UCM6510
IP PBX Appliance

Innovative IP Voice & Video

UCM6510 IP PBX Appliance

The UCM6510 is an innovative IP PBX appliance for E1/T1/J1 networks that brings enterprise-grade Unified Communications and security protection to small-to-medium businesses (SMBs) in an easy-to-manage fashion. Powered by an advanced hardware platform and revolutionary software functionalities, the UCM6510 offers a breakthrough turnkey solution for converged voice, video, data, fax, security surveillance, and mobility applications out of the box without any extra license fees or recurring costs.

Feature Highlights

- 1GHz 4-core Cortex A9 application processor, large memory (1GB DDR3 RAM, 32GB Flash), and dedicated high performance multi-core DSP array for advanced voice processing
- 1 Integrated T1/E1 Interface, 2 PSTN trunk FXO ports, 2 analog telephone FXS ports with lifeline capability in case of power outage, and up to 50 SIP trunk accounts
- Gigabit network port(s) with integrated PoE, USB, SD card; integrated NAT router with advanced QoS support
- Hardware DSP based 128ms-tail-length carrier-grade line echo cancellation (LEC), hardware based caller ID/call progress tone and smart automated impendance matching for various countries
- Automatic export of previous day's data, periodically cleans data
- Supports up to 2000 SIP endpoint registrations, up to 200 concurrent calls (up to 100 SRTP encrypted concurrent calls), and up to 32 conference attendees
- Flexible dial plan, call routing, site peering, call recording, central control panel for endpoints, integrated NTP server, and integrated LDAP contact directory
- Automated detection and provisioning of IP phones, video phones, ATAs, gateways, SIP cameras, and other endpoints for easy deployment
- Strongest-possible security protection using SRTP, TLS, and HTTPS with hardware encryption accelerator
- Manual and automatic recording for each SIP call and each trunk

Corporate Headquarters:
126 Brookline Avenue, 3rd Floor
Boston, MA 02215, USA

Regional Offices: Dallas, TX, USA | Los Angeles, CA, USA | Casablanca, Morocco | Valencia, Venezuela | Hangzhou, China | Shenzhen, China

Grandstream Networks, Inc. **www.grandstream.com** **info@grandstream.com**

UCM6510 Technical Specifications

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 Interface	1 RJ45 port
Network Interfaces	3 ports: 1 LAN, 1 WAN, 1 Heartbeat
NAT Router	Yes
Peripheral Ports	USB, SD
LED Indicators	Power 1/2, PoE, USB, SD, T1/E1/J1, FXS 1/2, FXO 1/2, LAN, WAN, Heartbeat
LCD Display	128x32 DOTS LCD with DOWN and OK buttons
Reset Switch	Yes, long press for factory reset an 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
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66/multicast SIP SUBSCRIBE/mDNS), 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, LADP
Security	
Media	SRTP, TLS, HTTPS, SSH
Physical	
Universal Power Supply	Input: 100 ~ 240VAC, 50/60Hz; Output: DC+12V, 1.5A;
Environmental	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)
Dimensions	Unit Weight: TBD Package Weight: TBD
Mounting	Wall mount & Desktop
Additional Features	
Multi-Language Support	English/Chinese/Spanish/French/Portuguese/German/Russian/Italian/Polish/Czech for Web UI Customizable IVR/extension to support English, Chinese, British English, German, Spanish, Greeks, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish,Turkish, Hebrew, Arabic
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)
Concurrent Calls	Up to 200 calls
Conference Bridges	Up to 5 bridges, up to 32 seats
Call Features	Call park, call forward, call transfer, DND, DISA, ring group, pickup group, blacklist, paging/intercom etc.
Defense	Fail2ban, Alert Events, Data Sync (automatically export previous day's data), cleaner (periodically deletes user data)
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) T1: TIA-968-B Section 5.2.4 E1: TBR12/TBR13, E1: AS/ACIF