Product External Specification

For VoIP Phone

Model number: DPH-400SE

Revision:1.0

Rev.	Date	Author	Reasons for changes
1.0	Sep 02, 2013	Archie Liu	

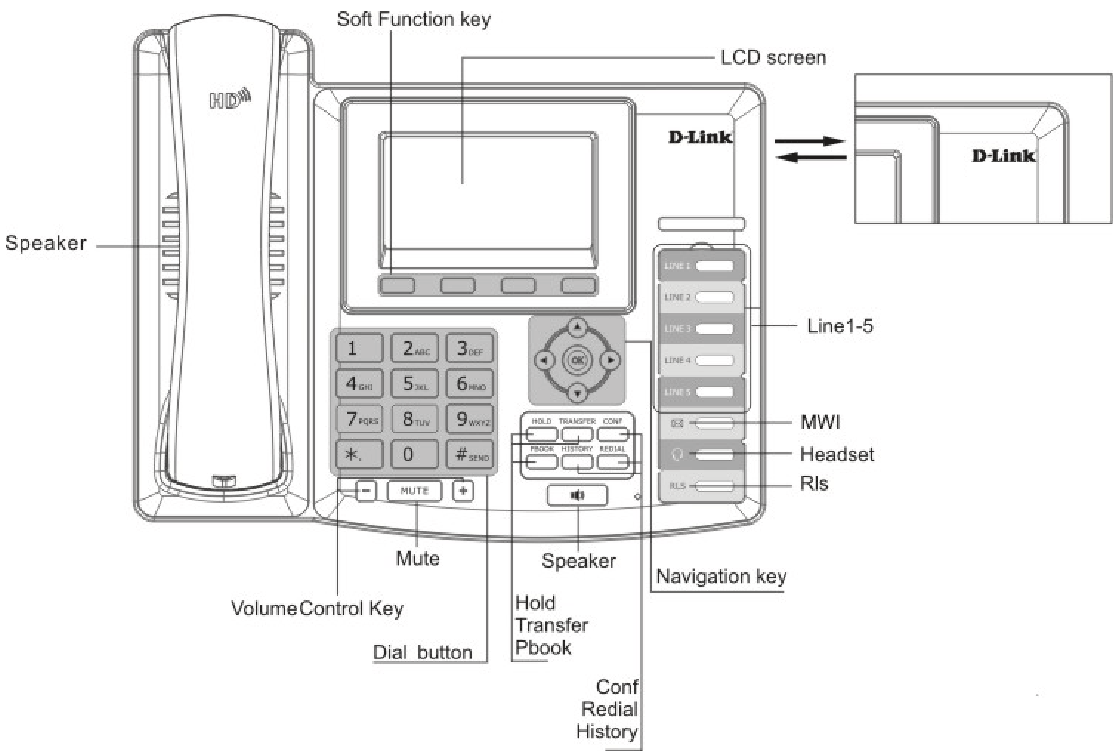
1. Production Introduction

DPH-400SE 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. A LCD display on the panel provides direct visual interface with user. User can use keypad/LCD or Web browser to configure this phone.

2. Appearance and User Interface Layout

2.1. Key

Keys: Dial pad 12 keys
Fixed function keys: 14 keys
Soft keys: 4 keys
Navigator keys: 4 keys
SIP Line key：5 keys



2.2. The Layout of I/O ports

WAN Port: to connect to 10/100Mbps Ethernet (toward Internet)
LAN Port: to connect to 10/100Mbps Ethernet (toward a local PC)
5V DC IN Jack: to connect to local power with a switching power adaptor
Handset Hook switch: for hang on/off control on handset cradle
Handset Jack: to connect with a handset
Headset Jack: to connect with a headset
External console Jack: to connect with expansion module

3. Hardware & Physical Specifications

Hardware:

Model	**Description**
DPH-400SE	Standard SIP phone with Power Adaptor support
Key components	Description
CPU	BCM 1190
SDRAM	16M
FLASH	4M
Port Name	**Functions**
WAN	1×10BaseT/100 BaseTX ports RJ45 Compliant to following standards: IEEE 802.3/802.3u Support Full-Duplex operations PoE Class 1
LAN	1×10/100 BaseT ports

	Compliant to following standards: IEEE 802.3 Support Full-Duplex operations

Physical:

