



IPAC 501 00 E

VoIP Door Entry Phone, « hands-free », vandal-resistant:

- Meeting the requirements for accessibility for people with disabilities
- with 1 direct call button.

IPAC 501 00 E for:

- a point-to-point communication (Peer to Peer) or
- the connection via a SIP server.

Functions

- Door entry phone
- Full duplex voice communication

Technical data

- Acoustic inductive loop amplifier as communication aid for hearing aid users
- 1 direct call button
- Redial if busy or if no answer (1 - 4 call numbers)
- Management of call parameters: communication time, button activation time, ring time for outgoing calls, volume...
- Management of time lock zones
- Pictograms display associated with product functions
- Automatic speech announcements (dialling, communication..., door opening)
- Audio-quality HD
- Data encryption (audio and video): SRTP / ZRTP / SIP-TLS
- 2 relays for door open command or remote control of external elements (lighting, etc...)
- 2 inputs for external contacts or voltage **with the possibility to define time lock zones**
- Day/night operation (adjustment of volume and brightness)
- Updates:
 - LDAP-update of the phonebook
 - system update by downloadable file
- Real-time monitoring of device status:
 - On access code keying, outgoing calls, door opening, loss of SIP server
 - In case of power failure
- Real-Time display of the device screen on the web page

Power supply

- Network: PoE+ or
 - PoE+ injector and **IP network access protection - AMPHITECH NetCut or**
 - External power supply unit: 24 VDC - **AMPHITECH BAS 2415**
- Power consumption:
- Idle: 180 mA
 - In communication/Audio adjustment medium level: 300 mA
 - In communication/Audio adjustment maximum level: 500 mA

Mechanical design

- Flush mount - Housing BM 500
- Dimensions 350 x 154,5 x 30 mm - weight 2,2 kg
- Degree of protection: IP 55 - IK 08
- Temperature range: -20°C to +60°C
- Stainless steel face plate 2,5 mm, ZAMAK housing



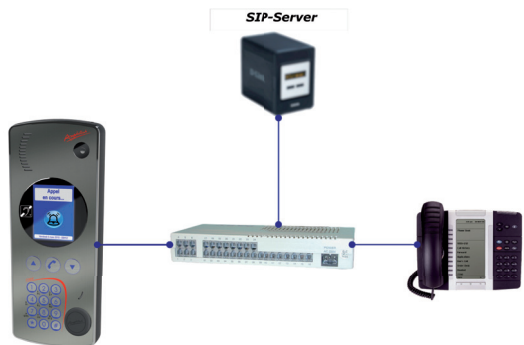


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Point-to-point communication (Peer to Peer)



Connection via a SIP server



Configuration

- Access via web server
- Basic programming; advanced programming via specific web interface
- Network parameter setting, SIP account, audio- and video codecs
- Point-to-point communication (Peer-to-Peer)
- Communication via SIP server (group call, conference call, queue management, messaging...)
- Management of 3 SIP accounts on different IP-PBX
- Communication protocols: SIP (RFC3261), IPV4, TCP/UDP, HTTP, HTTPS, RTP, DHCP/STATIC/ NAT, RFC 6086 INFO Method, DTMF RFC 2833, RFC 2976 SIP INFO, RADIUS 802.1x
- Audio codecs: G.722, G.711u, G.711a, GSM, Speex 8k, Speex 16k, Speex 32k, G.726-16, G.726-32, G.726-24, G.726-40, AAL2-G.726-16, AAL2-G.726-32, AAL2-G.726-24, AAL2-G.726-40, opus, AMR.-32
- Report of system events: downloadable files, SYSLOG, notifications by e-mail (SMTP-Client)
- Import / Export of configuration and name list (CSV format)
- Language selection (configuration, operation mode): French, German, English, Spanish, Portuguese
- Type of time setting: manually or via NTP server
- Automatic clock change – winter time / summer time

