

INNOMEDIA

ESBC 8328-4B

ENTERPRISE SESSION BORDER CONTROLLER

HIGHLY INTEGRATED ESBC IDEAL FOR SERVICE PROVIDERS OFFERING SIP TRUNKING, HOSTED VOICE, AND HIGH-SPEED DATA SERVICES

Designed for Service Providers offering SIP trunking, hosted voice, and high-speed data services, InnoMedia's ESBC 8328-4B is a highly integrated and highly manageable Enterprise Session Border Controller (ESBC) that can be auto-provisioned and remotely managed. It is ideally suitable for wide deployment by broadband service providers addressing SIP-PBX interoperability for SIP Trunking as well as providing simple NAT Traversal for Hosted PBX Services.

Key Benefits

A clear demarcation solution for broadband service providers to deliver SIP trunking, hosted voice, and broadband internet services to business customers with IP-PBX and IP-Phones

SIP trunking features and capabilities

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

SIP ALG/Proxy

- Allowing authorized non-SIP trunk SIP traffic
- NAT traversal with minimal configuration

Bridged/pass-through port

- Bridged to the WAN port
- For service provider to deliver high-speed data services

Stateful Inspection

- SIP-aware, access control
- Stateful packet inspection, IDS/IPS

Monitoring features including Test Call Agent, Calculated MOS Scores for every call, CDR, real-time UA & SIP trunk call states, SIP call trace, battery status, and packet loopback for server-based voice quality monitoring

Business environment friendly

- PBX (Ground start/Loop start & OSI)
- FAX (T.38 and G.711 fallback)
- House wiring with foreign voltage detection
- Credit card reader transaction

Security

- TLS for signaling
- Stateful Inspection, IDS/IPS
- Access control



A stateful inspection with IDS/IPS is placed in front of the FXS ports, the SIP trunk traffic path, as well as the non-SIP Trunk SIP traffic path to protect these paths from unauthorized access or attacks. The high-speed data path is not protected by the firewall as it is assumed that this path is connected to the enterprise firewall which has its protection policy.

The SIP trunk path provides SIP normalization, NAT traversal, topology hiding, and security for Service Providers offering SIP trunking service to enterprise customers with diverse IPPBX and network configurations. It includes B2BUA for SIP normalization, a Registrar for User Agent (UA) registration, TLS block for secured signaling, and NAT traversal for proper SDP address correction. The UA (e.g., IPPBX) registers to and communicates with the ESBC which terminates UA traffic and re-initiates normalized SIP packets to communicate with the Service Provider's network servers.

The SIP ALG path enables Service Providers to offer Hosted PBX Services with NAT traversal, TLS security for signaling,



and header manipulation. It allows SIP packets of registered UAs (e.g., IP Phones) to traverse through to communicate with the network servers. The UAs register to the designated network servers, and point to the ESBC as the default gateway.

The bridge path is a transparent pass-through port, allowing uninterrupted high-speed data to go through. It is intended for Service Providers to offer high-speed data services.

The ESBC 8328-4B, located at the edge of the HFC access networks, can be managed by the Service Provider with secured HTTP-based auto-provisioning and SNMP-based remote management. It offers an ideal demarcation between the Service Provider and its enterprise customers.

