INNOMEDIA

ESBC 9628

Enterprise Session Border Controller

HIGHLY INTEGRATED ESBC IDEAL FOR BROADBAND SERVICE PROVIDERS OFFERING SIP TRUNKING AND HOSTED VOIP SERVICES

Designed for Service Providers offering SIP Trunking, Hosted Voice and high-speed Gigabit data services, InnoMedia's ESBC 9628 is a highly integrated and highly manageable Enterprise Session Border Controller (ESBC) that can be auto-provisioned and remotely managed. It is ideally suited to wide deployment by broadband service providers addressing SIP-PBX interoperability for SIP Trunking as well as providing NAT Traversal for Hosted PBX Services.

KEY BENEFITS

An ideal solution for broadband service providers delivering SIP Trunking & Hosted Voice services to business customers with an IP-PBX and IP Phones.

SIP Trunking Voice Features

- B2BUA
- SIP registrar server
- SIP normalization for IP-PBX interoperability
- NAT traversal for SIP messages
- Special call handling for Emergency Calls
- 60 Concurrent Calls

Hosted Voice Features

- Allows authorized VoIP traffic from IP Phones
- SIP Header Manipulation for server interoperability
- NAT traversal with minimal configuration
- Hosted voice service local survivability
- 200 Concurrent calls

Security

- TLS for signaling
- SRTP for voice traffic
- VPN client support
- SIP-aware firewall
- Stateful packet inspection
- Access Control

Monitoring Features

- Test Call Agent
- Calculated MOS scores for every call
- Media & packet loopback for voice quality measurement (VQM)

Highly secure solution with TLS for secured signaling, SRTP for secure voice traffic and OpenVPN client support for secured access.

An ESBC stateful SIP firewall is also included placed in front of the FXS ports, SIP trunk traffic and Hosted service traffic to protect these voice streams from unauthorized access or attacks.

The Access Control feature further protects the ESBC on the specified interfaces and ports from undesired access attempts, scanning etc.

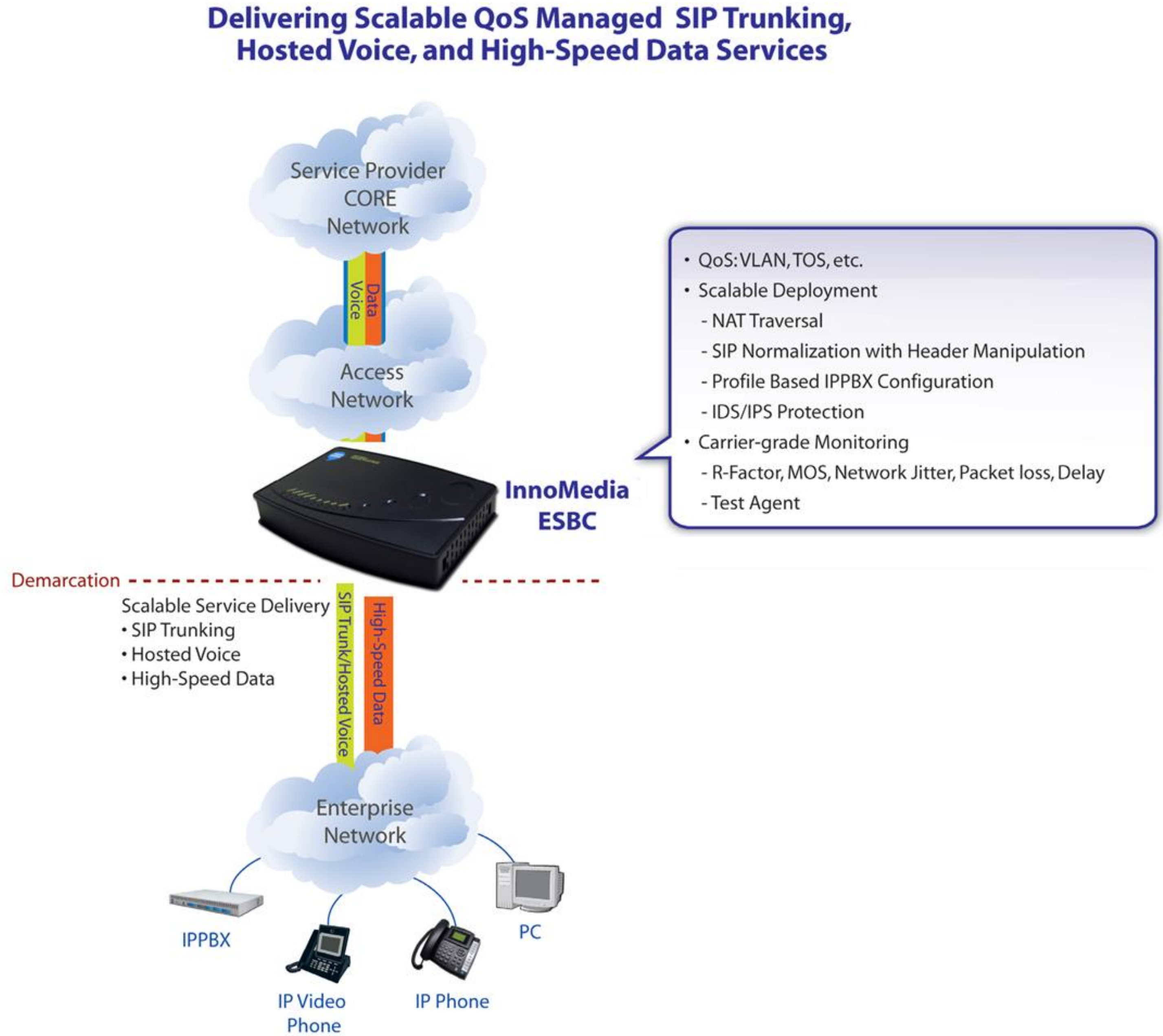
The SIP Trunk traffic path provides SIP normalization, NAT traversal, topology hiding, and security for Service Providers offering SIP Trunking service to enterprise customers with diverse IP-PBX and network configurations.

