

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

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.

Technical Specifications

IP Telephony Protocol

- SIP – RFC 3261
- MGCP/NCS – RFC 3435

Voice Processing

- Vcoders: 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 interface access.
- SRTP with MIKEY as key selection method.
 - Supported Cypher
 - AES – 128 bits
- MIKEY protocol using pre-shared keys (RFC 3830 and 4567) for negotiating SRTP keys.
- Certificate management.
- TLS transport method.
 - Supported Key Exchange Mechanism:
 - RSA\
 - Diffie-Hellman
 - Supported Cyphers (minimum):
 - AES (128 and 256 bits)
 - 3DES (168 bits)

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

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 (Auto-provisioning)
- TFTP, HTTP firmware upgrade
- SNMPv1/v2/v3 agent (MIB II and private MIB)

Data Features

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

Ethernet Connection

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

Analog Connection

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

QoS

- TOS/DiffServ
- IEEE 802.1p/Q

Operating Environment

- Operating temperature: 0°C to 45°C
- Storage temperature: -40°C to 85°C
- Humidity: up to 85%, non-condensing

Enhanced Telephony Features

- Multiple SIP Proxy support via DNS SRV
- Call Forward / Call Transfer / Conference Call / Call Waiting support
- T.38, fax tone detection and pass-through on G.711 and G.726
- 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



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