# ApplianX IP Gateway

The ApplianX IP Gateway can be used to realise many VoIP and TDM migration strategies. Whether connecting a TDM-based PBX to a SIP-based call manager, providing a PSTN front end to SIP-based platforms, or extending the life of a DPNSS-based PBX by connecting to IP-PBXs, which typically support QSIG, it offers a cost-effective solution. It is a 'plug & play' gateway that can reduce operational costs and provide extensibility between legacy equipment and IP-based services and endpoints.


## Network interfaces

The ApplianX IP Gateway provides one, two or four universal T1/E1 (worldwide) interfaces, with a wide range of signalling protocols, including PRI/ISDN types, various CAS types, and PBX protocols, such as QSIG and DPNSS. A different protocol can be selected for each trunk and, for DPNSS-to-QSIG, gateways can offer support for 30 or 60 conversions. The ApplianX IP Gateway provides dual-redundant traffic interfaces for SIP signalling and RTP voice streams, with support for G.711 and G.729AB codecs.

## Comprehensive management

As a closed appliance, the ApplianX IP Gateway provides a private (non-traffic) Ethernet port that gives access to an integral HTTP server and provides an intuitive HTML Web browser interface, with separate password protected access levels, to allow configuration, administration, traffic monitoring and diagnosis. An easy to use set-up wizard ensures its ‘plug & play’ capability. A comprehensive set of SNMP facilities (including traps) enable remote management within a traditional network management environment. Additionally, a range of ApplianX tools is freely available to assist with management of the gateway. The ApplianX Search Tool is used to aid the installation process by finding ApplianX devices on the network, revealing their serial number, type and IP address. The ApplianX Trace Tool is an easy to interrogate tool that captures and decodes/displays protocol trace.

## Traffic routing

A comprehensive call routing engine controls traffic between IP and TDM, enabling configuration of many routing strategies: implementing and routing on trunk groups; support for multiple dial plans; and strategies for handling call progress information. For integration with SIP systems supporting redundancy, the gateway provides load balancing between endpoints on a round robin basis. In addition, the gateway supports a wide range of failover options, including survivable branch appliance capability (survivable remote site telephony), for resilience to various network failure scenarios.

## Features

- 'Plug & play' solution that can bridge between various TDM and IP systems
- Web browser configuration with set-up wizard
- Extensive portfolio of worldwide protocol support
- Comprehensive support for PBX inter-working protocols with conversion of supplementary services between SIP, DPNSS and QSIG
- Comprehensive call routing engine, for controlling traffic between IP and TDM
- SIP load balancing and fault tolerance
- Survivable branch appliance (survivable remote site telephony) capability
- DTMF suppression (clamping)

## Benefits

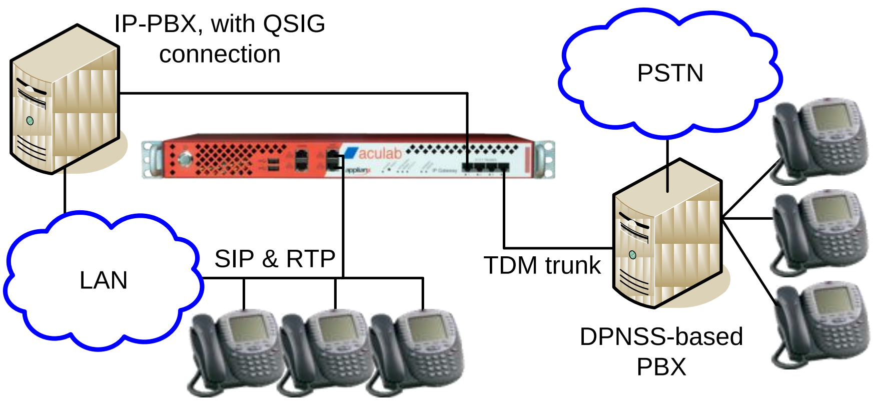
- Familiar interface tools; easy to install, configure, manage and maintain
- Global protocol support means gateways can connect to any TDM/PSTN network
- Protects investment and continued leverage of legacy equipment such as PBXs; users can readily gain advantage from new IP-based services and endpoints
- Offers complete control over routing strategies; SIP-to-TDM, or TDM-to-SIP
- Load balancing means calls can be routed away from unresponsive endpoints
- Calls can fallback to/from the TDM/PSTN when there is a problem with the IP network
- Enables PCI compliance for e.g., call recording

## Support for Supplementary Services

In addition to basic call control, a wide range of Supplementary Services can be converted between SIP, DPNSS and QSIG (see the Technical summary for details). All Supplementary Services are supported by default, however, should the ApplianX IP Gateway be connected to a PBX that does not support one or more of the Supplementary Services, users can readily disable the mappings for such functions, ensuring interoperability between all components in the network.

