INNOMEDIA

ESBC10K-MDX

ENTERPRISE SESSION BORDER CONTROLLER (ESBC)

HIGH DENSITY ENTERPRISE SESSION BORDER CONTROLLER IDEAL FOR BROADBAND SERVICE PROVIDERS OFFERING SIP TRUNKING SERVICES TO MID-TO-LARGE SIZE ENTERPRISE CUSTOMERS

OVERVIEW

ESBC10K-MDX is a high density and versatile Enterprise Session Border Controller (ESBC) solution with up to 200 transcoding processing sessions. It combines Enterprise Session Border Controller with Media Processing capabilities and Dual WAN Redundancy, all in one unit. ESBC10K-MDX can be used by service providers to offer SIP trunks to enterprises with up to 400 B2BUA Sessions for IP PBXs and 1000 SIP ALG Sessions for IP Centrex.

Figure 1. ESBC with B2BUA for IP PBXs

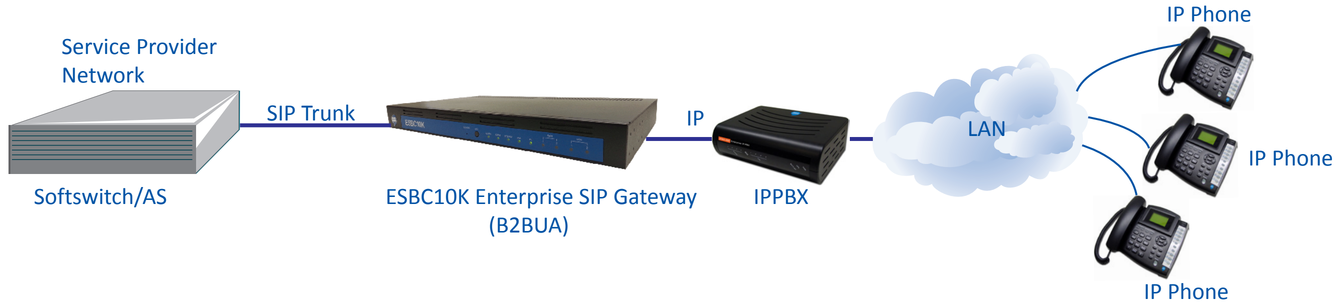
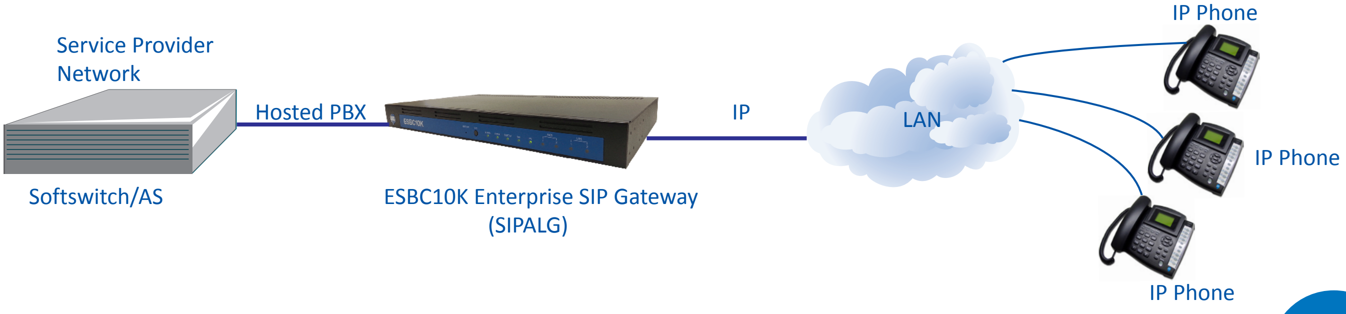


Figure 2. ESBC with SIP ALG for IP Centrex



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SERVICE USAGE EXAMPLES

Designed for Broadband Service Providers (BSPs) offering SIP trunking, and hosted voice services, InnoMedia's 19" rackmountable ESBC10K-MDX 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, ESBC10K-MDX is ideally suitable for BSPs offering SIP Trunking services with end-to-end quality of service over broadband service networks. It's 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 ESBC10K-MDX 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 ESBC10K-MDX:

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 desire hosted PBX (Figure 4).

