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 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.

3000 users

Supports up to 3000 users and up to 450 concurrent calls

zero CONFIG

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

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 (pending)

HD

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

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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; number of ports can be expanded by peering with an FXS gateway			
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; 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	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+, 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	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: 2490g; Package Weight: 3260g	Unit Weight: 2550g; Package Weight: 3320g
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 (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
Maximum Attendees of Conference Bridges	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)
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			
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			

GRANDSTREAM
CONNECTING THE WORLD

10-line Carrier-Grade IP Phone

GRP2615

The GRP2615 is a high-end carrier-grade IP phone designed with zero-touch provisioning for mass deployment and easy management. It supports 10 lines and 5 SIP accounts while featuring a sleek design and a suite of next-generation features including integrated Wi-Fi, Bluetooth support, 40 multi-purpose keys (MPKs), an available extension module, dual Gigabit ports and more. This device features a large 4.3 inch color LCD with swappable face plates to allow for easy logo customization. The GRP series includes carrier-grade security features to provide enterprise-level security, including secure boot, dual firmware images and encrypted data storage. For cloud provisioning and centralized management, the GRP2615 is supported by Grandstream's Device Management System (GDMS), which provides a centralized interface to configure, provision, manage and monitor deployments of Grandstream endpoints. Built for the needs of busy desktop workers and designed for easy deployment by enterprises, service providers and other high-volume markets, the GRP2615 offers an easy-to-use and easy-to-deploy voice platform.

HD audio, handset and speakerphone with support for wide-band audio

10 line keys with up to 5 SIP accounts

40 built-in digital BLF keys; available extension module offers 40 BLF/speed dial keys per module

Enterprise-level protection including secure boot, dual firmware images, and encrypted data storage

Integrated dual-band 802.11 a/b/g/n/ac Wi-Fi

Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE

Swappable face plates to allow for easy logo customization

Integrated Bluetooth

Equipped with noise shield technology to minimize background noise

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS(A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE
Bluetooth	Yes, integrated
Wi-Fi	Yes, integrated dual-band Wi-Fi 802.11 a/b/g/n/ac (2.4GHz & 5GHz)
Graphic Display	4.3 inch (480x272) TFT color LCD
Voice Codecs and Capabilities	Support for G7.29A/B, G.711µ/a-law, G.726, G.722(wide-band), G723, iLBC, OPUS, in-band and out-of-band DTMF(in audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC
Telephony Features	Hold, transfer, forward, 3-way conference, call park, call pickup, shared-call-appearance(SCA)/bridged-line-appearance(BLA), downloadable phonebook(XML, LDAP, up to 2000 items), call waiting, call log(up to 2000 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot-desking, personalized music ringtones and music on hold, server redundancy and fail-over
HD Audio	Yes, HD handset and speakerphone with support for wideband audio
Extension Module	Yes (GBX20)
Feature Keys	10 line keys with up to 5 SIP accounts, 5 XML programmable context sensitive softkeys, 5 navigation/menu keys, 9 dedicated function keys for: MESSAGE(with LED indicator), TRANSFER, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
Base Stand	Yes, 2 angle positions available, Wall Mountable (*wall mount sold separately)
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 QoS (ToS, DiffServ, MPLS)
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets), USB
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control, secure boot
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese and more
Upgrade/Provisioning	Firmware upgrade via FTP / TFTP / HTTP / HTTPS, mass provisioning with GDMS (Grandstream Device Management System), TR-069 or AES encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input: 100 - 240V; Output: +12VDC, 1A Integrated Power-over-Ethernet (802.3af) Max power consumption 6.3W (power adapter) or 7.4W (PoE)
Temperature and Humidity	Operation: 0°C to 40°C, Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing
Package Content	GRP2615 phone, handset with cord, phone stand, 12V power adapter, network cable, Quick Installation Guide, GPL license
Physical	Dimension: ; Unit weight: 970g ; Package weight: 1480g Dimension: 243mm x 210mm x 82.3mm
Compliance	FCC: Part 15 Class B; Part 15 Subpart C, 15.247; Part 15 Subpart E, 15.407; FCC Part 68 HAC CE: EN 55032; EN 55035; EN 61000-3-2; EN 61000-3-3; EN 62368-1; EN 301489-1; EN 301489-17; EN 300328; EN 301893; EN 62311 RCM: AS/NZS CISPR32; AS/NZS 4268; AS/NZS 62368.1; AS/CA S004. IC: ICES-003; CS-03; RSS-247; RSS-102.

GRANDSTREAM
CONNECTING THE WORLD

A simple and reliable IP Phone

GXP1610/1615

A simple IP phone for small business users, the GXP1610/1615 delivers a user-friendly VoIP calling experience in a very easy-to-use IP phone. The GXP1610/1615 offers support for 1 line, 2 call appearances and includes 3-way voice conferencing to maximize productivity. Additional features include a 132x48 (2.95”.) LCD screen for easy viewing, 3 XML programmable soft keys for customization, 10/100 mbps ports, integrated PoE on the GXP1615 model, EHS support for Plantronics headsets and multi-language support. These features allow the GXP1610/GXP1615 to be a high-quality small business IP phone that is simple and easy-to-use.

1 LINE

1 SIP account, up to 2 call appearances

TLS and SRTP security encryption technology to protect calls and accounts

3 WAY

3-way audio conferencing for easy conference calls

EHS

Electronic Hook Switch (EHS) support for Plantronics headsets

Automated provisioning options include TR-069 and XML config files

HD

Full-duplex speakerphone with HD audio to maximize audio quality and clarity

zero CONFIG

Use with Grandstream’s UCM series of IP PBXs for Zero Config provisioning

PoE

Built-in PoE 802.3af to power the device and give it a network connection (GXP1615 ony)

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP, CDP/SNMP/RTCP-XR
Network Interfaces	Dual switched auto-sensing 10/100 Mbps Ethernet ports, integrated PoE (GXP1615 only)
Graphic Display	132 x 48 (2.95”) LCD display
Feature Keys	2 line keys with dual-color LED and 1 SIP account, 3 XML programmable context sensitive soft keys, 5 (navigation, menu) keys. 13 dedicated function keys for PAGE/INTERCOM, PHONEBOOK, MESSAGE, HOME, HOLD, RECORD, MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME
Voice Codecs	Support for G.711µ/a, G.722 (wide-band), G.723,G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy), 3-way conferencing, call park/pickup, downloadable phone book (XML, LDAP, up to 1000 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over
Headset Jack	RJ9 headset jack (allowing EHS with Plantronics headsets)
Base Stand	Yes, 2 angled positions available, wall mountable
Wall Mountable	Yes
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese and more
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file, FTP/FTPS
Power & Green Energy Efficiency	Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA; PoE: IEEE802.3af Class 2, 3.84W-6.49W; IEEE802.3az (EEE) (GXP1615 Only)
Physical	Dimension: 209mm (L) x 184.5mm (W) x 76.2mm (H) (with handset) Unit weight: 0.74kg; Package weight: 1.1kg
Temperature and Humidity	Operation: 0°C to 40°C, Storage: -10°C to 60°C , Humidity: 10% to 90% Non-condensing
Package Content	GXP1610/1615 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, brochure, GPL License
Compliance	FCC: Part 15 (CFR 47) Class BCE : EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1