

INNOMEDIA

# ESBC 9380-4B

## T1/E1 ENTERPRISE SIP GATEWAY

HIGHLY INTEGRATED ESG WITH A BUILT-IN T1 PRI (or CAS) GATEWAY, IDEAL FOR BROADBAND SERVICE PROVIDERS OFFERING SIP TRUNKING AND HIGH-SPEED DATA SERVICES TO IP AS WELL AS TDM PBX

### Key Benefits

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

Multiple functions allowing BSPs to offer bundled services

- B2BUA and Registrar for SIP trunking
- SIP ALG for hosted voice service
- Transparent bridge port for high-speed data service

Highly integrated unit for easy installation and management

- Embedded Session Border Controller (eSBC)
- FXS ports
- T1/E1 PRI (or CAS) gateway
- Internal intelligent batteries

Combined IP and TDM solution

- Extending SIP trunking services to IP and TDM PBX customers
- Allowing enterprise customers easy transition from TDM to IP telephony

Flexible SIP normalization for scalable and rapid service deployment

- Header manipulation and flow adaptation eliminate user agent signaling variations
- Profile based IPPBX configuration for easy deployment
- SIPConnect compliant
- IMS compliant
- Special call handling and SIP Normalization for Emergency Calls
- Advanced media processing for DTMF and voice CODEC transcoding

Rich VoIP metrics for performance monitoring and quality analysis

- Voice metrics: R-Factors, MOS scores
- Network metrics: Network jitter, delay, packet loss
- CDR records
- SIP Endpoint Test Agent (SETA) for quality testing
- SNMP traps for quality alarms
- Battery status

Business environment friendly FXS lines

- 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



Designed for BSPs offering SIP trunking, hosted voice, and high-speed data services, InnoMedia's ESBC 9380-4B is a highly integrated and highly manageable Enterprise SIP Gateway (ESG) that can be auto-provisioned and remotely managed. Its B2BUA and SIP ALG capabilities enable wide deployment by BSPs addressing SIP-PBX interoperability for SIP Trunking as well as providing simple NAT Traversal for Hosted PBX Services. The built-in T1/E1 PRI (or CAS) gateway further extends BSPs SIP trunking service offering to traditional PBX customers who do not want to retrofit their existing TDM equipment, or are in transition from TDM to IP based telephony solutions.

Integrated with an embedded Session Border Controller (eSBC), a T1/E1 module, 4 FXS ports, intelligent internal battery, and an interface for external UPS, InnoMedia ESBC 9380-4B offers 4 business lines for fax, credit card readers, and POTS, a SIP trunk path for enterprise IP-PBXs or TDM-based PBXs, a SIP ALG path for Hosted voice or IP Centrex Services, and a bridge/pass-through path for high speed data.



The SIP trunk path provides SIP normalization, NAT traversal, topology hiding, and security for BSPs 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 translation. 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 BSPs' network servers. The ESBC9380 provides WAN port TOS/DSCP and LAN port VLAN to allow end-to-end QoS service delivery by the BSPs.

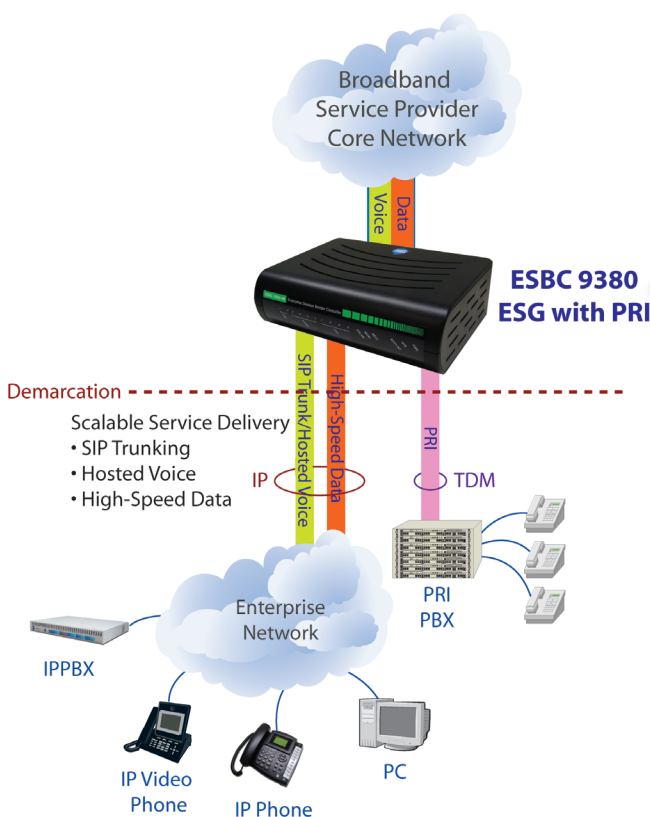
The SIP ALG path enables BSPs to offer Hosted Services with NAT traversal, TLS security for signaling, and header manipulation. It allows SIP packets of registered UAs (e.g., IP Phones) to traverse through to communicate with the network servers. The UAs register to the designated network servers, and 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 BSPs to offer high-speed data services.

