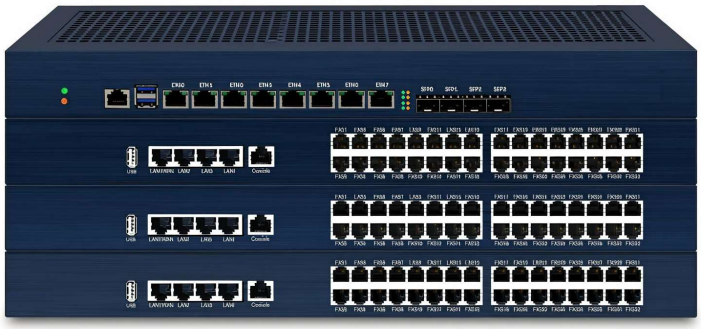
EDGE UC Series
Advanced Unified Communications

EDGE Call Manager (ECM)
EDGE Voice Gateway (EVG)

A Modular IP PBX (Disaggregated VoIP Architecture) breaks away from traditional closed-box appliances by splitting core telephony functions, security layers, and physical line terminations into distinct, independent hardware and software tiers over an IP network. Instead of a single monolithic machine managing everything, the architecture distributes workloads across specialized components—standalone FXS/FXO voice gateways, a centralized Session Border Controller (SBC), smart call manager core, and dedicated application servers for IVR and voicemail.



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Disaggregated VoIP Architecture Features

Disaggregated VoIP Architecture replaces a single, all-in-one IP PBX hardware box with an open network of specialized tools working together. Rather than overloading one modular with every task, physical phone lines, network security, call routing, and features like voicemail are divided across separate devices and servers. These modulars setup keeps your central system fast, prevents single points of failure, and allows you to expand or upgrade individual components easily without replacing the entire setup.

Disaggregated VoIP Architecture Features

- **Decoupled Call Processing & Routing:** The core engine (Smart Call Manager) focuses strictly on dialplans, extension state management, and call control logic, free from the hardware interrupt overhead of physical line cards.
- **Distributed Analog Termination**: Physical POTS lines and analog phone sets are terminated at the edge via independent multi-port FXO/FXS VoIP gateways connected over standard SIP/RTP data networks.
- **Centralized Security Edge (SBC):** A dedicated Session Border Controller handles perimeter defense, NAT traversal, SIP normalization, and cryptographic encryption (TLS/SRTP) for all incoming and outgoing trunks.
- **Modular Application Tier:** Voicemail, automated attendants (IVR), call recording, and CDR billing databases run as independent services or modular nodes that can scale separately from core call routing.

Scalability Dimensions

- **Independent Resource Scaling:** Because functions are disaggregated, components scale based on specific bottlenecks rather than upgrading an entire appliance.
- **Signaling & Call Control:** Scale call routing nodes horizontally using SIP proxies to handle thousands of call setups per second.
- **Media & Transcoding:** Scale media relays (e.g., RTP engine) independently to absorb heavy audio processing and conference bridge without burdening core signaling.
- **Analog Density:** Expand analog capacity seamlessly by plugging in additional VoIP FXO/FXS gateways without altering Call Manager CPU or core memory footprints.
- **Distributed Branch Deployments:** Seamlessly connect headquarters, regional offices, and remote facilities under a single unified numbering plan and shared global feature set.
- **Local Survivability (Lifeline Routing):** Remote branch gateways can operate autonomously if the Wide Area Network (WAN) link drops, ensuring internal branch extensions and local analog lifeline (PSTN) emergency calls remain active.

