INNOMEDIA

ESBC 9378-4B

ENTERPRISE SIP GATEWAY (ESG)

OVERVIEW

ESBC 9378-4B is a high performance and highly versatile Enterprise SIP Gateway (ESG) solution. It combines Enterprise Session Border Controller with Media Processing capabilities, business line FXS ports, switched data port, and internal battery, all in one unit. ESBC 9378-4B can be used by service providers to offer SIP trunks to enterprises with B2BUA for IP PBXs (Figure 1) or with SIP ALG for IP Centrex (Figure 2).

Figure 1. ESG with B2BUA for IP PBXs

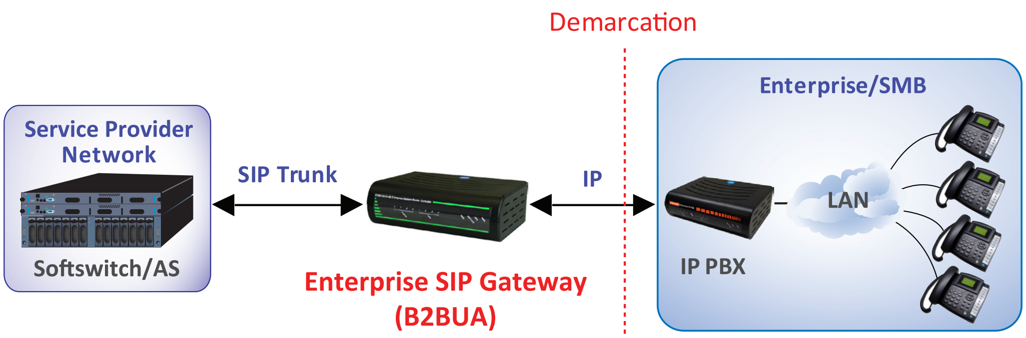
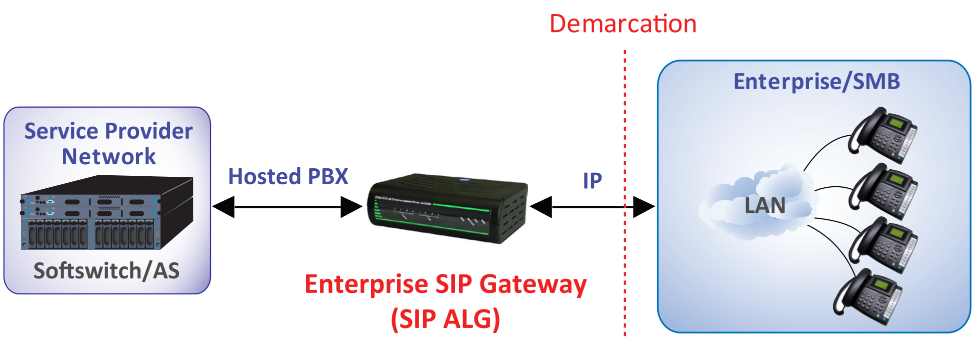


Figure 2. ESG with SIP ALG for IP Centrex



SERVICE USAGE EXAMPLES

Designed for Broadband Service Providers (BSPs) offering SIP trunking, hosted voice, and high-speed data services, InnoMedia's ESBC 9378-4B is a highly integrated and highly manageable Enterprise Session Border Controller (ESBC) that can be auto-provisioned and remotely managed. Equipped with VLAN and DSCP packet priority tagging, ESBC 9378-4B is ideally suitable for BSPs offering bundled services with end-to-end quality of service over broadband service networks. 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 media transcoding feature provides a solution to the problem where the Service Provider supports different media capabilities to those of the end device located at the enterprise. Specifically, the ESBC9378-4B provides the ability to transcode between the following media capabilities: Fax (T.38 and G.711), Voice CODECs (G.711, G.729, G.726), and DTMF (RFC2833 and In-band).

The two typical service scenarios for ESBC 9378-4B are:

1. BSPs delivering high-speed Internet access as well as SIP trunks to enterprises which use IP-based PBXs (Figure 3).
2. BSPs delivering high-speed Internet access as well as IP Centrex to enterprises which desires hosted PBX (Figure 4).

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

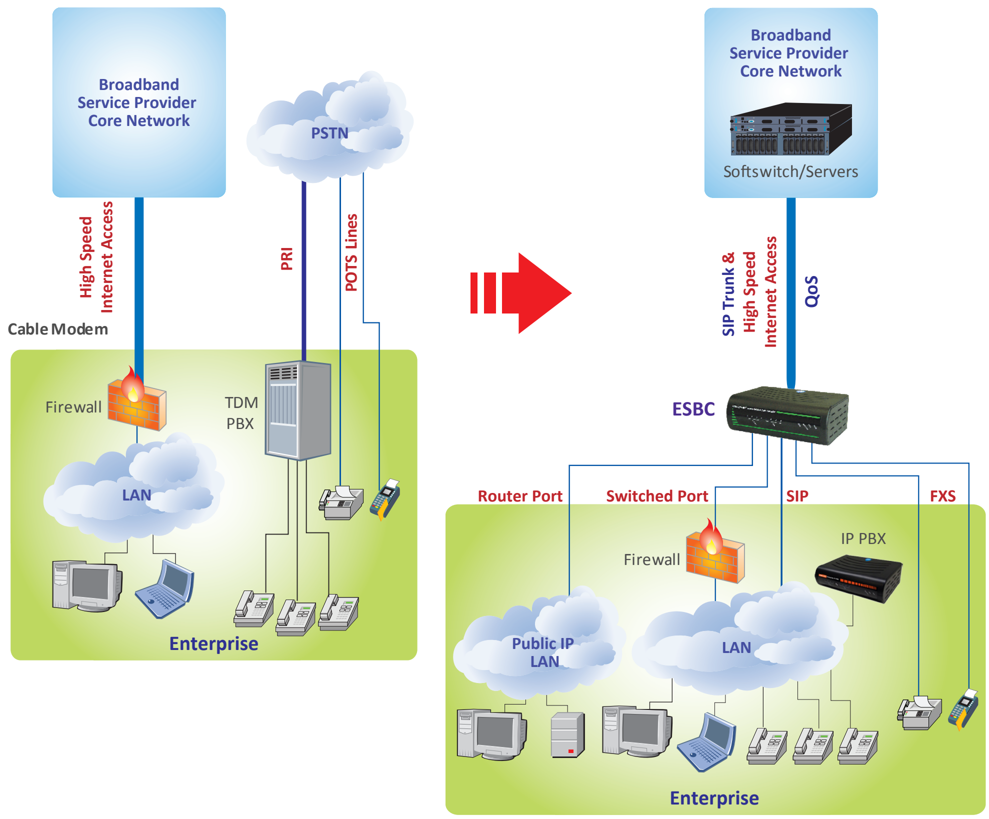
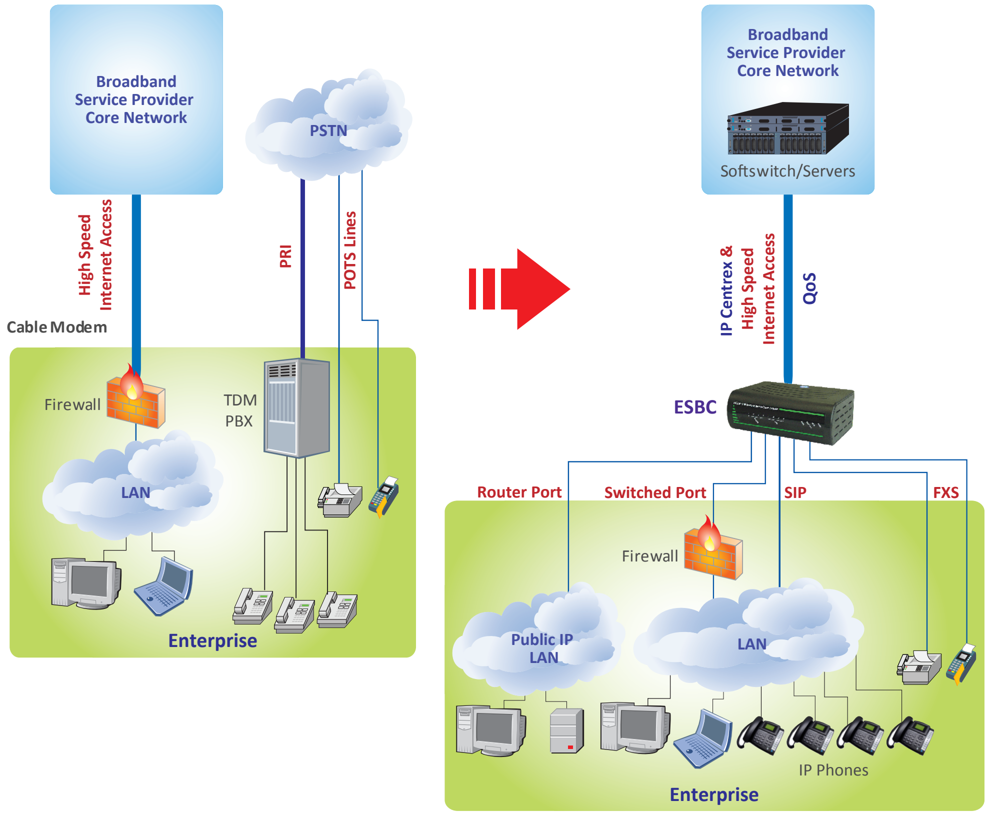