The ESBC includes B2BUA for SIP normalization, a Registrar for User Agent (UA) registration, and NAT traversal with full SDP address translation.

The SIP UA (e.g., IP-PBX) registers to and communicates with the ESBC, which terminates SIP UA traffic and re-initiates normalized SIP packets to communicate with the Service Provider's network servers.

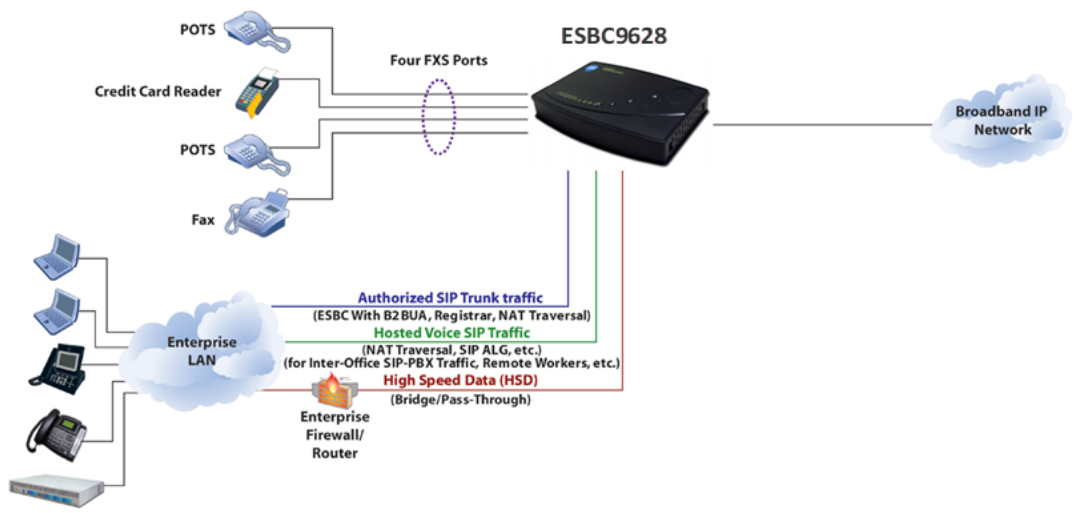
The SIP ALG traffic path enables Service Providers to offer Hosted Voice Services with NAT traversal and header manipulation. It allows authorized hosted SIP traffic from registered SIP UAs (e.g., IP Phones) to traverse through and communicate with the network servers. The SIP UAs register to the designated network servers, and point to the ESBC as the default gateway to route the packets. It can also route non-voice traffic between LAN and WAN interfaces (provisioning, NTP etc.). The service provider can also view a list of currently registered LAN-side Hosted clients.

The ESBC 9628, located at the edge of the broadband access network, can be managed by the Service Provider with secured HTTPS-based auto-provisioning and SNMP-based remote management. It offers an ideal demarcation between the Service Provider and enterprise customer networks.



