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Grandstream UCM6302A Audio Series IP PBX

Product Name: Grandstream UCM6302A Audio Series IP PBX
Manufacturer: Grandstream
Model Number: UCM6302A

Grandstream UCM6302A Audio Series IP PBX
The Grandstream UCM6302A allows businesses to build powerful and scalable unified communication and collaboration solutions. This series of IP PBXs provide a platform that unifies fundamental business communications needs, including voice, instant messaging (IM), voice meetings, audio web meetings, data, analytics, mobility, facility access,intercoms and more. The UCM6300 Audio Series supports up to 1500 users and includes a built-in instant messaging (IM), voice/web conferencing platform, and the free Wave App that allows users to communicate and collaborate from desktops, mobile devices, IP phones, and other SIP end points. It supports UCM RemoteConnect cloud service for remote users to offer a best-in-class hybrid platform that combines the control of an on-premise IP PBX withthe remote access and system manageability of a cloud solution. By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, instant messaging, voice conferencing andcollaboration tools, the UCM6300 Audio series provides a powerful business communication platform for any organisation.
Grandstream UCM6302A Key Features

ï¿½ Supports up to 1500 users and up to 200 concurrent calls
ï¿½ Zero configuration provisioning of Grandstream SIP endpoints
ï¿½ Built-in Instant Messaging (IM), Audio Conferencing & Web Meetings platform that supports access from computers, mobile devices, and SIP endpoints
ï¿½ Free Wave App allows easy voice & Instant Messaging (IM) communications using desktops, Web, and Android/ iOS devices
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 Enhanced reliability with support for Hot Standby High-Availability and local dual deployment
ï¿½ Supports Full-Band Opus voice 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

Grandstream UCM6302A - Technical Specifications
Analog Telephone FXS Ports

ï¿½ 2 RJ11 ports
ï¿½ All ports have lifeline capability in case of power outage

PSTN Line FXO Ports

ï¿½ 2 RJ11 ports
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

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

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Peripheral Ports

ï¿½ 1*USB 2.0
ï¿½ 1*USB 3.0
ï¿½ 1*SDcard 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, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, NetEQ, FEC 2.0, 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

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, RFC4733, and SIP INFO

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

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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: 725g
ï¿½ Package Weight: 1221g

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

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

Grandstream UCM6302A Audio Series IP PBX

skills/availability/ workload, 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): 75

Maximum Attendees of Conference Bridges

ï¿½ 5 meeting rooms and up to 75 parties

Wave Mobile App

ï¿½ Free; Available for desktop (Windows 10+, Mac OS 10+), web (Firefox and Chrome Browsers) and mobile (Android & iOS), allows usersto join UCM-hosted meetings, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 Audio series IP PBX

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, callwakeup, SCA, BLF, voicemail to email, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist,voice meeting, 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 centralised 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, 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

Price: £358.50