Core Benefits

- **Supported Protocol:** supported voice protocols (SIP RFC3261, IAX, H323), DTMF (RFC4733, SIPINFO, In-Band), Transparent Protocols (UDP, TCP, TLS, SRTP).
- **Fault Isolation & High Availability:** If a power surge or lightning strike damages an analog FXO gateway port, the failure is strictly contained to that peripheral device, leaving your central call manager, core database, and enterprise routing completely safe.
- **Simplified Maintenance:** System Administrators can patch, upgrade, or replace individual components (such as swapping a voip gateway or upgrading the SBC) without causing total system downtime or risking core dialplan-configurations.
- **Unified Central Management:** Administrators manage global extensions, security policies, and dialplans from a single centralized management panel.
- **Modular L2/L3 Routing Engine** delivers significant operational and performance benefits, L2 Virtual LANs & Trucking (802.1Q), L3 Static Routing, Network Address Translation (NAT), Integrated DHCP Server & Relay (Automated IP distribution with customizable lease pools, gateway options, and option 66/150 provisioning pointers for IP phones).

Why Businesses are Switching to EDGE Call Manager

1. Cost Efficiency: Traditional PBX systems have expensive maintenance and telecom dependencies.
2. Remote Work Enablement: EDGE Call Manager lets employees use the same extension. They can access it on desk phones, laptops, or mobile devices.
3. Multi-Branch Connectivity: all communications operate under one unified system.
4. Scalability: Adding new users requires configuration—not rewiring.



EDGE Call Manager (ECM) Features

Call Manager Services	
Voice Mail & Visual MWI	Records audio messages when extensions are busy/unanswered, supports voicemail-to-email routing, and triggers Message Waiting Indicators (MWI).
Group Call / Hunt Group	Rings multiple extensions simultaneously or sequentially (round-robin, linear, ring all) to ensure incoming calls to a department are answered.
Call Forward - Busy / No Answer / All	Forwards incoming calls conditionally based on extension state (unconditional/all, on busy, or on no-answer) to external or internal targets.
Call Transfer / Call Waiting	Supports both blind and attended/supervised call transfers, while maintaining secondary line call-waiting alerts for active endpoints.
Caller ID / Call Pickup	Transmits and displays caller identification (Name/Number) and allows users to answer ringing calls on another extension within their assigned pickup group.
Call Hold / Call Park	Temporarily place active calls on local hold with audio stream suspension or park them into virtual parking slots (e.g., ext: 701) for network-wide retrieval.
Do Not Disturb / Mute / Music on Hold	Toggles extension DND state to bypass ringing directly to voicemail, supports endpoint audio muting, and streams custom audio/chimes to held callers.
Call Queuing	Distributes incoming calls to call center agents based on strategies, featuring hold queue position announcements and estimated wait times.
Conference Bridging	Multi-party audio conferencing rooms feature PIN protection, entry/exit chime announcements, active speaker detection, and admin mute controls.
Direct Inward System Access (DISA)	Allows external callers to dial into the PBX, authenticate with a PIN code, and obtain internal dial tone to place outbound calls through system trunks.
Unlimited Auto-Attendant / IVR Menu	Unlimited multi-level automated attendant menu system that prompts callers with audio options and routes calls dynamically based on DTMF inputs.
Flexible Time-Based Call Routing	Automatically routes incoming calls based on working hours, weekends, holidays, or specific time schedules (routing to IVR during work hours, voicemail after hours).
Find Me / Follow Me	Dynamically routes incoming calls to a sequential sequence or simultaneous ring list of internal extensions and mobile numbers until answered.
SIP Trunk	Connects Smart Call Manager to external ITSP providers, VoIP gateways, or secondary PBXs over SIP/TLS for inbound and outbound call routing.
Call Detail Records (CDR)	Captures metadata for every call (start time, source, destination, duration, billsec, disposition) logged directly to SQL databases to retrieve calls reports.
Networking	Static IP, Dynamic IP, 802.1q VLAN and tagging, IEEE 802.1Q VLAN trunk, inter-VLAN routing, traffic management, DHCP Server/relay, Static Routing, Dynamic NAT, IP Access Control Lists (IP ACLs), MAC Access Control Lists (MAC ACLs).
Audio/Video Codec Support	ulaw, alaw, gsm, slin, ilbc, g.722, g.723 and g.729 A/B pass through. H.263, H.263+ and H.264



Flexible Deployment

ON-PREMISES ECM

The entire VoIP server and voice switching hardware are physically installed and hosted directly inside your local office or private data center.