The highly integrated ESBC 9628 includes the following key functional blocks:

1. Four FXS ports with business-friendly features
2. ESBC function supporting Service Provider's SIP Trunk business
3. ESBC SIP ALG for hosted voice SIP traffic
4. SIP Firewall and Access Control protecting the ESBC, FXS, and the SIP Proxy and ALG paths
5. Voice quality measurement and network monitoring



Four FXS Ports with Business-Friendly Features

InnoMedia's ESBC 9628 includes 4 voice ports that deliver revenue-generating telephony services to their enterprise customers. It has a rich set of business features including ground start/loop start and OSI for business PBX's, foreign voltage detection, T.38 and G.711 fallback fax support and reliable low-speed modem transmission for credit card reader transactions.

Network Port Flexibility

The ESBC 9628 offers 1x WAN Network port and 3x LAN Network ports. The LAN Network ports can be configured in multiple arrangements as follows:

- 3x VoIP ports
- 2x VoIP ports, 1x Management port
- 2x VoIP ports, 1x Backup WAN port
- 1x VoIP port, 1x Management port, 1x Backup WAN port

If one of the LAN ports is repurposed as a Backup WAN port, the ESBC will automatically detect if the primary WAN network is down and immediately switch to the Backup WAN. If revertive mode is selected, the unit will automatically revert back to the primary WAN network when it is available. Manual switchover is also possible between interfaces.

ESBC Function Supporting Service Provider's SIP Trunk Business

Using B2BUA, the ESBC 9628 supports the key functions needed by Service Providers to offer reliable and scalable SIP Trunk services to their enterprise customers. It supports up to 60 simultaneous B2BUA sessions. The key functions that are supported by the ESBC 9628 include:

SIP Normalization

Based on the B2BUA architecture, InnoMedia's ESBC 9628 provides Profile-based settings, high-level classification for interoperability with Service Provider SIP Servers, and low-level header manipulation for SIP signaling normalization.

- ***Profile based settings***. The ESBC 9628 allows parameter and option settings to adapt between the following two interfaces: the WAN interface to the Service Provider server, and the LAN interface to the SIP UAs/IP-PBX. The settings are pre-stored as a wide variety of SIP Trunk profiles and the SIP UA/IP-PBX profiles respectively, allowing the user to simply select the LAN and WAN interface SIP Profiles they wish to use.
- ***Normalization adaptation mechanisms.*** Each of the SIP Trunk profiles and SIP UA/IP-PBX profiles can be further customized by editing the profile. Settings available for customization include SIP Registration types (e.g., implicit, explicit, static/no registration), Security (TLS, SIP Digest), TCP versus UDP for SIP message transport, Redirect Handling (Out-of-dialog Diversion, 3xx, REFER, etc.), URI Formatting, anonymous calls, and others. See the discussion below under "Trunk SIP Profiles" and "IP-PBX Profiles".
- ***Low level SIP header manipulations.*** For very fine-grain adjustments, the ESBC includes the ability to create custom SIP header manipulation rules for adjusting SIP headers or SDP attributes.

Registration and Authentication

Acting as a registrar server to IP-PBXs, the ESBC 9628 supports the following IP-PBX registration methods.

- **Implicit registration.** IP-PBX with dynamic or fixed IP address sends registration for the Parent Number
- **Explicit registration.** IP-PBX with dynamic or fixed IP address sends registration for all SIP User accounts

NAT Traversal

Inspects and modifies headers, SDP, and implements media relay via RTP bridge control.

