Atcom AT620P SIP/IAX IP Phone

Product Name: Atcom AT620P SIP/IAX IP Phone
Manufacturer: -
Model Number: AT620P

Please Note: The Aastra AT620P has been discontinued. For an alternative, we highly recommend the Aastra AT820P.

Atcom AT620P SIP/IAX IP Phone
The AT620P is an IP Phone for businesses, providing high voice quality and stable performance while adopting a new fashion design. The Atcom AT-620P supports SIP and IAX2. It has 2 lines, 4 softkeys and the easier navigation and improved user friendliness helps to enhance user's IP phone experience. Atcom AT620P Phone Key Features

• Support SIP domain
• Adjustable user password and super password
• Wall Mountable
• Two lines SIP and one line IAX2
• RJ9 headset jack

Atcom AT620P IP Phone Technical Specification
VoIP

• Sip 2.0 (RFC3261)
• STUN
• Two lines SIP and one line IAX2
• Support multi language (LCD support Latin language system , web support all languages) and easy dynamic switch between different laguages
• Softkey * 4
• RJ9 headset jack
• Wall Mountable
• Jitter Buffer(200ms),VAD,CNG
• G.711A/u?G722?G.723?G.729 Codec
• G.168 compliant 96ms echo cancellation
• Support SIP domain,SIP authentication(none,basic, MD5).
• Support inbound audio, RFC2833 and SIP info , DTMF transmission way
• SIP Call Forward?Call transfer?Call hold?Call waiting, 3-way Talking?Pickup?Join call?Redial?Unredial?Call Park?vport?click to dial
• Dial without register
• Suppot Hotline?DND(Do Not Disturb)?BlackList?Call Limitation?DND?Incoming list
• Dial-peer calling rule ,IP to IP call
• SIP server conference
• Phone book 500 records; answered call?missed call 100 for each
• Support configuration and firmware updating by HTTP, FTP, TFTP
• Support decrypting configuration file during auto provision
• Support SIP signal and media(RTP) encryption
• Answering machine
• Support SNTP client
• Telnet, WEB visit terminal
• Support different level user management
• PoE
• Data Features:
• Static/Dynamic WAN-IP-Addressing

• PPPoE

Management

• Web, telnet and keypad management.
• Adjustable user password and super password
• Upgrade firmware through HTTP, FTP or TFTP.
• Telnet remote management.
• Upload/download setting file
• Auto-provisioning
• Safe mode provide reliability
• Phone book, maximum 100 entries.

Interface

• Sip 2.0 (RFC3261)
• STUN
• Two lines SIP and one line IAX2
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Interface

• Two RJ45 ports, one for WAN, one LAN.
• Power ports
• 2 Handset ports

Data networking

• MAC Address
• TCP:Transmission Control Protocol
• DHCP:Dynamic Host Configuration Protocol
• PPPoE:PPP Protocol over Ethernet
• POE(option)
• SNTP, Simple Network Time Protocol
• STUN - Simple Traversal of User Datagram ...
• MD5 Message-Digest Algorithm
• DNSDomain Name Server
• RTP: Real-time Transport Protocol
• RTCP:Real-time Control Protocol
• Telnet:Internet's remote login protocol
• HTTP:Hyper Text Transfer protocol
• FTP:File Transfer protocol
• TFTP:Trivial File Transfer Protocol

Call control /voip Features

• SIP RFC3261,RFC 2543
• Tone generation and Local DTMF re-generation according with ITU-T
• G.711(A-law or u-law)
• G.723.1(6.3kbps,5.3 kbps)
• G729
• AGC(Auto Gain Control)
• G.168/165 compliant 16ms echo cancellation
• AEC(Auto Echo Cancellation)
• VAD (Voice Activity Detection)
• CNG(Comfort Noise Generation
• Enviromental

Electric requirements

• Voltage: 9V ~ 24V
• Power adapter: output DC 12V/450 mA
• Operating requirement
• Operation temperature: 0 to 40 C (32 to 104F)
• Storage temperature: -30 to 65 C (-22 to 149F)
• Humidity: 10 to 90% no dew

Please Enquire