External Engineering Spec

For

DPH-150SE (Hw:C1) Series IP Phones

Model Name:

D-link DPH-150SE (Hw: C1)

Version: 1.0

2009/07/24

1.0 Product Introduction

DPH-150SE (C1) 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.

More distinct features including the PoE and Giga Ethernet ports, Multi-language LCD display, and Wideband Audio (G.722); Besides IPv4 environment, the DPH-150SE(C1) also is also complied in IPv6 environment.

2.0 Appearance and User Interface Layout

2.1 Key and LED layout

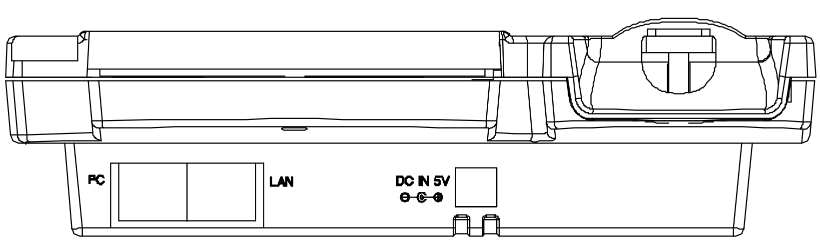
Keys: 25 Keys
- 12 x alphanumeric keys
- 9 x fixed functional keys keys
- 4 x Navigator keys including volume adjustment

LEDs:

Status x 1
Speaker x 1
Hold x 1

2.2 The Layout of I/O Ports

LAN Port: to connect to Gigabit Ethernet port (toward Internet)
PC Port: to connect to Gigabit Ethernet port (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-150SE (C1)	Standard SIP phone with Power Adaptor support only
Key Components	**Description**
CPU and DSP	SiTel SC 14452
SDRAM	16MB
Flash	4MB
Switch Controller	Realtek 3 ports gigabits switch RTL8363S
Port Name	**Functions**
LAN	1 x 10/100/1000 BaseT ports RJ45 Compliant to following standards: IEEE 802.3ab/802.3u Support 802.3x Full-Duplex operations PoE in IEEE 802.3af standard
PC	1 x 10/100/1000 BaseT ports Compliant to following standards: IEEE 802.3ab/802.3u Support 802.3x Full-Duplex operations

Physical :

Model	Physical Data
Dimension	196mm x 198mm x 100mm
Net Weight	760g & wall-mountable
Power Adaptor	AC-DC Switching Power switching Wall-Mount type Input: 100~120, 100~240VAC, 50~60Hz Output: DC 5V / 2000mA Max. Watt: 10 Watt.
Power Consumption	Typical: 2 Watt (Standby) Max.: 3.25 Watt (Talking)
Temperature	Operating: 0℃ to 40℃; Storage: -20℃ to 60℃
Related Humidity	Operating: 10% to 90 % (no-condensing) Storage: 10% to 95% (non-condensing)

4.0 Software Specifications

Software Components:

Part Item Name	**Description**
Operating System	ucLinux 2.6.19
SIP Stack	Cameo 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: 60 Missed Calls, 60 Received Calls, 60 Dialed Calls
  - Phone book (up to 200 contact names and phone number)
  - Day/Time display
  - Call timer display
  - 9 Selective Ring tones
  - 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.722 (wideband)

- **Advance Voice Quality Feature**
  - Silence Suppression
  - Acoustic Echo Cancellation (G.168)
  - Voice Activation Detection(VAD)
  - Comfort Noise Generation
  - Jitter Buffer
  - DTMF Transmitter(SIP info/RFC2976, RTP Payload/RFC4733 relay, and In-band)
  - Packet Lost Concealment (PLC)

- **Signal, Media & Network Protocols**
  - SIP (RFC 3261) & the related RFC standard in Appendix A
  - SDP (RFC 4566)
  - RTP (support both IPv4 and IPv6)
  - IP assignment: DHCP, Static IP and PPPoE
  - STUN, UPnP & static port mapping (for NAT traversal)
  - SNTP (RFC 4330)
  - DNS/Locating SIP server by RFC3263
  - DNS SRV (RFC 2782)
  - TFTP/FTP/HTTP for Auto Provision
  - Support IPv6 Auto-configuration (RFC 4862)
  - IP/TCP/UDP/ARP/ICMP
  - Support IPv6 standard- RFC 2460 (IPv6), RFC 4443 (ICMPv6)

- **Security**
  - HTTP 1.1 basic/digest authentication for Web setup
  - MD5 for SIP authentication (RFC2069/RFC2617)
  - AES+SHA256 for Configuration File (for Auto Provision)

- **Supplementary Call Feature**
  - Call Hold Resume
  - Call Mute
  - Call Transfer (Blind, Attend & Early Attended Transfer)
  - Call Forward (Busy, No answer, Unconditional)
  - Call Waiting
  - Call Waiting Indication (RFC3842)
  - Three Way Conferences
  - Anonymous Call / Blocking
  - Rejection
  - Message Waiting Indication
  - Do Not Disturb
  - Auto Answer (Support SIP Serer Required)
  - SMS (RFC 3428)

- **Network Capability**
  - QoS: IEEE 802.1Q & IEEE 802.1p QoS
  - DiffServ (RFC 2474 DSCP)/ToS
  - Full range VLAN ID Support,

- Class of Service Support by VLAN Tag
- "rport" parameter (RFC 3581)

- **User Interface and Network Management**
- LCD/Keypad UI in English & Chinese version
- HTTP/HTTPS (WEB) UI in English & Chinese 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/ ICES-003)		V 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)
(NO NTP RFP1305)
RFC 1889 RTP: A Transport Protocol for Real-Time Applications
RFC 2131 Dynamic Host Configuration Protocol
RFC 4566 SDP: Session Description Protocol
RFC 2782 A DNS RR for specifying the location of services (DNS SRV)
RFC 4733 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)

RFC 3581 “rport”

RFC 2460

RFC 4443

RFC 4862

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 ==============================