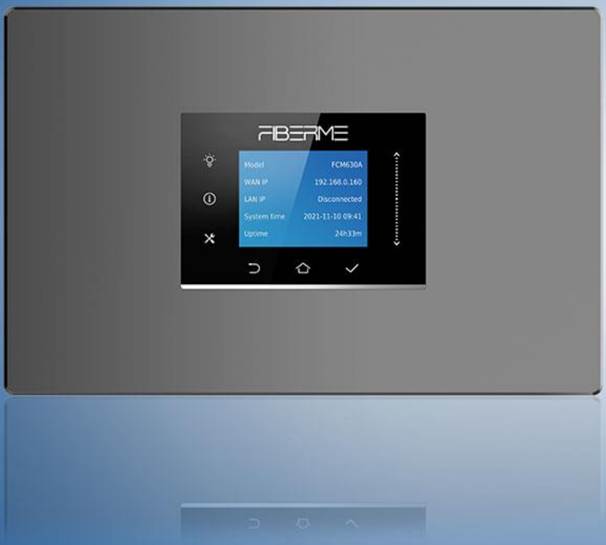

FIBERME

Unified Communication & Collaboration Solution

FCM630A IPPBX

The FCM630A Audio series 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 FCM630A Audio Series supports up to 250 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 endpoints.

By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, instant messaging, voice conferencing and collaboration tools, the FCM630A Audio series provides a powerful business communication platform for any organization.

250 users

Supports up to 250 users and up to 50 concurrent calls

zero CONFIG

Zero configuration provisioning of FIBERME SIP endpoints

Built-in Instant Messaging (IM), Audio Conferencing & Web Meetings platform that supports access from computers, mobile devices, and SIP endpoints

API

API available for third-party integrations, including CRM and PMS platforms

Advanced security protection with secure boot, unique certificate and random default password to protect calls and accounts

Gigabit

Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router

Enhanced reliability with support for Hot Standby High-Availability and local dual deployment

Supports Full-Band Opus voice codec,jitter resilience up to 50% packet loss

Based on Asterisk* version 16 open source telephony operating system

	FCM630A
Analog Telephone FXS Ports	None
PSTN Line FXO Ports	None
Network Interfaces	Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+
NAT Router	Yes (supports router mode and switch mode)
Peripheral Ports	1*USB 3.0, 1*SD card interface
LED Indicators	None
LCD Display	320x240 color LCD with touch screen for Shortcut Keys and Scroll Bar
Reset Switch	Yes, long press for factory reset and short press for reboot
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, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss
Voice and Fax Codecs	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	Layer 2 QoS (802.1Q, 802.1p) and Layer 3(ToS, DiffServ, MPLS) QoS
API	Full API available for third-party platform and application integration
Telephony Operating System	Based on Asterisk version 16
DTMF Methods	In-band audio, RFC4733, and SIP INFO
Provisioning Protocol & Plug-and-Play	Mass provisioning using AES encrypted XML configuration file, auto-discovery & auto-provisioning of FIBERME IP endpoints via ZeroConfig (DHCP 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, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending), IPv6, OpenVPN®
Disconnect Methods	Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect
Media Encryption	SRTP, TLS, HTTPS, SSH, 802.1X
Universal Power Supply	Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A
Dimensions	270mm(L) x 175mm(W) x 36mm(H)
Weight	Unit Weight: 705g; Package Weight: 1131g
Temperature & Humidity	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)
Mounting	Wall mount & Desktop
Multi-Language Support	-Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish -Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands -Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT
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/ workload, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages
Maximum Call Capacity	Users: 250, Concurrent calls (G.711): 50 Max concurrent SRTP calls (G.711): 50
Maximum Attendees of Conference Bridges	3 meeting rooms and up to 50 parties
Softphone	FCM630A Audio series IP PBX is compatible with any Windows, Mac, Android, and IOS
Call Features	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 meeting, eventlist, feature codes, busy camp-on/ call completion, voice control
Firmware Upgrade	Firmware upgrade via local firmware file upload with unified firmware for FCM63xA series
Compliance	FCC: Part 15 Class B; CE: EN 55032; EN 55035; EN 61000-3-2; EN 61000-3-3; EN 62368-1; RCM: AS/NZS CISPR32; AS/NZS 62368.1; AS/CA S004; IC: ICES-003; CS-03;