

# ESBC 8528-4B

ENTERPRISE SIP GATEWAY (ESG) WITH DOCSIS 2.0 CABLE MODEM

HIGHLY INTEGRATED ENTERPRISE SIP GATEWAY WITH A BUILT-IN DOCSIS 2.0 CABLE MODEM, IDEAL FOR MSOs OFFERING SIP TRUNKING, HOSTED PBX, AND HIGH-SPEED DATA SERVICES TO ENTERPRISE CUSTOMERS



## OVERVIEW

ESBC 8528-4B is a highly integrated and versatile Enterprise SIP Gateway (ESG) solution for the cable industry. It combines Enterprise Session Border Controller, business line FXS ports, switched data port, DOCSIS 2.0 cable modem, and internal battery in one unit. ESBC 8528-4B can be used by service providers as an ESG with B2BUA for IP PBXs (Figure 1) or with SIP ALG for IP Centrex (Figure 2).

Figure 1. ESG with B2BUA for IP PBXs

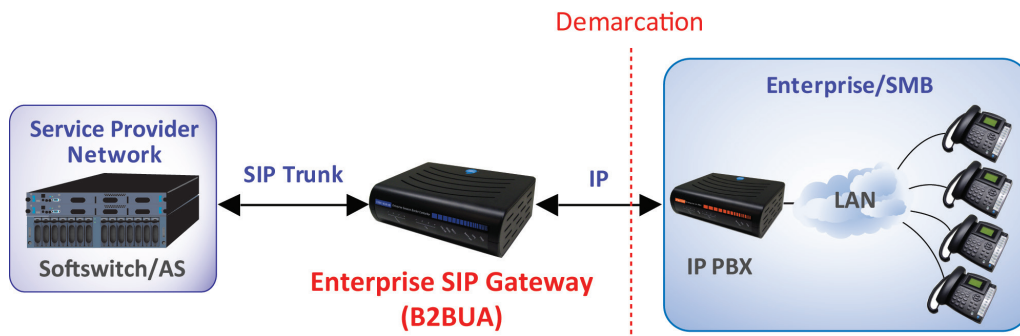
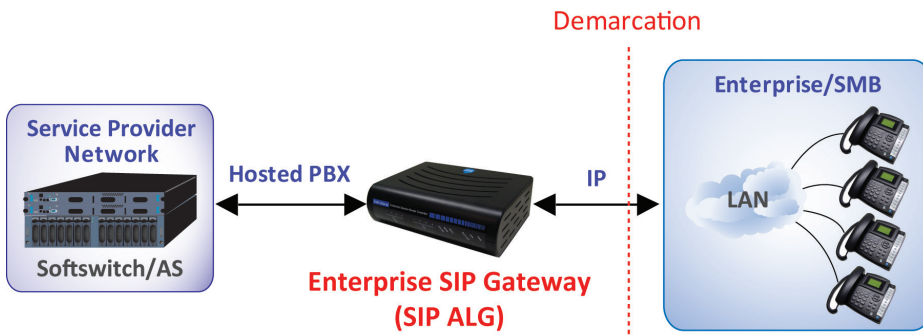


Figure 2. ESG with SIP ALG for IP Centrex



## SERVICE USAGE EXAMPLES

Designed for MSOs offering SIP trunking, hosted voice, and high-speed data services, InnoMedia's ESBC 8528-4B is a highly integrated and highly manageable Enterprise Session Border Controller (ESBC) that can be auto-provisioned and remotely managed. With InnoMedia's exclusive *Smart-DQoS*™ technology enabling device-initiated DQoS UGS service flow establishment, ESBC 8528-4B is ideally suitable for MSOs offering bundled services with end-to-end quality of service over HFC cable plants. Its B2BUA and SIP ALG capabilities enable wide deployment by MSOs addressing SIP-PBX interoperability for SIP Trunking as well as providing simple NAT Traversal for Hosted PBX Services, and its embedded DOCSIS 2.0 cable modem with 8x4 channel bonding and 14 UGS SIDs allows high speed data throughput and 14 simultaneous SIP sessions without requiring MGPI support.

The two typical service scenarios for ESBC 8528-4B are:

1. MSOs delivering high-speed Internet access as well as SIP trunks to enterprises with IP-based PBXs (Figure 3).
2. MSOs delivering high-speed Internet access as well as hosted PBX/IP Centrex to enterprises which will transition from legacy Centrex to IP Centrex (Figure 4).

Figure 3. MSOs delivering high-speed Internet access as well as SIP trunks to enterprises with IP-based PBXs