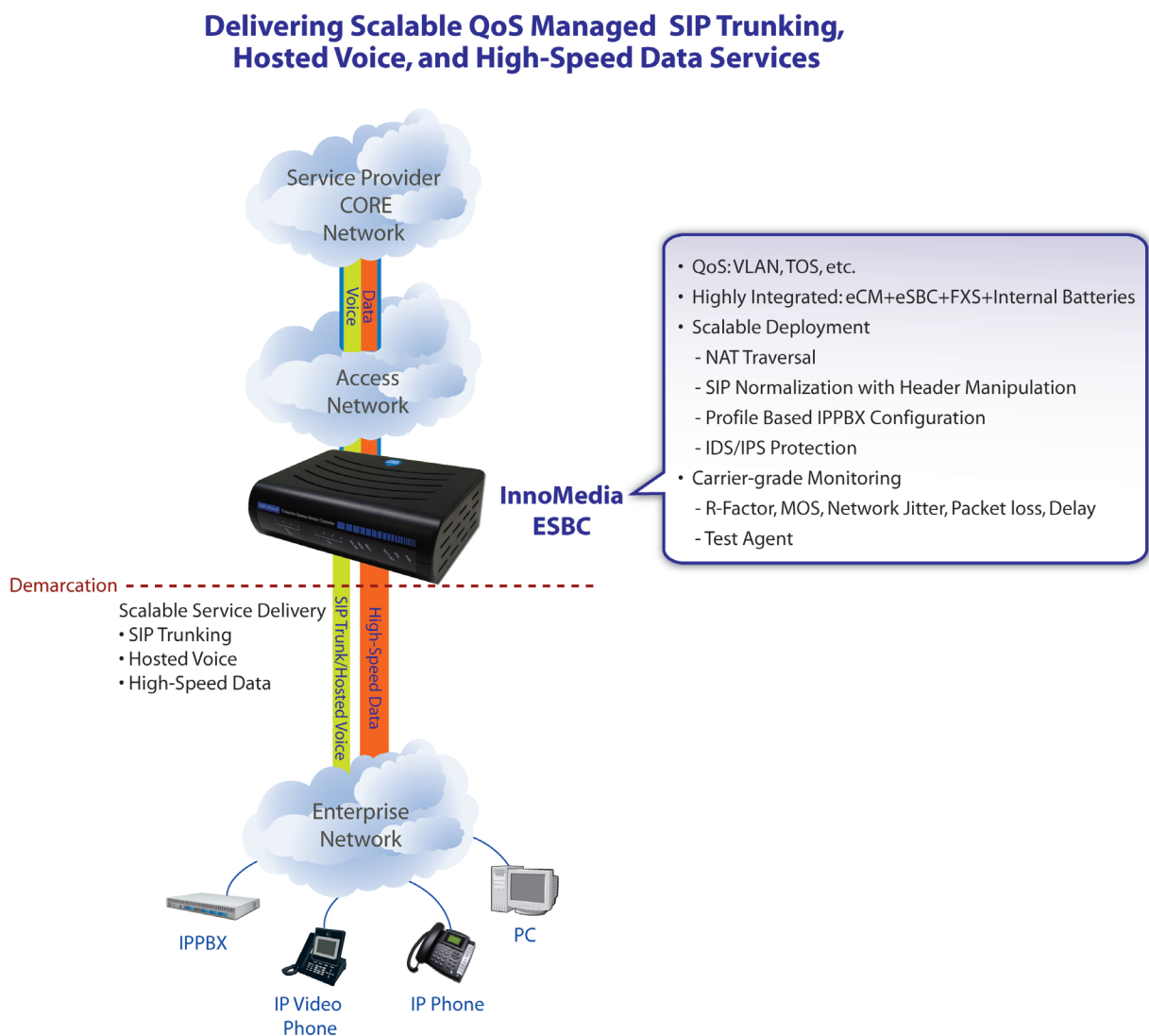


Figure 1

The highly integrated ESBC 8328-4B includes the following key functional blocks:

1. Intelligent internal battery as well as external UPS support
2. Four FXS ports with business friendly features
3. eSBC function supporting Service Provider's SIP trunk business
4. SIP ALG for hosted voice SIP traffic
5. Bridge/pass-through port for Service Provider's high-speed data services
6. Stateful inspection protecting the eSBC, FXS, and the SIP Proxy/ALG path
7. Voice and network Monitoring

