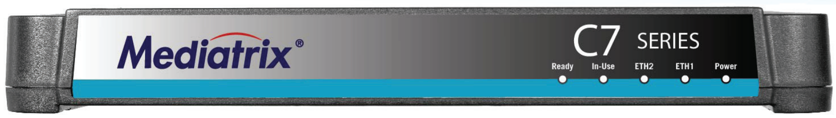
Mediatrix® C7 Series

The Mediatrix C7 Series are multi-function devices combining VoIP Analog Adapter, Gateway and QoS control in a secure and powerful platform.



This platform, featuring FXS/FXO interfaces, provides an ideal solution for enterprise voice applications or for connecting to a service provider's broadband access.

With FXO interfaces, it is the ideal solution to deploy private or hosted toll bypass networks. It provides a simple, transparent and cost-effective way of maintaining a connection to the PSTN.

The Mediatrix C7 Series also allows Enterprises, Service Providers, and System Integrators to deploy secure systems and generate additional revenue streams.

Key Benefits

Voice Functionalities
- Carrier-grade voice quality
- Analog FXS/FXO interface ports
- Fax over IP support, including T.38
- Multiple codec support
- Up to 8 simultaneous calls
- Battery Reversal for pay phones

Ease of configuration and management
- Automatic firmware and configuration file download
- SNMP and web management
- TFTP or HTTP auto-provisioning
- TR-069 for massive deployments (optional feature available at purchase time)

Security
- Support for SNMPv3
- Encrypted configuration files support
- HTTP Digest authentication
- Secured SIP signalling and media transmission

Network functionalities
- QoS features support
- DHCP client
- STUN Client
- Complete VoIP-enabled IP router
- 2 Ethernet ports
- Support for IPv6 Internet Protocol

Mediatrix C7 Series Overview

The Mediatrix C7 enables cost-effective VoIP deployments in small offices for both IP Centrex and private network applications.

The Mediatrix C7 has the additional benefit of supporting high compression codecs simultaneously on each voice ports, thus saving valuable bandwidth.

The Mediatrix C7 Series gateways make use of existing broadband access equipment to connect to any standards-based VoIP network.

The Mediatrix C7 offers security features such as TLS, SRTP, certificates management, and HTTPS designed to bring enhanced security for the network management, SIP signalling and media transmission aspects.

The Mediatrix C7 series is targeting:
- Enterprises with branch offices and remote offices, including SOHOs.
- Enterprises seeking access to the benefits of Voice over IP without discarding existing legacy equipment.
- Enterprises wishing to implement Voice over IP through a gradual, transparent, and cost-effective migration.
- Multi-tenant / multi-dwelling units and premises wishing to implement Voice over IP while maintaining existing wiring and handsets.
- Enterprises looking to implement secure RTP (SRTP) and Secure SIP (SIP over TLS).

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Applications

The Mediatrix C7 VoIP gateways allow any office to use an existing IP network for lower-cost voice communications.

The Mediatrix units link any standard FXS/FXO connection to the IP network and deliver the clarity of toll quality voice for a comprehensive VoIP solution.

T.38 FoIP, fax bypass, and modem bypass capabilities ensure that the Mediatrix C7 gateways seamlessly transport voice and data services. The Mediatrix C7 gateways offer flexibility and scalability for VoIP network integration and low bandwidth voice.

With configurable FXS and FXO ports, call-switching, and user-programmable call routing (including caller/called ID), Mediatrix C7 gateways integrate smoothly into existing PBX and PSTN networks. The Mediatrix C7 series provides a gateway to the PSTN for IP-based PBX and Key Systems. Thereby, it allows the deployment of VoIP Remote Line Extension and Branch Office Connectivity solutions without sacrificing any local PSTN access points.

By connecting CO lines from a selected site to VoIP networks, the Mediatrix C7 series enable service providers and enterprises to use VoIP connections between pre-determined local networks.

Variants

The Mediatrix C7 platform is available in two variants:

- Base model for the FXS and FXO gateway products.
- PBX optional model for the iPBX or proxy product types with FXS and FXO ports.

All variants offer the following:

- 2 x Fast-Ethernet ports
- External Power-supply
- 5 x LEDs (Power, Ready, In-Use, Eth1, Eth2)
- 1 x reset button
- DSP supporting a minimum of 8 bi-directional channels of encrypted and compressed media.

Optional for the PBX variant: External USB port for storage and extended solid state storage.