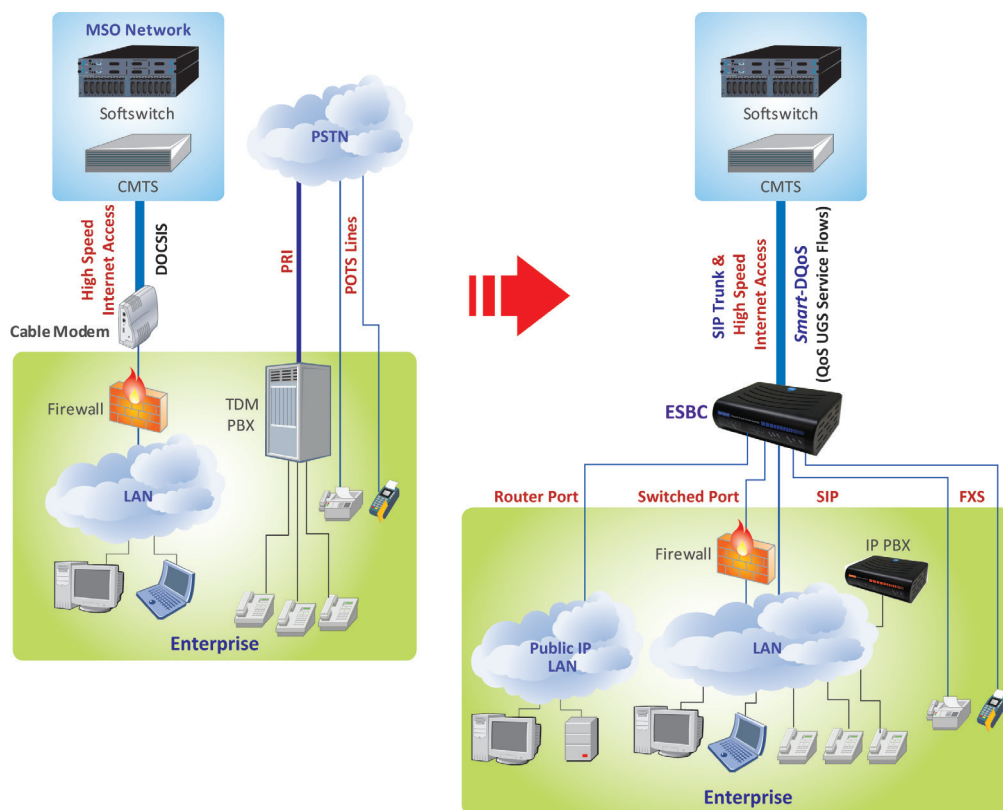
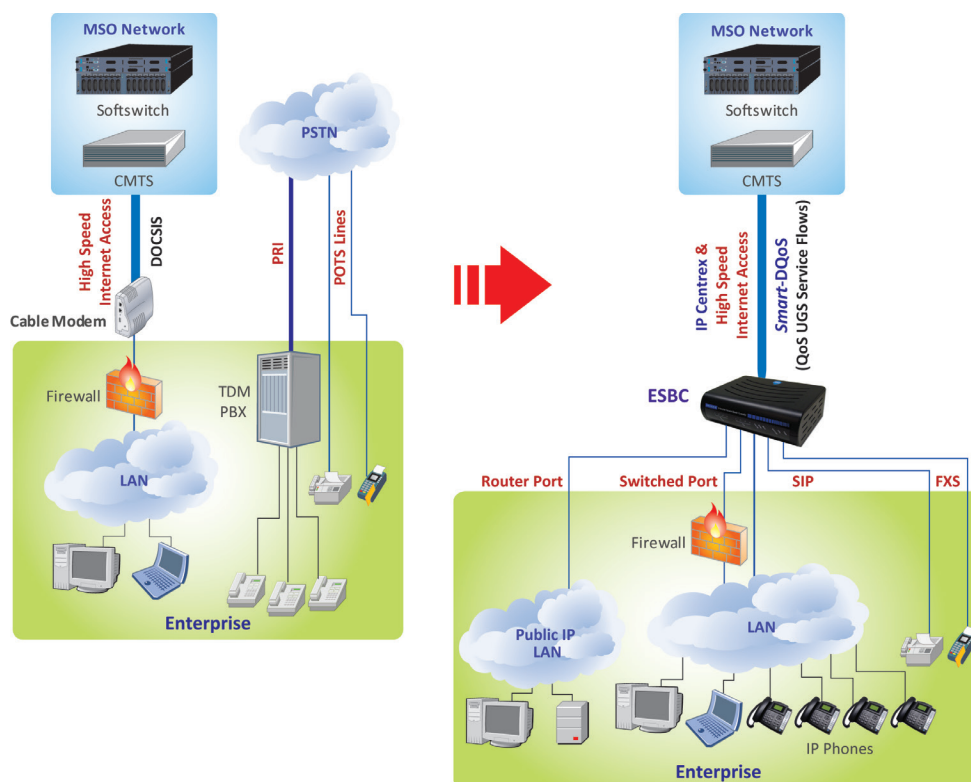


Figure 4. MSOs delivering high-speed Internet access as well as hosted PBX/IP Centrex to enterprises



## SUMMARY OF KEY FEATURES AND BENEFITS

Functional Blocks/Categories	Features	Benefits
DOCSIS 2.0 embedded Cable Modem (eCM)	DOCSIS 2.0 with 8x4 channel bonding and 14 UGS SIDs	1. 300 Mbps downstream and 120 Mbps upstream BW 2. 14 UGS SIDs for 14 DQoS ensured voice sessions
Quality of Service	QoS: 1. WAN: <i>Smart-DQoS™</i> for cable 2. LAN: VLAN with 1000 groups	Device initiated DQoS UGS for DQoS service flows ensured voice sessions a. Instant service quality improvement b. Minimum infrastructure investment c. Time to market

## SUMMARY OF KEY FEATURES AND BENEFITS CONT.

Functional Blocks/Categories	Features	Benefits
Embedded Session Border Controller (ESBC)	Registrar and B2BUA 1. Implicit, explicit, and static (no) registration 2. Header manipulation/SIP normalization 3. Advanced media processing for DTMF and voice CODEC transcoding 4. NAT traversal 5. SIPConnect compliant 6. IMS ready 7. Profile-based Multiple SIP Proxy Support 8. Built-in SIP Forking Feature for multiple SIP devices within the LAN	<ul style="list-style-type: none"> <li>• Highly interoperable between service provider and enterprise equipment</li> <li>• No interference with enterprise firewall setting</li> <li>• Reliable and scalable SIP trunking service delivery</li> <li>• Multiple Proxy Support allows separate and independent Softswitch (thus features, billings, etc.) to manage the FXS ports and LAN UA</li> </ul>
	SIP ALG 1. Header manipulation 2. NAT traversal 3. SIPConnect compliant 4. IMS ready	<ul style="list-style-type: none"> <li>• Highly interoperable between service provider and enterprise equipment</li> <li>• No interference with enterprise firewall setting</li> <li>• Reliable and scalable hosted PBX/IP Centrex service delivery</li> </ul>
High-speed bridge port	Transparent gigabit data port	High-speed Internet data service delivery
Dedicated router port	Separate router port with RIPv2 protocol	Data Service delivery for end-customer who request to use multiple public IP Addresses within the ESBC
FXS commercial voice ports	Four FXS ports with commercial line features 1. Low-speed (high-speed) modems 2. T.38	Business Friendly 1. Analog PBX interconnect 2. Fax 3. House wiring 4. Credit card reader
Internal battery	4 hours of talk time	Power backup for primary line services
Security Features	1. Stateful inspection 2. TLS for signaling 3. Access control	1. No intrusion into enterprise networks via SIP path 2. Secured signaling
Monitoring	1. VoIP performance metrics a. Voice: R-factor & MOS scores b. Network: jitter, delay, packet loss 2. CDR records 3. SIP End-point Test Agent 4. SNMP traps for quality alarms 5. Data monitoring throughput tools 6. VPN Server to manage SIP devices on the enterprise network	Allowing service providers to offer Service Level Agreement (SLA) based 1st tier quality services to enterprise customers
911 emergency call handling	1. Line pre-emption 2. SIP signaling a. Emergency caller ID b. Priority header 3. Media: a. G.711 b. Disable VAD 4. QoS a. DQoS pre-emption b. DiffServ 5. Syslog and SNMP trap	Allowing service providers to offer SIP trunking services with 911 support for primary line based services

## PRODUCT FEATURE DESCRIPTION

Integrated with DOCSIS 2.0 embedded cable modem (eCM), embedded Session Border Controller (eSBC), the *Smart-DQoS*™ technology, intelligent internal battery, and an interface for external UPS, InnoMedia ESBC 8528-4B offers 4 FXS ports, a SIP trunk path for enterprise IP-based UAs (IP-PBXs), a SIP ALG path for Hosted IP-PBX or IP Centrex Services, and a bridge/pass-through path for high speed data.