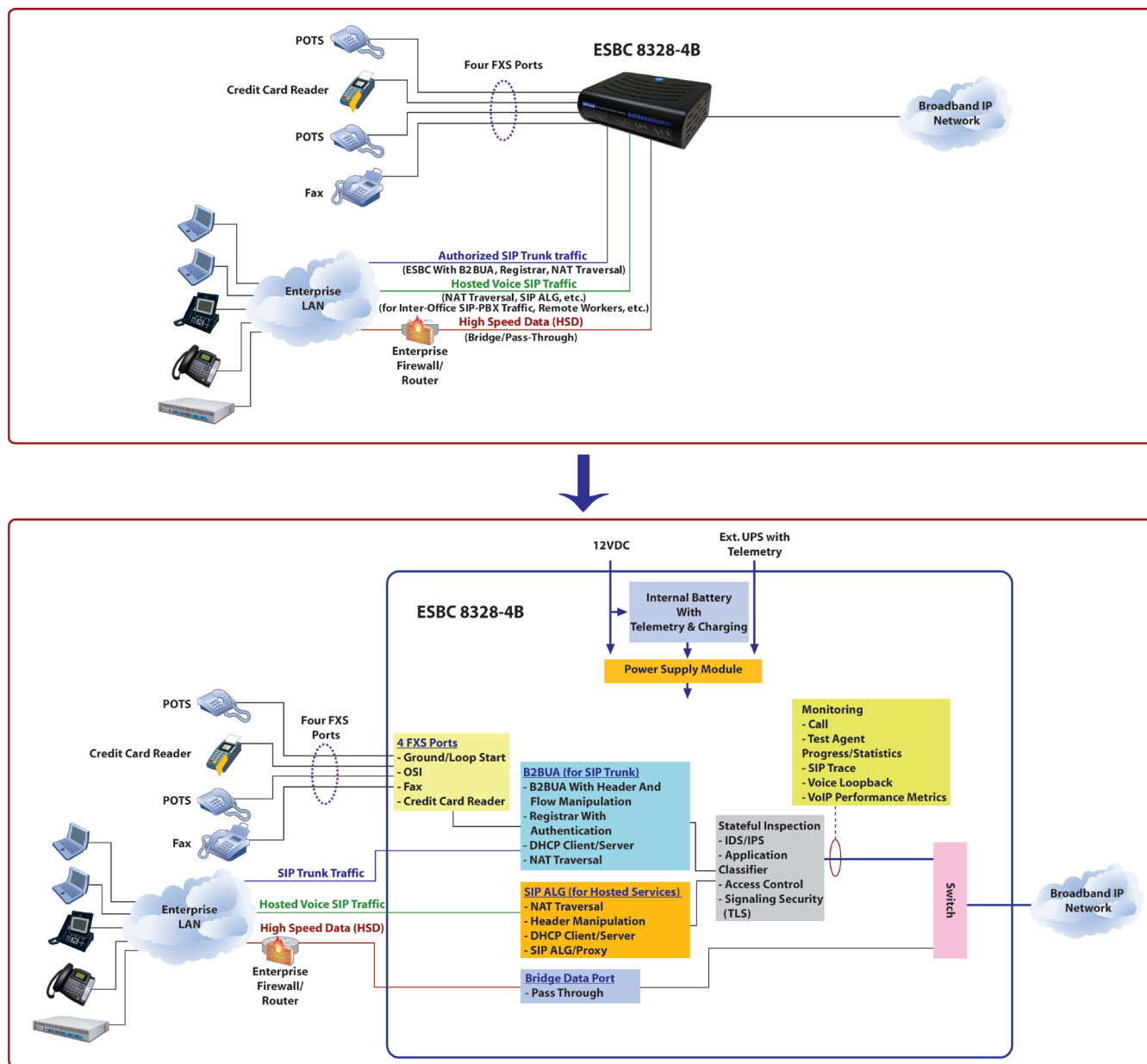


Figure 2

Integrated Internal Battery As Well As External UPS Support

The ESBC 8328-4B is equipped with an internal battery supporting up to 4 hours of continuous talk time for all 4 telephone lines, or up to 8 hours of standby time. It also has a UPS port to connect to external UPS batteries to allow service provider to offer primary line voice services. An Internal and External Battery LED as well as SNMP traps for remote monitoring indicates when the internal or external battery is in-use, charging, fully charged, faulty, or bad.

Four FXS Ports With Business Friendly Features

InnoMedia's ESBC 8328-4B includes 4 voice ports that deliver revenue generating telephony services to their enterprise customers. It has rich set of business features including ground start/loop start and OSI for business PBX's, foreign voltage detection to allow house wiring and prevent accidental connection of house wires to live PSTN, T.38 and G.711 fallback fax support, reliable Bell103/212A modem transmission for credit card reader information transaction, and RJ11 DC open loop for loss of voice link indication to allow alarm triggering.

eSBC Function Supporting Service Provider's SIP Trunk Business

Using B2BUA, the ESBC 8328-4B supports the key functions needed by the Service Providers and Service Providers to offer reliable and scalable SIP trunk services to their enterprise customers. It supports up to 50 simultaneous B2BUA sessions. The key functions that are supported by the ESBC 8328-4B include:

1. SIP Normalization:

Based on the B2BUA architecture, InnoMedia's ESBC 8328-4B provides Profile based settings, High-level classification for SIPConnect Adaptation, and Low-level header manipulation for SIP signaling normalization:

– Profile based settings:

ESBC 8328-4B allows parameter and option settings to adapt between the two interfaces: the WAN interface to the Service Provider servers, and the LAN interface to the UA/SIP-PBXs. The settings are stored as SIP Trunk profiles and the UA/SIP-PBX profiles respectively for selection.