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

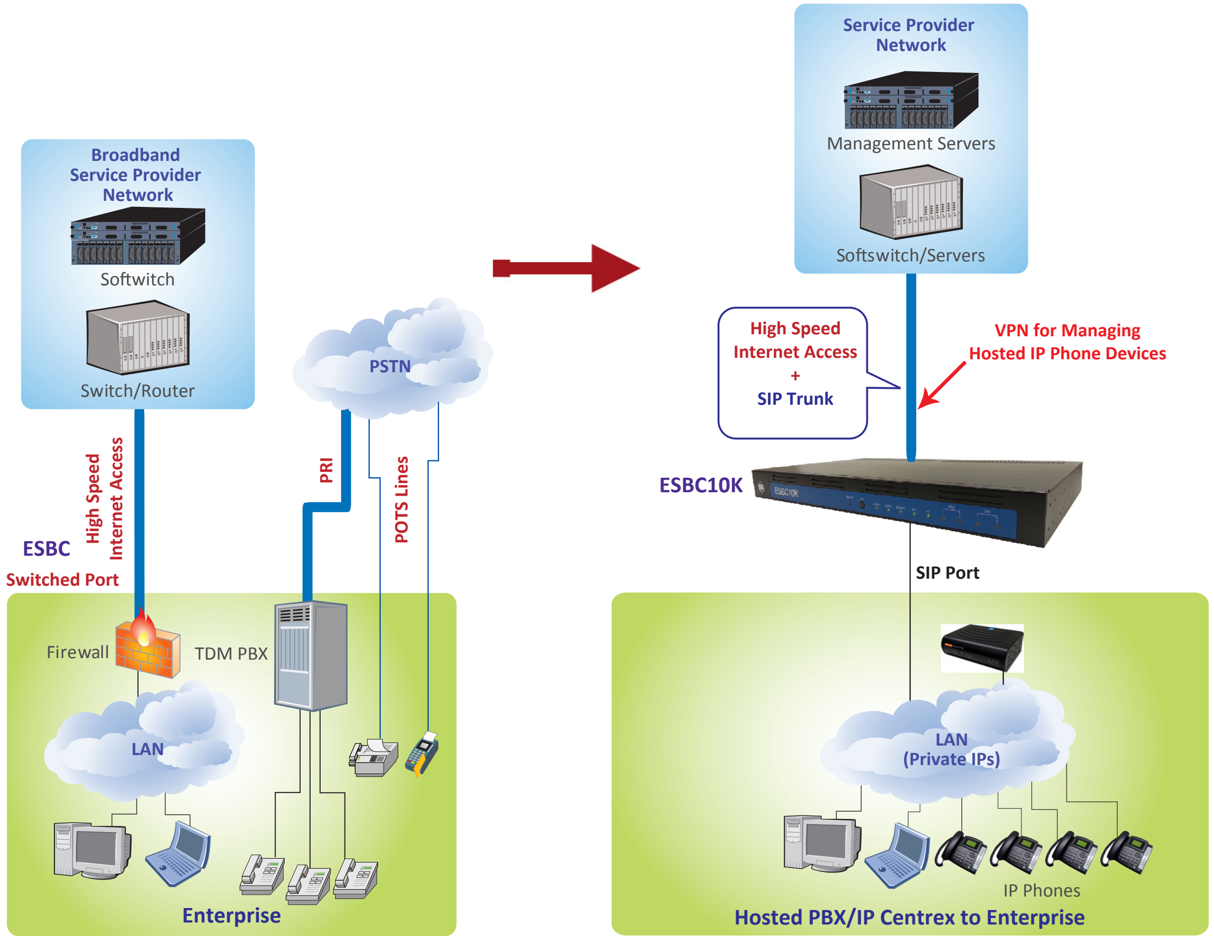
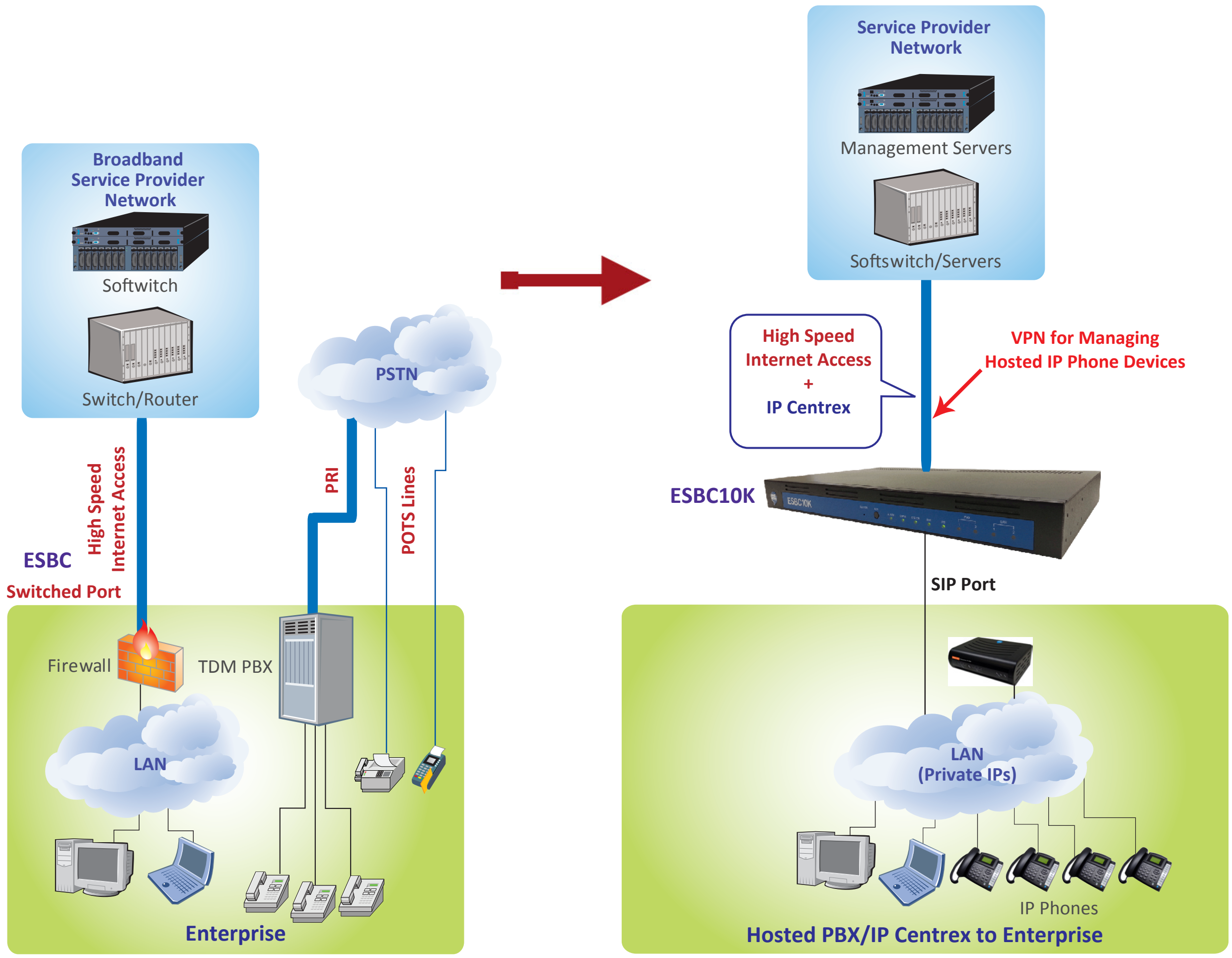


