Grandstream UCM6301 IP PBX

Product Name: Grandstream UCM6301 IP PBX
Manufacturer: Grandstream
Model Number: UCM6301

Grandstream UCM6301 IP PBX
The Grandstream UCM6301 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 centralised network, including voice, video calling, video conferencing, video surveillance, web meetings, data, analytics, mobility, facility access, intercoms and more.
Grandstream UCM6301 Key Features

ï¿½ Supports up to 500 users and up to 75 concurrent calls
ï¿½ Zero configuration provisioning of Grandstream SIP endpoints
ï¿½ Built-in conferencing & meetings platform; supports desktop, Wave app, and SIP endpoints
ï¿½ Wave for Android, iOS, Chrome and Firefox browsers allows communication with all UCM6300 users & solutions
ï¿½ API available for third-party integrations, including CRM and PMS platforms
ï¿½ Advanced security protection with secure boot, unique certificate and random default password to protect calls and accounts
ï¿½ Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router
ï¿½ Automated NAT firewall traversal service facilitates secure remote connections
ï¿½ Supports Full-Band Opus voice codec and H.264/H.263/ H.263+/H.265/VP8 video codec, jitter resilience up to 50% packet loss
ï¿½ Compatible with GDMS for cloud setup, management and monitoring
ï¿½ Based on Asterisk* version 16 open source telephony operating system

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 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 organisation.
Grandstream UCM6301 - Technical Specifications
Analog Telephone FXS Ports

ï¿½ 1 RJ11 Port
ï¿½ All ports have lifeline capability in case of power outage

PSTN Line FXO Ports

ï¿½ 1 RJ11 Port
ï¿½ All ports have lifeline capability in case of power outage

Network Interfaces

ï¿½ Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+

NAT Router

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ï¿½ Yes (supports router mode and switch mode)

Peripheral Ports

ï¿½ 1*USB 3.0
ï¿½ 1*SD card interface

LED Indicators

ï¿½ None

LCD Display

ï¿½ 320x240 colour LCD with touch screen for Shortcut Keys and Scroll Bar

Reset Switch

ï¿½ Yes, long press for factory reset and short press for reboot

Voice-over-Packet Capabilities

ï¿½ LEC with NLP Packetized Voice Protocol Unit
ï¿½ 128ms-tail-length carrier grade Line Echo Cancellation
ï¿½ Dynamic Jitter Buffer
ï¿½ Modem detection & auto-switch to G.711
ï¿½ NetEQ
ï¿½ FEC 2.0
ï¿½ Jitter resilience up to 50% audio packet loss

Voice and Fax Codecs

ï¿½ 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

ï¿½ H.264, H.263, H263+, H.265, VP8

QoS

ï¿½ Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS

API

ï¿½ Full API available for third-party platform and application integration

Telephony Operating System

ï¿½ Based on Asterisk version 16

DTMF Methods

ï¿½ In-band audio, RFC2833, and SIP INFO

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Provisioning Protocol & Plug-and-Play

ï¿½ 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

ï¿½ TCP/UDP/IP, RTP/RTCP, 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

ï¿½ Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect

Media Encryption

ï¿½ SRTP, TLS, HTTPS, SSH, 802.1X

Universal Power Supply

ï¿½ Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A

Dimensions

ï¿½ 270mm(L) x 175mm(W) x 36mm(H)

Weight

ï¿½ Unit Weight: 715g
ï¿½ Package Weight: 1211g

Temperature & Humidity

ï¿½ Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing)
ï¿½ Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)

Mounting

ï¿½ Wall mount & Desktop

Multi-Language Support

ï¿½ Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish
ï¿½ Customisable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands
ï¿½ Customisable language pack to support any other languages

Caller ID

ï¿½ Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT

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Polarity Reversal/Wink

ï¿½ Yes, with enable/disable option upon call establishment and termination

Call Center

ï¿½ Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ work-load, in-queue announcement

Customisable Auto Attendant

ï¿½ Up to 5 layers of IVR (Interactive Voice Response) in multiple languages

Maximum Call Capacity

ï¿½ Users: 500
ï¿½ Concurrent calls (G.711): 75
ï¿½ Max concurrent SRTP calls (G.711): 50

Maximum Attendees of Conference Bridges

ï¿½ 2 Video Conference rooms and up to 12 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & G.711)
ï¿½ Voice Conference: Up to 75 parties (G.711)

Wave Mobile App

ï¿½ Allows Android & iOS users to join UCM-hosted meetings & communicate with other users/solutions registered to the UCM6300

Call Features

ï¿½ 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

Firmware Upgrade

ï¿½ 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

ï¿½ FCC: Part 15 (CFR 47) Class B, Part 68
ï¿½ CE: EN 55032, EN 55035, EN 61000-3-2, EN 61000-3-3, EN 62368-1, ETSI ES 203 021, ITU-T K.21
ï¿½ IC: ICES-003, CS-03 Part I Issue 9
ï¿½ RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2
ï¿½ Power adapter: UL 60950-1 or UL 62368-1

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Price: £232.70