

# **DVX-1000**



## **FEATURES**

#### **Key Features:**

- Includes 25 User Extensions
- Supports 25 Simultaneous Inbound/Outbound Calls
- Supports Multiple Users Across Multiple Sites
- Add External Analog Trunk Gateways to Use Standard Phones
- Save Money by using Internet Phone Service (VoIP)

#### **Networking Features**

- SIP (RFC 3261) Compliant
- 10/100Mb IEEE 802.3 Compliant
- Integrated DoS/IDS Firewall

#### **Telephony Features**

- Caller ID
- Call Transfer
- Call Forward
- Call Park/Hold
- Voicemail
- IVR/Auto Attendant
- Hold Music
- · Customizable Greetings

#### **Integrated Conference Bridge**

- Dial In/Dial Out
- Access Control
- Conference Recording
- E-Mail Notifications

# S T A C K<sup>™</sup> IP Telephony SIP IP-PBX with Conferencing Server

D-Link, an industry leader in networking, introduces the DVX-1000, a SIP-based IP-PBX for up to 25 extensions.

Internet IP telephony, also called Voice over IP (VoIP), is defined as the transport of telephone calls over the Internet as standard Internet data packets. Internet telephone calls can originate from traditional phone handsets via phone line-to-Internet (Analog Trunk) gateways, by PCs using software, or embedded devices (IP Phones). Most of the interest in Internet telephony is motivated by cost savings and ease of developing and integrating new services. Internet telephony integrates a variety of services provided by the current Internet and the Public Switched Telephone Network (PSTN) infrastructure.

The DVX-1000 offers all of the essential telephony features required for small businesses. Features such as call forwarding, call hold, find me-follow me, and voicemail. Incoming calls are directed by the integrated auto-attendant and hunt groups to assist callers to their destinations. It can utilize standard phone lines via an external phone line gateway or cost effective Internet Telephony services.

One unit can support up to 25 extensions, which can be located anywhere with Internet access. Multiple units can be used to increase number of extensions or unite a company that has many locations under a single PBX system.

The PBX phone features are user adjustable via the DVX-1000's web configuration tool. The administrator assigns each extension a profile of telephony features, which allows the best match for a user's job function. Each user can fine-tune their assigned profile via the web to match their daily business schedule.

Phone conferencing is typically expensive external hardware or service. The DVX-1000 includes a phone conferencing bridge, which makes it unsurpassed for value and features. Users are able to schedule and invite parties to conferences via the web configuration. Conference Notifications are sent out by e-mail, which includes the phone number and access codes.

The DVX-1000 uses advanced security features to protect your voice network from unauthorized access. To prevent hackers from breaching the system, the DVX-1000 uses MD5 SIP authentication encryption encoder software. The DVX-1000 also includes an integrated firewall for intrusion detection and protection against denial of service attacks.

The DVX-1000 features a fanless solid-state design offering years of non-stop operation. The compact housing can be easily fastened to the wall of your distribution closet or stacked with your existing Ethernet switches or PSTN Gateways. The DVX-1000 is designed with dual processors for supporting up to 25 simultaneous calls. Its class leading performance allows a 1-to-1 extension to phone line mapping, allowing it to scale with your business.

Utilizing our 19+ years of networking design technology and manufacturing, D-Link created the new xStack IP Telephony family. The DVX-1000 was designed to include all the necessary features of a phone system a company can depend upon.

## **Product Data Sheet**



## **SIP IP-PBX with Conferencing Server**

### **Specifications**

## Management Features

- Includes 25 User Extensions
- Supports 25 simultaneous Inbound/Outbound calls
- Single IP PBX support multiple users across multiple sites
- · Add external Analog Trunk Gateways to use standard phone-lines
- Save Money by using Internet Phone service (VoIP)
- User-Friendly Administration Interface
- · Web-base Monitoring and Administration
- Call Statistics and Calling Detail Records

#### **Basic Calling Features**

- Basic Business Calling Features
- Caller ID, Call Transfer, Call History, Call Hold, Do Not Disturb, Call Forwarding (Always/on Busy/on No Answer/Follow me)

#### **IVR/Auto-Attendant Features**

- Music on Hold
- Attendant Override (Barge-In)
- Customizable Greetings
- · Configurable IVR Menu
- Holiday List Configuration

#### Voicemail

- Mailbox Access Control (PIN)
- Configurable Mailbox Size
- · Customizable Greetings
- · Message Priority
- Notification Via E-mail

#### **Security Features**

- Built-in Firewall
- . MD5 Authentication for SIP
- Secure Web Administrative and User Access for Configuration

#### **Conference Server**

- · Dial In/Dial Out Conferences
- Access Control (PIN)
- Conference Recording

#### Related products

- DIV-140: 4-Port Analog Trunk Gateway
- DVG-2001S: 1-Port Analog Terminal Adapter (ATA)
- DVG-1402S: Wired Router with 2-Port ATA
- DVG-G1402S: Wired/Wireless Router with 2-Port ATA
- DPH-140S: Wired Ethernet IP Phone

#### **Protocol Standards**

- SIP (RFC 3261)
- SDP (RFC 2327)
- RTP (RFC 1889)
- RTCP (RFC 1889)
- Out-Of-Band DTMF (RFC 2833)
- RTSP (RFC 2326)

#### Configuration

- Secure Web Based Management
- Configuration Backup/Restore
- Software Upgrade
- . D-Link Endpoint Provisioning
- · License Control for Advanced Features

#### Hardware

- Dual Intel IXP-425 533 MHz StrongARM Processors
- 64 MB SDRAM (Expandable to 256 MB)
- 1 GB of storage (VM, Announcements)
- 10/100Mb Ethernet Port (RJ-45)

#### **Physical**

- Power LED
- LAN Link/Act
- Dimensions: 9.25" x 6.49" x 1.3"
- Power Input: 5V DC, 3A
- Power Adapter: 90~265V AC
- · Power Consumption: 15 Watt Max
- . Operating: 32° to 122° F
- Humidity: 5% to 95% (Non-condensing)

#### Warranty

1-Year Limited Warranty

YOUR NETWORK SETUP (INTERGRATE BOTH VOIP AND PSTN CALLS,