Figure 4. BSPs delivering high-speed Internet access as well as hosted PBX/IP Centrex to enterprises



SUMMARY OF KEY FEATURES AND BENEFITS

Functional Blocks/Categories	Features	Benefits
Dual Gigabit WAN	Redundant Physical Gigabit WAN Ports	Suitable for BSPs offering high-availability and redundant connectivity
Management LAN Port	Dedicated Ethernet LAN Port for Web GUI Managements	Ability to manage the device without any service interruptions
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 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
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 supports different media capabilities from those of the end device located at the enterprise, resulting in highly scalable deployments.
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. 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), InnoMedia ESBC10K-MDX offers a SIP trunk path for enterprise IP-based UAs (IP-PBXs) or a SIP ALG path for Hosted IP-PBX or IP Centrex Services.

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.

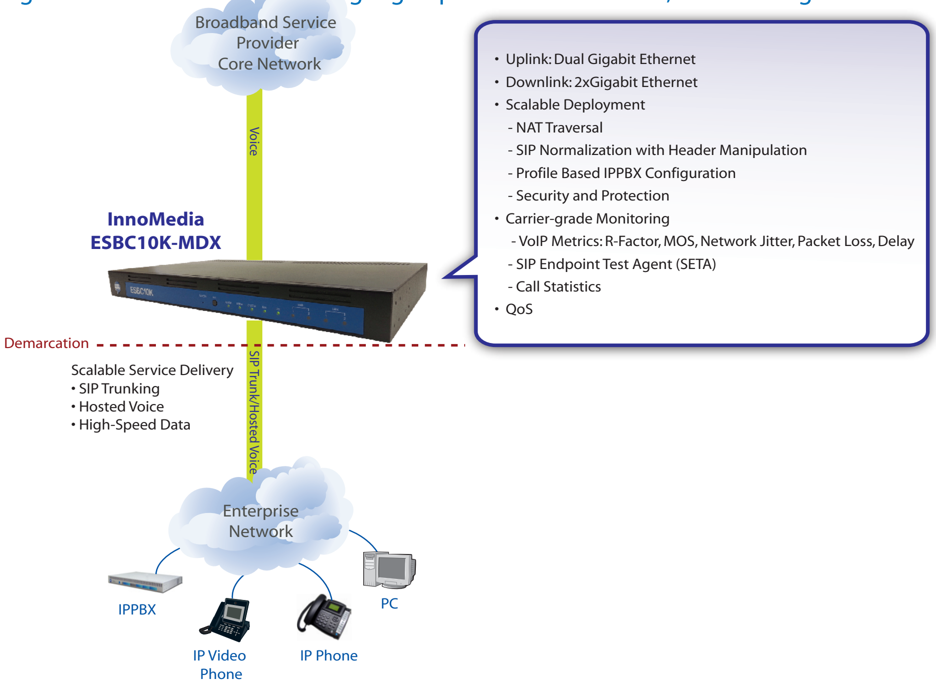
By configuring the ESBC10K-MDX, it is possible to allow different forms of SIP signaling negotiation between the Service Provider side and the Enterprise side before media transcoding takes place. There are maximum capabilities for the number of transcoding sessions per device:

- 200 maximum DTMF/CODEC/Fax Transcoding sessions

When "Allows calls even when no enough channels" is checked and the 200 FAX transcoding sessions are fully utilized, further FAX calls can still be made but without T.38-G.711 transcoding capability.

The ESBC10K-MDX, 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. A dedicated LAN Port for management may be enabled to allow access to the device without the need to interrupt any service.

Figure 5. BSP ESBC10K delivering high-speed Internet access, SIP trunking.



ESBC SUPPORTING SERVICE PROVIDER' SIP TRUNKING BUSINESS

Using B2BUA, the ESBC10K-MDX supports the key functions needed by the BSPs to offer reliable and scalable SIP trunk services to their enterprise customers. It supports up to 400 simultaneous B2BUA sessions. The key functions that are supported by the ESBC10K-MDX include:

1. <u>SIP Normalization</u>: Based on the B2BUA architecture, InnoMedia's ESBC10K-MDX provides Profile based settings, High-level classification for SIPConnect Adaptation, and Low-level header manipulation for SIP signaling normalization:
– Profile based settings: ESBC10K-MDX 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.
– 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 ESBC10K-MDX 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

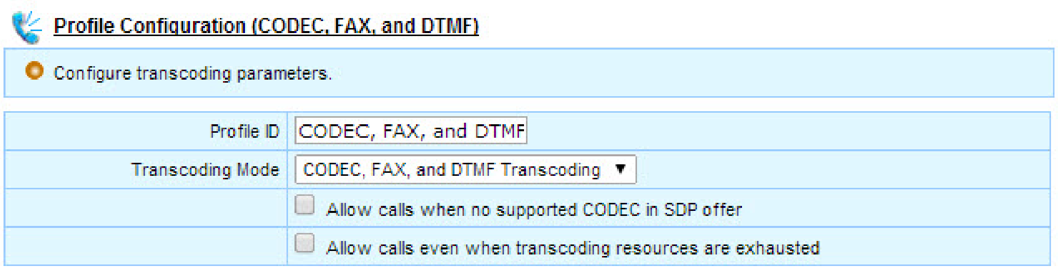
3. <u>Registration and Authentication</u>: Acting as a registrar server to SIP-PBXs, the InnoMedia ESBC10K-MDX supports the following SIP¬PBX registration methods:
a) Implicit registration – SIP-PBX with Dynamic or Static IP address sends registration of the Parent Number
b) Explicit registration – SIP-PBX 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. SIP signaling security:
– TLS: ESBC10K-MDX 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: ESBC10K-MDX validates all SIP messages

6. 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

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 (see section 0). 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 transcoding resources are exhausted	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.

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 1000 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 feature 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).

VOICE AND NETWORK PERFORMANCE MONITORING

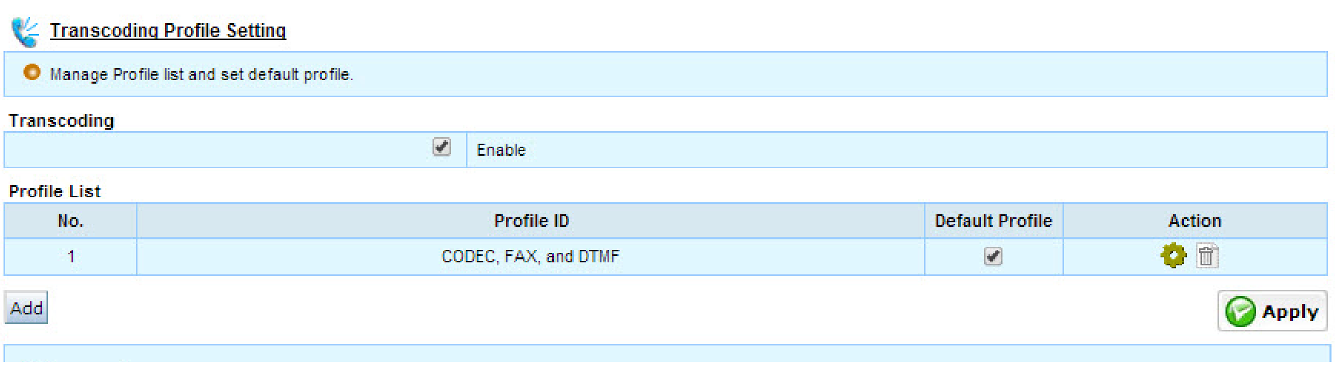
ESBC10K-MDX offers carrier-grade monitoring features, allowing service providers to offer SLA based SIP trunking services to their enterprise customers. The monitoring features including voice metrics with R-factor and MOS scores, network metrics with jitter, delay, and packet loss, CDR records and real-time UA & SIP trunk call states (Figure 9), SIP Call Trace (Figure 10), 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, making it easier for service providers to analyze the system performance bottlenecks (Figure 12). The ESBC10K-MDX also has an embedded SIP End-point Test Agent (SETA) that allows test calls to be made manually or programmed at scheduled times for quality tests or monitoring (Figure 13). 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.