Model	Physical Data
Dimension	240 × 185 × 45mm
Net Weight	0.99kg
Power Adaptor	AC-DC Switching Power switching Wall-Mount type Input: 100~120, 220~240VAC Output: DC 5V / 1000mA Max. Watt: 5 Watt.
Power Consumption	Typical: 2.5 Watt (Standby) Max.: 2.8 Watt (Talking)
Temperature	Operating: 0℃ to 40℃ Storage: -20℃ to 60℃
Related Humidity	Operating: 10% to 65 % (no-condensing) Storage: 15% to 85% (non-condensing)

4. Software Specifications

Software components:

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

Software Specifications:

■ **Keypad Features**

- 4 Soft keys for doing more functions
- 5 SIP line key
- MWI
- Headset
- RLS(Release key)
- Hold
- Transfer
- CONF
- Phone Book
- History

- Redial (Redial and the entry to access call history; finish dialing and call the number)
- Speaker
- 4 Navigator Keys for navigating in configuration; volume adjustment of Ringer, Speaker phone & Handset
- Enter
- 12 numeric keys with star & pound key
- Mute
- Vol-/Vol+

Phone Features

- Mult-user (5 SIP accounts)
- Caller ID display
- Call History: 300 Missed Calls, 300 Received Calls, 300 Dialed Calls
- Phone book (up to 500 contact names and phone numbers)
- Remote phonebook (up to 4 xml phonebook)
- Day/Time display
- Call/Time display
- 11 Selective Ring tones (9 tones & 3 melodies)
- 9 Speed dial number
- Incoming call indicator
- Flexible dial map
- Password control for Configuration
- Pre-dial before sending
- Connect with expansion module
- MWI
- SMS
- Keypad lock
- Emergency call

Voice Codec

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

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 Transfer (Blind, Attend & Semi Attended Transfer)
- Call Forward (Busy, No answer, Unconditional)
- Call Waiting
- Call waiting Indication
- Three Way Conference
- Anonymous Call/Rejection
- Message Waiting Indication
- Do Not Disturb
- Auto Answer
- Black list
- Limit list
- Auto hangup
- Auto Redial
- Ban outgoing
- Hotline
- BLF/Presence
- Intercom
- Call Pickup
- 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

5. Certification/Test Reports Requirement

EMC/RF Certificates and Test Reports

EMC Test Report	Class A	Class B	Remark
CE Report (EN55022/55024)		YES	
FCC Report (FCC CFR 47, Part 15B/ ICES-003)			
VCCI			
RoHS			
WEEE			

Safety Certificates and Test Reports

Certifications	Standard	Remark
CB (IEC 60950-1)		

6. Package Contents & weighting

Model	Content Description
DPH-400SE	1 × IP phone main set 1 × Handset Cord 1 × Handset 1× Quick Install Guide (paper printed) 1 × CD-ROM(User manual &QIG) 1 × Ethernet CAT5 Cable 1 × Switching power adaptor 1 × Warranty & Safety information(paper printed) 1 × Brown box Unit Packing Weight: 1.265 Kg Size of Gift box: 26.5*25*9CM The number of unit per carton:10 Carton Packing Weight:11.4Kg Size of Carton: 55*47.3*27CM

Appendix A: compliant RFC standards

RFC 1769 Simple Network Time Protocol (SNTP)
RFC 1889 RTP: A Transport Protocol for Real-Time Applications
RFC 2131 Dynamic Host Configuration Protocol
RFC 2327 SDP: Session Description Protocol
RFC 2782 A DNS RR for specifying the location of services (DNS SRV)
RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
RFC 2976 The SIP INFO Method
RFC 3261 SIP: Session Initiation Protocol
RFC 3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
RFC 3263 Session Initiation Protocol (SIP): Locating SIP Servers
RFC 3264 An Offer/Answer Model with the Session Description Protocol (SDP)
RFC 3265 Session Initiation Protocol (SIP)-Specific Event Notification

RFC 3311 The Session Initiation Protocol (SIP) UPDATE Method
RFC 3315 The Session Initiation Protocol (SIP) Refer Method
RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
RFC 3325 Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks

RFC 3420 Internet Media Type message/sipfrag
RFC 3428 Session Initiation Protocol (SIP) Extension for Instant Messaging
RFC 3489 STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
RFC 3665 Session Initiation Protocol Basic Call Flow Examples
RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
RFC 3891 The Session Initiation Protocol (SIP) "Replaces" Header
RFC 3892 The Session Initiation Protocol (SIP) Referred-By Mechanism
RFC 3960 Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
RFC 4028 Session Timers in the Session Initiation Protocol (SIP)
RFC 4325 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)