GRANDSTREAM
CONNECTING THE WORLD

Optimized UC Features for SMBs

UCM6208 series

Designed to provide a centralized solution for the communication needs of businesses, the UCM6208 IP PBX appliance combines enterprise-grade voice, video, data, and mobility features in an easy-to-manage solution. This IP PBX allows businesses to unify multiple communication technologies, such as voice, video calling, video conferencing, video surveillance, data tools, mobility options and facility access management onto one common network that can be managed and/or accessed remotely. The secure and reliable UCM6208 delivers enterprise-grade features without any licensing fees, costs-per-feature or recurring fees.

800 users
Supports up to 800 users, 50 SIP trunk accounts, up to 100 concurrent calls

zero CONFIG
Zero configuration provisioning of Grandstream SIP endpoints

Strongest-possible security protection using SRTP, TLS and HTTPS encryption

Gigabit
Dual Gigabit network ports with integrated PoE+

Supports up to a 5-level IVR (Interactive Voice Response)

REC
Built-in call recording server; recordings accessed via web user interface

Supports call queue for efficient call volume management

Built-in Call Detail Records (CDR) for tracking phone usage by line, date, etc.

Multi-language auto-attendant to efficiently handle incoming calls

Integrated LDAP and XML phonebooks, flexible dial plan

SIP
Supports any SIP video endpoint that using the H.264, H.263 and H.263+ codecs

Supports voicemail and fax forwarding to email

Analog Telephone FXS Ports	2 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	8 ports
Network Interfaces	Dual Gigabit RJ45 ports with integrated PoE Plus (IEEE 802.3at-2009)
NAT Router	Yes (supports router mode and switch mode)
Peripheral Ports	USB, SD
LED Indicators	Power/Ready, Network, PSTN Line, USB, SD
LCD Display	128x32 graphic LCD with DOWN & OK button
Reset Switch	Yes
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
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC, GSM, AAL2-G.726-32, ADPCM; T.38
Video Codecs	H.264, H.263, H263+
QoS	Layer 3 QoS, Layer 2 QoS
DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of Grandstream IP endpoints via Zero-Config (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, SIP (RFC3261), STUN, SRTP, TLS, LDAP
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Media Encryption	SRTP, TLS, HTTPS, SSH
Universal Power Supply	Output: 12VDC, 1.5A; Input: 100 ~ 240VAC, 50 ~ 60Hz
Dimensions	440(W) x 185(H) x 44(D) mm
Weight	Unit weight 2.23kg, Package weight 3.09kg
Environmental	Operating: 32 ~ 104ºF / 0 ~ 40ºC, 10 ~ 90% (non-condensing); Storage: 14 ~ 140ºF / -10 ~ 60ºC
Mounting	Rack mount & Desktop
Multi-Language Support	English/Simplified Chinese/Traditional Chinese/Spanish/French/Portuguese/German/Russian/Italian/Polish/Czech for Web UI; Customizable IVR/voice prompts for English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic; Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT
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/busy level, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response)
Maximum Call Capacity	-Registered SIP devices: supports up to 800 registered SIP devices/users -Concurrent SIP calls: Up to 100 or 66% performance if calls are SRTP encrypted
Conference Bridges	Up to 6 password-protected conference bridges allowing up to 32 simultaneous PSTN or IP participants
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom etc.
Compliance	FCC: Part 15 (CFR 47) Class B, Part 68 CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS A-TICK: AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 ITU-T K.21 (Basic Level); UL 60950 (power adapter)