SIP Signaling Security

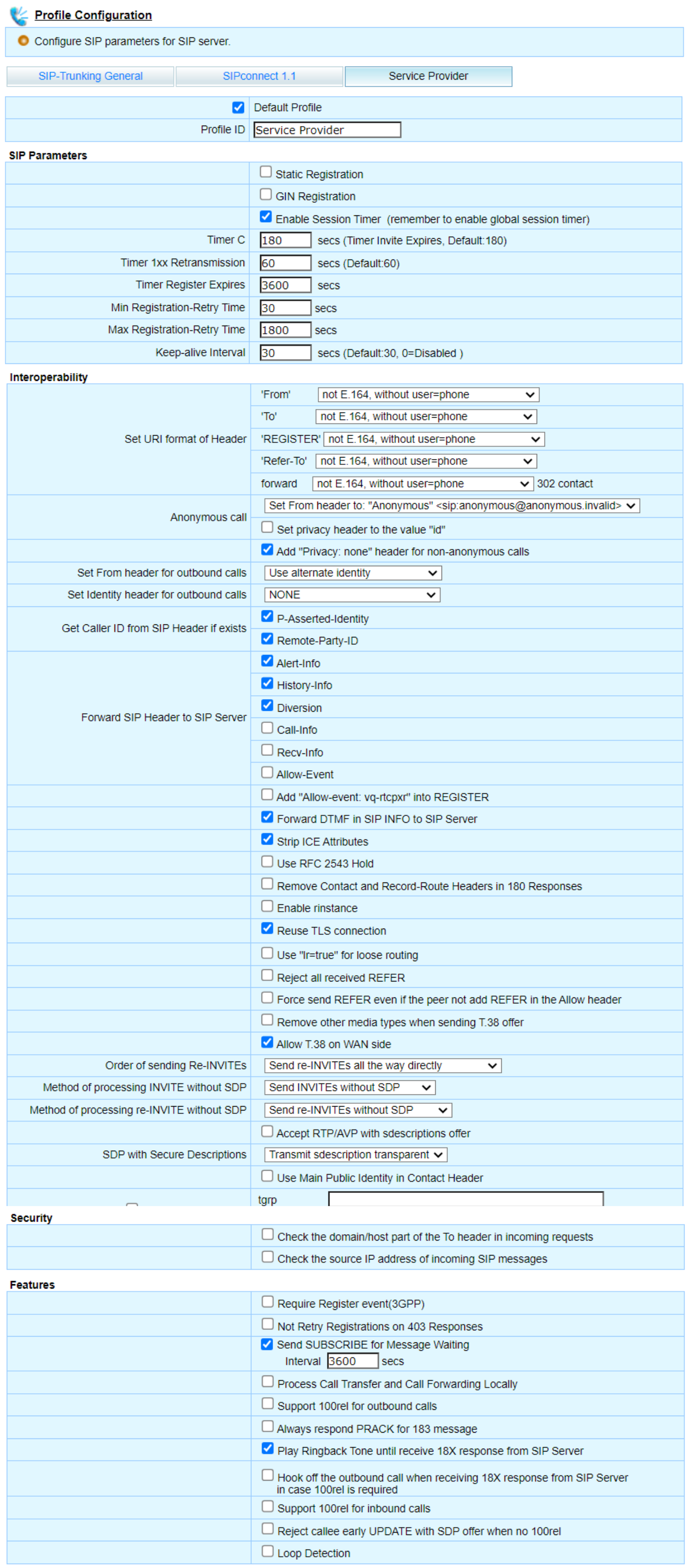
- TLS. The ESBC 9628 supports TLS connections with the Service Provider network for secured signaling transport, as well as SIP Digest authentication (challenged and authenticated by the Service Provider SIP servers).
- SIP Message validation. The ESBC 9628 validates all SIP messages.

Emergency Call Handling

- Special call handling and SIP header manipulation for emergency calls.
- Line Pre-emption to always allow emergency calls regardless of session limits.
- Media manipulation to force the codec used and disabling voice activity detection.
- Overriding caller ID and caller name information.

