Item	Explanation
Take a Backup	Take a backup of the current system configuration.
Restore	Restore system to the specified backup configuration.
Delete	Delete specified backup file.

Click the download button “” to download the specified backup file and manage locally.

Click **【Upload Backup File】** to upload the backup file here.

**Upload Backup File**

**Note: Don't change the backup file name.**

Please choose file to upload: Please choose

**Upload**

Click **【browse】** to select the local backup file, and click **【Upload】** to upload the backup file to

system.

### 7.5 Reset & Reboot

If you need reset the system to factory defaults or reset, please click **【System】** -> **【Reset & Reboot】** : **Restoring factory settings will make configuration data in the system lost.**

<b>Factory Defaults</b>
<b>Warning:</b> Restore factory settings,will lost all configuration data on the system!
<b>Factory Defaults</b>
<b>Reboot</b>
<b>Warning:</b> Rebooting the system will terminate all active calls!
<b>Reboot</b>

Click **【Factory Defaults】** to reset the system to factory defaults.

Click **【Reboot】** to reboot the system.

### 7.6 Upgrade

#### 7.6.1 WEB Upgrade

Click **【System】** -> **【Upgrade】** -> **【WEB Upgrade】** :

<b>Upgrade System Package</b>
<input checked="" type="radio"/> WEB Upgrade <input type="radio"/> TFTP Upgrade
Restore Default Set: <input type="checkbox"/>
Please choose file to upload: <input type="button" value="Please choose"/>
<b>Upload</b>

Click **【Browse】** to select the firmware file, then click **【Upload】** to upload the selected firmware to system and finish the upgrading automatically.

If check **【Restore Default Set】** , the system will clear all the configuration and reset to factory default.

**7.6.2 TFTP Upgrade**

Click **【System】** -> **【Upgrade】** -> **【TFTP Upgrade】** :

Upgrade System Package	
<input type="radio"/> WEB Upgrade	<input checked="" type="radio"/> TFTP Upgrade
Restore Default Set: <input type="checkbox"/>	
Enter The Package Name: _____	
TFTP Server IP address: _____	

**Reference:**

Item	Explanation
Restore Default Set	System will restore to factory defaults after checking this option.
Enter The Package Name	Enter the package name for upgrading.
TFTP Server IP address	Enter your TFTP server IP address.