The ESBC 8528-4B eCM is a DOCSIS 2.0 cable modem with 8x4 channel bonding and 14 UGS SIDs. It offers a maximum of over 300 Mbps of downstream throughput and 120 Mbps of upstream throughput. The 14 UGS SIDs make it possible to have 14 simultaneous UGS SIP trunk sessions or SIP ALG sessions without using the Multiple Grants per Interval (MGPI) scheme.

*Smart-DQoS*™ enables ESBC8528-4B to intelligently initiate and manage DQoS Unsolicited Grant Service (UGS) service flows based on user and SIP signaling events, while directing non-real time data traffic to DOCSIS Best Effort (BE) service flows. It instantly enables MSOs to offer bundled services with end-to-end QoS without having to wait for PacketCable Multimedia based network infrastructure enhancement.

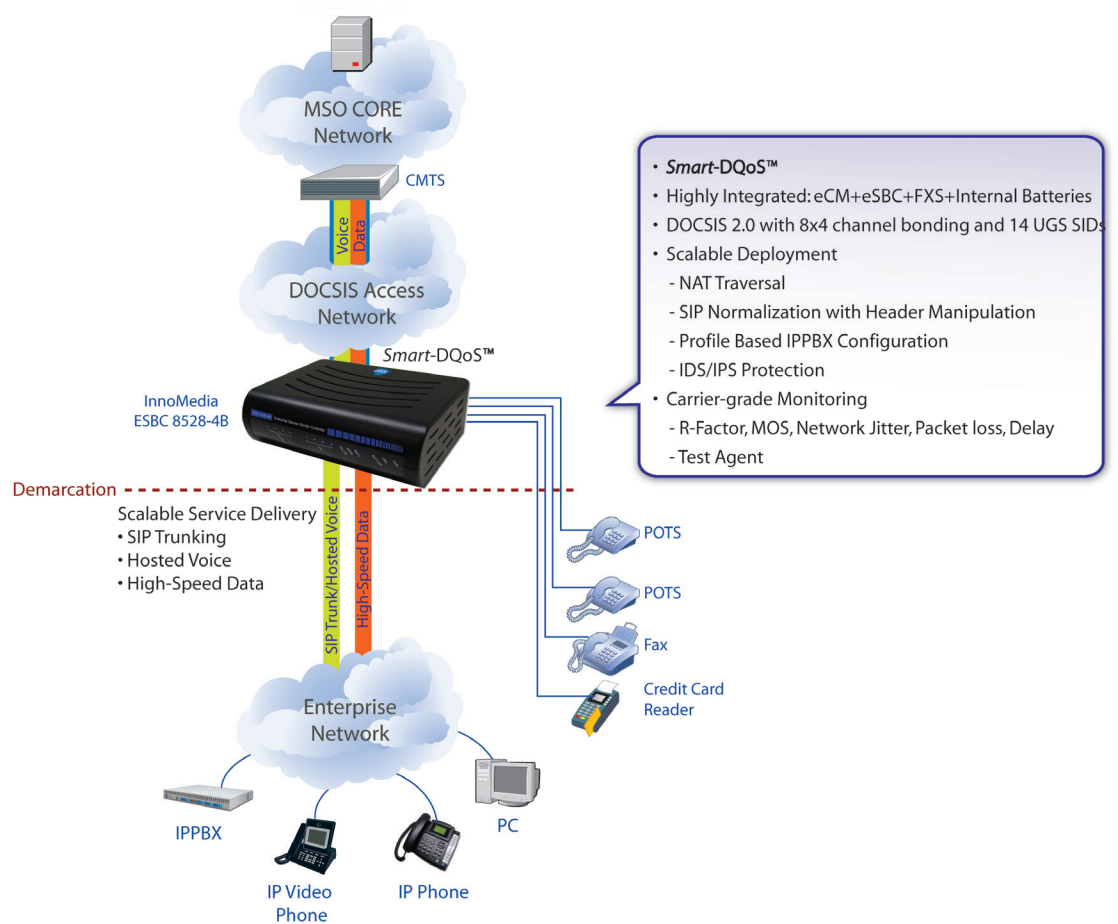
The SIP trunk path provides SIP normalization, NAT traversal, topology hiding, and security for MSOs offering SIP trunking service to enterprise customers with diverse IPPBX and network configurations. It includes B2BUA for SIP normalization, a Registrar for User Agent (UA) registration, TLS block for 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 MSOs' network servers. Together with *Smart-DQoS*™, the MSO is able to offer QoS ensured SIP trunking service.

The SIP ALG path enables MSOs 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. Together with *Smart-DQoS*™, the MSO is able to offer QoS ensured hosted voice/IP Centrex service.

The bridge path is a transparent pass-through port, allowing uninterrupted high-speed data to go through. It is intended for MSOs to offer high-speed data services. The router path is also a transparent pass-through port dedicated for MSOs to offer data services to enterprise customers requesting for multiple public IP Addresses.

The ESBC 8528-4B, located at the edge of the HFC access networks, can be managed by the MSO with secured HTTP-based auto-provisioning and SNMP-based remote management. It offers an ideal demarcation between the MSO and its enterprise customers.

Figure 5. ESBC 8528-4B delivering high-speed Internet access, SIP trunking, and FXS ports with Business Line Features