- For each SIP-PBX, the settings are captured in a UA/SIP-PBX specific Profile. Thus, an SI only needs to choose the profile corresponding to the specific SIP-PBX for easy system setup (see Figure 3).
- Based on the Service Provider's network servers, the parameters/options are captured in the corresponding SIP Trunk profile (see Figure 4).

The SIP normalization and adaptation mechanisms are:

– High-level classification for SIPConnect Adaptation (see Figure 4):

- Adapts between non-SIPConnect-compliant UA/SIP-PBXs and Service Provider's Servers which are compliant or non-compliant to SIPConnect
- Adaptation includes Registration (takes care of different forms of registration, 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.

– Low-level header manipulation for fine-grain adjustment (see Figure 3)

- Selectable header manipulation options, examples:
 - Remove headers in 180 responses, Remove RFC 2543 Hold, Strip ICE attributes, Loose routing, Expires header, Loose Username check, Force Remote TLS connection reuse, etc.

2. Registration and Authentication:

Acting as a registrar server to SIP-PBXs, the InnoMedia ESBC 8328-4B supports the following SIP-PBX registration methods:

- a) Implicit registration – SIP-PBX with Dynamic or Static IP address sends registration of the Parent Number
- b) Explicit registration – SIP-PBX with Dynamic or Static IP address sends registration of all SIP User Accounts
- c) Static registration – SIP-PBX with Dynamic or Static IP address does not send any registration messages.

3. NAT Traversal:

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

4. SIP signaling security:

- TLS: ESBC 8328-4B supports TLS connection with the Service Provider network (authenticate Service Provider servers) for secured signaling transport, as well as SIP Digest authentication (challenged and authenticated by the Service Provider servers).
- SIP Message Validation: ESBC 8328-4B validates all SIP messages

5. Emergency Call Handling

- Special call handling and SIP header manipulations for emergency calls
- Line Preemption to always allow emergency calls regardless of session limits
- Media manipulation to force CODEC and disabling voice activity detection
- Overriding caller ID and caller name information

Bridge/Pass-Through Port For Service Provider's High-Speed Data Services

The ESBC 8328-4B allows one of its LAN ports to be configured as a bridge to its WAN interface. This bridge port can be used by the Service Provider to offer high speed data services. The Service Provider can deliver global IP addresses to its enterprise customers who can connect this bridge port to the enterprise firewall.

SIP ALG For Hosted Voice SIP Traffic

The SIP Proxy/ALG path can be used for authorized non-SIP trunk SIP traffic. It will only allow SIP traffic from specific registered and authenticated devices to come in. This path can be used for devices that connect to Hosted PBX services.

Stateful Inspection

A stateful inspection with IDS/IPS can be enabled or disabled for the FXS ports, the SIP trunk traffic path, as well as the Non-SIP Trunk SIP traffic path to protect these paths from unauthorized access or attacks. The bridged/pass-through port is not protected by the firewall, and is typically connected to the enterprise firewall which has its protection policy.

Monitoring

The monitoring features including CDR, real-time UA & SIP trunk call states, SIP Call Trace, battery status, packet loopback for server-based Voice Quality Monitoring, R-Factor and MOS calculation for every call, and SNMP Traps based on thresholds of network call parameters. The ESBC also works in conjunction with InnoMedia's DMS Server for monitoring and analysis of MOS scores, Data Network Traffic and CDR information.

UA/SIP-PBX PROFILE

Profile Configuration (Cisco UC500)

Configure SIP parameters for SIP terminal.