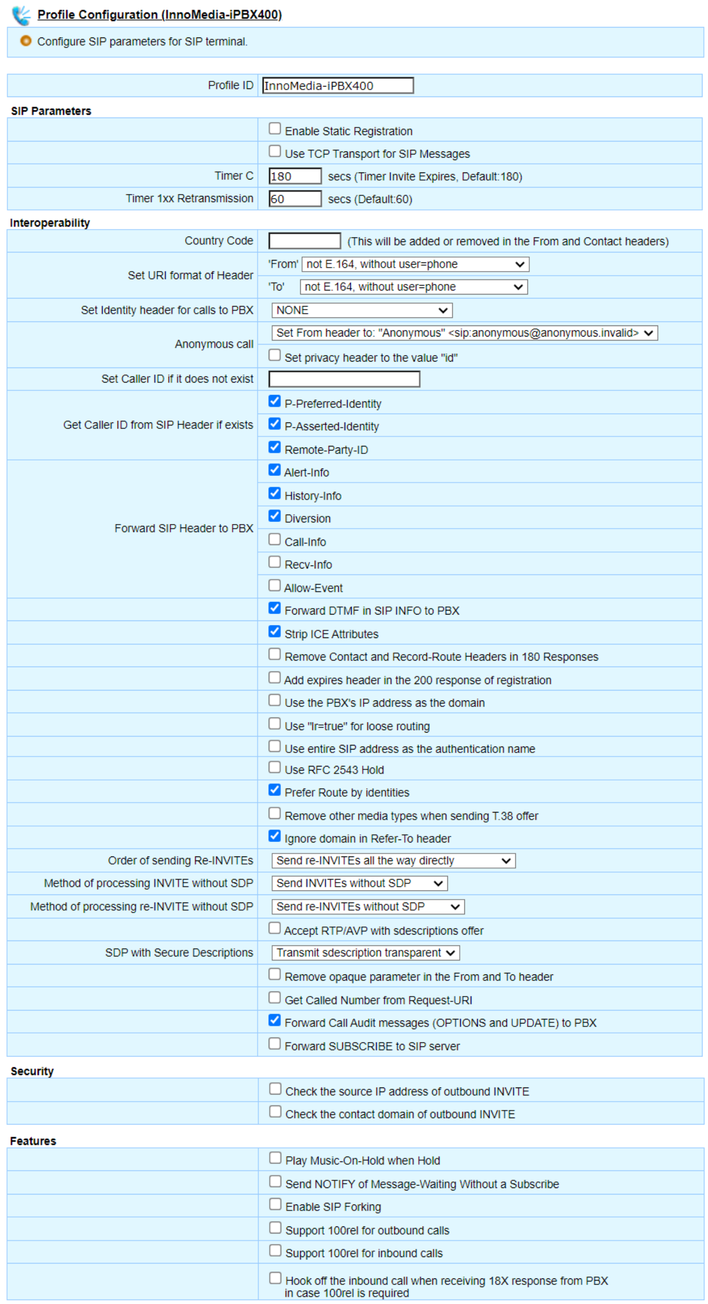
Trunk SIP Profiles

The ESBC normalizes SIP packets from the enterprise network to interwork with SIP servers in the service provider network. This screenshot shows an example of a single Trunk SIP profile that can be customized to meet the service provider's interoperability needs.



IP-PBX Profiles

IP-PBX models used in deployments, though generally conforming to SIP requirements, may also be designed with some deviations for specific needs. The ESBC normalizes SIP packets to/from the target IP-PBX to allow interconnection with the SIP server. The ESBC provides various pre-configured IP-PBX Profiles for commonly used IP-PBX models. The service provider can choose the target IP-PBX model from the profile list. If further customization is required, either the individual IP-PBX Profiles can be edited, or the "Generic" profile can be chosen and the associated configuration changes made according to the specific requirements of the PBX as shown in this screenshot.



Profile Configuration (InnoMedia-iPBX400)

Configure SIP parameters for SIP terminal.

