GRANDSTREAM
CONNECTING THE WORLD

Uni ied Communication & Collaboration Solution

UCM6304

The UCM6304 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 UCM6304 supports up to 2000 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 collaborting remotely, and UCM RemoteConnect, a cloud NAT traversal service for ensuring secure remote connections. The UCM6304 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 UCM6304 provides a powerful platform for any organization.

Supports up to 2000 users and up to 300 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

UCM RemoteConnect

Automated NAT firewall traversal service facilitates secure remote connections

Enhanced reliability with support for Hot Standby High-Availability

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

	UCM6304
Analog Telephone FXS Ports	4 RJ11 Ports All ports have lifeline capability in case of power outage; number of ports can be expanded by peering with an FXS gateway
PSTN Line FXO Ports	4 RJ11 Ports All ports have lifeline capability in case of power outage; number of ports can be expanded by peering with an FXO gateway
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	2*USB 3.0, 1*SD card interface
LED Indicators	Power 1/2, FXS, FXO, LAN, WAN, Heartbeat
LCD Display	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+, 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	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®
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	2x DC 12V Power Jack Input: 100~240VAC, 50/60Hz;Output: DC 12V, 2A
Dimensions	485mm(L) x 187.2mm(W) x 46.2mm(H)
Weight	Unit Weight: 2490g; Package Weight: 3260g
Temperature & Humidity	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)
Mounting	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: 2000 Concurrent calls (G.711): 300 Max concurrent SRTP calls (G.711): 200
Maximum Attendees of Conference Bridges	Up to 15 simultaneous video conference rooms, up to 200 simultaneous participants in all rooms combined, up to 9 video feeds in all conference rooms
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/conferences, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 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 conference, video conference, eventlist, feature codes, busy camp-on/ call completion, voice control, post-meeting reports, virtual fax sending/receiving, email to fax
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
Internet Protocol Standards	RFC 3261, RFC 3262, RFC 3263, RFC 3264, RFC 3515, RFC 3311, RFC 4028. RFC 2976, RFC 3842, RFC 3892, RFC 3428, RFC 4733, RFC 4566, RFC 2617, RFC 3856, RFC 3711, RFC 4582, RFC 4583, RFC 5245, RFC 5389, RFC 5766, RFC 6347, RFC 6455, RFC 8860, RFC 4734, RFC 3665, RFC 3323, RFC 3550
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

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