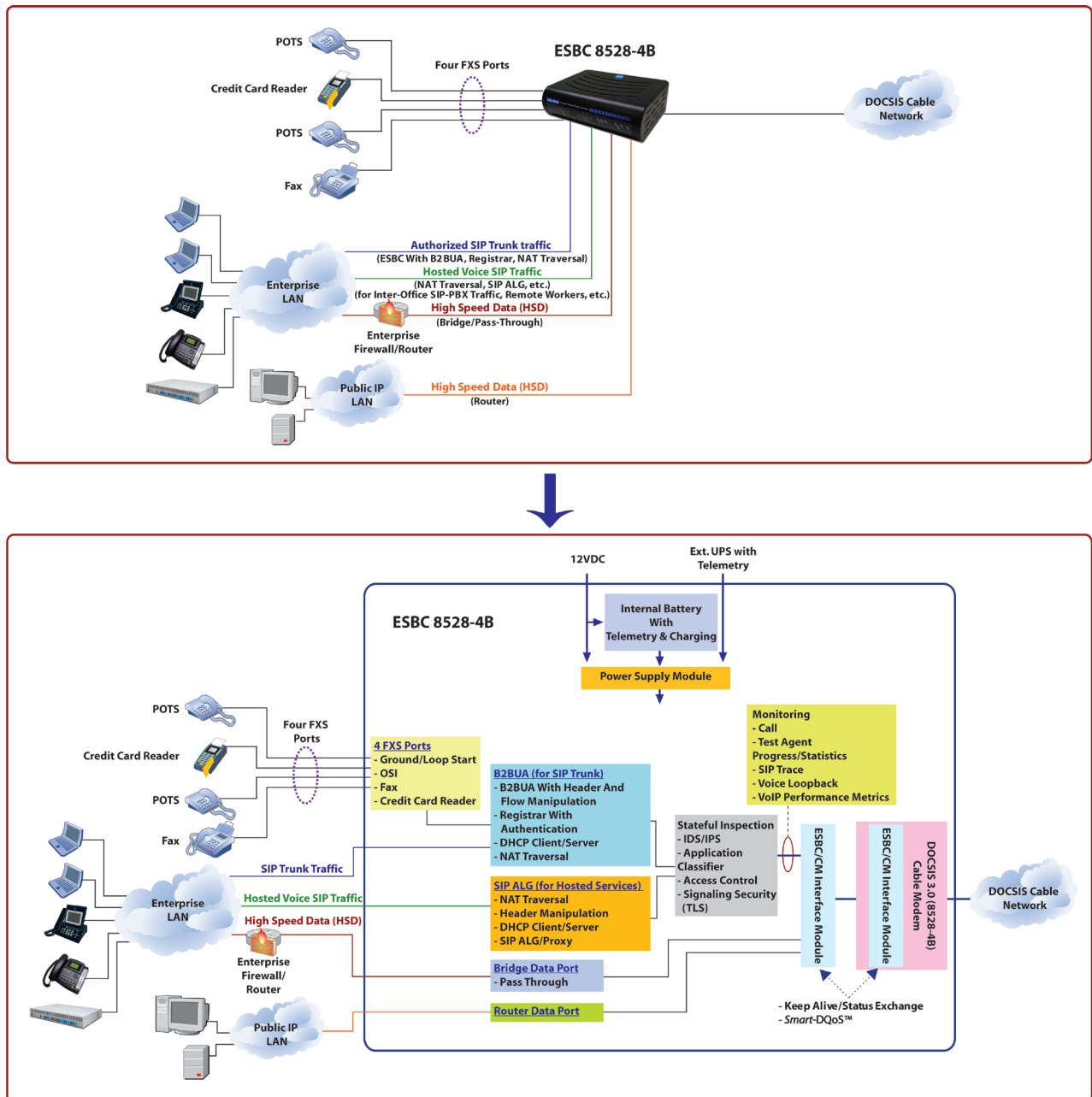




The highly integrated ESBC 8528-4B includes the following key functional blocks:

1. Embedded DOCSIS 2.0 cable modem with *Smart-DQoS™*
2. Intelligent internal battery as well as external UPS support
3. Four FXS ports with business friendly features
4. eSBC function supporting MSO's SIP trunk business
5. SIP ALG for hosted voice SIP traffic
6. Bridge/pass-through port for MSO's high-speed data services
7. Stateful inspection protecting the eSBC, FXS, and the SIP Proxy/ALG path
8. Voice and network Monitoring

Figure 6. ESBC 8528-4B functional block diagram



## Embedded DOCSIS 3.0 Cable Modem

Integrated with an embedded DOCSIS 2.0 cable modem module, the ESBC 8528-4B works with or without policy servers to manage DQoS service flows to ensure voice service QoS. With *Smart-DQoS™*, the ESBC 8528-4B can also intelligently initiate DQoS UGS service flows without the need of a policy server. The ESBC is aware of creating new SIP Trunking Sessions, therefore, can initiate and manage Dynamic Service Flows via DSX (DSA, DSC, and DSD) message exchanges with the CMTS.

With 8x4 channel bonding, the ESBC 8528-4B offers a maximum of over 300 Mbps of downstream throughput and 120 Mbps of upstream throughput. The eCM's 14 UGS SIDs allows 14 simultaneous UGS SIP trunk sessions or SIP ALG sessions without using the Multiple Grants per Interval (MGPI) scheme.

The ESBC 8528-4B also supports PacketCable Multimedia-based MGPI to allow multiple calls within one service flow, thus, allowing more than 14 simultaneous voice calls in the 14 available Unsolicited Grant Service (UGS) service flows. For example, 24 simultaneous calls can be supported with 14 UGS Service Flows. The embedded DOCSIS 2.0 cable modem module can be provisioned via standard DOCSIS provisioning.

## Integrated Internal Battery As Well As External UPS Support

The ESBC 8528-4B is equipped with an internal battery supporting up to 4 hours of continuous talk time for the SIP trunk traffic and all 4 telephone 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 8528-4B includes 4 voice ports that deliver revenue generating telephony services to their enterprise customers. It has rich set of business features including 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 Supporting Service Providers' SIP Trunking Business

Using B2BUA, the ESBC 8528-4B supports the key functions needed by the MSOs 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 8528-4B include:

1. SIP Normalization:  
Based on the B2BUA architecture, InnoMedia's ESBC 8528-4B provides Profile based settings, High-level classification for SIPConnect Adaptation, and Low-level header manipulation for SIP signaling normalization:
  - Profile based settings:  
ESBC 8528-4B allows parameter and option settings to adapt between the two interfaces: the WAN interface to the MSO servers with support for multiple SIP Proxies, 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 7).
    - Based on the MSO's network servers, the parameters/options are captured in the corresponding SIP Trunk profile (see Figure 8).

The SIP normalization and adaptation mechanisms are:





- High-level classification for SIPConnect Adaptation (see Figure 8):
    - Adapts between non-SIPConnect-compliant UA/SIP-PBXs and MSO'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 7)
    - 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 8528-4B 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 8528-4B supports TLS connection with the MSO network (authenticate MSO servers) for secured signaling transport, as well as SIP Digest authentication (challenged and authenticated by the MSO servers).
  - SIP Message Validation: ESBC 8528-4B validates all SIP messages
5. Emergency Call Handling (Figure 11)
- 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 MSOs 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 MSOs' hosted UAs (IP Phones).

## Bridge/Pass-Through and Router Ports For MSO's High-Speed Data Services

The ESBC 8528-4B allows its LAN ports to be configured as a bridge or a router to its WAN interface. This bridge port can be used by the MSO to offer pass-through high speed data services. The MSO can also deliver multiple public IP addresses to its enterprise customers by connecting to a dedicated Router port. The Router port is equipped to use RIPv2 protocol to advertise the used public IP Addresses to the service provider network.

## Stateful Inspection

A stateful inspection with IDS/IPS is used 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.