Mediatrix C7 Hardware Models

The following Mediatrix C7 hardware models are available:

Model	Telephony Ports
C710	4 FXS
C711	8 FXS
C730	4 FXO
C733	8 FXO
C731	4 FXS + 4 FXO

Technical Specifications

IP Telephony Protocol

- SIP – RFC3261
- Multiple Virtual Gateways
- Multiple SIP Proxy support via DNS SRV
- Dual-Stack IPv4/IPv6 support using ANAT (RFC 4091)
- OCSP (Online Certificate Status Protocol) revocation status verification for TLS links
- ANAT grouping in the SDP
- Media stream can be on a different network than SIP signalling
- Call router automatic routes can have multiple SIP gateways for sources and destinations

Voice Processing

- Vocoders: G.711 (A-law, µ-law), G.723.1, G.726, G.729a/b
- G.168 echo cancellation (64 ms)
- DTMF detection and generation
- Carrier tone detection and generation
- Silence detection / suppression and Comfort Noise Generation level software adjustable
- Configurable de-jitter buffer
- Configurable tones (dial, ringing, busy)
- Configurable transmit packet length
- RTP/RTCP - RFCs 1889 / 1890 / 2833 / 3389

Enhanced Security

- HTTPS, for web pages and for exchange of Configuration File.
- SRTP with MIKEY and SDES: Supported Cypher, AES – 128 bits.
- MIKEY key management protocol (RFC 3830 and 4567).
- SDES key management protocol (RFC 4568).
- X.509 Certificate management.
- TLS transport method: Supported Key Exchange Mechanism: RSA, Diffie-Hellman; Supported Cyphers (minimum): AES (128 and 256 bits), 3DES (168 bits).
- User Access Levels for units Management Observer, User and Admin user access rights, with the inclusion of flexible policies on the user password configuration.

Fax and Modem Support

- Group 3/Super G3 Fax real-time FoIP over clear channel (G.711), G.726 or T.38
- T.38 Fax relay (9.6 k, 14.4 k)
- G.711 Fax and Modem Bypass
- T.38, fax tone detection and pass-through on G.711 and G.726

Management

- Web-based GUI
- TFTP, HTTP configuration up- and download (Autoprovisioning)
- TFTP, HTTP firmware upgrade
- SNMPv1/v2/v3 agent (MIB II and private MIB)
- TR-069 for massive deployments (optional feature available at purchase time)
- Remote Services activation to purchase service licenses for deployed units, permitting to enable TR-069 feature in the field
- TR-104 to permit large provisioning systems to communicate with CPE on private networks
- Subscribes services for IMS: Transfer methods and real time media provisioning

Call Routing

- Local switching
- Interface hunt groups
- Routing Criteria; Interface, Calling/called party number, Time of day,

day of week, date
- Number manipulation functions; Replace numbers, Add/remove, digits, Multiple remote gateways, PLAR
- Call properties manipulations
- SIP header manipulations
- SIP Fallback on remote destinations - Redirect SIP calls to remote SIP Endpoints using the Call Router with SIP Redirect 302 Moved Temp messages.
- Support of field Remote Party ID.
- Call Router enhancement to support Privacy Header Remote-Party-ID.

Voice Signalling
- Session Description Protocol (SDP)
- Compliant with MMTEL requirements for 3GPP specification 24.615 for Call Waiting requests

QoS Marking
- TOS/DiffServ
- IEEE 802.1p/Q

Network Connection
- 2 x 10/100 BaseT Ethernet RJ-45 connectors

Enhanced Telephony Features
- Call Forward / Call Transfer / Conference Call / Call Waiting support
- Inter-digit timer and IP dialing
- Echo Cancellation / Dynamic Jitter Buffer / Voice Activity Detection / Silence Suppression
- Message Waiting Indication, via FSK
- Flash hook event signaling
- Caller ID Generation (Name & Number) as per Bellcore DTMF or FSK
- Call Completion (CCNR / CCBS)
- PRACK & UPDATE

Operating Environment
- Operating temperature: 0ºC to 40ºC
- Storage temperature: -20ºC to 70ºC
- Humidity: up to 85%, non-condensing

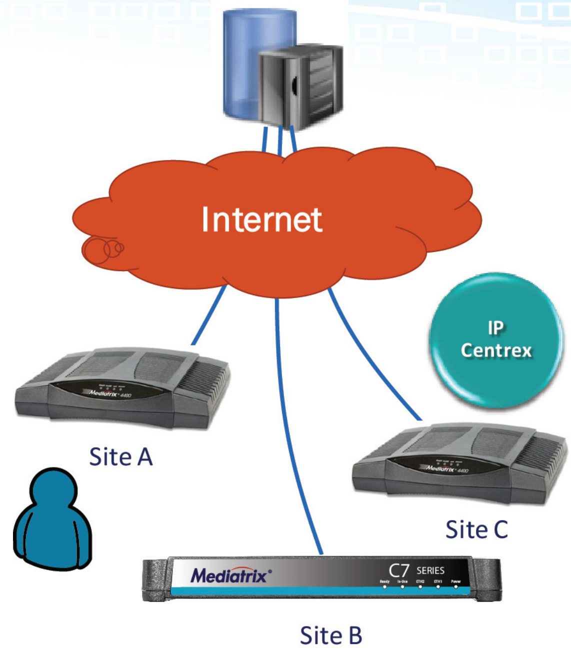
Power Supply
- External 100-240 VAC power supply.
- DC 12Vdc +/- 5% 1.5A (18W)

Dimensions
- 4.4 cm x 28.3 cm x 16.2 cm (1.73 in. x 11.1 in. x 6.4 in)

Mediatrix SDK (Software Development Kit)
- Enables developers and content authors to create rich, integrated VoIP applications for their specific requirements
- Available for download free of charge at: http://mediatrixsdk.media5corp.com

Solutions

Service Provider



Local Carriers, MSOs, MVNOs

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