WIFI Genius X1+ SIP Cordless Phone

WiFi Genius X1+ WIFI SIP cordless phone with video support and Bluetooth, based on a mobile platform, with seamless roaming between access points and flexible power management for long standby & talk time.

Aristel



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Highlights

- High compatibility to most WIFI network to transmit SIP voice
- Long standby and talk time

PHONE FEATURES

- Belt Clip
- 2 VoIP accounts
- Video display (one way video)
- Bluetooth or 3.5mm headset socket
- Call hold, mute, DND
- Call forward, call waiting, call transfer, Redial, 3-way conferencing
- Direct IP call without SIP proxy
- Ring tone selection
- Set date time manually or automatically
- Text message over SIP
- 1500mAH battery

AUDIO FEATURES

HD voice: HD handset, HD speaker CodecG711 A/ μ , G729, GSM, Ilibc, G722, OPUS, Speex DTMF: In-band, Out-of-band (AFC 2833) and SIPINFO DTMF: In-band, Out-of -band (AFC 2833) and SIPINFO Full-duplex hands-free speakerphone with AEC VAD,CNG,AEC,PLC,AJB,AGC

MANAGEMENT

Configuration: browser/phone/auto-provision Auto provision via TFTP/HTTP/HTTPS for mass deploy BroadSoft device management Phone lock for personal privacy protection Reset to factory, reboot, System log

Wi-Fi

Your Dealer

IEEE 802.11 b/g/n, IEEE 802.3, IEEE 802. 3u Frequency from 2.4GHz to 2.483GHz Data rate Speed up to 300Mbps Channel bandwidth: 20MHz/40MHz

Handset dimensions: 153mm H x 55mm W x 17mm D



E&OE Specifications subject to change without notice



Distributed by: Aristel Networks Pty Ltd 1/25 Howleys Rd Notting Hill Vic 3168 P: 03 8542 2300 F: 03 9544 3299 Freecall: 1800 002 133 www.aristel.com.au email sales@aristel.com.au

NETWORK AND SECURITY

Wi-Fi client

WAN supports static IP, dynamic IP NAT&NAPT with VPN pass-through Virtual Server Automatic receipt of IP address with DHCP Server Security through WEP, WPA-PSK, WPA2-PSK, WDS and build-in firewall Simple handling and management over web interface SIP v1 (RFC2543), v2 (RFC3261) NAT transverse: STUN mode Proxy mode and peer-to-peer SIP link mode IP assignment: static/DHCP/PPPoE HTTP/HTTPS web server Time and date synchronization using SNTP UDP/TCP/DNS-SRV(RFC 3263) SRTP for voice