## Voice and Network Performance Monitoring

ESBC 8528-4B offers carrier-grade monitoring features, allowing service providers to offer SLA based SIP trunking services to their enterprise customers. The monitoring features including voice metrics with R-factor and MOS scores (Figure 13), network metrics with jitter, delay, and packet loss, CDR records and real-time UA & SIP trunk call states (Figure 9), SIP Call Trace (Figure 10), battery status (Figure 12), packet loopback for server-based Voice Quality Monitoring, and SNMP Traps based on thresholds of network call parameters. The voice and network metrics are divided into ESBC LAN network and WAN network (Figure 13), making it easier for service providers to analyze the system performance bottlenecks. The ESBC 8528-4B also has an embedded SIP End-point Test Agent (SETA) that allows test calls to be made manually (Figure 14) or programmed at scheduled times (Figure 15) for quality tests or monitoring. The ESBC works in conjunction with InnoMedia's DMS Server for monitoring and analysis of MOS scores, Data Network Traffic and CDR information. The ESBC also has a built-in VPN server that allows the service provider to manage and troubleshoot end devices connected to the enterprise LAN network.

## UA/SIP-PBX PROFILE

Figure 7

**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

## SIP TRUNK PROFILE

Figure 8

**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

Forward SIP Header to SIP Server

☒ Alert-Info

☒ 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

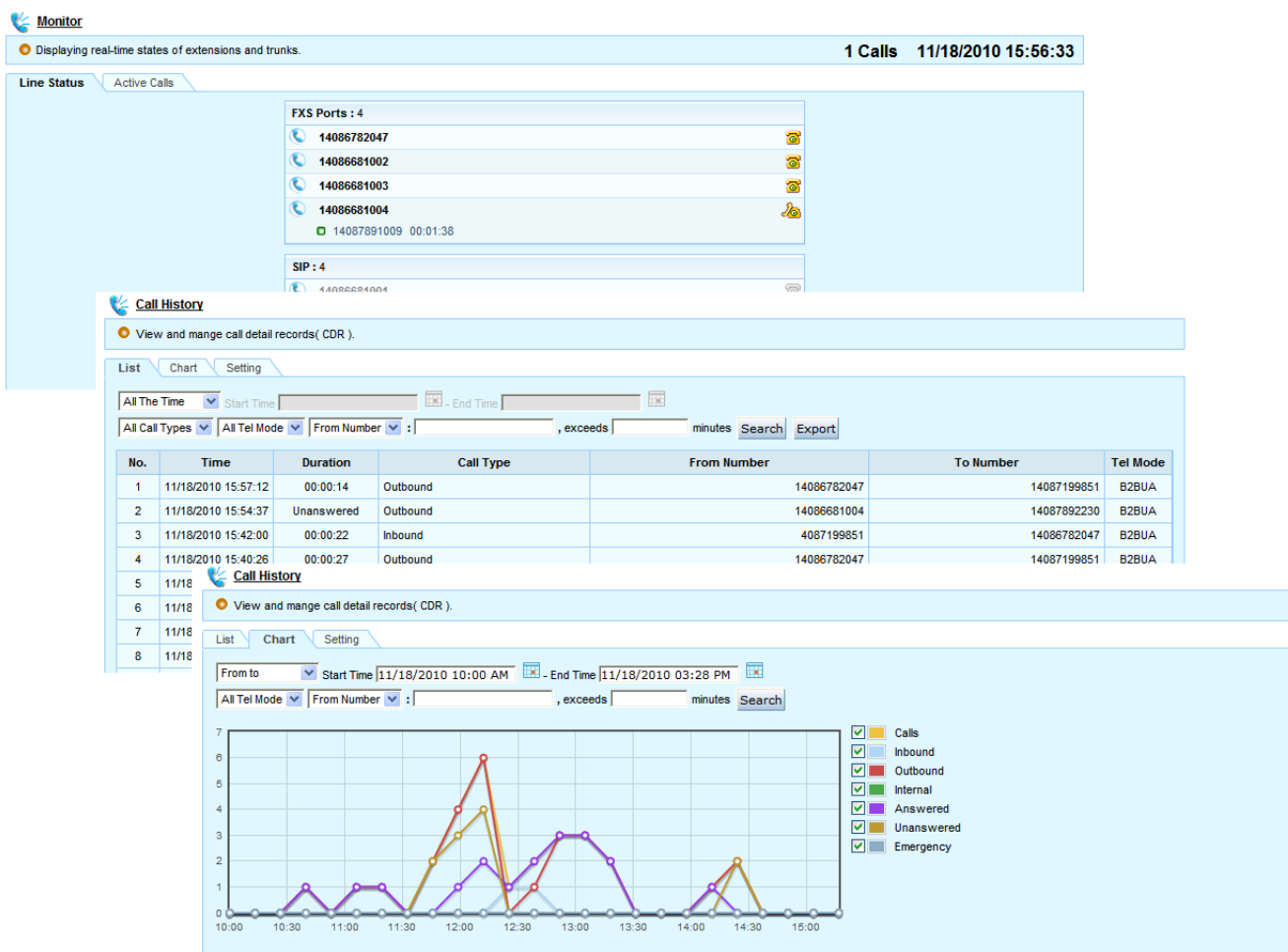
☐ Support 100rel for Outgoing calls

New Replicate Delete Restore Default

Apply Cancel

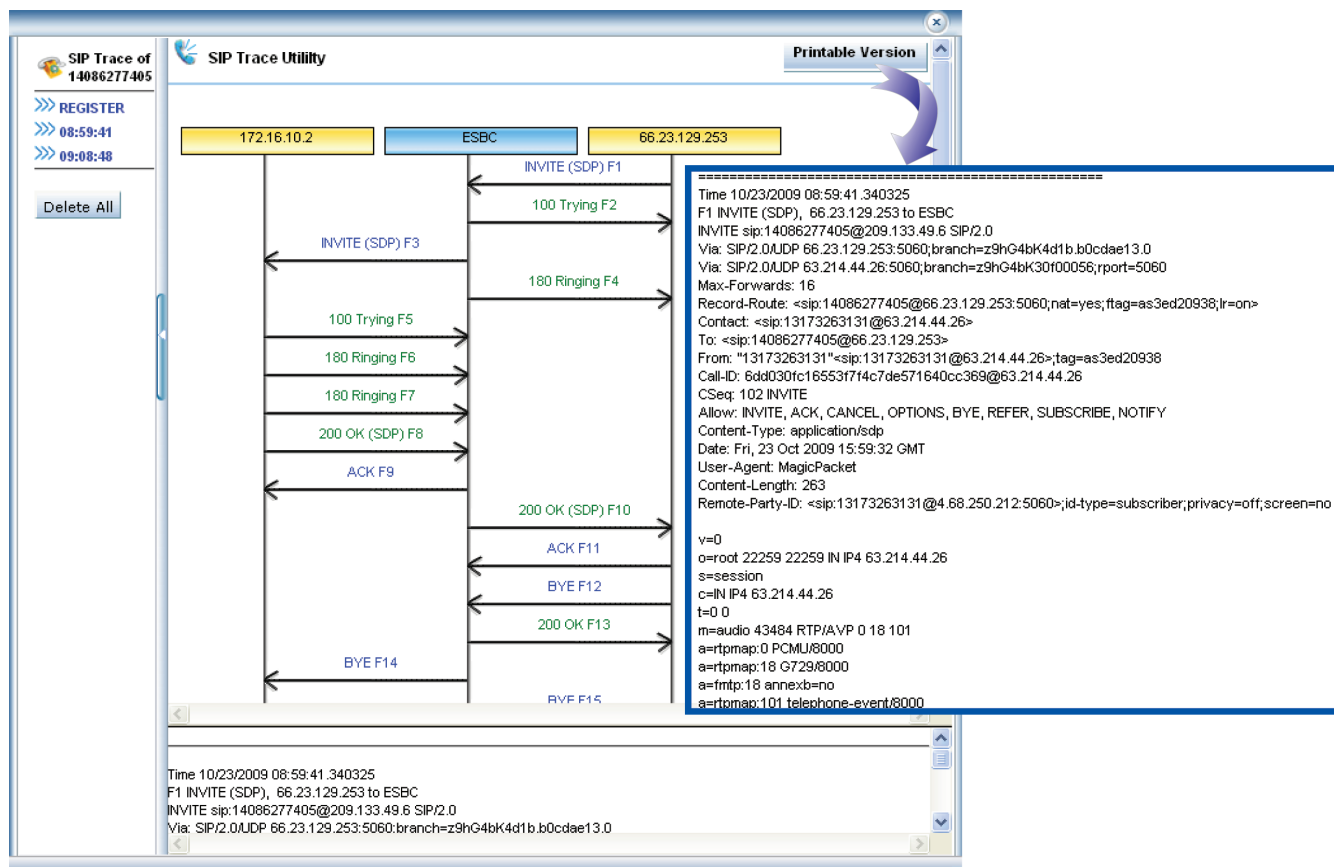
