The Mediatrix® 2102 is a high-quality and cost efficient VoIP gateway connecting SOHOs to an IP network, while preserving investment in analog telephones and faxes.

It allows Services Providers to deploy rapidly and economically their solutions in smaller premises and it is the ideal solution for remote line connection to larger private networks.

Key Benefits

Voice Functionalities

- Carrier-grade voice quality
- T.38 support
- High compression Codecs support
- PSTN bypass option available for emergency calls

Ease of configuration

- Automatic firmware and configuration file download
- SNMP and web management
- TFTP, HTTP or HTTPS auto-provisioning

Security

- Support for SNMPv3
- Encrypted configuration files support
- HTTP Digest authentication
- HTTPS support

Network functionalities

- Transparent IP Address Sharing
- PPPoE and DHCP client
- STUN Support
- Interoperable with equipment from leading industry vendors

Mediatrix® 2102
1-port and 2-port VoIP Access Device

Mediatrix 2102 Overview

The Mediatrix 2102 connects up to two analog phones and/or faxes, as well as a PC or a home router to a broadband modem.

The Mediatrix 2102 enables cost-effective VoIP deployments in residential and SOHO applications.

With an embedded PPPoE client and its innovative Transparent IP Address Sharing technology, the Mediatrix 2102 and the PC or router connected to the second Ethernet port have the same public IP address, eliminating the need for private IP addresses or address translations.

The Mediatrix 2102 has the additional benefit of supporting high compression codecs simultaneously on both analog voice ports, thus saving valuable bandwidth.

As all other Mediatrix devices, the 2102 provides web interface, giving users a convenient access to the unit for initial set-up. The devices can also auto-provision by fetching their encrypted configuration from a TFTP, HTTP or HTTPS server making installation secure and transparent to the end-users. To further facilitate deployments, factory loaded configurations are possible.

In addition, an optional intelligent PSTN bypass allows Mediatrix 2102 users to make emergency calls and maintain their phone service in the event of a power outage or network failure.

Sales Department
Americas **- Tel: 1-877-GET-VOIP / (877) 438-8647 Tel: (514) 285-0058 Fax: (514) 842-0125**
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CALA – Tel: **(954) 349-0394 Fax: (954) 349-8722**

Technical Specifications

IP Telephony Protocol
- SIP – RFC3261

Vocoders
- G.711 (a-law, µ-law) with optional VAD support, G.723.1, G.726, G.729a, ab

Echo Cancellation
- G.168

Silence Suppression
- Silence detection / suppression and Comfort Noise Generation level software adjustable.

Real-Time Transport Protocols
- RTP/RTCP – RFC1889, RFC1890, RFC2833, RFC3389

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

Ethernet Connection
- 2 10/100 Base T Ethernet Rj-45 connector

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

Bypass Connection
- 1 RJ-11 connector, PSTN bypass (opt.)

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

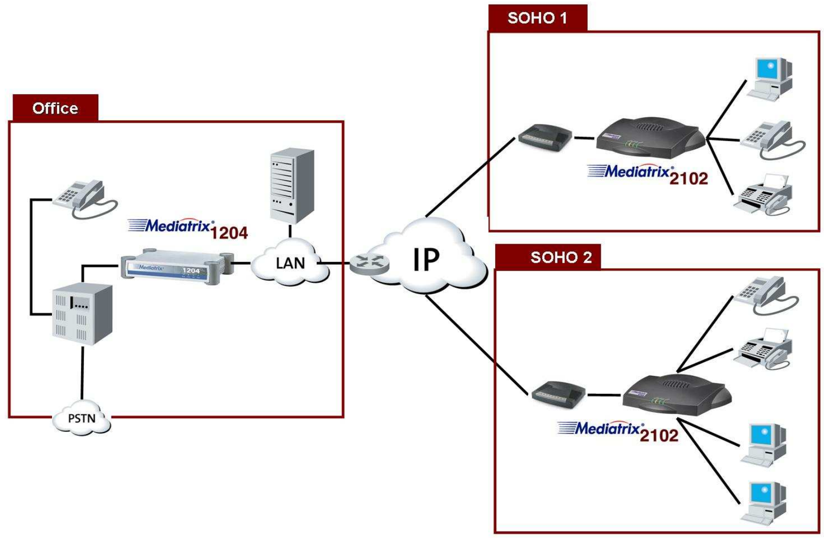
QoS
- ToS, DiffServ, 802.1p, 802.1Q

Unit Dimensions and Weight
- Height 5cm (2in)
- Width 20cm (8in)
- Depth 14 cm (5.5 in)
- Unit Weight 454g (1.2lbs)

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

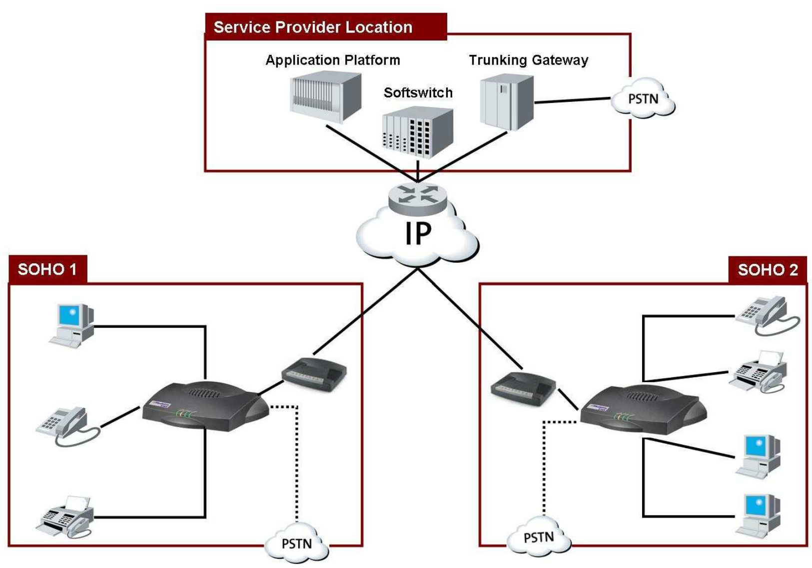
Applications

Remote Line Extension



Value Proposition
- Extend the reach of the IP PBX Analog Line Extensions to remote SOHO via IP network
- Connect your remote locations through VoIP while maintaining your PBX/KTS infrastructure
- Maintain most analog call features from via Hook Flash Relay and DTMF signalling
- Maintain 911 emergency capability

Hosted SOHO Applications



Value Proposition
- Deploy carrier-grade voice solutions
- Benefit from superior ease of configuration and remote management
- Take advantage from thorough interoperability with all major softswitches on the market

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