Complete Authority & Customization: Provides total control over your system configuration, local SIP trunks, security policies, and call data privacy.

Zero Dependency on Cloud Connectivity: Internal office-to-office calling remains operational even during internet service provider outages.

One-Time Capital Ownership: Eliminates recurring per-user monthly hosting fees, offering a lower total cost of ownership (TCO) over a long lifecycle.

Best For: Enterprises with IT teams, strict compliance/regulatory needs, or existing local telecom lines (FXO/PRI/ISDN).

HOSTED ECM

Voice services and call routing are fully managed and maintained off-site by the service provider over a secure cloud infrastructure.

Minimal Upfront Investment: Eliminates expensive PBX hardware purchases—requiring only IP phones or softphones on mobile and desktop devices.

Predictable Operational Expenses: Utilizes a simple, scalable subscription model where you pay only for the extensions and features you use.

Hassle-Free Maintenance: Automatic updates, system security, hardware maintenance, and uptime guarantees are fully handled by the provider.

Best For: Small-to-medium enterprises (SMEs), startups, and organizations with remote or distributed workforces looking for low setup costs and rapid deployment.

HYBRID ECM

A flexible, dual-architecture deployment that bridges local on-premises hardware with cloud-hosted voice features and remote connectivity.

Seamless Business Continuity: Combines the local survival capabilities of an on-premises system with cloud redundancy to ensure zero downtime.

Scalable Connectivity: Easily connect remote branches or work-from-home staff via the cloud while keeping central office communications on local infrastructure.

Migration Path: Allows organizations to preserve existing hardware investments while gradually transitioning legacy telecom setups to the cloud.

Best For: Mid-sized businesses, multi-branch organizations, or companies transitioning from traditional on-premises telephony toward full cloud adoption.

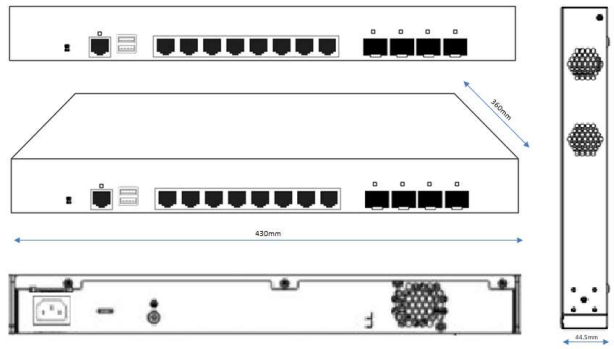
Appliance Call Manager

A dedicated 1U/2U on-premises hardware appliance for enterprise voice management, offering full physical control over call data. Pre-loaded with PBX software, the appliance features integrated physical interfaces for PSTN lines, analog desk phones, SIP trunk gateways. That provides enterprises with complete physical control and security over their voice infrastructure and call data.

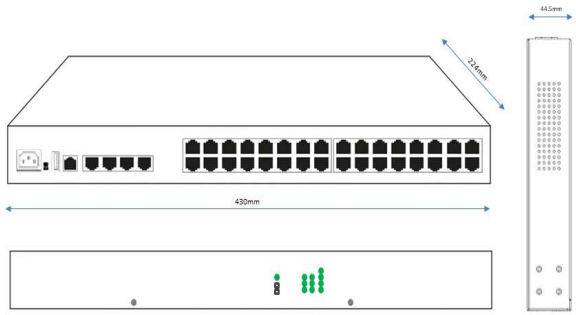
Virtual Call Manager

A flexible, software-based PBX solution built for virtual machines and cloud servers without dedicated hardware. Built on the exact same PBX codebase as physical EDGE appliances, it eliminates dedicated hardware requirements while delivering full enterprise-grade telephony capabilities—including intelligent call routing, voicemail, auto-attendants, and SIP trunk—over an IP connection.