## REAL-TIME LINE CALL STATES, CDR, AND CALL STATISTICS

Figure 9



## CALL TRACE GUI

Figure 10





## EMERGENCY CALL HANDLING



Figure 11

**Emergency Call Setting**

Configure the Emergency Call basic settings.

Numbers Setting

Override Caller Information	<input checked="" type="checkbox"/> Enabled
	Caller ID <input type="text" value="14086681000"/> Display Name <input type="text" value="emergency"/>
Override Trunk Group Identifier	<input checked="" type="checkbox"/> Set SIP Priority Header to "emergency"
	<input type="checkbox"/> Enabled
	tgrp <input type="text"/> trunk-context <input type="text"/>
DSCP for Media packet	<input type="checkbox"/> Enabled
	Value <input type="text" value="aa"/> (Hex, 00-FF)
	<input type="checkbox"/> Send SNMP Trap

### Call History

View and manage call detail records( CDR ).

List Chart Setting

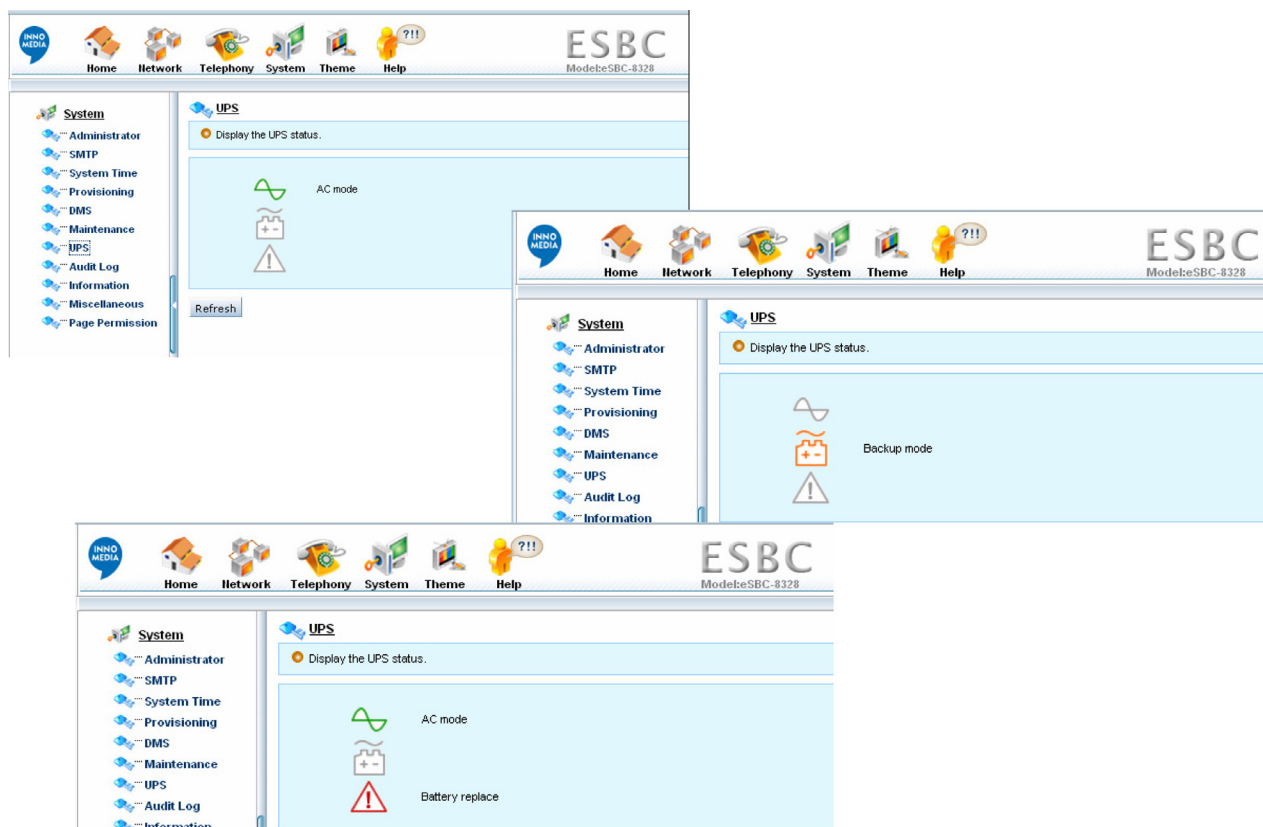
All The Time  Start Time  - End Time

All Call Types  All Tel Mode  From Number  :  , exceeds  minutes

No.	Time	Duration	Call Type	From Number	To Number	Tel Mode
1	11/16/2010 16:13:48	Unanswered	Outbound	14086782049	14085288811	B2BUA
2	11/16/2010 16:11:25	Unanswered	Outbound	14086782049	14087891009	B2BUA
3	11/16/2010 16:09:25	Unanswered	Outbound	14086782049	14087891007	B2BUA
4	11/16/2010 15:54:04	Unanswered	Emergency Call	14086782049	911	B2BUA
5	11/16/2010 15:51:50	Unanswered	Emergency Call	14086782049	911	B2BUA
6	11/16/2010 14:01:11	00:00:16	Emergency Call	14086782049	911	B2BUA
7	11/16/2010 14:00:41	Unanswered	Emergency Call	8000	911	B2BUA
8	11/16/2010 13:59:13	00:00:15	Emergency Call	14086782049	911	B2BUA
9	11/16/2010 13:59:06	Unanswered	Emergency Call	8000	911	B2BUA
10	11/16/2010 13:58:57	Unanswered	Emergency Call	14086782049	911	B2BUA