Figure 4. BSPs delivering high-speed Internet access as well as hosted PBX/IP Centrex to enterprises



SUMMARY OF KEY FEATURES AND BENEFITS

Functional Blocks/Categories	Features	Benefits
Gigabit WAN and LAN ports	1 Gigabit WAN ports and 4 Gigabit LAN ports	1. Suitable for BSPs offering high-speed connectivity 2. 4 LAN ports can be used as 1 high-speed data port and 3 SIP ports
Quality of Service	QoS: 1. WAN: VLAN or DSCP 2. LAN: VLAN with 1000 groups	Enabling end-to-end QoS based service offering
Embedded Session Border Controller (ESBC)	Registrar and B2BUA 1. Implicit, explicit, and static (no) registration 2. Header manipulation/SIP normalization 3. NAT traversal 4. SIPConnect compliant 5. IMS ready 6. Profile-based Multiple SIP Proxy Support 7. Built-in SIP Forking Feature for multiple SIP devices within the LAN.	• Highly interoperable between service provider and enterprise equipment • No interference with enterprise firewall setting • Reliable and scalable SIP trunking service delivery • Multiple Proxy Support allows separate and independent Softswitch (thus features, billings, etc.) to manage the FXS ports and LAN UA
	SIP ALG 1. Header manipulation 2. NAT traversal 3. SIPConnect compliant 4. IMS ready	• Highly interoperable between service provider and enterprise equipment • No interference with enterprise firewall setting • Reliable and scalable hosted PBX/IP Centrex service delivery

SUMMARY OF KEY FEATURES AND BENEFITS CONT.

Functional Blocks/Categories	Features	Benefits
Media Transcoding	Transcoding between the following media capabilities: 1. Fax: T.38 and G.711 2. DTMF: inband and RFC2833 3. CODECs (G.711, G.729, G.726)	Allows the service provider to support different media capabilities from those of the end device located at the enterprise, resulting in highly scalable deployments.
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. 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. 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 embedded Session Border Controller (eSBC), intelligent internal battery, four FXS ports with business line features, and an interface for external UPS, InnoMedia ESBC 9378-4B offers 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 SIP trunk path provides SIP normalization, Media Processing for DTMF, Voice, or Fax Transcoding, 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. Together with VLAN and DSCP, the BSP is able to offer QoS ensured SIP trunking service.

The SIP ALG path enables BSPs to offer Hosted PBX Services with NAT traversal, 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 VLAN and DSCP, the BSP is able to offer QoS ensured hosted voice/IP Centrex service.

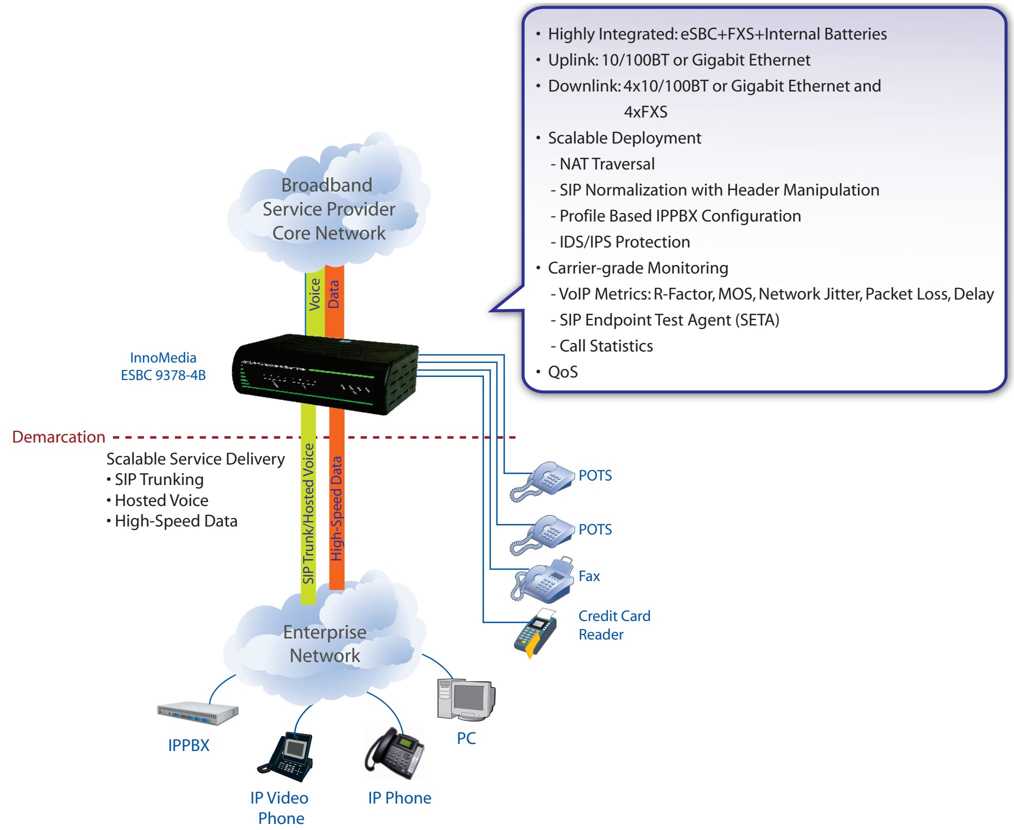
By configuring the ESBC9378-4B, it is possible to allow different forms of SIP signaling negotiation between the Service Provider side and the Enterprise side before media transcoding takes place. There are maximum capabilities for the number of transcoding sessions per device:

- 60 maximum DTMF/CODEC Transcoding sessions.
- 24 maximum Fax Transcoding Sessions,
- When "Allows calls even when no enough channels" is checked and the 24 FAX transcoding sessions are fully utilized, further FAX calls can still be made but without T.38-G.711 transcoding capability.

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

The ESBC 9378-4B, located at the edge of broadband service providers' 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.

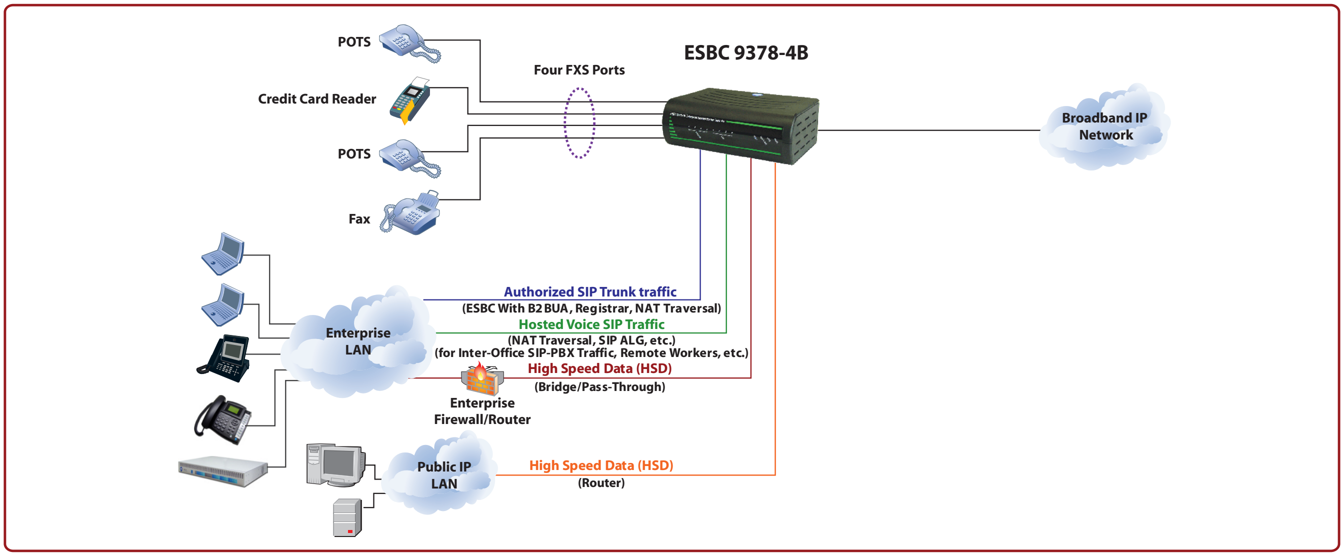
Figure 5. ESBC 9378-4B delivering high-speed Internet access, SIP trunking, and FXS ports with Business Line Features

