GRANDSTREAM
CONNECTING THE WORLD

Enterprise-Grade Unified Communication Solution

UCM6300 series

The UCM6300 series of IP PBXs allows businesses of all sizes to build powerful and scalable unified communication solutions in an easy-to-manage fashion with no licensing fees. This enterprise-grade IP PBX supports up to 6000 users and allows businesses to unify all of their communication technologies on to one centralized network, including voice, video calling, video conferencing, video surveillance, data tools, analytics, mobility options, facility access, intercom and more. The UCM6300 series can be integrated with third party applications and platforms, including customer relationship management (CRM) and property management systems (PMS), and offers an API for additional custom integrations. For cloud-based setup, management and monitoring, the UCM6300 series is compatible with the Grandstream Device Management System (GDMS). This unified communications manager also offers a built-in video conferencing platform with support for desktop users through the Grandstream Wave desktop app. By offering thousands of secure and reliable features at an all-inclusive price point without any licensing fees, costs-per-feature, or recurring fees, the UCM6300 series allows enterprises, offices, hotels, retail chains, call centers and more to create a seamless unified communication solution.

6000 users

Supports up to 6000 users and up to 400 concurrent calls

zero CONFIG

Zero configuration provisioning of Grandstream SIP endpoints

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

API

API available for third-party integrations, including CRM and PMS platforms

Built-in collaboration server (MCU) with conferencing, chats, screen sharing, white board, recording, and support for the Grandstream Wave app

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

Built-in call center suite and call queue for efficient call traffic management

Enhanced reliability with support for Hot Standby High-Availability

5-level IVR and multi-language auto-attendant to efficiently handle incoming calls

Compatible with GDMS for cloud setup, management and monitoring

Based on the latest 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 card 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		
LCD Display	320x240 color 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			
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		Input: 100~240VAC, 50/60Hz;Output: DC+12V, 2A	
Dimensions	270mm(L) x 175mm(W) x 36mm(H)			270mm(L) x 175mm(W) x 43mm(H)
Weight	Unit Weight: 700g; Package Weight: 1185g	Unit Weight: 705g; Package Weight: 1190g	Unit Weight: 850g; Package Weight: 1335g	Unit Weight: 950g; Package Weight: 1435g
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			
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: 50 Concurrent calls: 15 Max concurrent SRTP calls: 15	Users: 1000 Concurrent calls: 200 Max concurrent SRTP calls: 120	Users: 2000 Concurrent calls: 300 Max concurrent SRTP calls: 200	Users: 6000 Concurrent calls: 400 Max concurrent SRTP calls: 240
Maximum Attendees of Conference Bridges	Video Conference: None Voice Conference: Up to 15	Video Conference: Up tp 20 Voice Conference: Up to 80	Video Conference: Up to 30 Voice Conference: Up to 120	Video Conference: Up to 40 Voice Conference: Up to 160
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, EN61000-3-2, EN61000-3-3, EN 62368.1, ES 203 021, ITU K.21 IC: ICES-003, CS-03 Part I Issue 9 RCM: AS/NZS CISPR32, AS/NZS 61000.3.2, AS/NZS 61000.3.3, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2 TBR4, UL 60950 (power adapter)			

*Asterisk is a Registered Trademark of Digium, Inc.