GRANDSTREAM
CONNECTING THE WORLD

Unified Communication & Collaboration Solution

UCM6300 Audio Series

The UCM6300 Audio series 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 endpoints. 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 organization.

1500 users

Supports up to 1500 users and up to 200 concurrent calls

zero CONFIG

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

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

Gigabit

Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router

UCM RemoteConnect

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

	UCM6300A	UCM6302A	UCM6304A	UCM6308A
Analog Telephone FXS Ports	None	2 RJ11 ports	4 RJ11 ports	8 RJ11 ports
	All ports have lifeline capability in case of power outage			
PSTN Line FXO Ports	None	2 RJ11 ports	4 RJ11 ports	8 RJ11 ports
	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)			
Peripheral Ports	1*USB 3.0, 1*SD card interface	1*USB 2.0, 1*USB 3.0, 1*SD card interface	2*USB 3.0, 1*SD card interface	
LED Indicators	None			Power 1/2, FXS, FXO, LAN, WAN, Heartbeat
LCD Display	320x240 color LCD with touch screen for Shortcut Keys and Scroll Bar			128x32 dot matrix graphic LCD with DOWN and OK buttons
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), eventlist 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 (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		1x DC 12V Power Jack Input: 100~240VAC, 50/60Hz;Output: DC 12V, 2A	2x DC 12V Power Jack Input: 100~240VAC, 50/60Hz;Output: DC 12V, 2A
Dimensions	270mm(L) x 175mm(W) x 36mm(H)			485mm(L) x 187.2mm(W) x 46.2mm(H)
Weight	Unit Weight: 705g; Package Weight: 1131g	Unit Weight: 725g; Package Weight: 1221g	Unit Weight: 775g; Package Weight: 1621g	Unit Weight: 2538g; Package Weight: 3463g
Temperature & Humidity	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)			
Mounting	Wall mount & Desktop			Rack mount & Desktop
Multi-Language Support	-Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish -Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands -Customizable 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			
Customizable Auto Attendant	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	Users: 500 Concurrent calls (G.711): 75 Max concurrent SRTP calls (G.711): 75	Users: 1000 Concurrent calls (G.711): 150 Max concurrent SRTP calls (G.711): 120	Users: 1500 Concurrent calls (G.711): 200 Max concurrent SRTP calls (G.711): 150
Maximum Attendees of Conference Bridges	3 meeting rooms and up to 50 parties	5 meeting rooms and up to 75 parties	7 meeting rooms and up to 120 parties	9 meeting rooms and up to 150 parties
Wave 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, eventlist, 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 centralized 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			

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