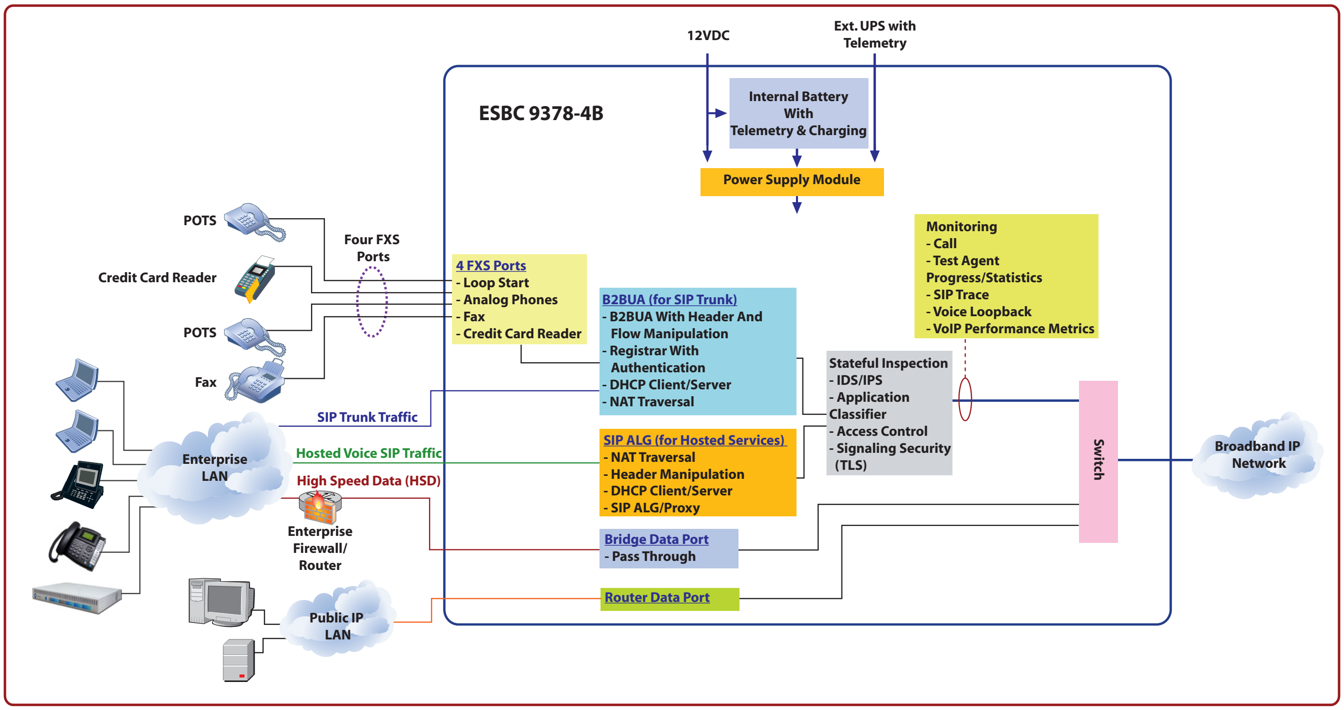


The highly integrated ESBC9378-4B includes the following key functional blocks:

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

Figure 6. ESBC 9378-4B functional block diagram





Integrated Internal Battery As Well As External UPS Support

The ESBC 9378-4B is equipped with an internal battery supporting up to 4 hours of continuous talk time for the SIP trunk traffic and the 4 FXS 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 9378-4B 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 9378-4B 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 9378-4B include:

1. SIP Normalization:
   Based on the B2BUA architecture, InnoMedia's ESBC 9378-4B provides Profile based settings, High-level classification for SIPConnect Adaptation, and Low-level header manipulation for SIP signaling normalization:
   - Profile based settings:
     ESBC 9378-4B allows parameter and option settings to adapt between the two interfaces: the WAN interface to the BSP 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 BSPs 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 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 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. <u>Media Processing:</u>
   InnoMedia's ESBC 9378-4B provides the ability to process and transcode media for interoperability between the SIP-PBX and the BSP servers:
   - Fax transcoding between T.38 and G.711 Pass through
   - DTMF transcoding between In-band DTMF and RFC 2833
   - Codec transcoding between various CODEC selections

3. <u>Registration and Authentication:</u>
   Acting as a registrar server to SIP-PBXs, the InnoMedia ESBC 9378-4B 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.

4. <u>NAT Traversal:</u>
   – Inspects and modifies headers, SDP, and implement media relay via RTP bridge control.

5. <u>SIP signaling security:</u>
   – TLS: ESBC 9378-4B supports TLS connection with the BSP network (authenticate BSPservers) for secured signaling transport, as well as SIP Digest authentication (challenged and authenticated by the BSP servers).
   – SIP Message Validation: ESBC 9378-4B validates all SIP messages

6. <u>Emergency Call Handling (Figure 11):</u>
   – 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 And Router Ports For BSP's High-Speed Data Services

The ESBC 9378-4B 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

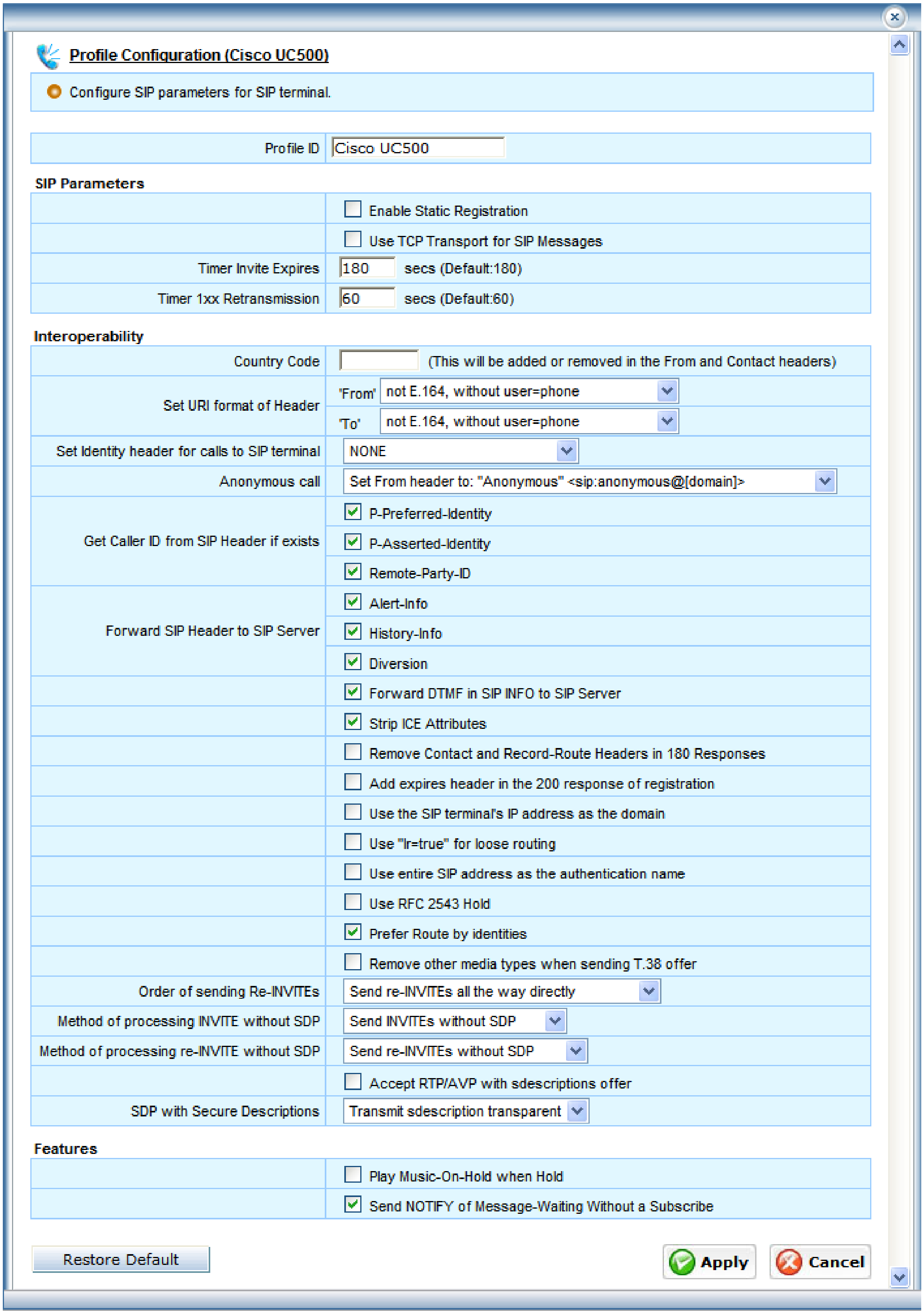
A stateful inspection with IDS/IPS is used for the FXS ports, the SIP trunk traffic path, as well as the Non-SIP Trunk SIP traffic path to protect these paths from unauthorized access or attacks. The bridged/pass-through port is not protected by the firewall, and is typically connected to the enterprise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

Voice and Network Performance Monitoring

ESBC 9378-4B 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 9378-4B 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.

