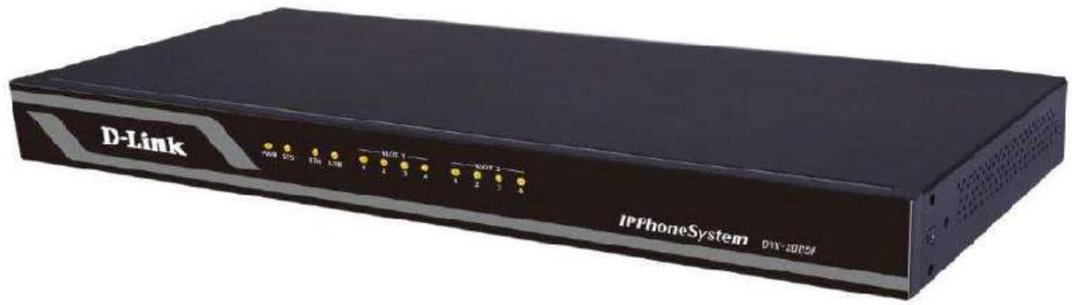

SAVE COST

Save money by using Internet Phone Service, integrates well with known ITSPs, connect to your office on the Go

MODULAR & SCALABLE

Expand your existing analog telephony infrastructure, supports modular analog interface & Features SIP/IAX trunks.

SIMPLE INTEGRATION

Provides user-friendly administration Interface.

IP TELEPHONY

The all-in-one DVX-2005FIP PBX can not only provide the traditional basic PBX features (call hold, call forwarding, call waiting, video call, etc.), but also provide enhanced features such as visual operator, voice mail to mail, multi-media music on hold, and auto attendant, etc. In addition, it's very convenient for SMEs' management and maintenance, also easy to upgrade. SMEs can set up own phone system to improve the company image and office efficiency.

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-2005F offers all of the essential telephony Features such as call forwarding, call hold, follow me, and voice mail. 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.

EXTENSIONS ANYWHERE

The DVX-2005F supports up to 100 extensions, which can be located anywhere with Internet access. Multiple units can be used to increase the number of extensions or unite a company that has many locations under a single PBX system.

EASY WEB CONFIGURATION

The PBX phone features are user adjustable via the DVX-2005'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.

KEY BENEFITS OF THE DVX 2005F:

AS PBX:

- Configurable as core IP or hybrid
- PBX.
- Switches calls & Manages routes.
- Connects callers with the
- outside world over IP/analog (POTS)
- Support build-in 4 FXO ports
- expandable up to 8 FXO ports

AS GATEWAY:

- Configurable as media gateway.
- Bridges legacy PSTN to the expanding world of IP telephony.
- Conversion between a wide range of communications protocols and media codecs.

AS MEDIA/FEATURE SERVER:

- Provides IVR and Conference
- Bridge.
- Automated attendant and unified messaging.
- Can replace aging legacy voicemail systems.

IN CALL CENTER:

- Features built-in ACD systems.
- Additional remote IP agent capabilities.
- Advanced skills-based routing.

Call Features

- Call Back
- Call Forward
- Call Group
- Call Hold
- Call Paging and Intercom
- Call Park
- Call Pickup
- Call Queue
- Call Recording
- Call Routing
- Blind Transfer
- Attendant Transfer
- Call Waiting
- Caller ID
- Dial by Name
- Music On Hold/Transfer
- 3-Way Conference
- Video Calls

TECHNICAL SPECIFICATIONS

PBX Features

Black List
BLF (Busy Lamp Field)
CDR (Call Detailed Record)
Conference Room (20 Rooms)
Call Monitoring
DID (Direct Inward Dialing Number)
DISA (Direct Inward System Access)
Distinctive Ringtone
DND (Do Not Disturb)
DNIS (Dialed Number Identification Service)
Feature Codes
FOP (Flash Operation Panel)
Status Monitoring
Follow Me
IVR (Interactive Voice Responses)
Mobility Extension
Multi-Language Prompts
Multi-Language GUI
One Touch Recording
Phone Book (LDAP Server)
Phone Provisioning
Pin Set
Record File Download
Ring group
SIP Register with UDP/TCP/TLS
SIP Trunk
Skype for SIP
Smart DID
Speed Dial
Spy
SRTP (Secure Real-time Transport Protocol)
T.38 Fax (Pass-through)
Time Based Rule
Fax to Email
WebRTC/ Web Dial
Voicemail & Voicemail to Email

VOIP GATEWAYS	
DVG-2102S	2 port Analog VoIP Telephone Adapter (ATA), 1WAN, 2FXS, with
DVG-5004S	VoIP Gateway with built-in 4 FXS, 1 10/100Mbps WAN & 4-port
DVG-5008S	VoIP Gateway with built-in 8 FXS, 1 10/100Mbps WAN & 4-port
DVG-6004S	VoIP Gateway with built-in 4 FXO, 1 10/100Mbps WAN & 4-port
DVG-6008S	VoIP Gateway with built-in 8 FXO, 1 10/100Mbps WAN & 4-port

DVX-2020 Additional 4 Port FXO Module

Network Features

DDNS Client
DHCP Server
IPv4/IPv6/IEEE802.1Q
IP Assignment (PPPoE/DHCP/Static)
SNMP v1/v2
TR069
Static Route Table
Trouble Shooting (Ping/Traceroute)
VPN Client (L2TP/PPTP/OpenVPN/N2N/IPSec)
VPN Server (L2TP/PPTP/OpenVPN/IPSec)

Logs

PBX Log
Web Access Log
PBX Debug Log

Hardware Interfaces

1 Reset Button
1 Power Interface
2 Ethernet Interfaces
1 Console Interface
1 USB Interface
Slot 1/2 for Analog

Hardware Specifications

Processor: Dual core A7 1Ghz
SDRAM: DDR3 1GB
Storage: 8GB SD Card
Power Supply: Input AC 100~240V, 50/60Hz;
Output DC 12V/2A

Environment

Working Temperature: 0 ~ 40°C
Storage Temperature: -20 ~ 55°C
Humidity: 5 ~ 95% Non-condensing

Packings

Dimension: 280 × 150 × 65mm
Weight/Unit: 0.8kg
Unit/Carton: 10 Units
Weight/Carton: 10kg

System Capacities

30 Concurrent Calls
Up to 100 IP Phone Registers/Extensions
Recording: 36,000 mins (.gsm); 4,000 mins (.wav)
Voicemail: 36,000 mins (.gsm); 4,000 mins (.wav)

Codecs & Protocols

Audio Codecs: G.722/G.711-Ulaw/G.711-Alaw/
G.726/G.729/GSM/SPEEX
Video Codecs: H.261/H.263/H.263+/H.264
Protocols: SIP (RFC3261)/IAX2
DTMF: RFC2833/SIP INFO/In-Band