Mediatrix® 4102 / 4102S

The Mediatrix® 4102 is a Security-Ready, VoIP gateway allowing Service Providers and Enterprise Networks to connect SOHOs, Remote Workers and Branch Offices to an IP network, while preserving investment in analog telephones and faxes.

Key Benefits

New Package for VoIP FXS Interfaces
- Hardware ready to support Security (SIP over TLS, SRTP, MIKEY)
- Secured SIP signaling and media transmission
- Replacement of Mediatrix 1102/2102

Best Total Cost of Ownership
- Ease of deployment & management with autoprovisioning
- Protect analog telephony investments with the VoIP-benefits

Best Price Quality Ratio
- High Voice Quality and Reliability
- Industry-proven T.38 fax
- Wide support of countries

Network functionalities
- QoS features support
- DHCP client
- STUN Client
- Support for IPv6 Internet Protocol (Dgw v2.0 application)

Models
- Mediatrix 4102: For non-secure applications.
- Mediatrix 4102S: For security-enabled applications.

Mediatrix 4102 Overview

The Mediatrix 4102 connects up to two analog phones and/or faxes, as well as a PC or a home router to a broadband modem.

The Mediatrix 4102 offers security features such as SIP over TLS, SRTP, certificates management, and HTTPS designed to bring enhanced security for the network management, SIP signalling and media transmission aspects. It interfaces seamlessly with the full Mediatrix portfolio of products in secure networks.

The Mediatrix 4102 also uses its innovative TAS (Transparent IP Address Sharing) technology and an embedded PPPoE client to allow the PC (or router) connected to the second Ethernet port to have the same public IP address, eliminating the need for private IP addresses or address translations. The 4102 also supports high compression codecs simultaneously on both analog voice ports, saving valuable bandwidth.

As with all Mediatrix devices, the 4102 provides a web interface, giving users convenient access to the unit for initial set-up. The devices can also auto-provision by fetching their encrypted configuration from a TFTP or HTTP server making installation secure and transparent to the end-users. To further facilitate deployments, factory loaded configurations are possible.

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IP Telephony Protocol
- SIP – RFC3261
- MGCP/NCS – RFC 3435.
- Multiple Virtual Gateways
- Multiple SIP Proxy support via DNS SRV
- Dual-Stack IPv4/IPv6 support using ANAT (RFC 4091)
- FIPS on the Mediatrix 4102S
- 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.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

Network Management Protocols
- SNMPv3, DHCP – RFC2131, RFC2132, TFTP – RFC1350, RFC2347, RFC2348, RFC2349, Syslog – RFC3164, HTTP 1.0 – RFC1945, HTTP 1.1 – RFC2616, Basic and digest HTTP authentication – RFC2617

Management
- Web-based GUI (SIP only)
- 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.

Data Features
- PPPoE client – RFC1332, RFC1661, RFC1334, RFC1994, RFC2516, RFC1471, RFC1472, RFC1473, RFC1877
- DHCP server (planned)
- STUN client

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

Analog Connection
- 2 x RJ-11 connectors, analog phone/fax (FXS) interface

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

QoS
- TOS/DiffSery
- IEEE 802.1p/Q

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 45ºC
- Storage temperature: -40ºC to 85ºC
- Humidity: up to 85%, non-condensing

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

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