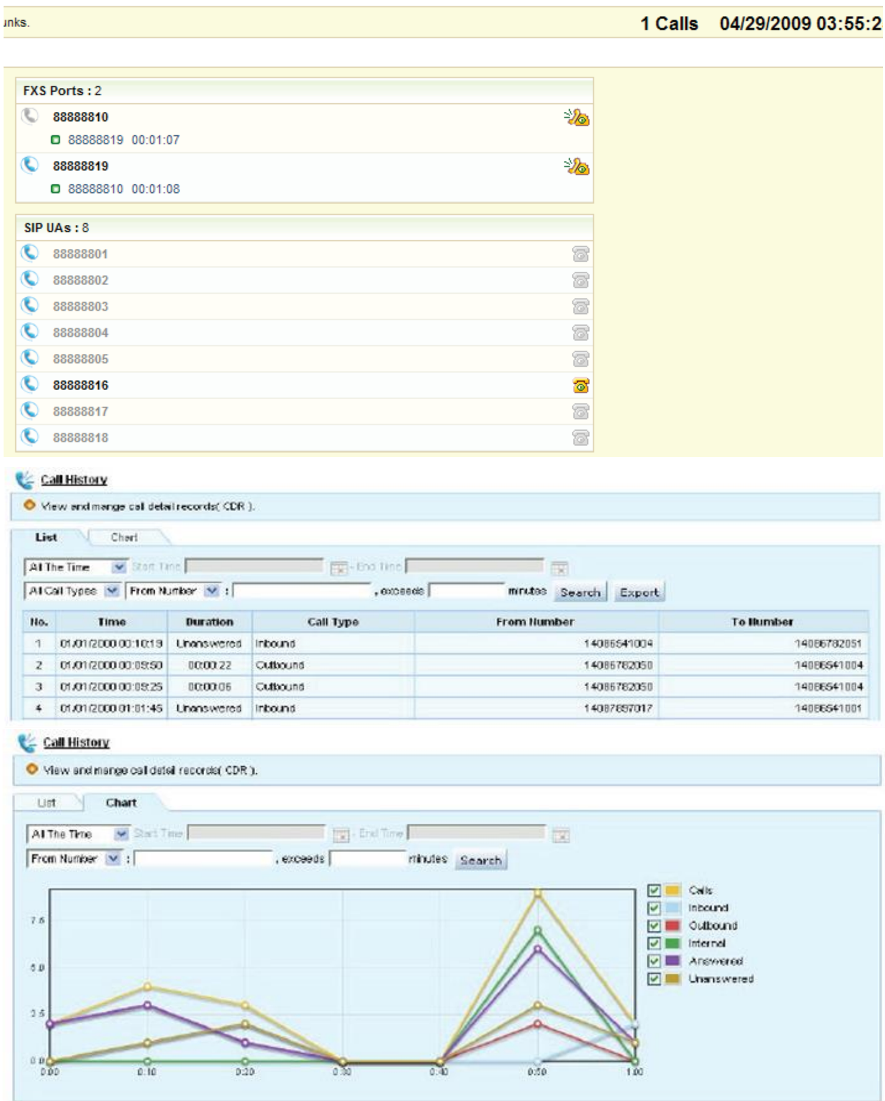
Profile ID	InnoMedia-iPBX400
SIP Parameters	
	☐ Enable Static Registration
	☐ Use TCP Transport for SIP Messages
Timer C	180 secs (Timer Invite Expires, Default:180)
Timer 1xx Retransmission	60 secs (Default:60)
Interoperability	
Country Code	(This will be added or removed in the From and Contact headers)
Set URI format of Header	'From' not E.164, without user=phone
	'To' not E.164, without user=phone
Set Identity header for calls to PBX	NONE
Anonymous call	Set From header to: "Anonymous" <sip:anonymous@anonymous.invalid>
	☐ Set privacy header to the value "id"
Set Caller ID if it does not exist	
Get Caller ID from SIP Header if exists	☑ P-Preferred-Identity
	☑ P-Asserted-Identity
	☑ Remote-Party-ID
Forward SIP Header to PBX	☑ Alert-Info
	☑ History-Info
	☑ Diversion
	☐ Call-Info
	☐ Recv-Info
	☐ Allow-Event
	☑ Forward DTMF in SIP INFO to PBX
	☑ Strip ICE Attributes
	☐ Remove Contact and Record-Route Headers in 180 Responses
	☐ Add expires header in the 200 response of registration
	☐ Use the PBX's IP address as the domain
	☐ Use "lr=true" for loose routing
	☐ Use entire SIP address as the authentication name
	☐ Use RFC 2543 Hold
	☑ Prefer Route by identities
	☐ Remove other media types when sending T.38 offer
	☑ Ignore domain in Refer-To header
Order of sending Re-INVITEs	Send re-INVITEs all the way directly
Method of processing INVITE without SDP	Send INVITEs without SDP
Method of processing re-INVITE without SDP	Send re-INVITEs without SDP
	☐ Accept RTP/AVP with sdescriptions offer
SDP with Secure Descriptions	Transmit sdescription transparent
	☐ Remove opaque parameter in the From and To header
	☐ Get Called Number from Request-URI
	☑ Forward Call Audit messages (OPTIONS and UPDATE) to PBX
	☐ Forward SUBSCRIBE to SIP server
Security	
	☐ Check the source IP address of outbound INVITE
	☐ Check the contact domain of outbound INVITE
Features	
	☐ Play Music-On-Hold when Hold
	☐ Send NOTIFY of Message-Waiting Without a Subscribe
	☐ Enable SIP Forking
	☐ Support 100rel for outbound calls
	☐ Support 100rel for inbound calls
	☐ Hook off the inbound call when receiving 18X response from PBX in case 100rel is required

