



# Unified Communication & Collaboration Solution

## UCM6300 series

The UCM6300 series allows businesses to build powerful and scalable unified communication and collaboration solutions. This series of IP PBXs provide a platform that unifies all business communication on one centralized network, including voice, video calling, video conferencing, video surveillance, web meetings, data, analytics, mobility, facility access, intercoms and more. The UCM6300 series supports up to 3000 users and includes a built-in web meetings and video conferencing solution that allows employees to connect from the desktop, mobile, GVC series devices and IP phones. It can be paired with the UCM6300 ecosystem to offer a hybrid platform that combines the control of an on-premise IP PBX with the remote access of a cloud solution. The UCM6300 ecosystem consists of the Wave app for desktop, web and mobile, which provides a hub for collaborating remotely, and UCM RemoteConnect, a cloud NAT traversal service for ensuring secure remote connections. The UCM6300 series also offers cloud setup and management through GDMS and an API for integration with third-party platforms. By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, meeting and collaboration tools, the UCM6300 series provides a powerful platform for any organization.



Supports up to 3000 users and up to 450 concurrent calls



Zero configuration provisioning of Grandstream SIP endpoints



Built-in conferencing & meetings platform; supports desktop, Wave app, and SIP endpoints



Wave App allows communication with all UCM6300 users & solutions



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 (pending)



Supports Full-Band Opus voice codec and H.264/H.263/H.263+/VP8 video 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

	UCM6301	UCM6302	UCM6304	UCM6308
<b>Analog Telephone FXS Ports</b>	1 RJ11 Port	2 RJ11 Ports	4 RJ11 Ports	8 RJ11 Ports
	All ports have lifeline capability in case of power outage; number of ports can be expanded by peering with an FXS gateway			
<b>PSTN Line FXO Ports</b>	1 RJ11 Port	2 RJ11 Ports	4 RJ11 Ports	8 RJ11 Ports
	All ports have lifeline capability in case of power outage; number of ports can be expanded by peering with an FXO gateway			
<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>Video Codecs</b>	H.264, H.263, H263+, VP8			
<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, RFC2833, 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>	SIP, TCP/UDP/IP, RTP/RTCP, IAX, 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: 715g; Package Weight: 1211g	Unit Weight: 725g; Package Weight: 1221g	Unit Weight: 2490g; Package Weight: 3260g	Unit Weight: 2550g; Package Weight: 3320g
<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/ work-load, 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: 500 Concurrent calls (G.711): 75 Max concurrent SRTP calls (G.711): 50	Users: 1000 Concurrent calls (G.711): 150 Max concurrent SRTP calls (G.711): 100	Users: 2000 Concurrent calls (G.711): 300 Max concurrent SRTP calls (G.711): 200	Users: 3000 Concurrent calls (G.711): 450 Max concurrent SRTP calls (G.711): 300
<b>Maximum Attendees of Conference Bridges</b>	4 Video Conference rooms and up to 20 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & Opus) Voice Conference: Up to 75 parties (G.711)	6 Video Conference rooms and up to 30 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & Opus) Voice Conference: Up to 150 parties (G.711)	8 Video Conference rooms and up to 60 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & Opus) Voice Conference: Up to 200 parties (G.711)	10 Video Conference rooms and up to 80 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & Opus) Voice Conference: Up to 300 parties (G.711)
<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/conferences, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 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 conference, video conference, eventlist, feature codes, busy camp-on/ call completion, voice control, post-meeting reports, virtual fax sending/receiving, email to fax			
<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, ETSI 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			