D-Link®

DPH-120SE/F1

EFFORTLESS CONFIGURATION

The Ip phone features web-based configuration via phone and computer, bringing additional convenience and increasing productivity

SUPERIOR SOUND

Speakerphone IP phone with Echo Cancellation, CNG, VAD and Dynamic Jitter Buffer features for excellent sound quality

COMPLETE PHONE FEATURES

Extensive phone functions: Call Transfer, Forward and Hold, 3-Way Conference, Speed Dial/Phone Book, Caller ID

DPH-120SE/F1 IP Phones are mainly designed for general office users (from VSE, SOHO to SMB) in VoIP communication. With sophisticated and elegant design, this platform has high performance and can offer versatile features and specifications to meet different environment requirements. It can be installed on LAN /DSL/Cable network environment and registered to SIP registrar Server(s), soft switch(es), IP-PBX(s), or IMS-based system and let SIP-enabled terminals to communicate with. Beyond this, user's PC can be connected to this phone instead of LAN directly. The phone comes with a plastic Housing and some accessories, including handset, handset cord, keypad, keys and wall-mounting kit. LCD display on the panel provides direct visual interface with user. User can use keypad/LCD or Web browser to configure this phone.

DPH-120SE

WHAT THIS PRODUCT DOES

An entry-level, cost-effective professional desktop IP Telephone. Coupled with basic features including 2-lines, 3-party conference function, easy-to-read backlight Lattice display, it deftly meets the affordability and reliability requirements of any budget.
10/100 Mbps with PoE Integrated , free of power cables reduce number of wires on your desk and enabled you to install the IP phone with ease.

KEYPAD FEATURES

+ 4 Soft keys for doing more functions
+ 7 Function keys
+ 4 Navigator Keys & 1 ok key for navigating in configuration
+ Hold
+ Call forward
+ Three-party conference
+ MWI
+ Headset
+ Redial
+ Speaker
+ 12 numeric keys with star & pound key
+ Mute
+ Vol - / Vol +

PHONE FEATURES

+ Multi-user (2 SIP accounts)
+ Caller ID display
+ Call History: 300 Calls
+ Phone book (up to 500 contact names & phone numbers)
+ Day/Time display
+ Call/Time display
+ 9 Selective Ring tones
+ 9 Speed dial number
+ Incoming call indicator
+ Flexible dial map
+ Password control for Configuration
+ Pre-dial before sending
+ Connect with expansion module
+ Memo
+ MWI
+ SMS
+ Keypad lock
+ Emergency call
+ Customize DSS key/ softkey

TECHNICAL SPECIFICATIONS

NETWORK INTERFACE AND I/O PORTS

+ WAN Port: to connect to 10/100 Mbps Ethernet
+ LAN Port: to connect to 10/100 Mbps Ethernet
+ 5V DC IN Jack: to connect to local power with a switching power adaptor

HARDWARE & PHYSICAL SPECIFICATIONS

Model	Description
	Standard SIP phone with Power Adaptor support
Key components	Description
CPU	Broadcom
	Description
Screen	Resolution: 128x48 dot-matrix
Port Name	Functions
WAN	1×10BaseT/100 BaseTX ports RJ45 Compliant to following standards: IEEE 802.3/802.3u Support Full-Duplex operations
LAN	1×10/100 Base T ports Compliant to following standards: IEEE 802.3 Support Full-Duplex operations

Model	Description
Dimension	195 × 188 × 51 mm
Net Weight	0.51kg
Power Adaptor	AC-DC Switching Power switching Wall-Mount type Input: 100~120, 220~240VAC Output: DC 5V / 1000mA Max. Watt: 5 Watt. IEEE802.3af POE Class 1
Power Consumption	Typical: 1.3 Watt (Standby) Max.: 4.3 Watt (Talking)
Temperature	Operating: 0˚C to 40˚C Storage: -20˚C to 60˚C
Related Humidity	Operating: 10% to 65 % (no-condensing) Storage: 15% to 85% (non-condensing)

Software Requirement	Description
Browser for Web of Phone	Microsoft Windows IE, or PC-based general web browser
Auto Provisioning Server	General compatible TFTP, FTP , HTTP & HTTPs Server Software

VOICE CODEC

+ G.711a/u (64k bps)
+ G.729A/B (8k bps)
+ G.723.1 high/low
+ G.726-32
+ G.722

ADVANCE VOICE QUALITY FEATURE

+ Silence Suppression
+ Acoustic Echo Cancellation (G.167)
+ Voice Active Detection (VAD)
+ Comfort Noise Generation
+ Jitter Buffer
+ DTMF Transmitter (SIP info, Transparent, RFC 2833)
+ Packet Lost Concealment (PLC)
+ HD Voice handset

SIGNAL, MEDIA & NETWORK PROTOCOLS

+ SIP RFC 3261 & the related RFC standard in Appendix A
+ SDP RFC 2327
+ RTP RFC 1889
+ IP assignment: Static IP, DHCP and PPPoE
+ STUN, static port mapping (for NAT traversal)
+ SNTP
+ DNS & DNS SRV
+ TFTP/FTP/HTTP/HTTPS for Auto Provision
+ IP/TCP/UDP/ARP/ICMP
+ Route and Bridge mode

SUPPLEMENTARY CALL FEATURE

+ Call Hold Resume
+ Call Mute
+ Call Waiting
+ Call waiting Indication
+ Three Way Conference
+ Anonymous Call/Rejection
+ Message Waiting Indication
+ Auto Answer
+ Black list
+ Limit list
+ Auto hangup
+ Auto Redial
+ Ban outgoing
+ Hotline
+ BLF/Presence
+ Action url/Active uri

NETWORK CAPABILITY

+ QoS: IEEE 802.1Q & IEEE 802.1p Compliant
+ Diffserv (DSCP)/ToS
+ Full range VLAN ID Support
+ Class of Service Support by VLAN Tag
+ LLDP
+ L2TP VPN/OpenVpn

USER INTERFACE AND NETWORK MANAGEMENT

+ LCD/Keypad UI in English & other Languages
+ HTTP(WEB) UI in English version & other Languages
+ FTP/TFTP/HTTP for Firmware remote update
+ Auto-provisioning (APS) for firmware and profile upgrade
+ Emergence upgrade if firmware corrupted

CE FC