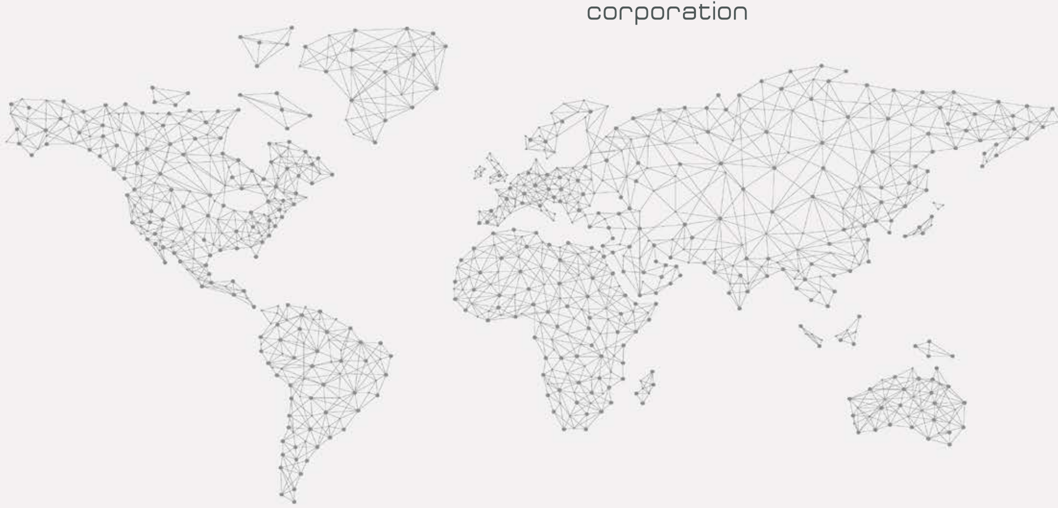

Mediatrix® G7 Series

The Mediatrix G7 Series is a reliable and secure VoIP Analog Adaptor and Media Gateway platform for SMBs. Featuring PRI, FXS, and FXO interfaces; the Mediatrix G7 Series provides the best solution to connect legacy equipment to cloud telephony services and IP PBX systems to PSTN landlines.

Widely interoperable with SIP softswitch and IMS vendors, the Mediatrix G7 Series provides transparent integration of legacy PBX systems for SIP Trunking and PSTN replacement applications.

Interconnects any device to SIP

The Mediatrix G7 Series links any analog or digital connection to an IP network and delivers a rich feature set for a comprehensive VoIP solution.

PSTN access and legacy PBX system gateway

With FXS, FXO, configurable NT/TE PRI ports, local call switching, and user-defined call properties (including caller/calling ID), Mediatrix G7 Series gateways smoothly integrate into legacy PBXs and incumbent PSTN networks.

Highly reliable Fax and Modem Transmissions over IP

With T.38 and clear channel fax and modem pass-through capabilities, the Mediatrix G7 Series ensures seamless transport of voice and data services over IP networks.

Advanced Mass Management

Our advanced provisioning capabilities deliver remarkable benefits to Mediatrix customers. Mediatrix enables centralised CPE management, a definite advantage to monitor the network, ensure service, and reduce operational costs.

Applications

Operators

- ✓ Connect legacy equipment in PSTN replacement/TDM replacement projects
- ✓ Provide SIP termination for cloud telephony services
- ✓ Convert ISDN signaling to SIP for SIP trunking
- ✓ Convert analog signaling to SIP for Hosted Unified Communications and IP-Centrex

System Integrators

- ✓ Integrate Unified Communications with legacy systems
- ✓ Connect Skype for Business with IP/TDM trunks and legacy telephony equipment
- ✓ Keep existing telephony equipment in SIP migrations
- ✓ Inter-connect branch offices to headquarters
- ✓ Survivability for branch offices in case of WAN failure

Key Features

Carrier-Grade Voice Quality

T.38 and clear channel fax over IP
High performance processing of up to 120 voice channels
Survivability for IP-Phones in Hosted UC/PBX deployments
Battery reversal for pay phones

Robust Security

Encrypted media, signaling, and management

Deep packet inspection firewall with DoS protection

Easy Configuration and Management

Zero-touch configuration

Intuitive Web GUI

Customisable factory settings

Networking

Dual-stack IPv6 and IPv4

Multiple IP addresses and VLANs

NAT, firewall, and router capabilities

Benefits

- ✓ High quality built and carrier-grade validation standards contribute to the industry’s most reliable platform.
- ✓ Extensive TR-069 support for an easy management of large-scale deployments with a centralised EMS.
- ✓ Superior call routing and manipulation allow greater flexibility in the implementation of complex deployment scenarios.