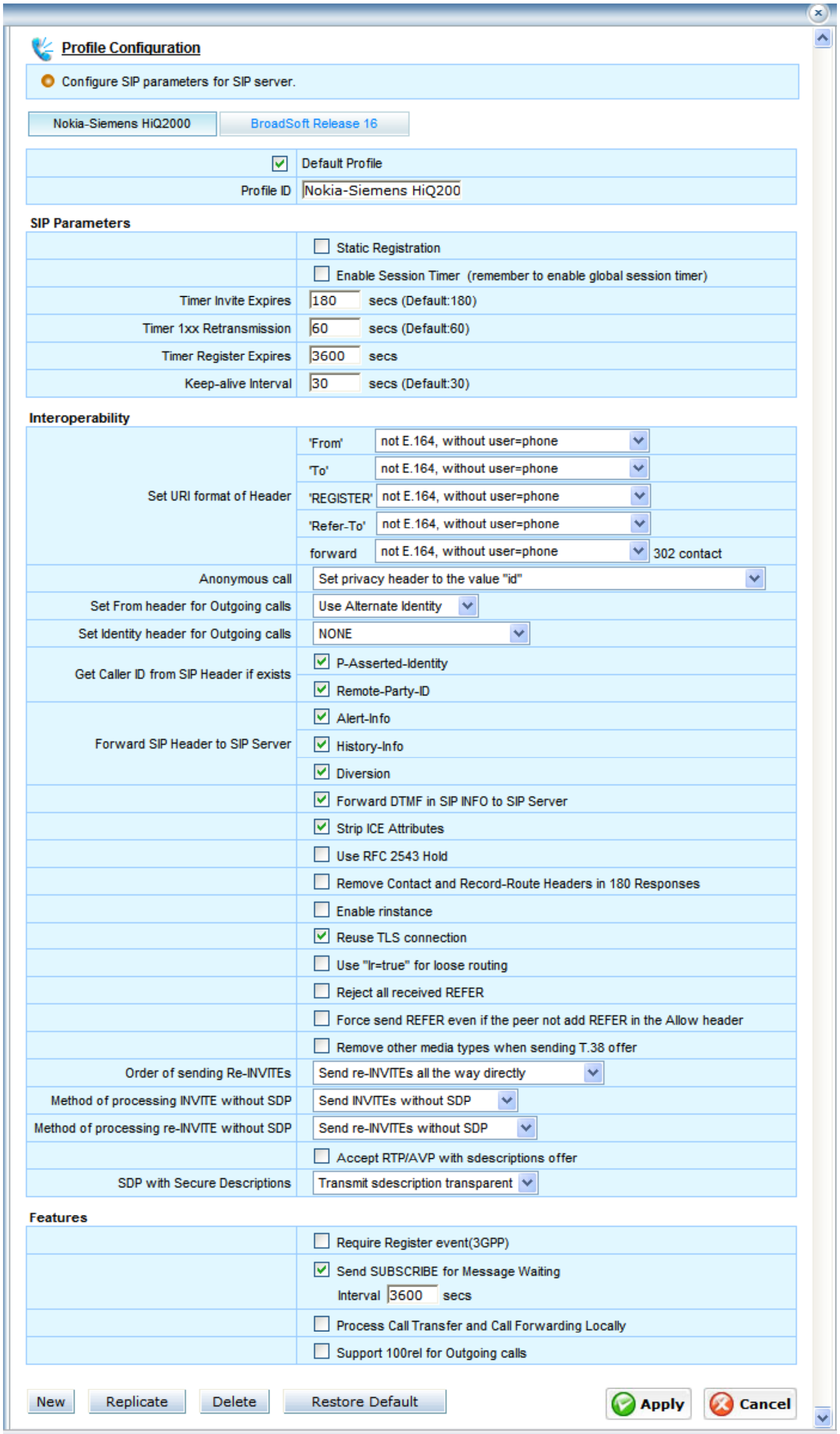
UA/SIP-PBX PROFILE

Figure 7



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 HiQ200

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 rinstance
		☑ 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