The T1/E1 module can be configured to support either ISDN PRI or CAS signaling. It provides one or two T1/E1 ports for enterprise's legacy TDM PBXs. The T1/E1 TDM voice traffic is converted to VoIP and processed by the ESBC 9380-4B B2BUA module to connect to service provider's SIP trunks.

The ESBC 9380-4B, located at the edge of the BSP's access networks, can be managed by the BSP with secured HTTP-based auto-provisioning and SNMP-based remote management. It offers an ideal demarcation between the BSP and its enterprise customers.

## Delivering Scalable QoS Managed SIP Trunking, Hosted Voice, and High-Speed Data Services



- Highly Integrated: eSBC+FXS+Internal Batteries+PRI
- Uplink: 10/100BT or Gigabit Ethernet
- Downlink: 4x10/100BT or Gigabit Ethernet and 2xT1/E1PRI and 4xFXS
- Scalable Deployment
  - NAT Traversal
  - SIP Normalization with Header Manipulation
  - Profile Based IPPBX Configuration
  - Security and Protection
- Carrier-grade Monitoring
  - VoIP Metrics: R-Factor, MOS, Network Jitter, Packet Loss, Delay
  - SIP Endpoint Test Agent (SETA)
  - Call Statistics
- QoS

Figure 1

The highly integrated ESBC9380-4B 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. Two T1/E1 Trunk Ports Interfacing Traditional PBXs
4. eSBC function supporting BSP's SIP trunk business
5. SIP ALG for hosted voice SIP traffic
6. Bridge/pass-through port for BSP's high-speed data services
7. Stateful inspection protecting the eSBC, FXS, and the SIP Proxy/ALG path
8. Voice and network Monitoring