Built-in Firewall

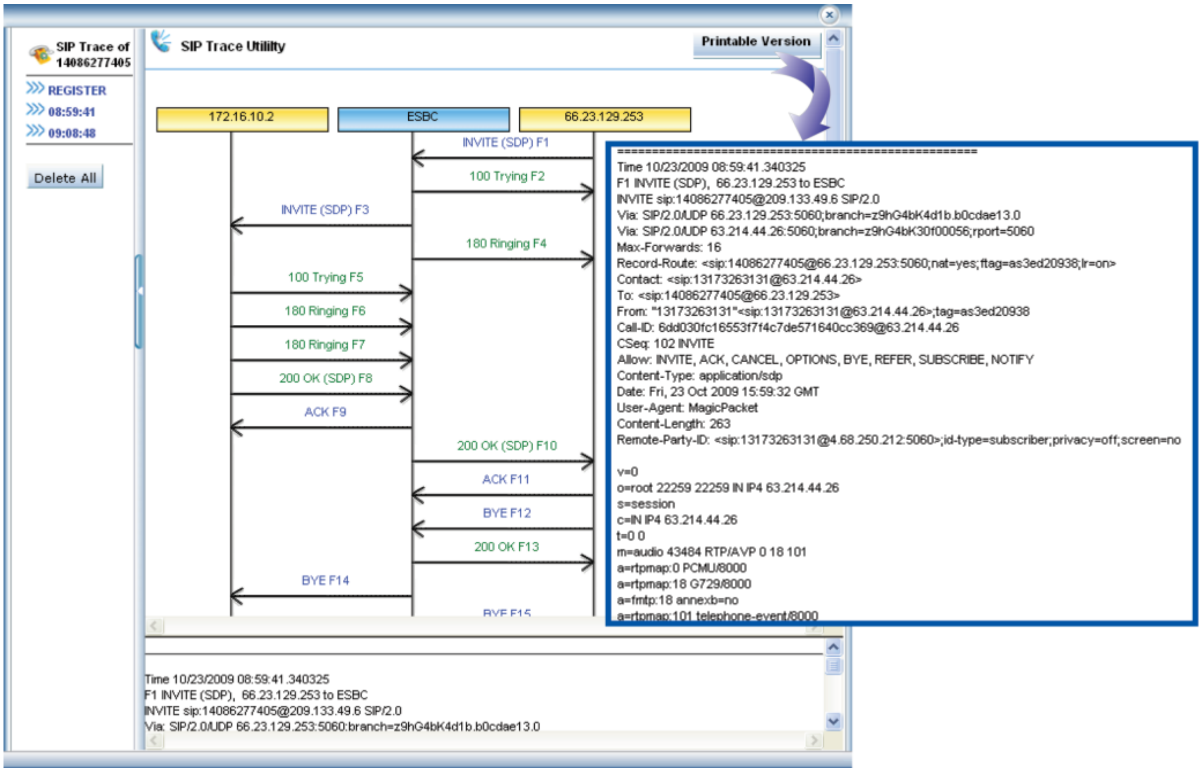
A built-in firewall protects the FXS ports, the SIP trunk traffic path, as well as the ALG-hosted SIP traffic path from unauthorized access or attacks. The bridge port which connects to the enterprise network will typically be protected by a separate data firewall provided by the enterprise.

Monitoring

ESBC monitoring features include CDR-like functionality, real-time SIP UA & SIP Trunk call states, SIP Call Trace, media and packet loopback for server-based Voice Quality Measurement, R-Factor and MOS calculations for every call, and SNMP Traps based on configurable thresholds of network call parameters. The ESBC also works in conjunction with the InnoMedia EMS Server for monitoring and analysis of MOS scores, Data Network Traffic and CDR information. See the screenshots below for illustrations of Real-Time Line Call States, CDR-like information and Call Statistics.



Call Trace Diagrams



Specifications

ESBC 9628 Interfaces

- Four standard FXS ports
- Two Gigabit RJ-45 Ethernet WAN ports
- Three Gigabit RJ-45 Ethernet LAN ports
- 12V DC Power
- RSTR (restore to factory)

Software Features

Category	Specifications
SIP Trunking Features	• SIP Connect Compliant • Implicit, Explicit and Static Registration support • SIP User account authentication - Digest and RADIUS • Secured SIP Signaling – TLS • NAT Traversal • SIP Normalization • Emergency Call Handling • SIP Header Manipulation • SIP Proxy and Registrar • SIP Method Filtering • Monitoring Features - SIP Call Trace, Call Statistics, and Voice Quality Monitoring, Test Agent for test calls, R-Factor and MOS calculation • Call capacity: 60 concurrent calls
Hosted Voice Features	• SIP Application Layer Gateway (ALG) • NAT Traversal • SIP Header Manipulation • Monitoring Features - SIP Call Trace, Call Statistics, and Voice Quality Monitoring, Test Agent for test calls, R-Factor and MOS calculation • Local Survivability • Call capacity: 200 concurrent calls
Data Networking Features	• Built-in DHCP server • Backup WAN redundancy • Static IP routing and port forwarding • IP Addressing modes: IPv4 only, IPv6 only, IPv4/IPv6 dual-stack • Network Access Control by Applications, IP address, Subnet, Port Number, MAC address, or Destination Domain Name • Web GUI with 3 levels of page permissions • Auto-Backup of Configurations • Port Forwarding with ACL support
Other Security Features	• Secured Voice Traffic – SRTP • VPN client support
VoIP Protocols	SIP 2.0, RFC2833