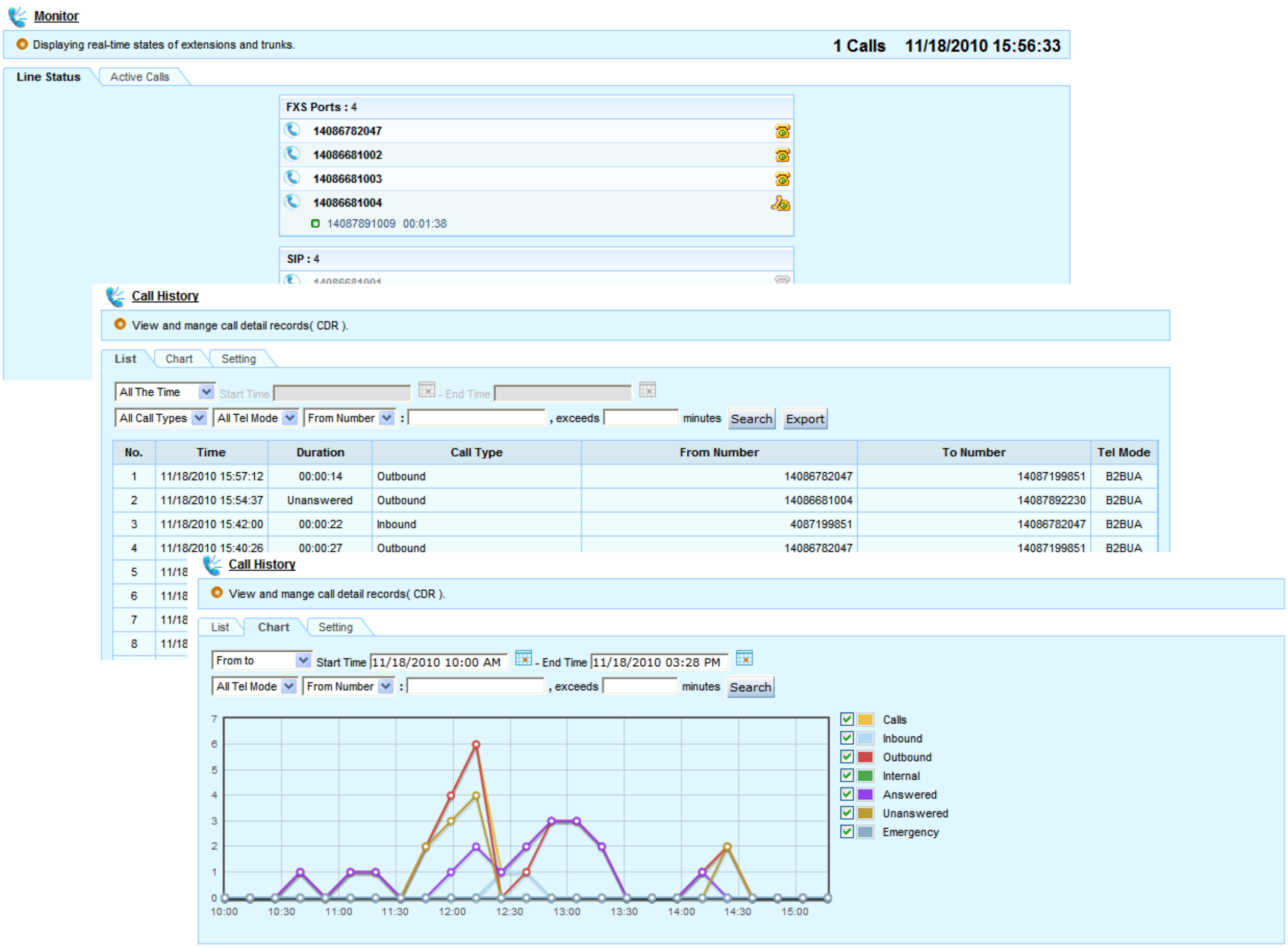
Features

	☐ 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

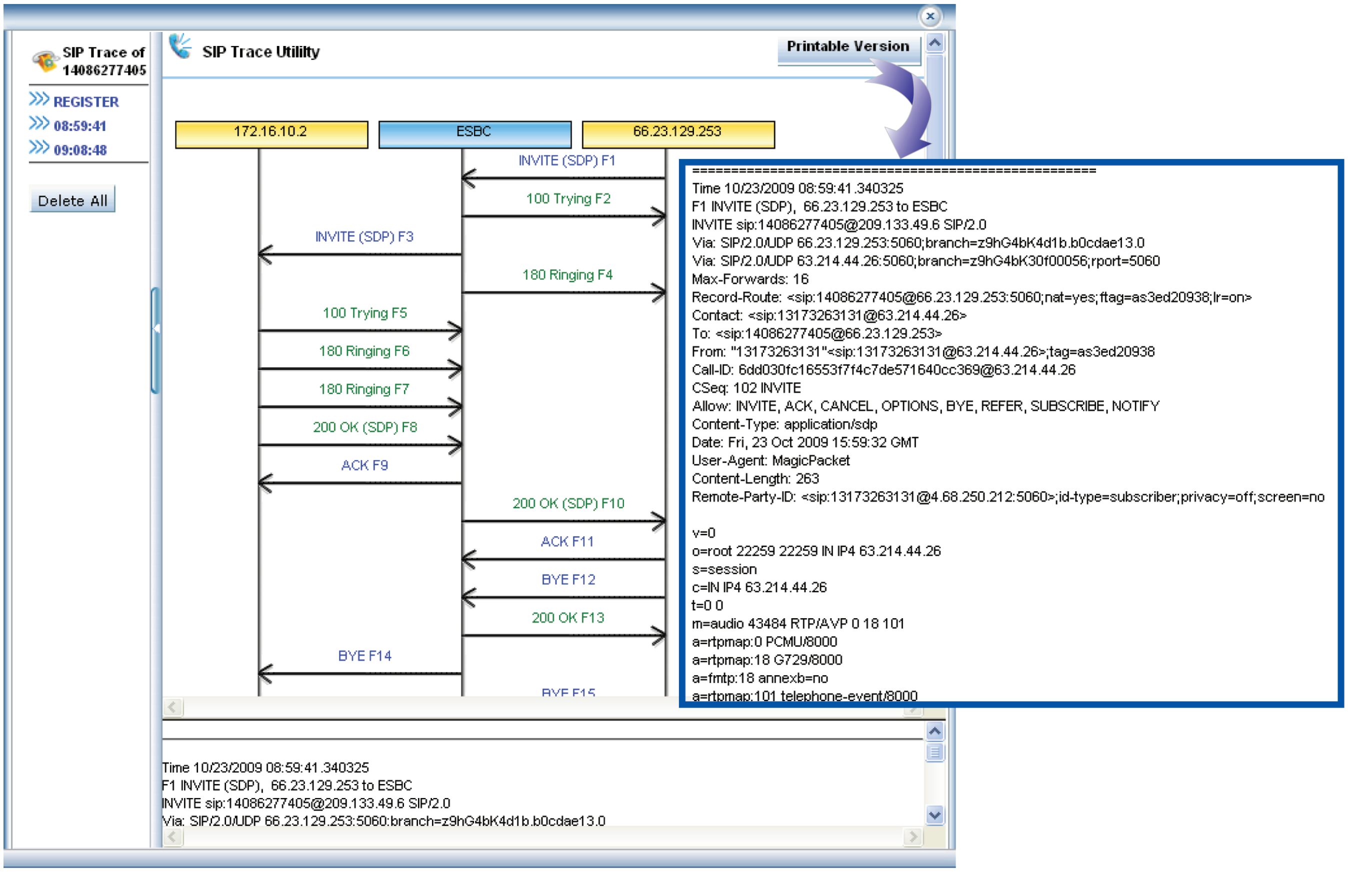
REAL-TIME LINE CALL STATES, CDR, AND CALL STATISTICS

Figure 9



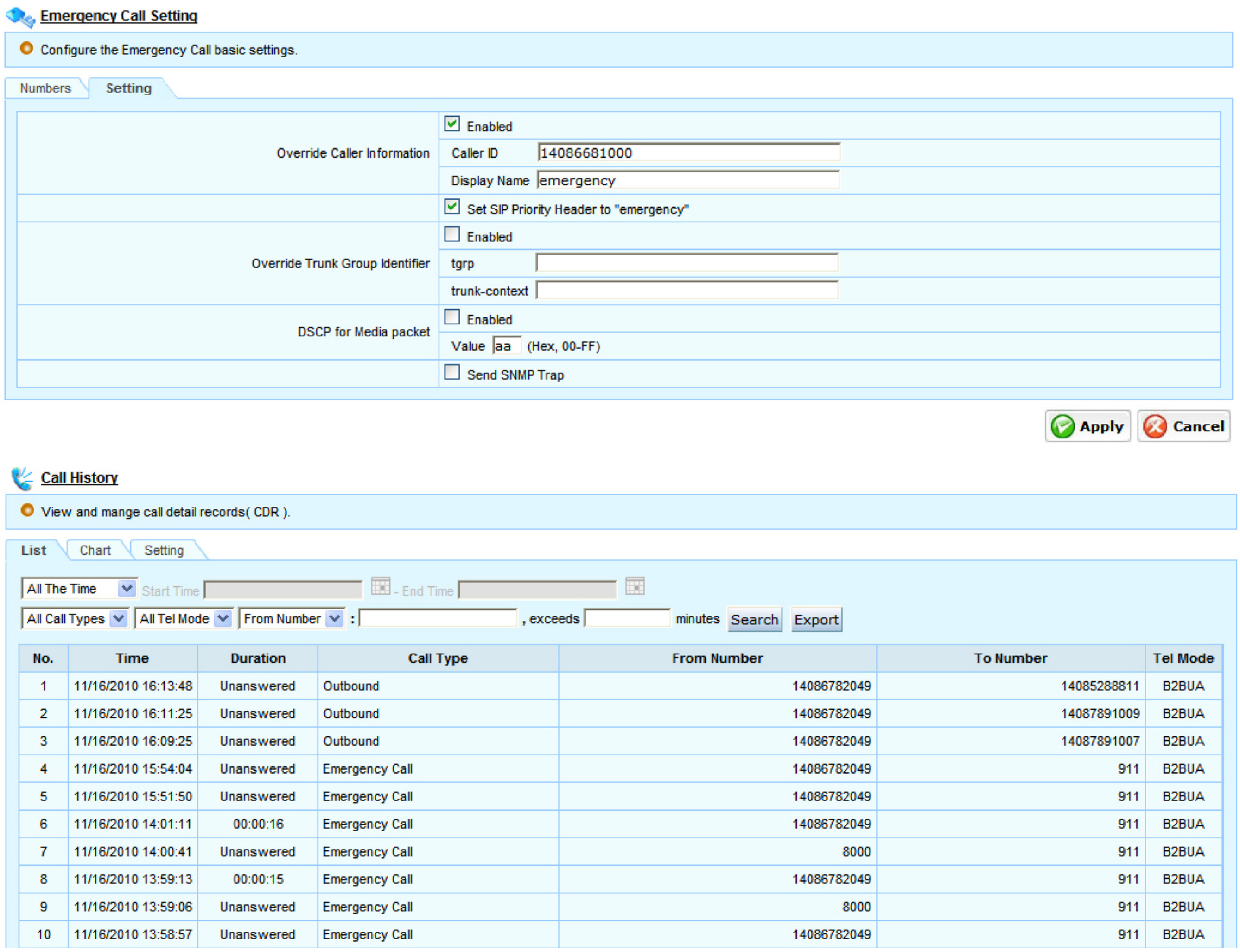
CALL TRACE GUI

Figure 10



EMERGENCY CALL HANDLING

Figure 11



Emergency Call Setting

Configure the Emergency Call basic settings.

Numbers | Setting

Override Caller Information	☑ Enabled	
	Caller ID	14086681000
	Display Name	emergency
	☑ Set SIP Priority Header to "emergency"	
Override Trunk Group Identifier	☐ Enabled	
	tgrp	
	trunk-context	
DSCP for Media packet	☐ Enabled	
	Value aa (Hex, 00-FF)	
	☐ Send SNMP Trap	

Apply | Cancel

Call History

View and mange call detail records(CDR).

