UCM 6510

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.


Features

- Supports up to 2000 SIP endpoint registrations, up to 200 concurrent calls and up to 64 conference attendees
- 1GHz quad-core Cortex A9 processor
- 1GB DDR3 Ram, 32GB Flash
- 1 Integrated T1/E1/J1 interface, 2PSTN trunk FXO ports, 2 analog telephone/Fax FXS ports with lifeline capability
- Gigabit network ports with Integrates PoE, USB, SD card, integrated NAT router
- Comprehensive security protection using SRTP, TLS and HTTPS with hardware encryption accelerator

- Quickly setup and provision Grand stream endpoints using the Auto-Discovery and Zero Config feature within the product's web user interface.

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	s Dual Gigabit ports (switched or routed) with PoE+
NAT Router	Yes (user configurable)
Peripheral Ports	USB, SD
LED Indicators	Power 1/2, PoE, USB, SD, T1/E1/J1, FXS 1/2, FXO 1/2, LAN, WAN
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
Plug-and-Play	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,

Disconnect Methods	HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending) 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/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/ 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.
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 RCM: AS/NZS CISPR 22, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002 ITU-T K.21 (Basic Level); UL 60950 (power adapter) T1: TIA-968-B Section 5.2.4 E1: TBR4/TBR12/TBR13, E1: AS/ACIF