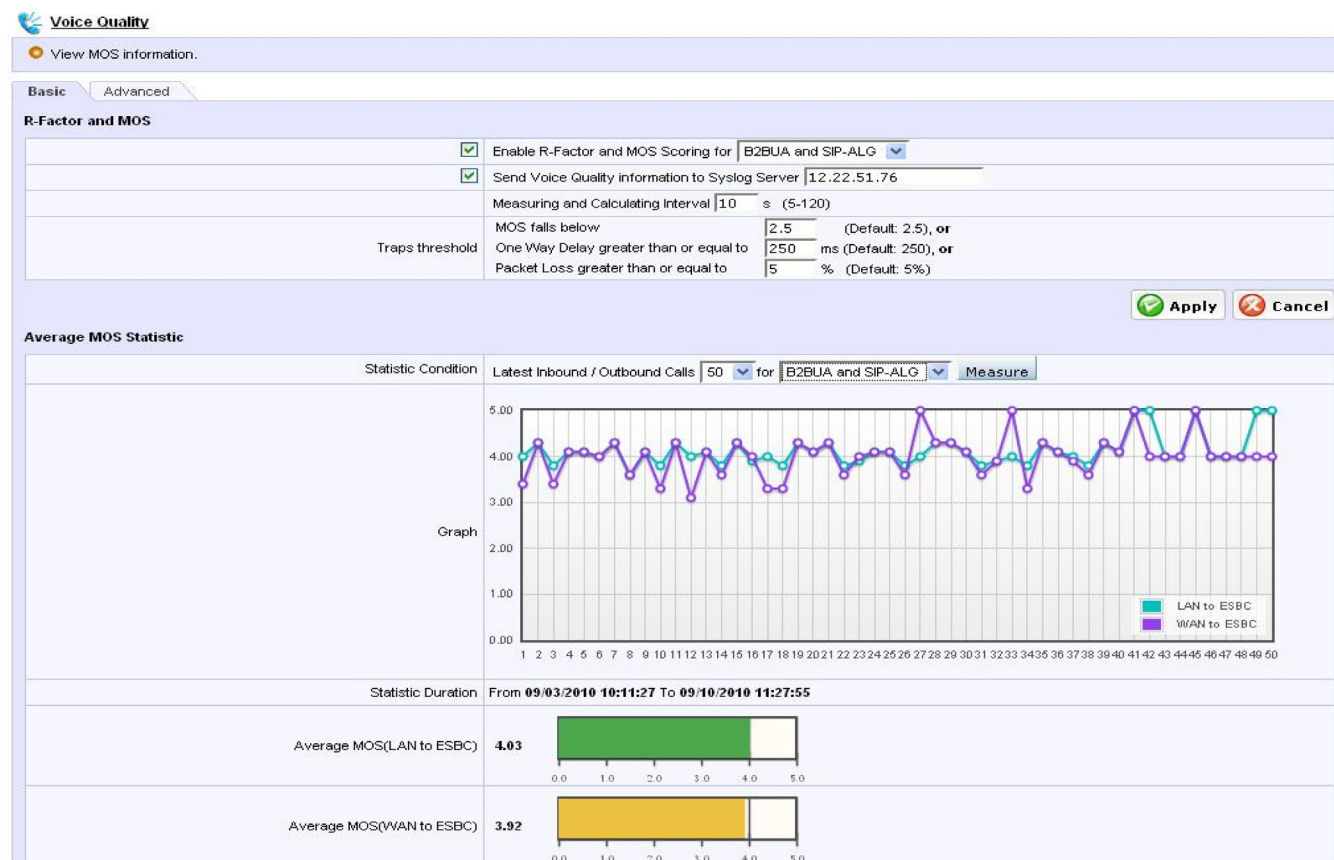
## BATTERY STATUS DISPLAY

Figure 12



## ESBC VOICE QUALITY MONITORING AND SNMP TRAP THRESHOLD SETTING

Figure 13. Performance threshold settings for SNMP traps, and averaged MOS scores for LAN and WAN networks



## TEST AGENT MANUAL TEST CALL GUI AND WAN MOS DISPLAY

Figure 14

**Test Agent Call Control**  
Configure Test Agent Call Control.

Call control    Setting

**Test Agent**

Number	14087898810		
Registration State	Connected	Register	De-Register
Schedule Test	Disabled		
State	<b>14087898810</b> 14084325470    00:00:05 <b>Hang Up</b>		

**Manual Test**

Destination Number	14084325470
--------------------	-------------

**Latest Test Result**

	<b>Show</b>
Test Type	Manual call
State	Successful
Time	09/10/2010 11:24:59
From	14087898810
To	14084325470
Call Type	Outbound
Duration	00:00:13
Voice Quality	WAN Side Average MOS = 4.0

## TEST AGENT SCHEDULED TEST CALL SETTINGS

Figure 15

**Test Agent Parameters**  
Configure Test Agent parameters.

Call control    Setting

**Test Agent**

	<input checked="" type="checkbox"/> Enabled
User ID	14087898810
Display Name	Test Agent
Auth ID	14087898810
Auth Password	•••••
Trunk SIP Profile	SIP-Trunking
Registration State	Connected    Register    De-Register

**Audio File**

Codec	G.729A/G.729
File used during calls	<input checked="" type="radio"/> Default <input type="radio"/> Customize    Upload    (No uploaded file)