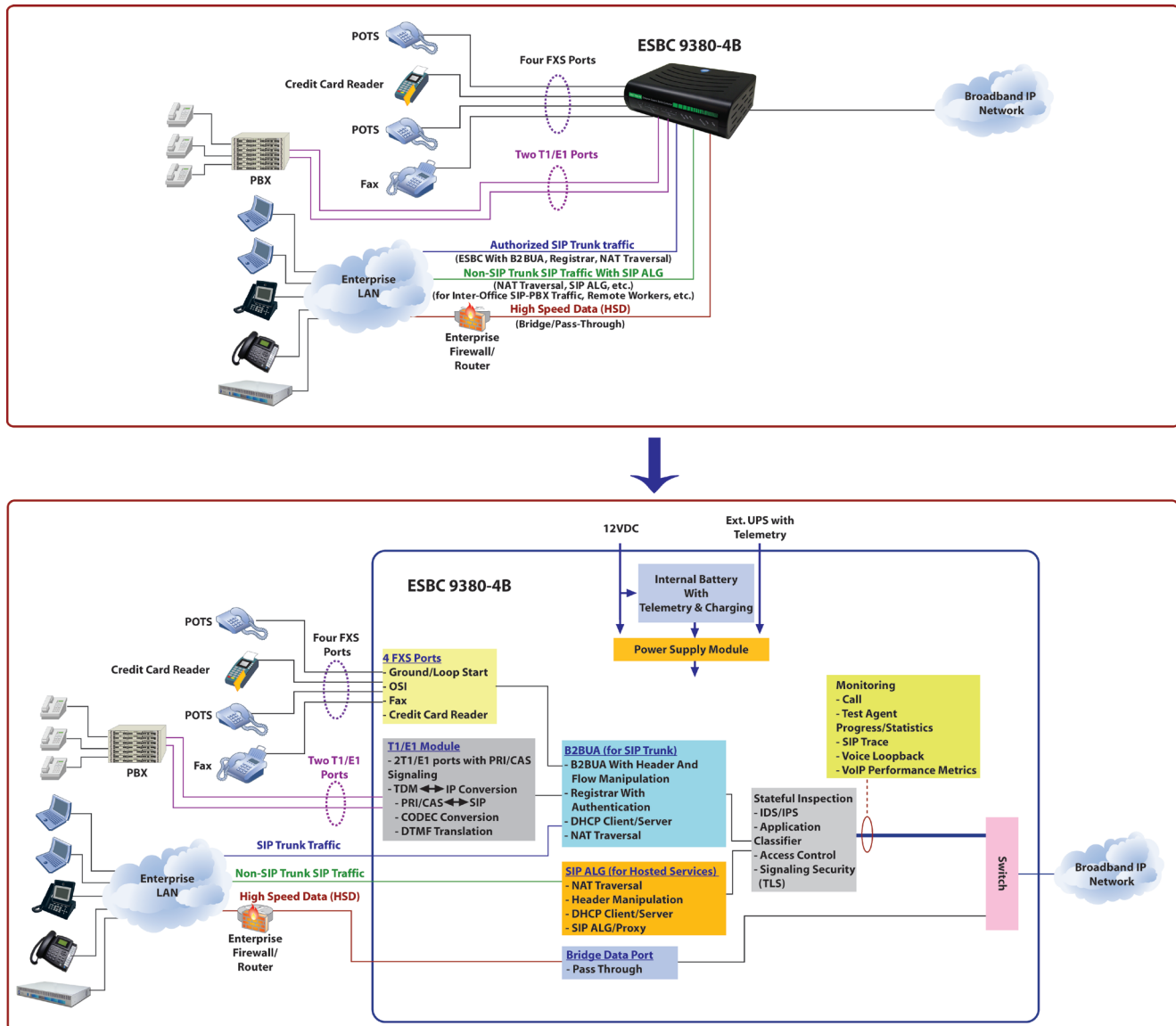


Figure 2

## Integrated Internal Battery As Well As External UPS Support

The ESBC 9380-4B is equipped with an internal battery supporting up to 4 hours of continuous talk time for all 4 telephone lines and the TDM PBX trunk lines. 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 9380-4B 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.

## Two T1/E1 Trunk Ports Interfacing Traditional PBXs

The ESBC 9380-4B includes one or two T1/E1 trunk interfaces to connect to traditional PBXs running various network interface protocols for various switch types – 4ESS, 5ESS, DMS100, NI-1, NI-2, NET5 or NTT. These trunk interfaces can be configured for either PRI or CAS signaling. The ESBC 9380-4B is responsible to convert TDM traffic into its B2BUA SIP traffic and vice versa.

## eSBC Function Supporting BSP's SIP Trunk Business

Using B2BUA, the ESBC 9380-4B supports the key functions needed by the BSPs 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 9380-4B include:

1. SIP Normalization:

Based on the B2BUA architecture, InnoMedia's ESBC 9380-4B provides Profile based settings, High-level classification for SIPConnect Adaptation, and Low-level header manipulation for SIP signaling normalization:

- Profile based settings:

ESBC 9380-4B allows parameter and option settings to adapt between the two interfaces: the WAN interface to the BSP 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 BSP'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 BSP'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 9380-4B 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 9380-4B supports TLS connection with the BSP network (authenticate BSP servers) for secured signaling transport, as well as SIP Digest authentication (challenged and authenticated by the BSP servers).
  - SIP Message Validation: ESBC 9380-4B 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

## SIP ALG for Hosted Voice SIP Traffic

The SIP ALG path is intended for BSPs offering hosted voice or IP Centrex service. It is equipped with NAT traversal and TLS signaling security, and supports up to 200 simultaneous SIP ALG sessions. The SIP ALG inspects SIP messages and states, and allows SIP packets of successfully registered UAs (e.g., IP Phones) with legitimate SIP states to communicate with the network servers. The NAT traversal module makes necessary modifications to the headers and SDPs to allow SIP packets to successfully traverse through NAT.