Profile ID: Cisco UC500

SIP Parameters

	<input type="checkbox"/> Enable Static Registration
	<input type="checkbox"/> Use TCP Transport for SIP Messages
Timer Invite Expires	180 secs (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 SIP terminal	NONE
Anonymous call	Set From header to: "Anonymous" <sip:anonymous@[domain]>
Get Caller ID from SIP Header if exists	<input checked="" type="checkbox"/> P-Preferred-Identity
	<input checked="" type="checkbox"/> P-Asserted-Identity
	<input checked="" type="checkbox"/> Remote-Party-ID
Forward SIP Header to SIP Server	<input checked="" type="checkbox"/> Alert-Info
	<input checked="" type="checkbox"/> History-Info
	<input checked="" type="checkbox"/> Diversion
	<input checked="" type="checkbox"/> Forward DTMF in SIP INFO to SIP Server
	<input checked="" type="checkbox"/> Strip ICE Attributes
	<input type="checkbox"/> Remove Contact and Record-Route Headers in 180 Responses
	<input type="checkbox"/> Add expires header in the 200 response of registration
	<input type="checkbox"/> Use the SIP terminal's IP address as the domain
	<input type="checkbox"/> Use "lr=true" for loose routing
	<input type="checkbox"/> Use entire SIP address as the authentication name
	<input type="checkbox"/> Use RFC 2543 Hold
	<input checked="" type="checkbox"/> Prefer Route by identities
	<input type="checkbox"/> Remove other media types when sending T.38 offer
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
	<input type="checkbox"/> Accept RTP/AVP with sdescriptions offer
SDP with Secure Descriptions	Transmit sdescription transparent

Features

	<input type="checkbox"/> Play Music-On-Hold when Hold
	<input checked="" type="checkbox"/> Send NOTIFY of Message-Waiting Without a Subscribe

Restore Default

Apply Cancel

Figure 3

SIP TRUNK PROFILE

Profile Configuration

Configure SIP parameters for SIP server.

Nokia-Siemens HiQ2000 BroadSoft Release 16

☒ Default Profile

Profile ID: Nokia-Siemens HiQ2000

SIP Parameters

☐ Static Registration

☐ Enable Session Timer (remember to enable global session timer)

Timer Invite Expires: 180 secs (Default:180)

Timer 1xx Retransmission: 60 secs (Default:60)

Timer Register Expires: 3600 secs

Keep-alive Interval: 30 secs (Default:30)

Interoperability

Set URI format of Header

'From': not E.164, without user=phone

'To': not E.164, without user=phone

'REGISTER': not E.164, without user=phone

'Refer-To': not E.164, without user=phone

forward: not E.164, without user=phone 302 contact

Anonymous call: Set privacy header to the value "id"

Set From header for Outgoing calls: Use Alternate Identity

Set Identity header for Outgoing calls: NONE

Get Caller ID from SIP Header if exists

☒ P-Asserted-Identity

☒ Remote-Party-ID

☒ Alert-Info

Forward SIP Header to SIP Server

☒ History-Info

☒ Diversion

☒ Forward DTMF in SIP INFO to SIP Server

☒ Strip ICE Attributes

☐ Use RFC 2543 Hold

☐ Remove Contact and Record-Route Headers in 180 Responses

☐ Enable rinstate

☒ Reuse TLS connection

☐ Use "lr=true" for loose routing

☐ Reject all received REFER

☐ Force send REFER even if the peer not add REFER in the Allow header

☐ Remove other media types when sending T.38 offer

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

Features

☐ Require Register event(3GPP)