List | Chart | Setting

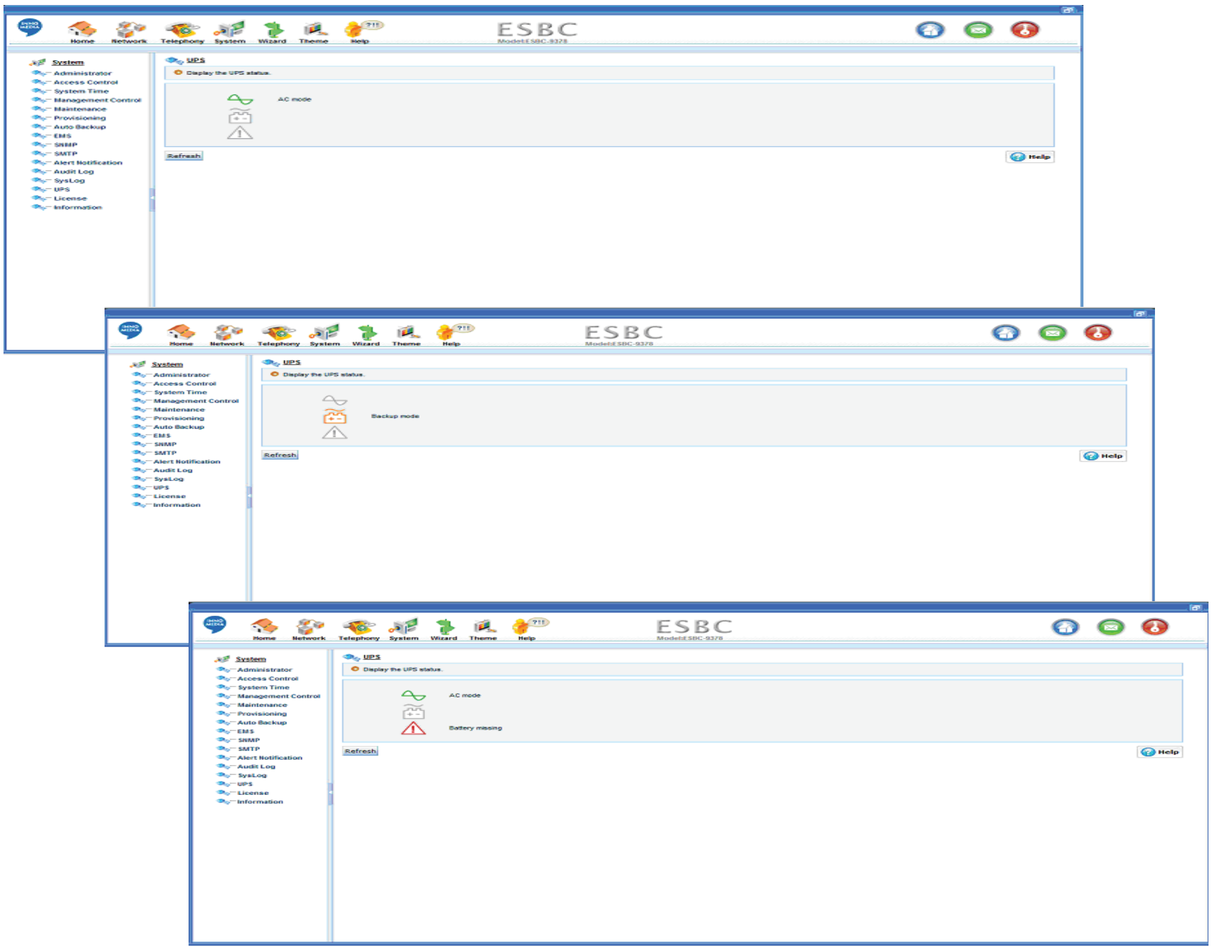
All The Time | Start Time | - End Time

All Call Types | All Tel Mode | From Number | : | , exceeds | minutes | Search | Export