SIP RFC Support	RFC 1847, RFC 2045, RFC 2046, RFC 2181, RFC 2617, RFC 2782, RFC 2915, RFC 2976, RFC 3261, RFC 3263, RFC 3265, RFC 3311, RFC 3325, RFC 3326, RFC 3420, RFC 3428, RFC 3486, RFC 3515, RFC 3581, RFC 3761, RFC 3824, RFC 3891, RFC 3892, RFC 3903, RFC 4028, RFC 4320, RFC 4474, RFC 4508, RFC 4566, RFC 3264, RFC 3313, RFC 3323, RFC 3327, RFC 3329, RFC 3388, RFC 3605, RFC 3608, RFC 3841, RFC 3911, RFC 3966, RFC 4483, RFC 4488
Network RFC Support	RFC 768, RFC 783, RFC 791, RFC 792, RFC 793, RFC 826, RFC 854, RFC 1157, RFC 1256, RFC 1332, RFC 1349, RFC 1519, RFC 1570, RFC 1631, RFC 1661, RFC 1812, RFC 1918, RFC 2131, RFC 2571, RFC 2572, RFC 2573, RFC 2574, RFC 2575, RFC 2578, RFC 2579, RFC 2580, RFC 2865
Speech Codec Capabilities	SIP Trunking/Hosted ports: G.711, G.729, G.723.1, Opus FXS ports: G.711, G.729
Signal Processing	• G.168 Echo cancellation • FAX (T.38 and G.711 pass through) • Caller ID FSK signal regeneration • Foreign voltage detection • Media and Packet loopback • Ground Start/Loop Start
Tones	• Ring back tone • Reorder tone • Dial tone • Ring splash • Off-hook warning tone • Caller ID generation • Call Waiting tone • Busy tone • Distinctive ringing (5 rings) • Confirmation tone • Stutter tone • Message waiting indicator (MWI) • Configurable ring frequency
DTMF Tone	DTMF tone detection and generation
Announcements	Play out any voice stream sent by a Call Agent-controlled Announcement Server
Monitoring	• SIP call traces • CDR (Call Data Record)-like functionality • Real-time SIP UA and SIP Trunk call states • Voice Quality Measurement (VQM) based on packet & media loopback • Configurable SNMP traps for voice quality threshold breaches
OAM&P	• Access components implemented: ○ TFTP, FTP, HTTP (HTTPS) 1.1/1.2/1.3, SNMP, Telnet, DHCP & DNS • Works with any SNMP (v.2c or v3) based EMS • Web GUI, SSH and local console for administration • Provisioning via HTTP, HTTPS, FTP or TFTP • Syslog support • Per-customer Customization of units at Production • Dual image capability • TACACS+ support for User Logins
QoS	• Voice Bandwidth Reservation QoS • Type of Service (ToS) • Traffic Shaping

Regulatory Compliance

Category	Specifications
Certifications/Compliance	UL, FCC Part15B

Hardware and Environmental Specifications

Category	Specifications
Power Consumption	• Idle: 3.18W • Ringing: 14.1W (max for all ports @ 2 REN each) • Talking: 2.28W (max for all ports)
Loop Current	SNMP-settable to 25 mA(default) or 40 mA(max.)
Ring Voltage	• 60 Vrms • Maximum 2 REN per port • 24 AWG loop
Power Supply	Output: DC 12V, 4A / Input: AC 100~240V, 50~60Hz
Dimensions	H x W x D: 1.625 in x 8 in x 5.125 in / 40 mm x 204 mm x 128 mm
Weight	Unit: 0.4 kg (0.89 lb) Packaging: 1.17 kg (2.6 lb)
Operating Temperature	32ºF to 104ºF (0ºC to 40ºC)
Storage Temperature	-4ºF to 158ºF (-20ºC to 70ºC)
Operating Humidity	10 to 90% RH
Storage Humidity	5 to 95% RH

www.innomedia.com

InnoMedia Pte Ltd.
Blk 22 Sin Ming Lane #05-87
Midview City, Singapore 573969

InnoMedia, Inc
48531 Warm Springs Blvd., Suite 417
Fremont, CA 94539, USA

InnoMedia Technology Inc.
3F, No. 3, Industrial East Road IX Hsinchu Science-Based
Industrial Park, Hsinchu 300, TAIWAN, R.O.C

InnoMedia Technology China, Ltd.
Room 302 Housha Yu Bailu Guangchang
Shunyi District Beijing, 101300 CHINA