External Engineering Spec

For

DPH-150S (Hw:B1) Series IP Phones

Model Name:

D-link DPH-150S (Hw: B1)

Version: 1.0

2007/10/29

1.0 Product Introduction

DPH-150S (B1) 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.

2.0 Appearance and User Interface Layout

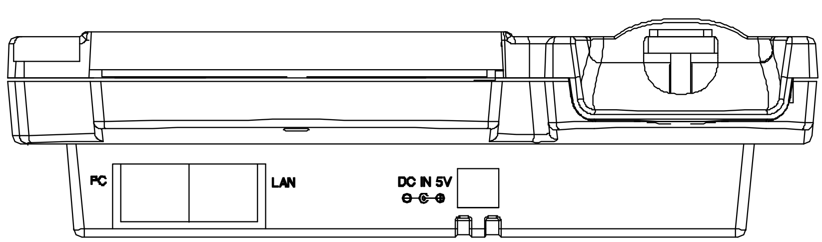
2.1 Key and LED layout

Keys: Dial pad 12 keys
Fixed functional keys: 9 keys
Navigator keys: 4

LEDs:
Status x 1
Speaker x 1
Hold x 1

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.
Engineering Service Port: NOT used by end user(for Vendor only)
Handset Hook switch: for hang on/off control on handset cradle
Handset Jack: to connect with a handset



3.0 Hardware & Physical Specifications

Hardware:

Model	Description
DPH-150S (B1)	Standard SIP phone with Power Adaptor support only
Key Components	**Description**
CPU	Silicon Data 9216
DSP	Silicon Data 8616
SDRAM	16MB
Flash	4MB
Port Name	**Functions**
WAN	1 x 10/100 BaseT ports RJ45 Compliant to following standards: IEEE 802.3/802.3u Support Full-Duplex operations
LAN	1 x 10/100 BaseT ports Compliant to following standards: IEEE 802.3 Support Full-Duplex operations

Physical :

Model	Physical Data
Dimension	196mm x 198mm x 100mm
Net Weight	760g
Power Adaptor	AC-DC Switching Power switching Wall-Mount type Input: 100~120, 220~240VAC Output: DC 5V / 1200mA Max. Watt: 6 Watt.
Power Consumption	Typical: 3.5 Watt (Standby) Max.: 4 Watt (Talking)
Temperature	Operating: 0℃ to 40℃ Storage: -20℃ to 60℃
Related Humidity	Operating: 20% to 80 % (no-condensing) Storage: 15% to 85% (non-condensing)

4.0 Software Specifications

Software Components:

Part Item Name	Description
Operating System	Linux Kernel 2.4.17
SIP Stack	ACT SIP stack

Software Platform	Linux software platform
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 Server Software

Software Specifications:

Main Parts	**Specification Details**

- **Keypad Features**
  - Menu
  - Phone Book
  - 4 Navigator Keys for navigating in configuration; volume adjustment of Ringer, Speaker phone & Handset; OK & Cancel
  - MWI
  - Conference
  - Mute
  - Function (stay with Mute; integrated with numeric keys for configuration and information)
  - Transfer
  - Redial (Redial and the entry to access call history)
  - Hold
  - Speaker
  - 12 numeric keys with star & pound key

- **Phone Features**
  - Multi-user (4 SIP accounts)
  - Caller ID display
  - Call History: 20 Missed Calls, 20 Received Calls, 20 Dialed Calls
  - Phone book (up to 200 contact names and phone number)
  - Day/Time display
  - Call timer display
  - 8 Selective Ring tones (4 tones & 4 melodies)
  - 10 Speed dial number
  - Incoming call Indicator
  - Password control for Configuration
  - Pre-dial before sending

- **Voice Codec**
  - G.711a/u (64k bps)
  - G.729A/B(8 k bps)
  - G.723.1 (*Optional)

- **Advance Voice Quality Feature**
  - Silence Suppression
  - Acoustic Echo Cancellation (G.167)
  - Voice Activation Detection(VAD)
  - Comfort Noise Generation

- Jitter Buffer
- DTMF Transmitter(SIP info, Transparent, RTP 2833 relay)
- Packet Lost Concealment (PLC)

- **Signal, Media & Network Protocols**
- SIP RFC 3261 & the related RFC standard in Appendix A
- SDP RFC 3266
- RTP RFC 1889
- IP assignment: DHCP, Static IP and PPPoE
- STUN, UPnP & static port mapping (for NAT traversal)
- SNTP
- DNS & DNS SRV
- TFTP/FTP/HTTP for Auto Provision
- IP/TCP/UDP/ARP/ICMP

- **Supplementary Call Feature**
- Call Hold Resume
- Call Mute
- Call Transfer (Blind, Attend & Early AttendedTransfer)
- Call Forward (Busy, No answer, Unconditional)
- Call Waiting
- Call Waiting Indication
- Three Way Conferences
- Anonymous Call / Rejection
- Message Waiting Indication
- Do Not Disturb
- Auto Answer (doable)

- **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

- **User Interface and Network Management**
- LCD/Keypad UI in English & Japanese version
- HTTP(WEB) UI in English version
- FTP/TFTP/HTTP for Firmware remote update
- APS auto-provisioning for firmware and profile upgrade
- Emergency upgrade if firmware corrupted

5.0 Certifications/Test Reports Requirement

EMC/RF Certificates and Test Reports

EMC Test Report	Class A	Class B	Remarks
CE Report (EN55022/55024)		V	
FCC Report (FCC CFR 47, Part 15B/		V	

ICES-003)			
VCCI		V	
RoHS			V (certificated)
WEEE			

Safety Certificates and Test Reports

Certifications	Standards	Remarks
CB (IEC 60950-1)	V	

6.0 Package Contents & Weighting

Model	Content Description
DPH-150S (B1)	1 x IP phone main set 1 x Handset Cord 1 x Handset 1 x Quick Install Guide (paper printed) 1 x CD-ROM 1 x Ethernet CAT5 Cable 1 x Wall-mount kit 1 x Switching power adaptor (optional for different country/area) 1 x Gift box Unit Packing Weight: 1.2 Kg Size of Gift box: 269mm x 210mm x 92mm The number of unit per carton: 5pcs Carton Packing Weight: 7.5Kg Size of Carton : 48.5 x 22.2 x 28.5 cm^3

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 3361 Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers

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)

draft-ietf-sipping-service-examples-10 Session Initiation Protocol Service Examples
draft-worley-sipping-dialog-01 Guidelines for Implementing the Dialog Event Package in User Agents
draft-worley-sipping-pickup-01 Call Pickup Examples in SIP

=========The End of Document =============================