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

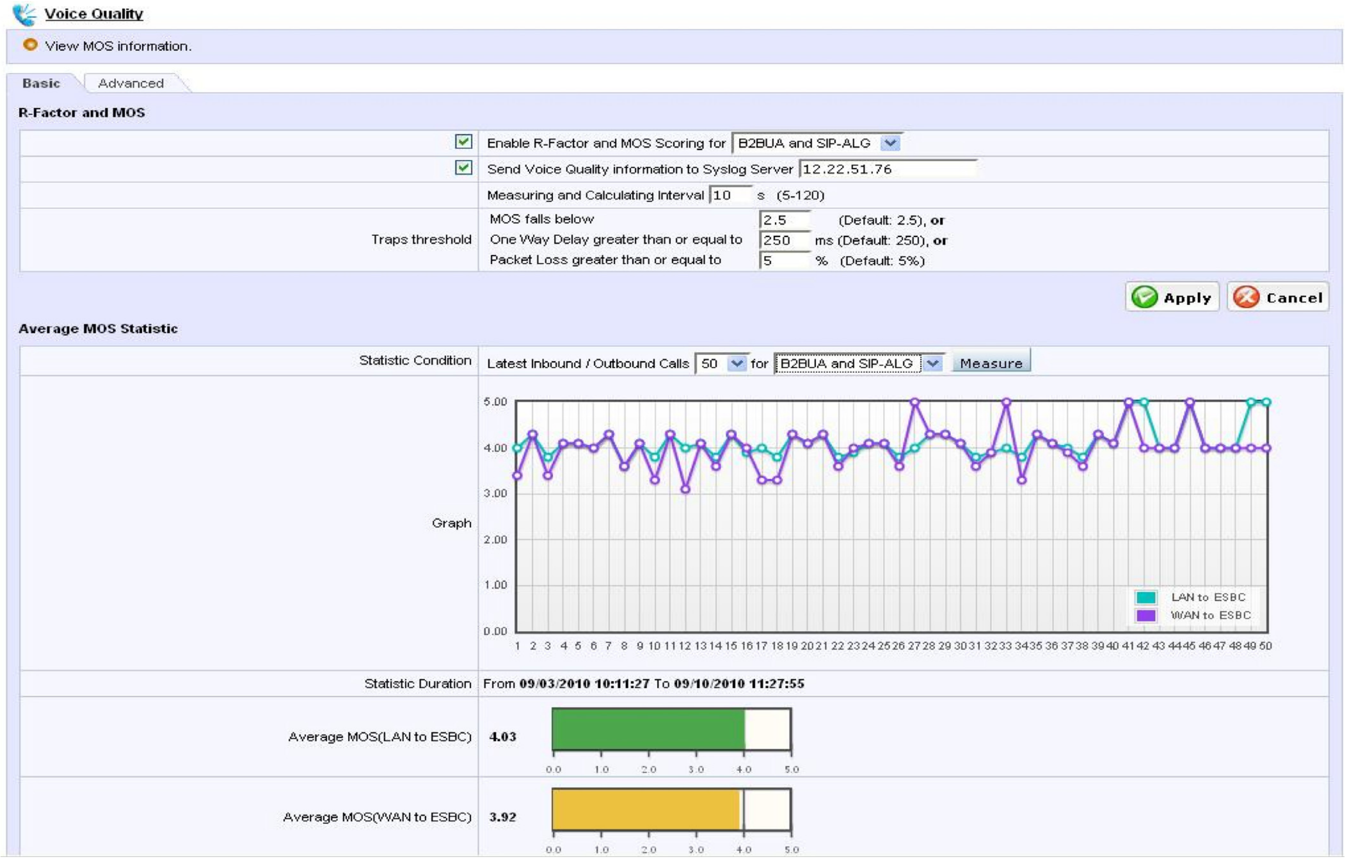
BATTERY STATUS DISPLAY

Figure 12



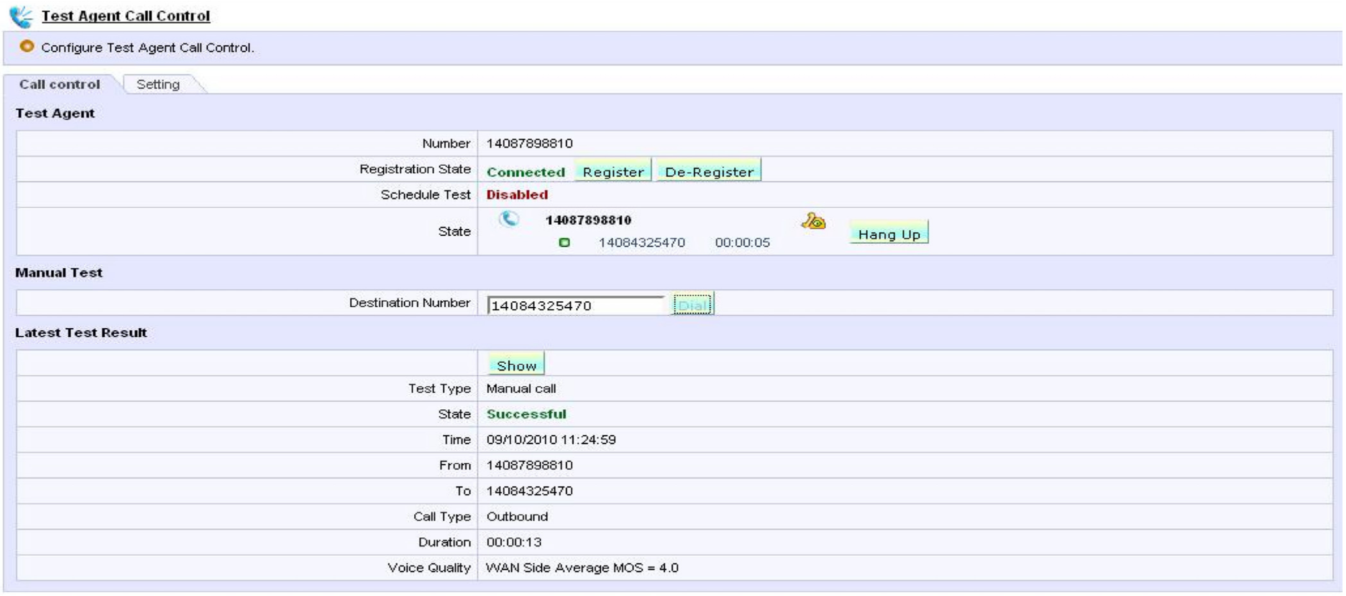
ESBC VOICE QUALITY MONITORING AND SNMP TRAP THRESHOLD SETTING

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



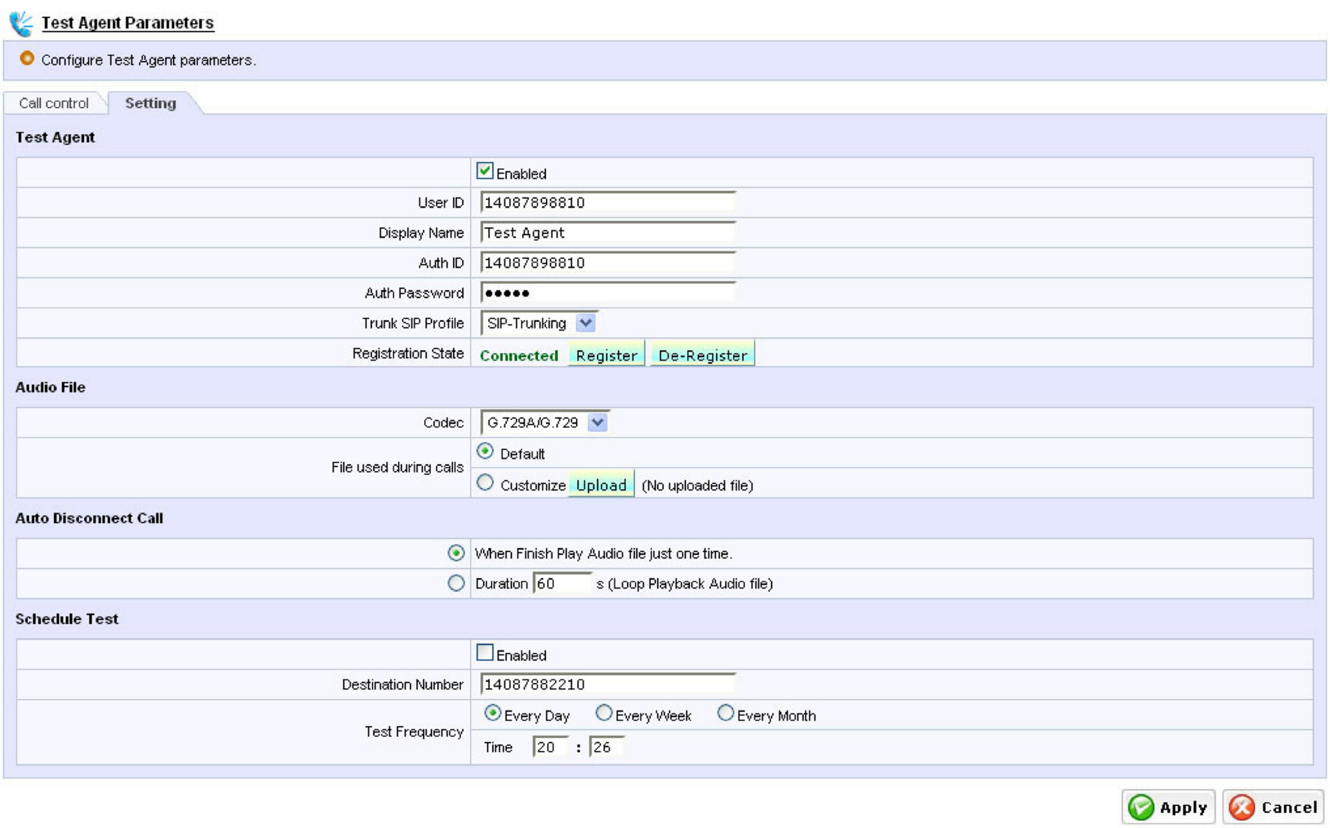
TEST AGENT MANUAL TEST CALL GUI AND WAN MOS DISPLAY

Figure 14



TEST AGENT SCHEDULED TEST CALL SETTINGS

Figure 15

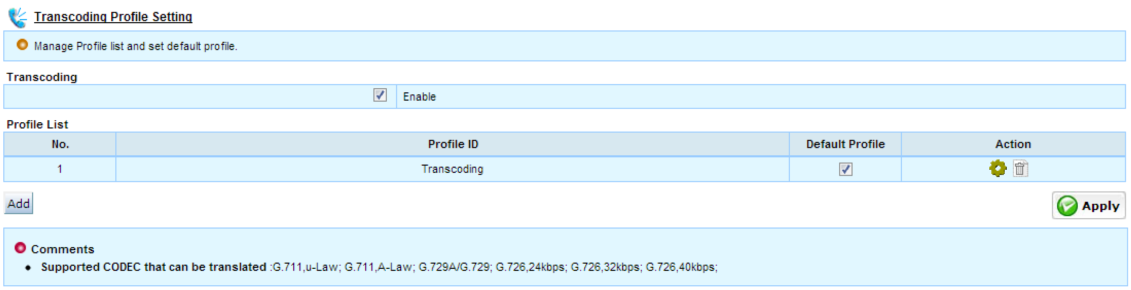


Media Transcoding

The media transcoding can be configured via the Transcoding Profile. The Transcoding Profile screen allows the profile list and the default profile to be managed. It also provides access to the Profile configuration screen, which allows the system administrator to configure Fax, CODEC or DTMF transcoding settings between the WAN and LAN side of the ESBC.

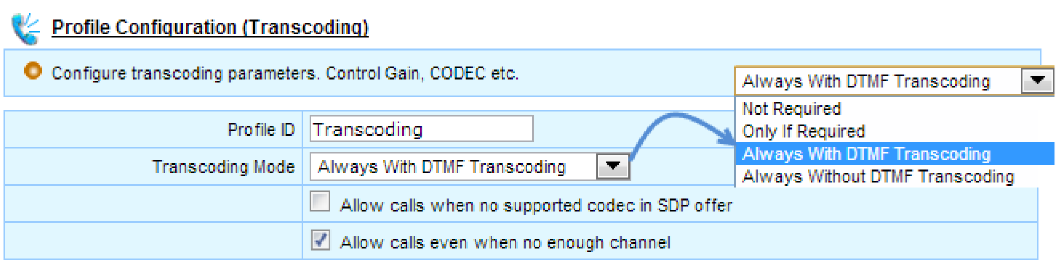
To configure the Transcoding Profile, follow these steps:

1. Login to the ESBC as "admin" through the web console.
2. Go to the page at “Telephony à ADVANCED à Transcoding Profile”.



Adding or Editing Profiles

To add a Transcoding Profile, click the <Add> button and then click the <Setting> button to configure Transcoding parameters. Individual profiles can be created with different configurations for a specific SIP UA or a group of SIP UAs to use.



In the profile Configuration screen, modify the Profile ID and select one Transcoding Mode option from the drop-down list that a SIP UA group can use:

Transcoding Mode	
Item Name	**Description**
Not Required	Transcoding is disabled on these UAs.
Only If Required	This setting only allows CODEC transcoding. CODEC transcoding only happens if there are no common CODECs between the caller and called UAs. Fax and DTMF transcoding will not be performed.
Always With DTMF Transcoding	Fax, DTMF and CODEC transcoding for all calls.
Always Without DTMF Transcoding	Fax, CODEC for all calls, but no DTMF transcoding.