The SIP ALG block also contains a DHCP server with Option control (e.g., Option 66) which can be used as the designated DHCP server for the BSPs' hosted UAs (IP Phones).

## Bridge/Pass-Through Port For BSP's High-Speed Data Services

The ESBC 9380-4B allows one of its LAN ports to be configured as a bridge to its WAN interface. This bridge port can be used by the BSP to offer high speed data services. The BSP can deliver global IP addresses to its enterprise customers who can connect this bridge port to the enterprise firewall.

## 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**

☐ Enable Static Registration

☐ 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

☒ P-Preferred-Identity

☒ P-Asserted-Identity

☒ Remote-Party-ID

Forward SIP Header to SIP Server

☒ Alert-Info

☒ History-Info

☒ Diversion

☒ Forward DTMF in SIP INFO to SIP Server

☒ Strip ICE Attributes

☐ Remove Contact and Record-Route Headers in 180 Responses

☐ Add expires header in the 200 response of registration

☐ Use the SIP terminal'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

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**

☐ Play Music-On-Hold when Hold

☒ 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**

	<input type="checkbox"/> Static Registration
	<input type="checkbox"/> 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	<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"/> Use RFC 2543 Hold	
	<input type="checkbox"/> Remove Contact and Record-Route Headers in 180 Responses	
	<input type="checkbox"/> Enable rinstance	
	<input checked="" type="checkbox"/> Reuse TLS connection	
	<input type="checkbox"/> Use "lr=true" for loose routing	
	<input type="checkbox"/> Reject all received REFER	
	<input type="checkbox"/> Force send REFER even if the peer not add REFER in the Allow header	
	<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"/> Require Register event(3GPP)
	<input checked="" type="checkbox"/> Send SUBSCRIBE for Message Waiting Interval 3600 secs
	<input type="checkbox"/> Process Call Transfer and Call Forwarding Locally
	<input type="checkbox"/> Support 100rel for Outgoing calls

