

Grandstream UCM6300A Audio Series IP PBX

Product Name: Grandstream UCM6300A Audio Series IP PBX

Manufacturer: Grandstream

Model Number: UCM6300A

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The Grandstream UCM6300A 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.

Grandstream UCM6300A Key Features

- Supports up to 1500 users and up to 200 concurrent calls
- Zero configuration provisioning of Grandstream SIP endpoints
- Built-in Instant Messaging (IM), Audio Conferencing & Web Meetings platform that supports access from computers, mobile devices, and SIP endpoints
- Free Wave App allows easy voice & Instant Messaging (IM) communications using desktops, Web, and Android/ iOS devices
- 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
- Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router
- Automated NAT firewall traversal service facilitates secure remote connections
- Enhanced reliability with support for Hot Standby High-Availability and local dual deployment
- Supports Full-Band Opus voice codec, jitter resilience up to 50% packet loss
- Compatible with GDMS for cloud setup, management, and monitoring
- Based on Asterisk* version 16 open source telephony operating system

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. 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 UCM6300A - Technical Specifications

Analog Telephone FXS Ports

- None
- All ports have lifeline capability in case of power outage

PSTN Line FXO Ports

- None
- All ports have lifeline capability in case of power outage

Network Interfaces

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

NAT Router

- Yes (supports router mode and switch mode)

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Peripheral Ports

• 1*USB 3.0
• 1*SD card interface

LED Indicators

• None

LCD Display

• 320x240 colour LCD with touch screen for Shortcut Keys and Scroll Bar

Reset Switch

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

Voice-over-Packet Capabilities

• LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & 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

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

API

• Full API available for third-party platform and application integration

Telephony Operating System

• Based on Asterisk version 16

DTMF Methods

• In-band audio, RFC4733, and SIP INFO

Provisioning Protocol & Plug-and-Play

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

Network Protocols

• 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

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(pending), IPv6, OpenVPN®

Disconnect Methods

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

Media Encryption

SRTP, TLS, HTTPS, SSH, 802.1X

Universal Power Supply

Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A

Dimensions

270mm(L) x 175mm(W) x 36mm(H)

Weight

Unit Weight: 705g

Package Weight: 1131g

Temperature & Humidity

Operating: 32 - 113°F / 0 ~ 45°C, Humidity 10 - 90% (non-condensing)
Storage: 14 - 140°F / -10 ~ 60°C, Humidity 10 - 90% (non-condensing)

Mounting

Wall mount & Desktop

Multi-Language Support

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

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

Customisable language pack to support any other languages

Caller ID

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

Polarity Reversal/Wink

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

Call Center

Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ workload, in-queue announcement

Customisable Auto Attendant

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

Maximum Call Capacity

Users: 250

Concurrent calls (G.711): 50

Max concurrent SRTP calls (G.711): 50

Maximum Attendees of Conference Bridges

3 meeting rooms and up to 50 parties

Wave Mobile App

Free; Available for desktop (Windows 10+, Mac OS 10+), web (Firefox and Chrome Browsers) and mobile (Android & iOS), allows users to 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

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, call wakeup, SCA, BLF, voicemail to email, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, voice meeting, event list, feature codes, busy camp-on/ call completion, voice control

Firmware Upgrade

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

FCC: Part 15 (CFR 47) Class B, Part 68

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

IC: ICES-003, CS-03 Part I Issue 9

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

Power adapter: UL 60950-1 or UL 62368-1

Price: 207.28EUR