ALCATEL-LUCENT 8001 DESKPHONE

The Alcatel-Lucent 8001 DeskPhone is a cost-effective, business-grade phone offering SIP telephony for essential communications. Its graphical display can show up to five lines of text simultaneously, and it features an elegant user interface and clear operations to provide an excellent user experience. Moreover, the Alcatel-Lucent 8001 DeskPhone creates a unique, vivid audio experience. Its rich business functions greatly improve work efficiency.

FEATURES

- Standard SIP telephony based on IETF standards (RFCs)
- 132×64 graphic dot matrix display, 5 lines shown simultaneously, with Chinese font support
- RJ9 and 3.5 mm jack for headset connection
- USB connectivity (for example, for charging smartphones)
- 2 x 100BASE-TX Ethernet interfaces (1 LAN access and 1 PC port)
- Support static and dynamic IP allocation
- Power over Ethernet (IEEE 802.3af)*
- Two SIP accounts.
- 3-party conference
- Support voice mail
- XML/LDAP phone book
- Redial of last called numbers
- Third-party call control (3PCC) support
- Three configuration methods: (web page, phone configuration, auto-provision)

BENEFITS

- Business-grade design
- Cost-effective
- Always-on connectivity to ensure continuity of critical communications
- Support Call Center and Contact Center
- Simplified operations, reduced operational expenditure and total cost of ownership through a seamless single management platform shared with other devices, applications and networking elements
- Integration with Alcatel-Lucent communication servers

*A local power supply adaptor is available for Chinese market

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TECHNICAL SPECIFICATIONS

Physical characteristics

- Height: 205 mm
- Width: 197 mm
- Depth: 54 mm
- Weight: 800 g
- Color: gray
- 2-level adjustable foot stand: 40° or 60°
- Wall-mountable (an adaptor is required for compliance with standard TIA-570-C, section 8)
- Operating temperature: 0°C to 45°C
- Operating humidity: 10% to 90%

Display

- 132 x 64 pixel graphic monochrome display, 5 lines shown with backlight

Audio characteristics

- Audio codec: G. 711(A-law and μ-law), G723.1, G.729AB
- Supports VAD, CNG, AEC, AJB
- Volume controls
- Mute control

Keys and navigation

Application keys

- 2 SIP account keys
- Conference key
- Redial key
- Transfer key
- Hold key

Audio keys

- Volume control keys (+ and -)
- Hands-free on/off key with LED
- Mute on/off key

Indication

- Message key with LED
- Headset key with LED
- Call indication LED

Navigation keys

- Cancel key
- 4-way navigation and OK key

Power

- Supports 802.3AF Power over Ethernet (POE) - Class 0*

*Maximum power for the 8001 in use without USB charging <= 3 W

Connectivity

- LAN: 2 x RJ45 10/100M Ethernet ports
- 1 x 3.5 mm headset port
- 1 x RJ9 handset port
- 1 x RJ9 headset port
- 1 x USB por (DC 5V/1A output)

Management

- Auto-provisioning via FTP/TFTP/HTTP/HTTPS (for settings and firmware upgrade/download)
- Configuration via Browser/phone/auto-provisioning
- Supports TR069 (optional)
- Supports Telnet
- Supports LLDP
- Trace package and system log export

QoS

- 802.1p (SIP and RTP QoS)
- DSCP

Security

- 802.1x
- Supports VPN
- VLAN tagging (802.1q)
- Transport Layer Security (TLS)
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES, encryption
- Admin/user 2-level configuration mode

Multi-language

- English, Chinese (simplified and traditional), French, Hungarian, Italian, Polish, Portuguese, Russian, Spanish, Turkish, Slovak, Dutch, Norwegian, Danish and Finnish.
- Display of Unicode characters for supported languages

Regulatory and standards safety

- EN 60950-1
- IEC 60950-1

EMC

- EN 55022 Class B
- CISPR 22 Class B
- EN 55024
- CISPR 24

Eco-design

- ROHS Directive 2011/65/EU
- WEEE Directive 2012/19/EU

Phone features

- Support two SIP account lines, hotline
- Support call holding, call waiting, call forwarding and transfer (blind/busy/ask)
- Caller ID, redial, mute, do not disturb (DND)
- Auto answer, conference
- Speed-dial, voicemail
- IP direct dial
- Custom ring tone
- Call records(50 logs each): dialed calls/received calls/missed calls

Phonebook

- Individual phonebook (300 entries)
- Enterprise phonebook (800 entries)
- LDAP/XML directory service

Advanced feature

- SIP server redundancy
- Third party call control (3PCC)

System interoperability

- Alcatel-lucent OmniPCX® Enterprise R11.0.1 and later
- Alcatel-Lucent OmniPCX Office RCE R9.2 and R.10

Standards references (non-exhaustive list)

- RFC 3261: Session Initiation Protocol (SIP)
- RFC 2131: Dynamic Host Configuration Protocol (DHCP)
- RFC 2617: HTTP Authentication; Basic and Digest Access Authentication
- RFC 2833: RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 2976: The SIP INFO Method
- RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264: An Offer/Answer Model with the Session Description Protocol
- RFC 3515: The SIP Refer Method
- RFC 3892: The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 4028: Session Timers in the Session Initiation Protocol (SIP)

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