



# 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.



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

	UCM6300A	UCM6302A	UCM6304A	UCM6308A
<b>Analog Telephone FXS Ports</b>	None	2 RJ11 ports	4 RJ11 ports	8 RJ11 ports
	All ports have lifeline capability in case of power outage			
<b>PSTN Line FXO Ports</b>	None	2 RJ11 ports	4 RJ11 ports	8 RJ11 ports
	All ports have lifeline capability in case of power outage			
<b>Network Interfaces</b>	Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+			
<b>NAT Router</b>	Yes (supports router mode and switch mode)			
<b>Peripheral Ports</b>	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	
<b>LED Indicators</b>	None			Power 1/2, FXS, FXO, LAN, WAN, Heartbeat
<b>LCD Display</b>	320x240 color LCD with touch screen for Shortcut Keys and Scroll Bar			128x32 dot matrix graphic LCD with DOWN and OK buttons
<b>Reset Switch</b>	Yes, long press for factory reset and short press for reboot			
<b>Voice-over-Packet Capabilities</b>	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			
<b>Voice and Fax Codecs</b>	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			
<b>QoS</b>	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS			
<b>API</b>	Full API available for third-party platform and application integration			
<b>Telephony Operating System</b>	Based on Asterisk version 16			
<b>DTMF Methods</b>	In-band audio, RFC4733, and SIP INFO			
<b>Provisioning Protocol &amp; Plug-and-Play</b>	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			
<b>Network Protocols</b>	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®			
<b>Disconnect Methods</b>	Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect			
<b>Media Encryption</b>	SRTP, TLS, HTTPS, SSH, 802.1X			
<b>Universal Power Supply</b>	Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A			2x DC 12V Power Jack Input: 100~240VAC, 50/60Hz; Output: DC 12V, 2A
<b>Dimensions</b>	270mm(L) x 175mm(W) x 36mm(H)			485mm(L) x 187.2mm(W) x 46.2mm(H)
<b>Weight</b>	Unit Weight: 705g;	Unit Weight: 725g;	Unit Weight: 775g;	Unit Weight: 2538g;
	Package Weight: 1131g	Package Weight: 1221g	Package Weight: 1621g	Package Weight: 3463g
<b>Temperature &amp; Humidity</b>	Operating: 32 - 113°F / 0 ~ 45°C, Humidity 10 - 90% (non-condensing) Storage: 14 - 140°F / -10 ~ 60°C, Humidity 10 - 90% (non-condensing)			
<b>Mounting</b>	Wall mount & Desktop			Rack mount & Desktop
<b>Multi-Language Support</b>	-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			
<b>Caller ID</b>	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 - BT, NTT			
<b>Polarity Reversal/Wink</b>	Yes, with enable/disable option upon call establishment and termination			
<b>Call Center</b>	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ workload, in-queue announcement			
<b>Customizable Auto Attendant</b>	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages			
<b>Maximum Call Capacity</b>	Users: 250	Users: 500	Users: 1000	Users: 1500
	Concurrent calls (G.711): 50 Max concurrent SRTP calls (G.711): 50	Concurrent calls (G.711): 75 Max concurrent SRTP calls (G.711): 75	Concurrent calls (G.711): 150 Max concurrent SRTP calls (G.711): 120	Concurrent calls (G.711): 200 Max concurrent SRTP calls (G.711): 150
<b>Maximum Attendees of Conference Bridges</b>	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
<b>Wave App</b>	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			
<b>Call Features</b>	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			
<b>Firmware Upgrade</b>	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			
<b>Compliance</b>	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 S003.1/1.2 Power adapter: UL 60950-1 or UL 62368-1			