MEDIA TRANSCODING

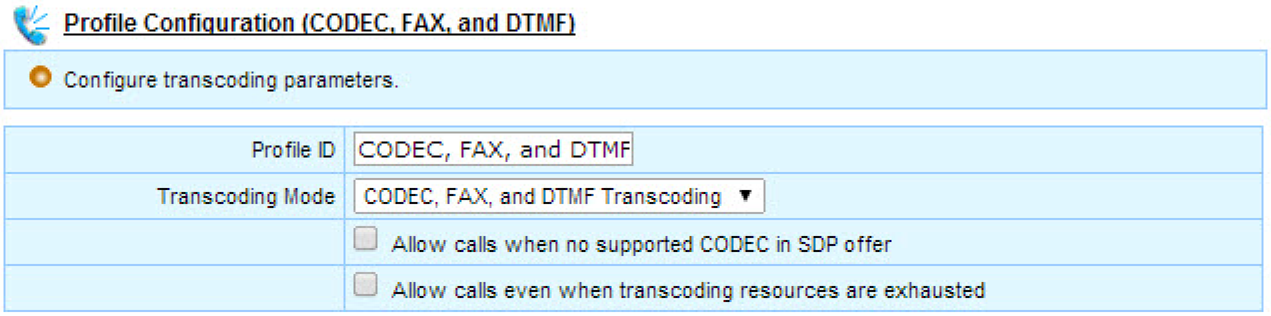
The media transcoding can be configured via the Transcoding Profile. The Transcoding Profile screen allows the profile list and the default profile to be managed. It also provides access to the Profile configuration screen, which allows the system administrator to configure Fax, CODEC or DTMF transcoding 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 -> ADVANCED -> 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.



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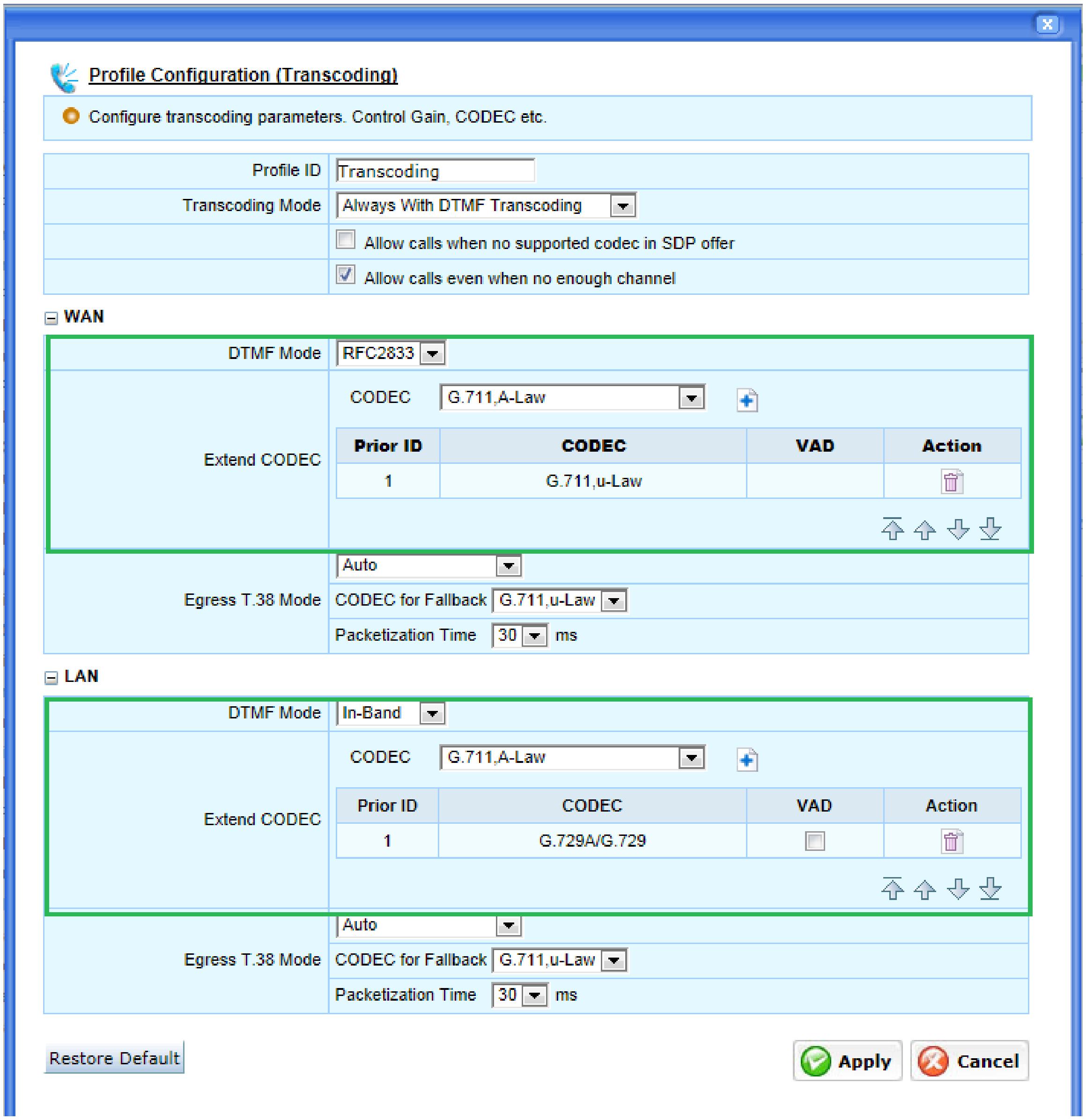
Only If Required	This setting only allows CODEC transcoding. CODEC transcoding only happens if there are no common CODECs between the caller and called UAs. Fax and DTMF transcoding will not be performed.
Always With DTMF Transcoding	Fax, DTMF and CODEC transcoding for all calls.
Always Without DTMF Transcoding	Fax, CODEC for all calls, but no DTMF transcoding.