New Replicate Delete Restore Default

Apply Cancel

Figure 4



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

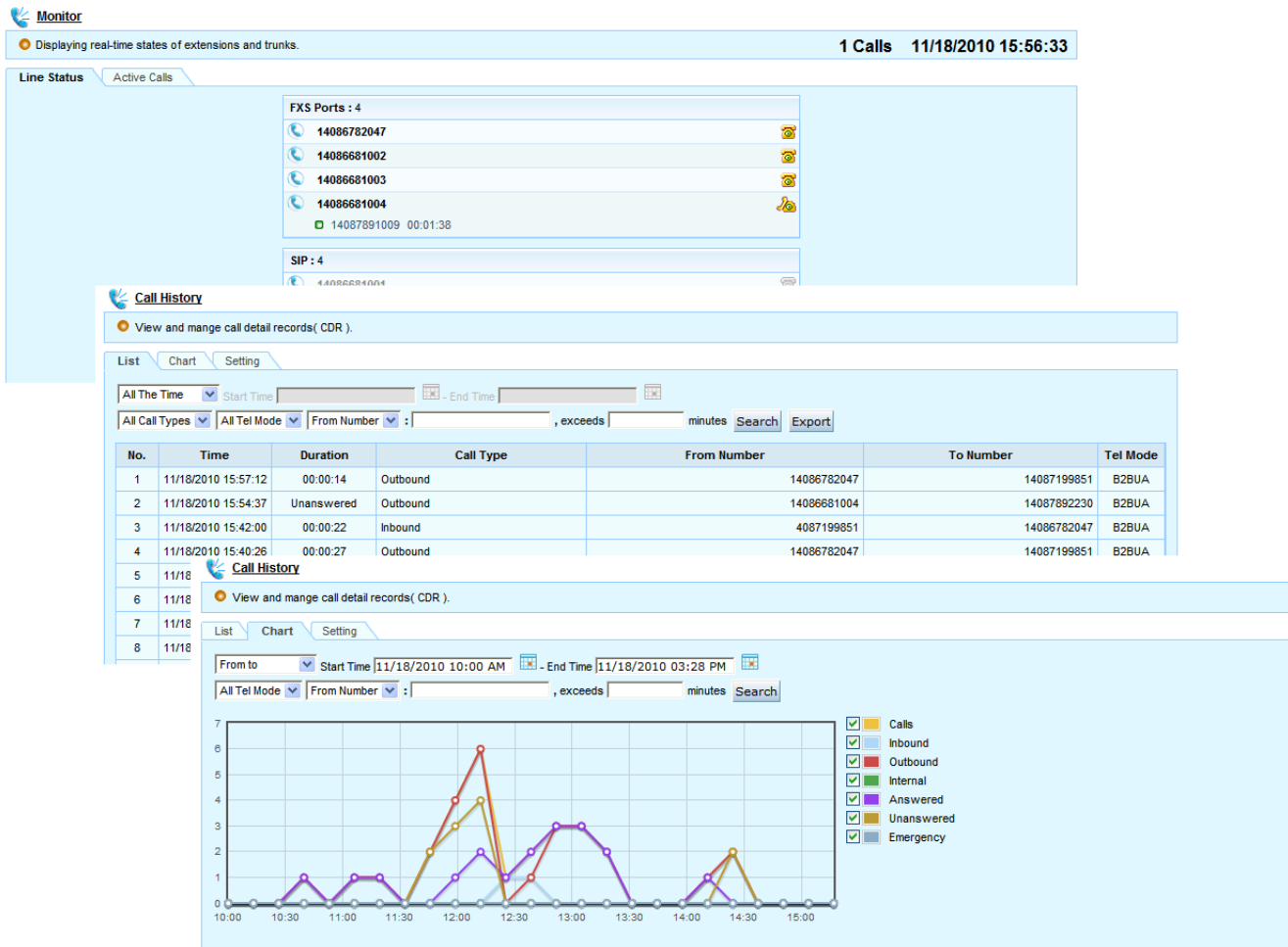


Figure 5

## CALL TRACE GUI

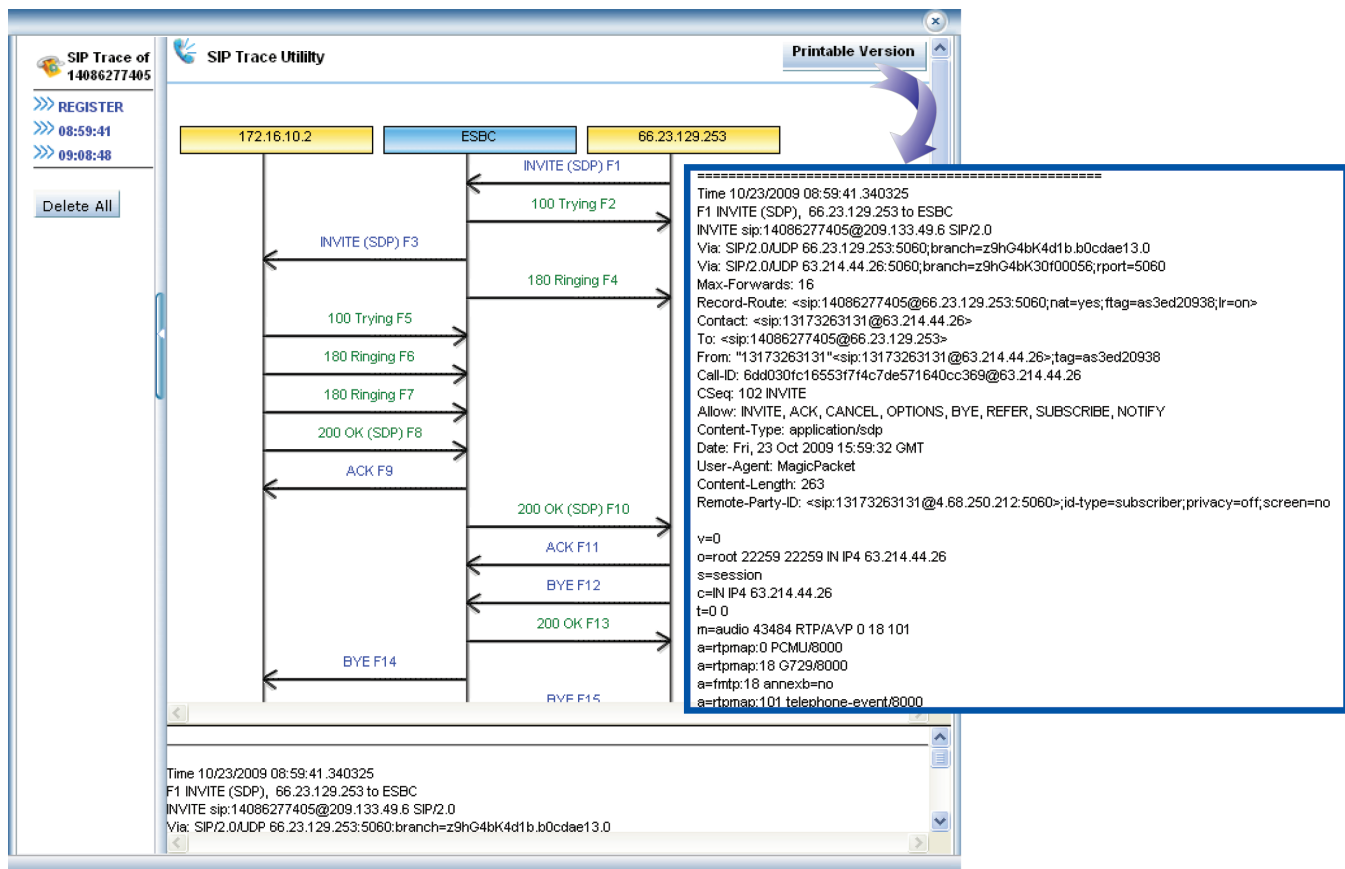
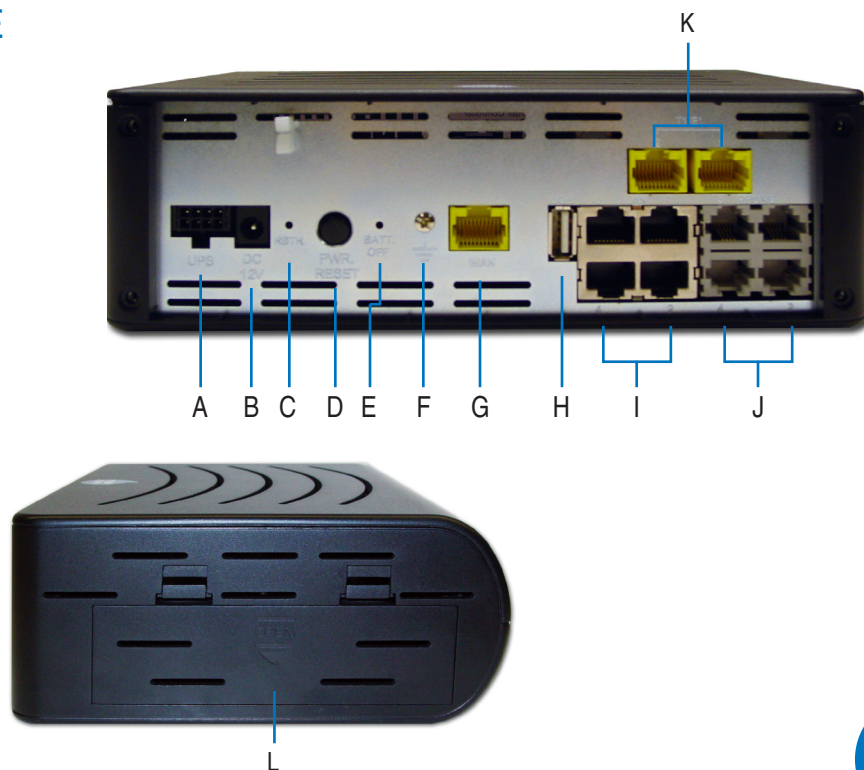