☒ Send SUBSCRIBE for Message Waiting

Interval: 3600 secs

☐ Process Call Transfer and Call Forwarding Locally

☐ Support 100rel for Outgoing calls

New Replicate Delete Restore Default

Apply Cancel

Figure 4

REAL-TIME LINE CALL STATES, CDR, AND CALL STATISTICS

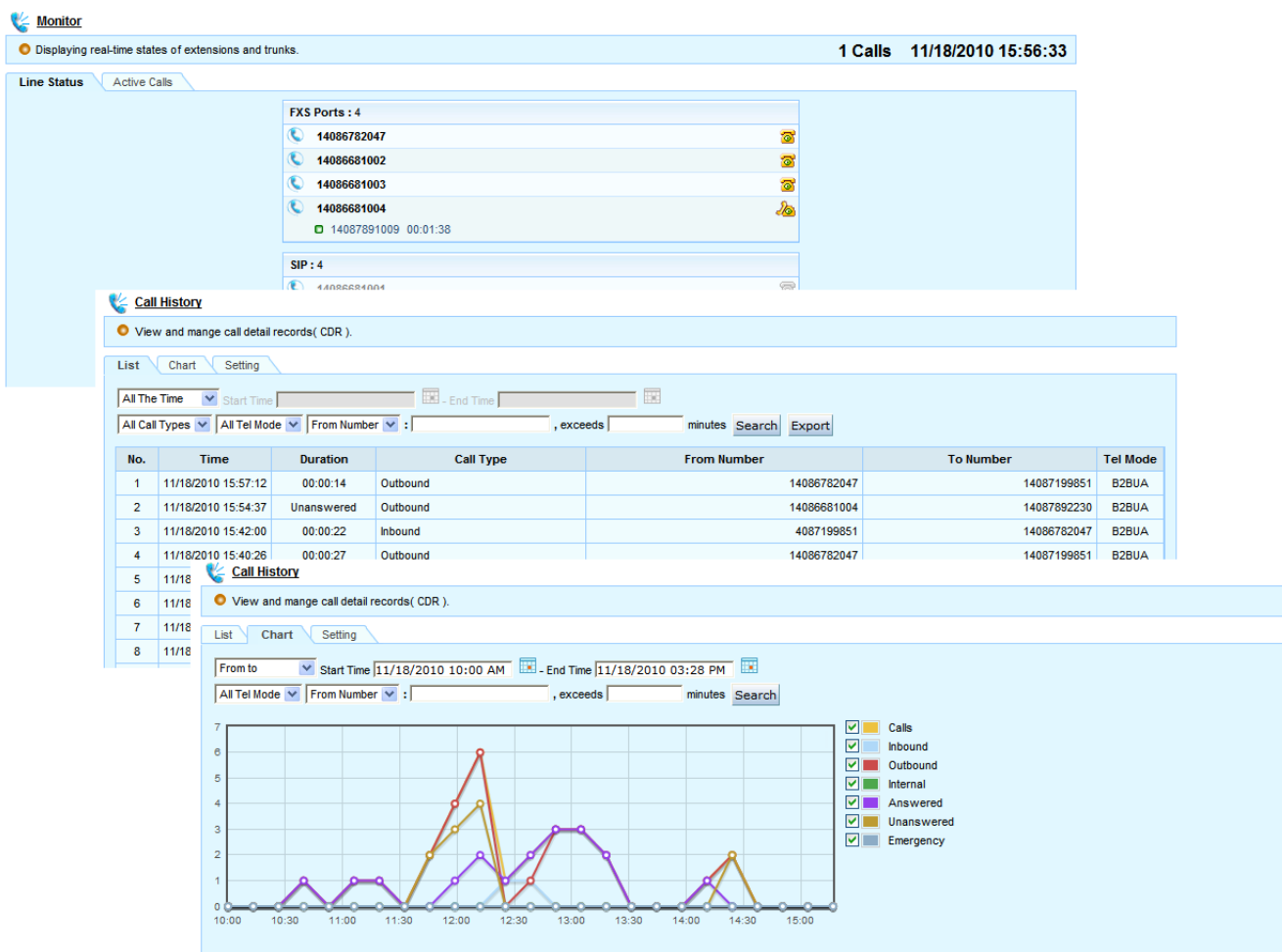


Figure 5

CALL TRACE GUI

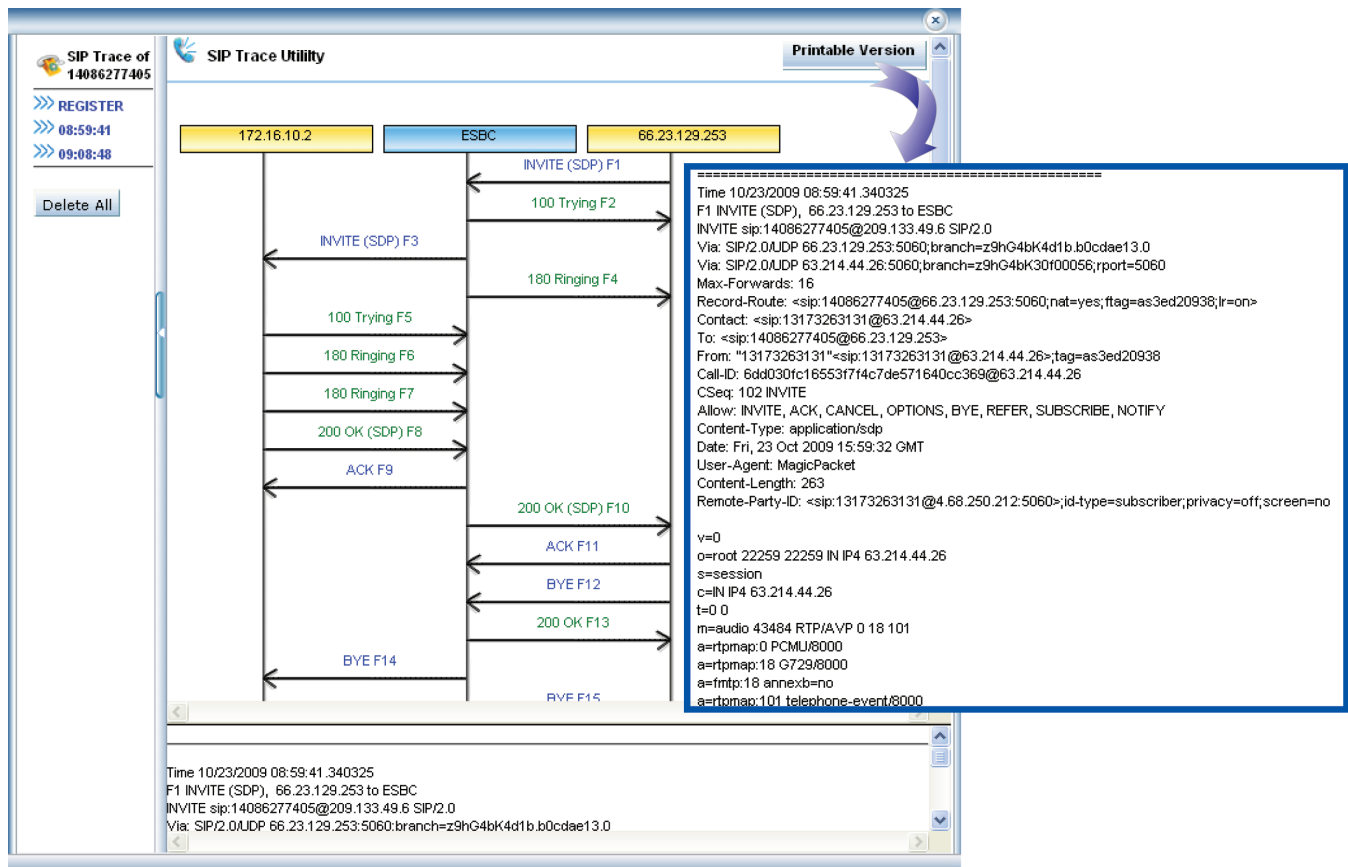
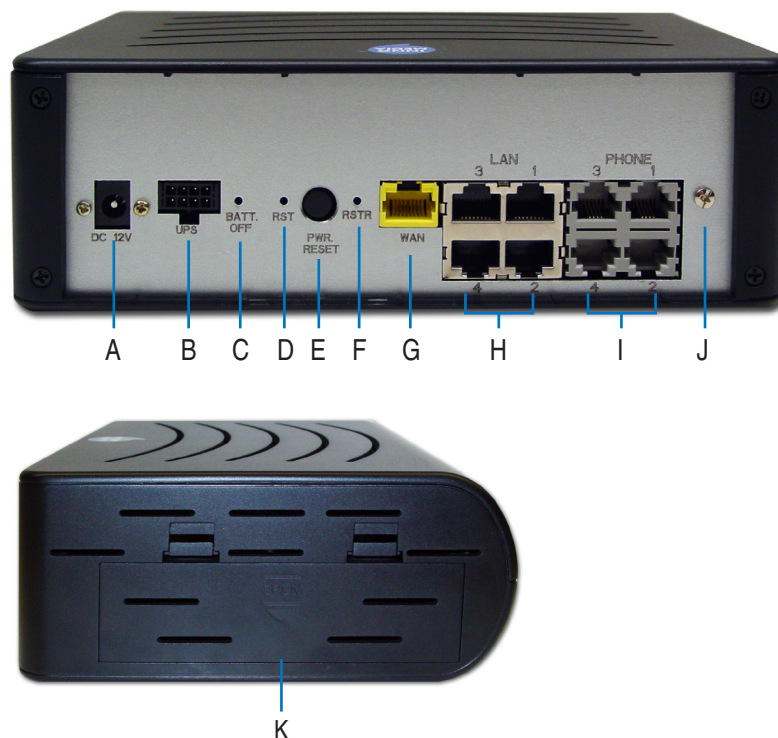


