Atcom AT610 SIP/IAX IP Phone

Product Name: Atcom AT610 SIP/IAX IP Phone
Manufacturer: -
Model Number: AT610

Please Note: The Aastra AT610 has been discontinued. For an alternative, we highly recommend the Aastra AT810P.

Atcom AT610 SIP/IAX IP Phone
The Atcom AT610 IP Phone adopts SIP protocols and multiple voice compression codec to directly convert analog voice into IP packet for internet transport, meaning it is effectively using the existing bandwidth to provide PSTN quality voice service.
Atcom AT610 SIP/IAX IP Phone Key Features

• Full duplex hands-free speakerphone
• Call control features
• Echo cancellation
• Support Voice Gain Setting
• Support PPPoE for xDSL

The style of the AT610 is in accordance with other phones of Atcom AT6 series, meaning the phone is solid and reliable in a business environment. With Broadcom solution, it offers high quality voice stream and is compatible with various platforms including Asterisk, FreePBX, Broadsoft and Cisco call manager etc.
Atcom AT610 IP Phone Technical Specifications
VoIP

• Support SIP 2.0 (RFC3261) and correlative RFCs
• Full duplex hands-free speakerphone
• NAT transverse：support STUN client
• SIP support SIP domain, SIP authentication (none, basic, MD5), DNS name of server, Peer to Peer/ IP call
• SIP support 2 SIP lines. Can connect to SIP1 and SIP2 server at the same time
• DTMF：Support SIP info, DTMF Relay, RFC2833
• SIP application: support Call forward/ transfer/ holding/ waiting / 3 way talking/ paging and intercom/pickup/join call/click to dial/call park
• Call control features: Flexible dial map, Hotline, Empty calling reject, Black list for reject authenticated call, limit call, No disturb, Caller ID
• Support Phonebook 500 records
• Incoming calls / Outgoing calls / Missing calls. Each support 100 records
• Support conference and voice record on SIP server
• 10 kind of ring type
• Support SRTP
• Support MWI
• Redundancy sip server capable.
• Hotline.
• Call Forward、Call transfer、Call hold、Call waiting, 3-way Talking、Pickup、Join call、Redial、Unredial、
• Call Park、vport、click to dial
• DND(Do Not Disturb),
• Black List,Limit List
• E.164 dial plan and customized dial rules

Voice

- Codec：G.711 A/U Law, G.723.1, G.729a/b, G.722,G.722.1,
- Echo cancellation: Support G.168, and Hands-free can support 96ms, Hand free Speaker Phone
- Support Voice Gain Setting, VAD, CNG
- Tone generation and Local DTMF re-generation according with ITU-T
- AGC(Auto Gain Control)
- AEC(Auto Echo Cancellation)
- VAD (Voice Activity Detection)
- CNG(Comfort Noise Generation)

Networking

- WAN/LAN: Support Bridge and Router model（optional）
- Support basic NAT and NAPT（optional）
- Support PPPoE for xDSL
- Support DHCP Client on WAN
- Support DHCP Server on LAN（optional）
- Support VLAN (optional: voice vlan/data vlan)
- QoS with DiffServ
- Support DMZ（optional）
- Support VPN (L2TP/UDP TUNNEL)（optional）
- Support main DNS and secondary DNS server
- Support DNS Relay（optional）
- Support SNTP Client, Firewall
- Network tools in telnet server: Including ping, trace route, telnet client

Protocols

- 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
- DNS: Domain 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

Management

- Web ,telnet and keypad management
- Management with different account right
- Upgrade firmware through POST mode
- Upgrade firmware through HTTP, FTP or TFTP.
- Telnet remote management/Upload/download setting file
- Safe mode provide reliability
- Support Auto Provisioning （configuration file)
- Support Syslog

Please Enquire