Figure 6

## ESBC INTERFACE

- A. UPS Port
- B. 12V DC Power
- C. RSTR Button
- D. Power Reset Button
- E. Battery Off
- F. External Ground
- G. WAN Port
- H. USB Port
- I. LAN 1-4
- J. Phone 1-4
- K. T1/E1 1-2
- L. Battery Compartment



## SPECIFICATIONS

### Product Interfaces

Category	Specification
Service Provider Interface	1 10/100/1000 BaseT Ethernet
Telephone Interface	4 FXS Voice Ports
PBX Interface	Up to 2 T1/E1 trunk ports with PRI or CAS signaling
User Data Interface	4 10/100/1000 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, Voice Quality Monitoring, Test Agent for Test Calls, R-Factor and MOS Calculation Media Processing (DTMF and Voice CODEC Transcoding)
Networking Features	Built-in DHCP Server NAT Capabilities for Simultaneous SIP User Accounts Static IP Routing NAT Traversal UPnP DMZ SIP Application Layer Gateway Network Access Control by IP Address, Subnet, Port Number, MAC Address or Destination Domain Name Web GUI with 3 Levels of Page Permissions Auto-Backup of Configuration
T1/E1 Protocols	<b>Frames and Formats:</b> T1: AMI/B8ZS E1: AMI/HDB3 <b>Signalling:</b> CAS: Loop-start, Ground-start, E&M Wink start, E&M Immediate CCS: Q.921, Q.931, Q.932 PRI: 4ESS, 5ESS, DMS100, NI-1, NI-2, NET5, NTT
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

## SPECIFICATIONS cont.

Category	Specification												
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)												
Signal Processing	<table> <tr> <td>G.168 Echo cancellation</td><td>Loop Back</td></tr> <tr> <td>FAX (T.38 and G.711 fall-back)</td><td>FXS voltage drop when CA or RF fails</td></tr> <tr> <td>Caller ID FSK signal regeneration</td><td>Pulse Dialing</td></tr> <tr> <td>Line reversal</td><td>Foreign voltage detection</td></tr> <tr> <td>Ground Start/Loop Start</td><td></td></tr> </table>	G.168 Echo cancellation	Loop Back	FAX (T.38 and G.711 fall-back)	FXS voltage drop when CA or RF fails	Caller ID FSK signal regeneration	Pulse Dialing	Line reversal	Foreign voltage detection	Ground Start/Loop Start			
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Tones	<table> <tr> <td>Ring back tone</td><td>Busy tone</td></tr> <tr> <td>Recorder tone</td><td>5 distinct rings</td></tr> <tr> <td>Dial tone</td><td>Confirmation tone</td></tr> <tr> <td>Ring splash</td><td>Stutter tone</td></tr> <tr> <td>Off hook warning tone</td><td>Message waiting indicator (MWI)</td></tr> <tr> <td>Caller ID generation &amp; call waiting tone</td><td>Configurable ring frequency</td></tr> </table>	Ring back tone	Busy tone	Recorder tone	5 distinct rings	Dial tone	Confirmation tone	Ring splash	Stutter tone	Off hook warning tone	Message waiting indicator (MWI)	Caller ID generation & call waiting tone	Configurable ring frequency
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Off hook warning tone	Message waiting indicator (MWI)												
Caller ID generation & call waiting tone	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	<p>Access components implemented:</p> <p>TFTP, FTP, HTTP 1.0, SNMP, Telnet, DHCP &amp; DNS</p> <p>Works with any SNMP (v.1-3) -based EMS</p> <p>Offers web-based access as well as TFTP-based remote software downloads or upgrades</p> <p>Data monitoring throughput tools</p>												
QoS	WAN: TOS, DSCP, LAN: VLAN Tagging of up to 1,000 VLAN groups												

## SPECIFICATIONS cont.

## 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
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

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