EDGE Call Manager Dimensions



EDGE Voice IAD Dimensions



EDGE Call Manager Hardware Specification	
Processor	Supports Intel Gen6/Gen7 Xeon E3-1200 i3/i5/i7 Core, Pentium processor
Chipset	Intel C236
System memory	Two 2133/2400MHz DDR4 slots, up to 64GB
Ports	8x 10/100/1000Mbps RJ-45 ports
NIC Modules	2x 10G SFP LAN ports NIC module Lan card based on Intel 82599ES controllers 4x 1G SFP LAN ports NIC module Lan card based on Intel 82599ES controllers
Console Port	One RJ-45 COM
USB Ports	Two USB3.0
Indicator	One Power LED, one HDD LED
Power consumption	Built-in 250W power supply
Power input	100 ~ 240V AC input (DC power or Redundant power)
Mounting	Supports rack-mounting or desktop-mounting
Operate Temperature	-10°C ~ 50°C (5% ~ 85% non-condensing @ 40°C)
Storage Temperature	-20°C ~ 75°C (5% ~ 95% non-condensing)
Construction	1U chassis, Heavy duty metal
Dimensions (D x W x H)	360x430x44.5mm
weight	11.6 kg

EDGE Voice Gateway Specifications

EDGE Voice Gateway Specification

Hardware Specification	FXS interface:4/8/16/32 (RJ11) Console interface: 1*RJ45, 115200bps USB interface: 1*USB 2.0, Full speed Network interface: 1 WAN, 3 LAN full Gigabit Ethernet ports (1000 Base-T Ethernet) Size: 440*224*43.5mm Input voltage: 100VAC-240VAC; 50/60Hz Power: 45W Weight: <2.2KG Chassis: 1U, metal material Working environment: temperature -5°C-55°C, environmental relative humidity≦95%, no performance degradation within 3000m above sea level; atmospheric pressure 86KPa-106KPa airborne particle concentration≦180mg/M3
Software specifications	Support static IP, DHCP, PPPoE. Support WAN sub-interface Support DHCP server, NAT, DNS, IGMP, VLAN. Support static routing, dynamic routing, policy routing Support PPTP, L2TP, IPSec SSL, VPN Support Switch VLAN, Trunk Port, support NTP Client Support Bandwidth Control, QoS Policy Support Security Firewall, APR Defense, DDoS protection
Voice services	basic calling, calling number display, call forwarding, three-way calling, finding malicious calls, outgoing restrictions, do not disturb, abbreviated dialing, hotline service, alarm clock, busy callback, conference call, support IMS SIP-based voice services, NGN SIP voice services, voice services based on the H.248 protocol
Voice codec	G.711U, G.711A, G.723, G.729
Voice quality assurance	support echo cancellation, mute detection, mute compression, comfortable background sound generation, support jitter buffer, support voice frame compensation function, support gain adjustment Log function: support local log record, log upload to log server
Local management	Support local HTTP web management, remote HTTS web management SSH console interface, support gateway configuration file import and export, support gateway configuration file import and export, support device reset, software upgrade, system debugging, support fault detection, equipment Status monitoring
Remote management	remote upgrade function, SNMP network management protocol, TR069 network management protocol, SYSLOG remote network management protocol.

EDGE Voice Gateway

EDGE Voice Gateway 4 FXS	EDGE Voice Gateway 8 FXS	EDGE Voice Gateway 16 FXS	EDGE Voice Gateway 32 FXS
			

LANEX

LANEX is a leader in cloud-native networking software and network disaggregation solutions. Powered by TeleSources Solutions, founded in 2015 based in London. For more information, visit us at www.lanex.uk