GRANDSTREAM
CONNECTING THE WORLD

An easy-to-use 1 port ATA

HT801

The HT801 is a single port analog telephone adapter (ATA) that allows users to create a high-quality and manageable IP telephony solution for residential and office environments. Its ultra-compact size, voice quality, advanced VoIP functionality, security protection and auto provisioning options enable users to take advantage of VoIP on analog phones. It also allows service providers to offer high quality IP service to their market. The HT801 is an ideal ATA for individual use as well as commercial IP voice deployments worldwide.

1 LINE

Supports 1 SIP profile through a single FXS port and a single 10/100Mbps port

TLS and SRTP security encryption technology to protect calls and accounts

Automated provisioning options include TR-069 and XML config files

3 WAY

Supports 3-way voice conferencing

Failover SIP server automatically switches to secondary server if main server loses connection

Supports T.38 Fax for creating Fax-over-IP

Supports a wide range of caller ID formats

zero CONFIG

Use with Grandstream's UCM series of IP PBXs for Zero Configuration provisioning

Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more

Interfaces	
Telephone Interfaces	One (1) FXS port
Network Interfaces	One (1) 10/100Mbps auto-sensing ethernet port (RJ45)
LED Indicators	POWER, INTERNET, PHONE
Factory Reset Button	Yes
Voice, Fax, Modem	
Telephony Features	Caller ID display or block, call waiting, flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference
Voice Codecs	G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.722, G.723.1, G.729A/B, G.726-32, iLBC, Opus, dynamic jitter buffer, advanced line echo cancellation
Fax Over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
Short/Long Haul Ring Load	5 REN: Up to 1km on 24 AWG
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Dial Methods	DTMF, Pulse
Disconnect Methods	Busy Tone, Polarity Reversal/Wink, Loop Current
Signaling	
Network Protocols	TCP/IP/UDP, RTP/RTCP (RFC1889,1890), HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SSH, Telnet, STUN (RFC3489, 5389), SIP (RFC3261), SIP over TCP/TLS, SRTP, SNMP, TR-069, IMS/3GPP, IPoE
QoS	Layer 2 (802.1Q VLAN, SIP/RTP 802.1p) and Layer 3 (ToS, DiffServ, MPLS)
DTMF Method	In-audio, RFC2833 and/or SIP INFO
Provisioning and Control	HTTP, HTTPS, SSH, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog
Security	
Media	SRTP
Control	TLS/SIPS/HTTPS
Management	Syslog support, SSH, remote management using web browser
Physical	
Universal Power Supply	Input: 100-240VAC, 50-60Hz Output: 5.0VDC/1.0A
Environmental	Operational: 32º – 104ºF or 0º – 40ºC Storage: 14º – 140ºF or -10º – 60ºC Humidity: 10 – 90% Non-condensing
Dimension and Weight	Dimensions: 100mm x 100mm x 29.5mm Weight: 102 g
Compliance	FCC: Part15B CE: EN55032, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/NZS CISPR22, AS/NZS60950.1, S003 K.21