Figure 6

ESBC INTERFACE

- A. 12V DC Power
- B. UPS Port
- C. Battery Off Button
- D. Reset Button
- E. Power Reset Button
- F. Restore Button
- G. WAN Port
- H. LAN 1-4
- I. Phone 1-4
- J. External Ground
- K. Battery Compartment



SPECIFICATIONS

Product Interfaces

Category	Specification
Broadband Uplink Interface	10/100 BaseT Ethernet (RJ-45)
Telephone Interface	4 FXS Voice Ports
User Data Interface	4 10/100 BaseT Ethernet (RJ-45)

Software Specifications

Category	Specification
SIP Trunking Features	Implicit, Explicit, and Static Registration support SIP User Account Authentication - Digest and RADIUS Secured Registration - TLS SIP 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
Data Networking Features	Built-in DHCP server NAT capabilities for simultaneous SIP User Accounts Static IP routing and port forwarding NAT traversal UPnP DMZ SIP Application Layer Gateway Network access Control by application, IP address, Subnet, Port Number, MaC address, or Destination Domain Name Web GUI with 3 levels of page permissions Auto-Backup of Configuration.
VoIP Protocols	SIP 2.0, RFC 2833
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	G.711, G.726 (No compression & simple compression) G.728, G.729E (High quality high complexity codecs) G.723.1, G.729A (Low bit rate codecs)

SPECIFICATIONS cont.

Category	Specification
Signal Processing	G.168 Echo cancellation FAX (T.38 and G.711 fall-back) Caller ID FSK signal regeneration Line reversal Ground Start/Loop Start Loop Back FXS voltage drop when CA or RF fails Pulse Dialing Foreign voltage detection
Tones	Ring back tone Recorder tone Dial tone Ring splash Off hook warning tone Caller ID generation & call waiting tone Busy tone 5 distinct 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 Call Agent controlled announcement server
OAM&P	Access components implemented: TFTP, FTP, HTTP 1.0, SNMP, Telnet, DHCP & DNS Works with any SNMP (v.1-3) -based EMS Offers web-based access as well as TFTP-based remote software downloads or upgrades Dual image capability
QoS	Voice Bandwidth Reservation QoS, Type of Service, VLAN Tagging

Physical Specifications

Category	Specification
Loop Current	For load of 520Ω, SNMP-settable to 23 mA (default) or 32 mA (max.)
Ring Voltage	> 40 Vrms @ 2000 ft. 5 REN max. per port 24 AWG loop
On Battery	Li-ion battery providing 4 hrs Talk Time / 8 hrs Standby Time
Power Supply	AC 100~240V/50~60Hz (DC 12V @ 4.0 Amps)
Dimensions	2.5 in (H) x 7.8 in (W) x 6.0 in (D) / 63.5 mm (H) x 198 mm (W) x 152 mm (D)
Approval	UL, FCC Part15B, cUL
Operating Temperature	32°F to 104°F (0°C to 40°C)
Storage Temperature	-4°F to 140°F (-20°C to 60°C)
Operating Humidity	Up to 80% RH
Storage Humidity	Up to 80% RH

www.innomedia.com

InnoMedia Pte Ltd.

10 Science Park Road #03-04
The Alpha, Singapore Science Park II, SINGAPORE 117684
Ph: (65) 6872 0828; Fax: (65) 6872 0900

InnoMedia Technology Inc.

3F, No. 3, Industrial East Road IX
Hsinchu Science-Based Industrial Park, Hsinchu TAIWAN 300
Ph: (886) 3 564 1299; Fax: (886) 3 564 1589

InnoMedia, Inc.

128 Baytech Drive
San Jose, CA 95134
Ph: (408) 432-5400; Fax: (408) 941-8152

InnoMedia, Inc. Beijing Rep. Office

Room 1328, JingXin Building
Jia 2# North Road Dong San Huan Chao Yang District
Beijing 100027 CHINA
Ph: (86) 10 65261186, (86) 10 65261189
Fax: (86) 10-65261186, (86) 10-65261189 ext 210