**Auto Disconnect Call**

	<input checked="" type="radio"/> When Finish Play Audio file just one time. <input type="radio"/> Duration 60 s (Loop Playback Audio file)
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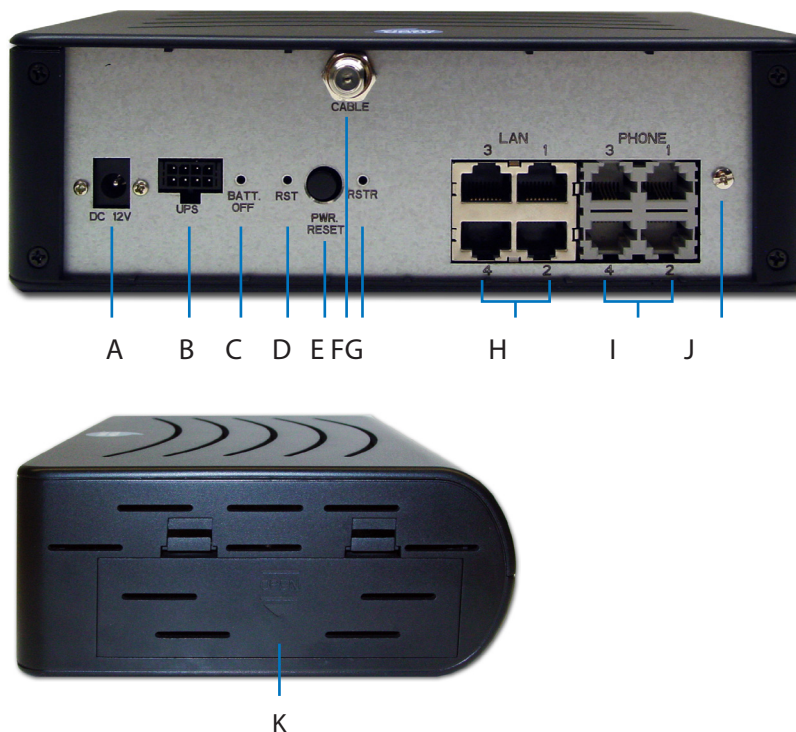
**Schedule Test**

	<input type="checkbox"/> Enabled
Destination Number	14087882210
Test Frequency	<input checked="" type="radio"/> Every Day <input type="radio"/> Every Week <input type="radio"/> Every Month Time 20 : 26

☒ Apply    ☐ Cancel

## ESBC INTERFACE

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



## SPECIFICATIONS

### Product Interfaces

Category	Specification
Service Provider Interface	DOCSIS Standard CATV coaxial cable, 75 Ohms "F" type connector
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 SIP Forking support 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) Profile-based Multiple Proxy support

## SPECIFICATIONS cont.

Category	Specification
Networking Features	Dedicated Bridge Port Dedicated Router Port with RIPv2 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
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)
Signal Processing	G.168 Echo cancellation FAX (T.38 and G.711 fall-back) Loop Back Caller ID FSK signal regeneration
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 VPN Server for remote management of end devices Dual image capability Data monitoring throughput tools
QoS	Voice Bandwidth Reservation QoS, <i>Smart-DQoS™</i> , DQoS using Packetcable Multimedia, Type of Service, VLAN Tagging



## SPECIFICATIONS cont.

### Cable Modem Technical Specifications

- DOCSIS 1.1 and 2.0 compliant.
- Integrated A-TDMA and S-CDMA technology - Capable of providing 30 Mbps upstream data rate
- 8/16/32/64/128/256 QAM auto detection

### Cable Transmit/Receive Specifications

Item	Downstream	Upstream
Frequency Range	DOCSIS: 88~860MHz Euro-DOCSIS*: 112~858 Mhz	DOCSIS: 5~65Mhz Euro-DOCSIS*: 5~42Mhz
Modulation	QPSK, 16/ 32/ 64/ 128/ 256QAM	QPSK, 8/16/32/64/128 QAM
Data Rate	DOCSIS: 64 QAM: 30 Mbps 256 QAM: 42.8 Mbps Euro-DOCSIS*: 64 QAM: 41 Mbps 256 QAM: 55 Mbps	QPSK 0.32 ~ 10.24 Mbps 8 QAM 0.48 ~ 15.36 Mbps 16 QAM 0.64 ~ 20.48 Mbps 32 QAM 0.80 ~ 25.60 Mbps 64 QAM 0.96 ~ 30.72 Mbps 128 QAM/TCM 30.72 Mbps
Bandwidth	Euro-DOCSIS*: 8 MHz; DOCSIS: 6 MHz	TDMA: 200, 400, 800, 1600, 3200 and 6400 kHz S-CDMA: 1600, 3200 and 6400 kHz
FEC	RS (128,122) GF128 with Trellis coding	Reed Solomon
Signal Level	Receive Power Level: DOCSIS: -15 ~ +15 dBmV Euro-DOCSIS*: 64 QAM: -17 dBmV ~ +13 dBmV 256 QAM: -13 dBmV ~ +17 dBmV	Transmit Power Level : TDMA: +8 ~ +54 dBmV (32QAM, 64QAM) +8 ~ +55 dBmV (8QAM, 16QAM) +8 ~ +58 dBmV (QPSK) S-CDMA: +8 ~ +53 dBmV (all modulation)

### Cable Modem Other Specifications

Signal-to-NoiseRatio (SNR)	DOCSIS: 64 QAM: >23.5 dB 256 QAM: >30 dB	Euro-DOCSIS*: 64QAM: >= 25.5 dB 256QM: -13 dBmV ~ -6 dBmV >= 34.5 dB -6 dBmV ~ +17 dBmV >= 31.5 dB
Security	DOCSIS Baseline Privacy Plus: 1024-bit RSA and 128-bit Tripple-DES for BPKM protocol 56 -bit DES for data encryption X.509 v3 certificates	
DOCSIS	Compliant to DOCSIS 2.0 and Euro-DOCSIS 2.0*	
Protocol	TCP/IP, UDP, ARP, ICMP, DHCP, SNMP, TFTP, TOD, BOOTP, SYSLOG	
Configuration	Ease of configuration and privacy control provided by resident or downloaded code from a Cable Modem Termination System (CMTS)	
Bridging	Support for unicast, broadcast, and multicast IP packetsVariable-length packet cable Media Access Control (MAC) transport layerMix of contention and reservation-based upstream transmission	
Quality of Service	Quality of service of MAC layer	
SIDs	16	

## SPECIFICATIONS cont.

Management Operations (SNMPv1/v2c/v3)	RFC1157, RFC1901, RFC3416, RFC3417, RFC2578, RFC2570, RFC3411, RFC3412, RFC3413, RFC3414, RFC3415, RFC2576
MIBs support	RFC1493, RFC3418, RFC2011, RFC2013, RFC2233, RFC3411, RFC3412, RFC3413, SNMP-NOTIFICATION-MIB, RFC3414, RFC3415, RFC2576, RFC2665, RFC2669, RFC2786, RFC2851, RFC2933, RFC3083, DRAFT: DOCS-IF-MIB, DRAFT: USB-MIB, DRAFT: DOCS-BPI2-MIB, DRAFT: DOCS-QOS-MIB, Append L/Annex H: DOCS-IF-EXT-MIB, Append L/Annex H: DOCS-CABLE-DEVICE-TRAP-MIB

\* Check for availability

### 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 Part15A, 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

### About Smart-DQoS™

*Smart-DQoS™* is InnoMedia's exclusive Device-initiated DQoS technology which enables edge devices to intelligently initiate and manage DOCSIS DQoS UGS service flows based on user and signaling events without the need for PacketCable Multimedia. *Smart-DQoS™* instantly allows end-to-end quality of service without having to wait for network infrastructure modifications.

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