<u>DTMF Mode: RFC2833 and In-band DTMF</u>
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.

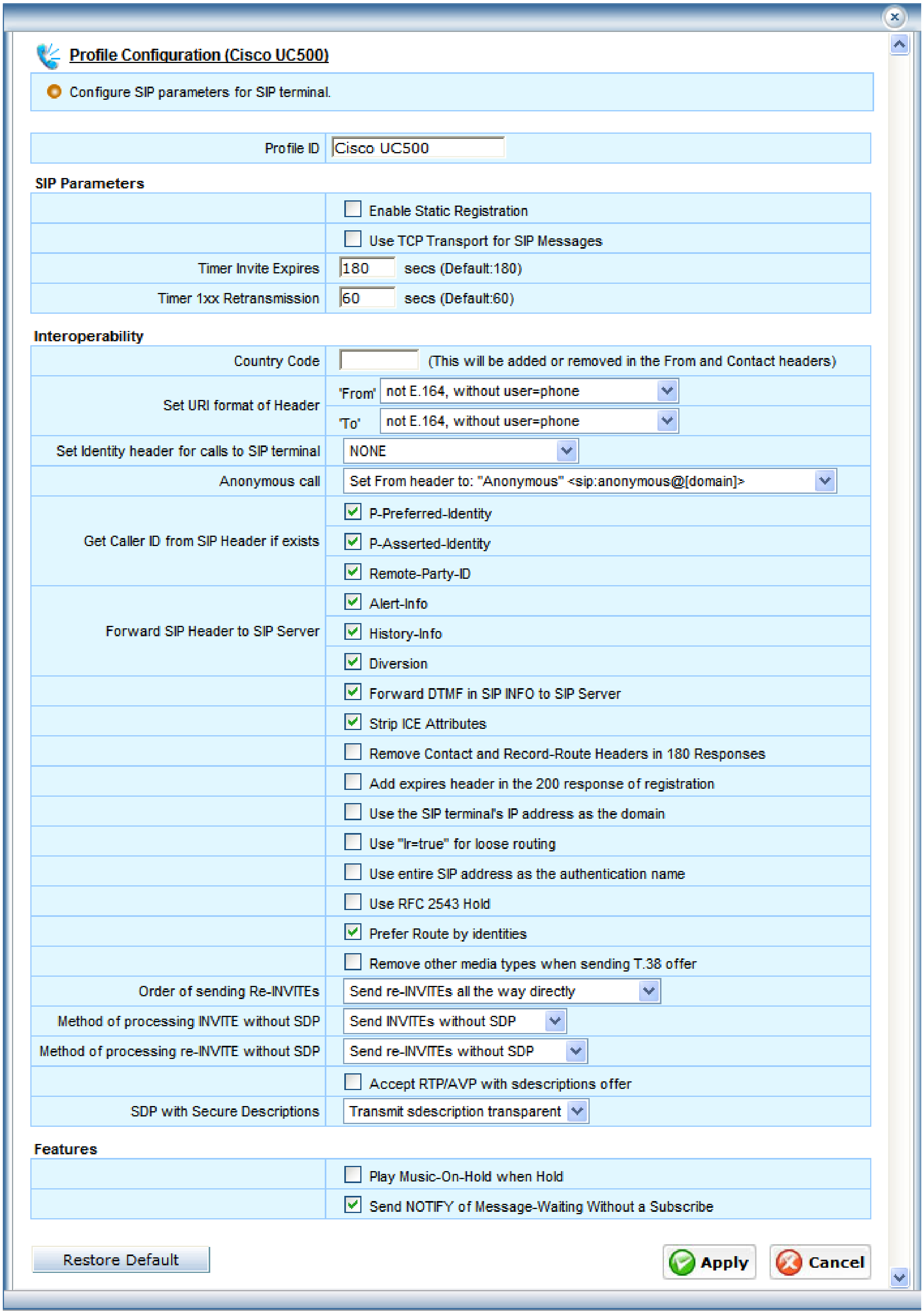
<u>Extend Codec:</u>
When configuring the 'Extend CODEC', the selection is based on the supported CODEC for that side of the ESBC's interface. The ESBC can be configured to add certain codec capabilities to transcoding profiles, and then perform transcoding in cases where the selected codec in the answer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