Technical Specifications

Media Processing
G.711 (A-law and µ-law), G.726, and G.729a/b;
G.168 echo cancellation
DTMF detection and generation
Carrier tone detection and generation
Silence detection/suppression and comfort noise
Configurable de-jitter buffer and packet length

Enhanced Security
Denial of Service (DoS) protection
SIP over TLS
SRTP with AES cipher – 128 bits
MIKEY key management protocol (RFC 3830 and 4567)
SDES key management protocol (RFC 4568)
TLS-encrypted configuration and management
X.509 certificate management
OCSP (Online Certificate Status Protocol) revocation status verification
Supported TLS key exchange mechanism:
• RSA
• Diffie-Hellman
Supported TLS ciphers (minimum):
• AES (128 and 256 bits)
• 3DES (168 bits)

Management
Zero-touch provisioning
TR-069, TR-104, and TR-111
Web GUI
SSH and TELNET
SMNP v1, v2c, and v3
Scripts/firmware files uploaded via HTTP, HTTPS, FTP, and TFTP
Multiple levels of management access rights
Customisable CDR
Event notifications via Syslog, SIP, log file, and SNMP traps
Remote activation of service licenses

Monitoring and Troubleshooting
Alarms and traps
Call Details Record (CDR)
Media quality statistics
System: CPU and memory usage
PCM capture
IP network capture
Diagnostic traces

Quality of Service (QoS)
Bandwidth limitation and traffic shaping
TOS/DiffServ
IEEE 802.1p/Q
RTCP-XR – special order

IP Telephony Protocol
SIP (RFC 3261) over UDP, TCP, and TLS
IMS (3GPP TS 24.229)
RTP (RFC 3550)
SDP (RFC 4566)
Multi-part body support
Redundancy support via DNS SRV
Multiple trunk support
Survivability for IP-Phones
IPv4 and IPv6 dual stack signaling and media

Digital Telephony
Euro ISDN EDSS-1/ETSI PRI/NET5
ISDN NI-2 (US T1 PRI)
ISDN DMS100 (US T1 PRI)
ISDN 5ESS (US T1 PRI)
ISDN speech, audio, and data (Fax Gr 4, UDI 64, and RDI 64)
ECMA-143 (QSIG-BC)
E1 R2 digital line signaling (ITU-T Q.421)
E1 R2 MFC interregister signaling (ITU-T Q.441)
Presets for: Brazil, Argentina, Mexico, Saudi Arabia, Venezuela, Philipines, and ITU-T
T1/E1 E&M (Immediate, Wink-Start, Feature Group-B, and Feature Group-D), MF-R1, DTMF
Advice of Charge AOC-D, AOC-E (ETS 300 182)

Analog Telephony
Support for call forward, call transfer, conference call, call waiting, CCNR, and CCBS
Multiple country presets
Customisable tones and ring patterns
Echo cancellation
Message Waiting Indication (MWI), via FSK
Caller ID detection (name & number) as per Bell-core FSK
On-hook/off-hook caller ID generation (name & number) as per Bell-core DTMF or FSK and Telebras BINA
Answer and disconnect signaling

Call Routing
Local switching
Call filtering and blocking
Calling/called number manipulation using regular expressions
Routing Criteria:
• Interface
• Calling/called party number
• Calling/called URI
• Time of day, day of week, and date
• Many others
Mapping and transformation of call properties to/from SIP headers
Hunt groups

Fax and Modem Support
Group 3/super G3 fax real-time fax over IP
T.38 fax relay (9.6 k and 14.4 k)
Clear channel (G.711) fax and modem pass-through

Networking

IPv4 – IPv6
Multiple IP addresses per link or VLAN
Multiple VLANs per link
DHCP client
PPPoE (RFC 2516)
IEEE 802.1q + DSCP QoS tagging (media, signaling, and mgmt)
IEEE 802.1x wired authentication
LLDP-med (ANSI/TIA-1057)
QoS traffic shaping
Firewall with stateful inspection, rate-limitation, and automatic black-listing
Static routing
NAPT
DHCP Server

Dimensions

Height: 4.4 cm
Width (mounting brackets): 48.5 cm
Depth: 19.5 cm
Weight: 3Kg

Physical Interfaces

5 x 10/100/1000 Base-T Ethernet RJ-45 connectors
2 x TDM sync RJ-45 connectors

Power Supply

Internal 100-240 VAC power supply

Operating Environment

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

Digital Ports	Up to 4 E1/T1
Analog Ports	Up to 24 FXS Up to 24 FXO
Mounting	Rack
Network	5 x 10/100/1000 Base-T
Survivability	✓

This datasheet applies to model: M.

A Trusted Partner

Media5 Corporation is a global supplier of multimedia communication solutions, offering a complete set of SIP-based products and technologies.

With a focus on innovation and excellence in customer support, we deliver highly adaptive hardware and software components as well as ready-to-market SoftClients. This allows our customers and partners to take advantage of secure, reliable, and comprehensive communication solutions.

Mediatrix access devices include a complete set of VoIP Adaptors, Media Gateways, and Session Border Controllers customer premise equipment to connect any network to cloud telephony services.

For additional information, contact your Media5 representative.

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