Transcoding Option	
Item Name	**Description**
Allow calls when no supported codec in SDP offer	ESBC will allow SDP offer to pass through even if the codec is not in the Extend Codec list. In this case, no transcoding will take place, but this feature allows unsupported transcoding codecs (eg G.723.1) to be negotiated end-to-end between Enterprise and Service Provider SIP UA's.
Allow calls even when no enough channel	When selected, the ESBC will allow calls to be processed even if there may not be enough channels to process transcoding. The calls will go through but the media may not be transcoded.

DTMF Mode

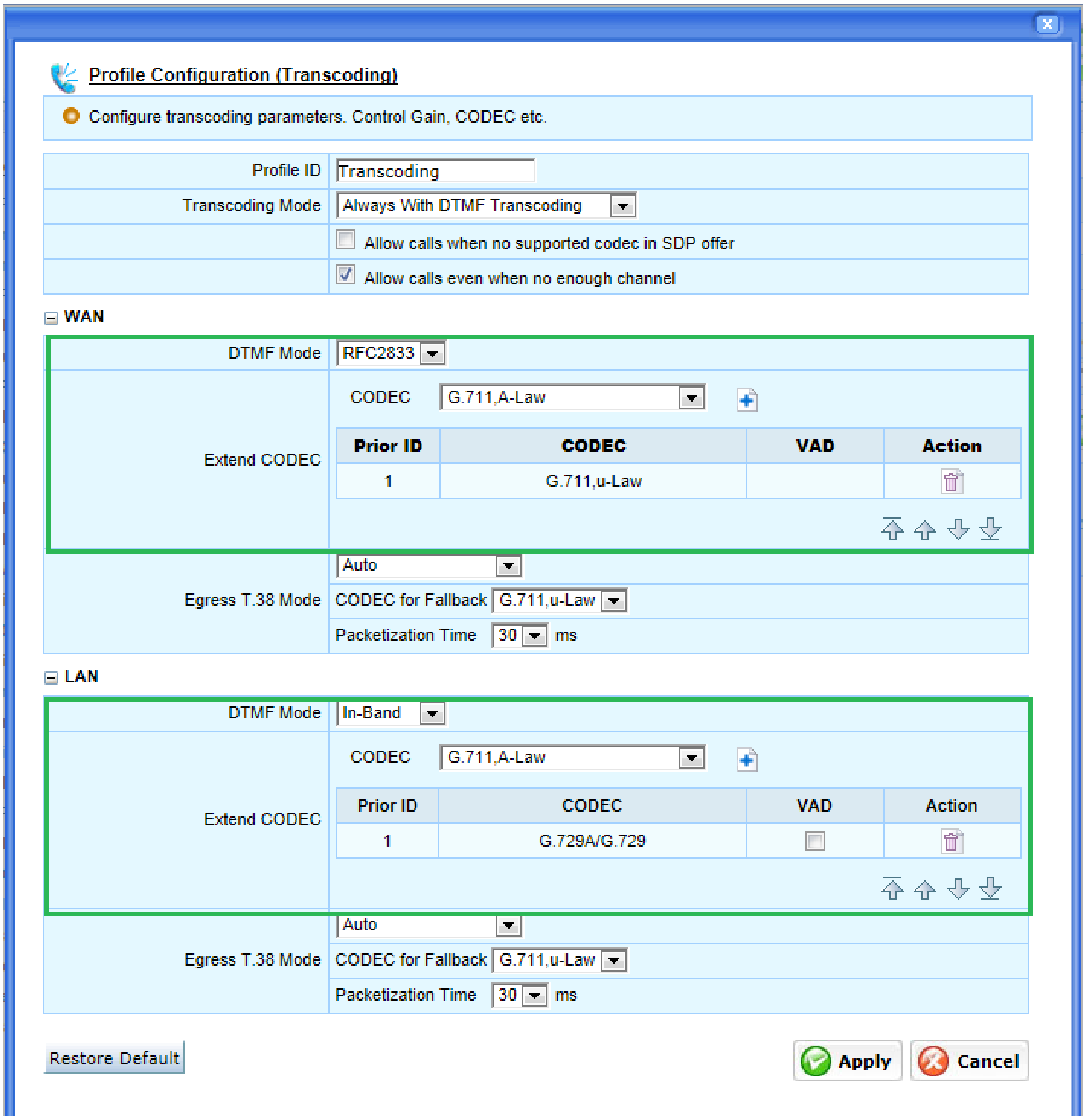
Supported DTMF modes: RFC2833 and In-band DTMF.
DTMF transcoding can be activated only when "Always with DTMF Transcoding" is selected. Select the appropriate "DTMF Mode" for the ESBC LAN side SIP UA and WAN side SIP UA as follows:

·If the SIP UA supports only in-band DTMF, then "In-band" should be selected towards this UA.

·If the SIP UA supports both in-band DTMF and RFC2833, then either "In-band" or "RFC2833" may be selected, depending on the desired result.

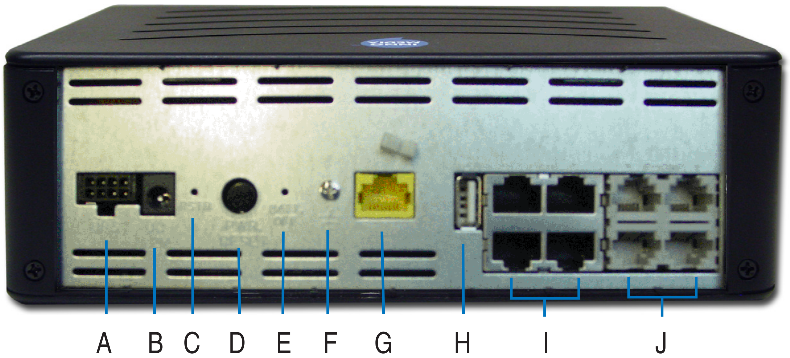
Extend CODEC

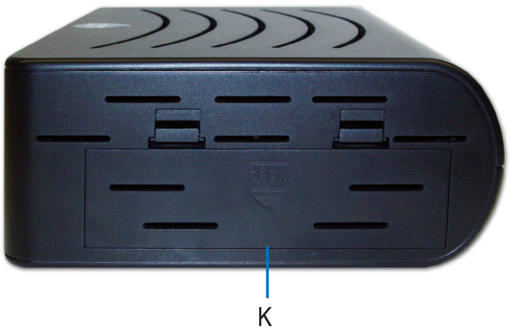
When configuring the 'Extend CODEC', the selection is based on the supported CODEC for that side of the ESBC's interface. The ESBC can be configured to add certain codec capabilities to transcoding profiles, and then perform transcoding in cases where the selected codec in the answer SDP is not available in the original offer.


Profile Configuration (Transcoding)
Configure transcoding parameters. Control Gain, CODEC etc.
Profile ID
Transcoding
Transcoding Mode
Always With DTMF Transcoding
Allow calls when no supported codec in SDP offer
Allow calls even when no enough channel
WAN
DTMF Mode
RFC2833
Extend CODEC
CODEC
G.711,A-Law
Prior ID
CODEC
VAD
Action
1
G.711,u-Law
Egress T.38 Mode
Auto
CODEC for Fallback
G.711,u-Law
Packetization Time
30
ms
LAN
DTMF Mode
In-Band
Extend CODEC
CODEC
G.711,A-Law
Prior ID
CODEC
VAD
Action
1
G.729A/G.729
Egress T.38 Mode
Auto
CODEC for Fallback
G.711,u-Law
Packetization Time
30
ms
Restore Default
Apply
Cancel

ESBC INTERFACE

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





SPECIFICATIONS

Product Interfaces

Category	Specification
Service Provider Interface	Gigabit Ethernet RJ45 Connector
Telephone Interface	4 FXS Voice Ports
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 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 (Fax, DTMF and Voice CODEC Transcoding) Profile-based Multiple Proxy support

SPECIFICATIONS cont.

Category	Specification
Media Transcoding	1. 60 maximum DTMF/CODEC Transcoding sessions 2. 24 maximum fax Transcoding sessions. 3. When “Allows calls even when no enough channels” is checked and the 24 FAX transcoding sessions are fully utilized, further FAX calls can still be made but without T.38-G.711 transcoding capability. 4. Simultaneous CODEC Transcoding sessions and FAX Transcoding sessions will reduce the maximum transcoding capability per device.
Category 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	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

SPECIFICATIONS cont.

Category	Specification
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, Type of Service, VLAN Tagging, DSCP

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

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