

Product Name: Grandstream UCM6308A Audio Series IP PBX

Manufacturer: Grandstream Model Number: UCM6308A

#### Grandstream UCM6308A Audio Series IP PBX

The Grandstream UCM6308A allows businesses to build powerful and scalable unified communication and collaboration solutions. This series of IP PBXs provide a platform that unifies fundamental business communications needs, including voice, instant messaging (IM), voice meetings, audio web meetings, data, analytics, mobility, facility access, intercoms and more. The UCM6300 Audio Series supports up to 1500 users and includes a built-in instant messaging (IM), voice/web conferencing platform, and the free Wave App that allows users to communicate and collaborate from desktops, mobile devices, IP phones, and other SIP end points. Grandstream UCM6308A Key Features

- � Supports up to 1500 users and up to 200 concurrent calls
- า๊¿½ Zero configuration provisioning of Grandstream SIP endpoints
- تَىٰ٪ Built-in Instant Messaging (IM), Audio Conferencing & Amp; Web Meetings platform that supports access from computers, mobile devices, and SIP endpoints
- าั¿½ Free Wave App allows easy voice & Instant Messaging (IM) communications using desktops, Web, and Android/ iOS devices
- آذِرًا API available for third-party integrations, including CRM and PMS platforms
- � Advanced security protection with secure boot, unique certificate and random default password to protect calls and accounts
- าั¿½ Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router
- 17.1/2 Automated NAT firewall traversal service facilitates secure remote connections
- เว้น Enhanced reliability with support for Hot Standby High-Availability and local dual deployment
- آذِلًا Supports Full-Band Opus voice codec, jitter resilience up to 50% packet loss
- ī¿½ Compatible with GDMS for cloud setup, management, and monitoring
- ī¿1/2 Based on Asterisk\* version 16 open source telephony operating system

It supports UCM RemoteConnect cloud service for remote users to offer a best-in-class hybrid platform that combines the control of an on-premise IP PBX with the remote access and system manageability of a cloud solution. By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, instant messaging, voice conferencing and collaboration tools, the UCM6300 Audio series provides a powerful business communication platform for any organisation.

Grandstream UCM6308A - Technical Specifications Analog Telephone FXS Ports

� 8 RJ11 ports

ī సై All ports have lifeline capability in case of power outage

**PSTN Line FXO Ports** 

تزير 8 RJ11 ports

า๊¿½ All ports have lifeline capability in case of power outage

Network Interfaces

آذِيًّ Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+

**NAT Router** 

� Yes (supports router mode and switch mode)



Peripheral Ports

� 2\*USB 3.0 � 1\*SD card interface

**LED Indicators** 

ï¿1/2 Power 1/2, FXS, FXO, LAN, WAN, Heartbeat

LCD Display

าั¿½ 128x32 dot matrix graphic LCD with DOWN and OK buttons

Reset Switch

آذِيً Yes, long press for factory reset and short press for reboot

Voice-over-Packet Capabilities

Τ¿½ LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & amp; auto-switch to G.711, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss

Voice and Fax Codecs

� Opus, G.711 A-law/U-law, G.722, G722.1 G722.1C, G.723.1 5.3K/6.3K, G.726-32, G.729A/B, iLBC, GSM; T.38

QoS

آزا Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS

API

าั¿½ Full API available for third-party platform and application integration

**Telephony Operating System** 

ï¿1/2 Based on Asterisk version 16

**DTMF Methods** 

ï¿1/2 In-band audio, RFC4733, and SIP INFO

Provisioning Protocol & Plug-and-Play

� Mass provisioning using AES encrypted XML configuration file, auto-discovery & amp; auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE mDNS), eventlist between local and remote trunk

**Network Protocols** 



Τέ½ TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending), IPv6, OpenVPN®

Disconnect Methods

าั¿½ Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect

Media Encryption

ï¿1/2 SRTP, TLS, HTTPS, SSH, 802.1X

**Universal Power Supply** 

� 2x DC 12V Power Jack Input: 100~240VAC, 50/60Hz;Output: DC 12V, 2A

**Dimensions** 

� 485mm(L) x 187.2mm(W) x 46.2mm(H)

Weight

า๊ะไน Unit Weight: 2538g า๊ะไน Package Weight: 3463g

Temperature & amp; Humidity

� Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) � Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)

Mounting

ï¿1/2 Rack mount & amp; Desktop

Multi-Language Support

آذِ½ Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish

ī¿½ Customisable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands

า๊¿½ Customisable language pack to support any other languages

Caller ID

تز½ Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT

Polarity Reversal/Wink

آذِرُ Yes, with enable/disable option upon call establishment and termination

Call Center

 $\ddot{i}_{c}$  Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ workload, in-queue announcement



#### Customisable Auto Attendant

រ៉េះ½ Up to 5 layers of IVR (Interactive Voice Response) in multiple languages

Maximum Call Capacity

� Users: 1500

ï¿1/2 Concurrent calls (G.711): 200

ï¿1/2 Max concurrent SRTP calls (G.711): 150

Maximum Attendees of Conference Bridges

� 9 meeting rooms and up to 150 parties

Wave Mobile App

T¿½ Free; Available for desktop (Windows 10+, Mac OS 10+), web (Firefox and Chrome Browsers) and mobile (Android & DS), allows usersto join UCM-hosted meetings, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 Audio series IP PBX

#### Call Features

ῖ¿½ Call park, call forward, call transfer, call waiting, caller ID, call record, call history, ringtone, IVR, music on hold, call routes, DID, DOD,DND, DISA, ring group, ring simultaneously, time schedule, PIN groups, call queue, pickup group, paging/intercom, voicemail, callwakeup, SCA, BLF, voicemail to email, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist,voice meeting, eventlist, feature codes, busy camp-on/ call completion, voice control

### Firmware Upgrade

ī¿½ Supported by Grandstream Device Management System (GDMS), a zero-touch cloud provisioning and management system, It provides a centralised interface to provision, manage, monitor and troubleshoot Grandstream products

### Compliance

ï¿1/2 FCC: Part 15 (CFR 47) Class B, Part 68

าั¿½ CE: EN 55032, EN 55035, EN 61000-3-2, EN 61000-3-3, EN 62368.1, ES 203 021, ITU-T K.21

ï¿1/2 IC: ICES-003, CS-03 Part I Issue 9

ϊ¿½ RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2

ï¿1/2 Power adapter: UL 60950-1 or UL 62368-1

Price: £747.10