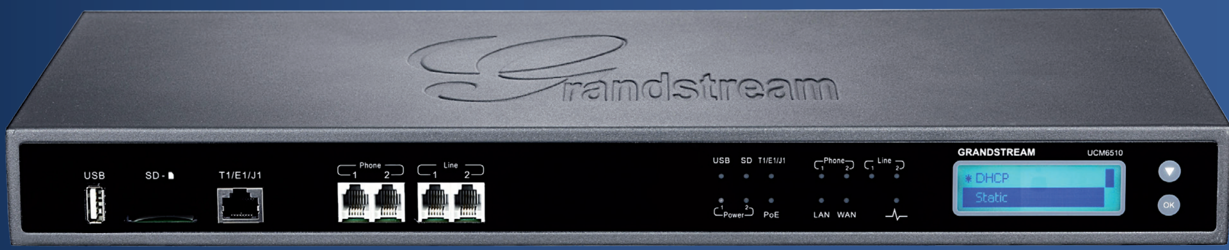
GRANDSTREAM
CONNECTING THE WORLD



UCM6510

UC features for up to 2000 users

The UCM6510 IP PBX appliance is designed to bring leading edge voice, video, data, and mobility features to enterprises, small and medium businesses, retail and residential environments in an easy-to-manage fashion. This enterprise-grade on premise IP PBX supports E1, T1 and J1 networks and offers scalability by supporting up to 2000 users. The UCM6510 sports a 1GHz quad-core Cortex A9 processor, 1GB RAM and 32GB flash. This secure and reliable IP PBX delivers unified communication features at an unprecedented price point without any licensing fees, costs-per-feature, or recurring fees.

2000 users

Supports up to 2000 users, unlimited SIP trunk accounts, up to 200 concurrent calls

zero CONFIG

Zero configuration provisioning of Grandstream SIP endpoints

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

Gigabit

Gigabit netwokr ports with integrated PoE; Integrated NAT router

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

Interfaces	
Analog Telephone FXS Ports	2 RJ11 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	2 RJ11 ports (both with lifeline capability in case of power outage)
T1/E1/J1 Interface	1 RJ45 port
Network Interfaces	Dual Gigabit ports (switched or routed) with PoE; A 3rd Gigabit port for Hot-Standby Clustering
NAT Router	Yes (user configurable)
Peripheral Ports	USD, SD
LED Indicators	Power 1/2, PoE, USB, SD, T1/E1/J1, FXS 1/2, FXO 1/2, LAN, WAN, Cluster 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/Video Capabilities	
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
Signaling & Control	
DTMF Methods	In Audio, RFC2833, and SIP INFO
Digital Signaling	TPRI, SS7, MFC/R2, RBS (pending)
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE m/DNS), eventlist between local and remote trunks
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending)
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Security	
Media Encryption	SRTP, TLS, HTTPS, SSH
Advanced Defense	Fail2ban, alert events, Whitelist, Blacklist, strong password based access control
Physical	
Universal Power Supply	Input: 100 ~ 240VAC, 50/60Hz; Output: DC+12V, 1.5A
Physical	Unit Weight: 2.165 kg; Package Weight: 3.012 kg
Dimensions	440mm(L) x 185mm(W) x 44mm(H)
Environmental	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)
Mounting	Rack mount & Desktop
Additional Features	
Multi-Language Support	English/Simplified Chinese/Traditional Chinese/Spanish/French/Portuguese/German/Russian/Italian/Polish/Czechfor Web UI; Customizable IVR/voice prompts for English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT Japan (pending)
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)
Maximum Call Capacity	Up to 2000 registered SIP endpoints, up to 200 concurrent calls
Conference Bridges	Up to 8 bridges, up to 64 simultaneous conference attendees
Call Features	Call park, call forward, call transfer, DND, DISA, ring group, pickup group, blacklist, paging/intercom etc.