GRANDSTREAM

CONNECTING THE WORLD

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 5000 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 and mobile, which provides a hub for collaborting 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.

5000 users

Supports up to 5000 users and up to 500 concurrent calls

zero CONFIG

Zero configuration provisioning of Grandstream SIP endpoints

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

Wave for Android & iOS allows communication with all UCM6300 users & solutions

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

HD

Supports Full-Band Opus voice codec and H.264/H.263/ H.263+/H.265/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
Analog Telephone FXS Ports	1 RJ11 Port	2 RJ11 Ports	4 RJ11 Ports	8 RJ11 Ports
	All ports have lifeline capability in case of power outage			
PSTN Line FXO Ports	1 RJ11 Port	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			
Video Codecs	H.264, H.263, H263+, H.265, VP8			
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, RFC2833, 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		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: 715g; Package Weight: 1211g	Unit Weight: 725g; Package Weight: 1221g	Unit Weight: 2510g; Package Weight: 3435g	Unit Weight: 2540g; Package Weight: 3465g
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/ work-load, in-queue announcement			
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages			
Maximum Call Capacity	Users: 500 Concurrent calls: 100 Max concurrent SRTP calls: 75	Users: 1000 Concurrent calls: 200 Max concurrent SRTP calls: 100	Users: 2500 Concurrent calls: 350 Max concurrent SRTP calls: 150	Users: 5000 Concurrent calls: 500 Max concurrent SRTP calls: 250
Maximum Attendees of Conference Bridges	2 Video Conference rooms and up to 12 parties with 1080p HD Voice Conference: Up to 75 parties	3 Video Conference rooms and up to 20 parties with 1080p HD Voice Conference: Up to 150 parties	4 Video Conference rooms and up to 40 parties with 1080p HD Voice Conference: Up to 200 parties	8 Video Conference rooms and up to 60 parties with 1080p HD Voice Conference: Up to 300 parties
Wave Mobile App	Allows Android & iOS users to join UCM-hosted meetings & communicate with other users/solutions registered to the UCM6300			
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 conference, video conference, 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, 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			