## Target applications

- Interfacing enterprise VoIP telecoms to the PSTN, legacy PBXs, or private networks
- Enabling a transitional, migration strategy from TDM to VoIP
- Allowing customers with TDM-based PBXs to take advantage of SIP trunks
- Interconnecting legacy TDM infrastructure with IP-based call centres
- Providing telephony access points for multi-tenanted managed facilities
- Providing PSTN access points and local survivability for corporate, wide area VoIP networks



## Technical summary

| Feature | | Feature detail |
|---|---|---|
| **Network interfaces** | | |
| SIP and RTP (VoIP) | | 2 x Gb (1000 BASE-T) Ethernet; front panel RJ45 (RJ48C) |
| TDM telephony | | 1, 2, or 4 E1/T1 trunks; front panel RJ45 (RJ48C); co-axial converters available |
| Management | | 2 x Gb (1000 BASE-T) Ethernet; front panel RJ45 (RJ48C) |
| **Signalling and control protocols** | | |
| IP | | SIP, RTP, RTCP, TCP, UDP |
| PSTN | ISDN[1] | Including: Q.931 PRI/ISDN types, Euro ISDN/ETS300, QSIG (ISO, ECMA), DPNSS, DASS 2, NI2, and AT&T |
| | CAS[1] | Including: MFC R2, T1 robbed bit, E1, R1, R2 and DTMF CAS types |
| Capacity | | 24/30, 48/60 or 96/120 (E1/T1) independent voice calls; up to 60 independent DPNSS-to-QSIG conversions |
| Security | | SRTP[2], TLS[2], SIPS[2], HTTPS |
| **Feature support** | | |
| Voice codecs | | G.711, G.729AB; dynamically selected on a per call basis |
| DTMF | | DTMF detection, generation, suppression; in-band; pass-through; relay and user indications (RFC 2833); DTMF out-of-band (SIP INFO, RFC 2976) |
| Fax | | T.30 fax receive and transmit; fax over G.711; and T.38 fax relay |
| Echo cancellation | | G.168 compliant |
| QoS | | Enhanced jitter buffer; packet loss concealment; comfort noise generation; voice activity detection; silence suppression |
| Supplementary services | | Transfer (incl. SIP Update and REMOTE-PARTY-ID), Diversion (Immediate, On Busy, On No Reply); Route Optimisation; Loop Avoidance; Call Back When Free; Call Back When Next Available; Message Waiting Indicator; Hold; Call Offer; Three Party; Redirection; Data Call; Bearer Service Selection; Network Address Extension; Add-on Conference; Do Not Disturb; and Centralised Operator |
| **Operational management** | | |
| Configuration | | Via easy to use embedded Web/HTTP interface |
| Management and monitoring | | Via comprehensive SNMP traps, and DHCP |
| Diagnostics | | Full protocol trace feature with decoder inlcluded |
| **Hardware** | | |
| Dimensions | | 1U high, 19” wide rack mountable |
| Power | | 110-230V AC power supply |
| Regulatory | | EC Directives: 2014/35/EU (EMC); 2014/30/EU (LVD); 2011/65/EU (ROHS) |
| EMC standards | | EU – EN55032 and EN55024; USA – FCC part 15 |
| Safety | | CB certification: UL/CUL; EN60950 |
| Operating environment | | Operating temperature: 0°C to +40°C; humidity: 20% to 80% RH non-condensing |

**Notes:**
1 Visit www.aculab.com to view the full range of protocols supported.
2 Export restrictions apply; please contact your Account Manager for details.

For more information, please contact your Account Manager or view our website