Figure 6


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
CODEC
G.711,A-Law
Extend CODEC
Prior ID
CODEC
VAD
Action
1
G.711,u-Law
Auto
Egress T.38 Mode
CODEC for Fallback
G.711,u-Law
Packetization Time
30
ms
LAN
DTMF Mode
In-Band
CODEC
G.711,A-Law
Extend CODEC
Prior ID
CODEC
VAD
Action
1
G.729A/G.729
Auto
Egress T.38 Mode
CODEC for Fallback
G.711,u-Law
Packetization Time
30
ms
Restore Default
Apply
Cancel

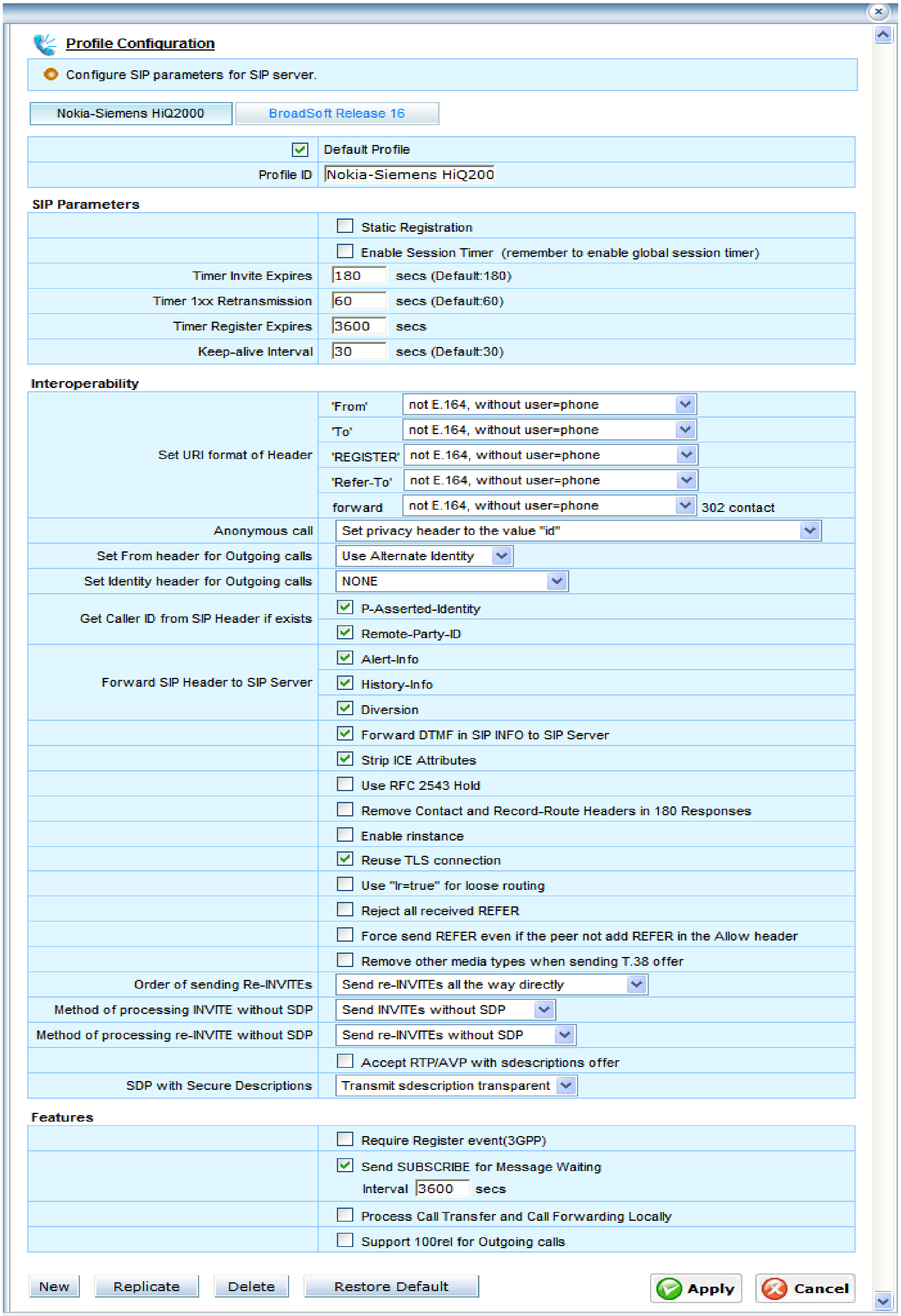
UA/SIP-PBX PROFILE

Figure 7



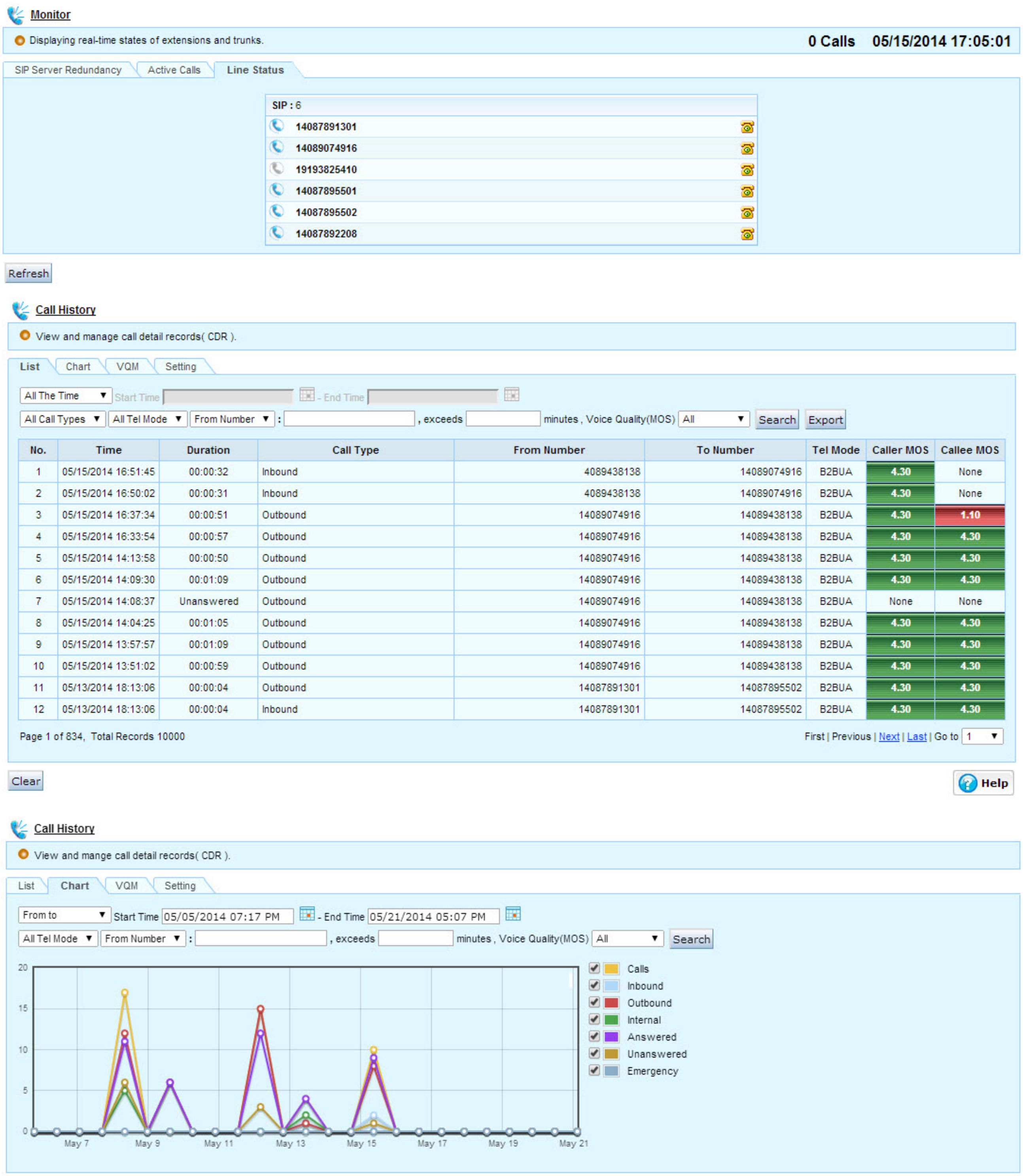
SIP TRUNK PROFILE

Figure 8



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

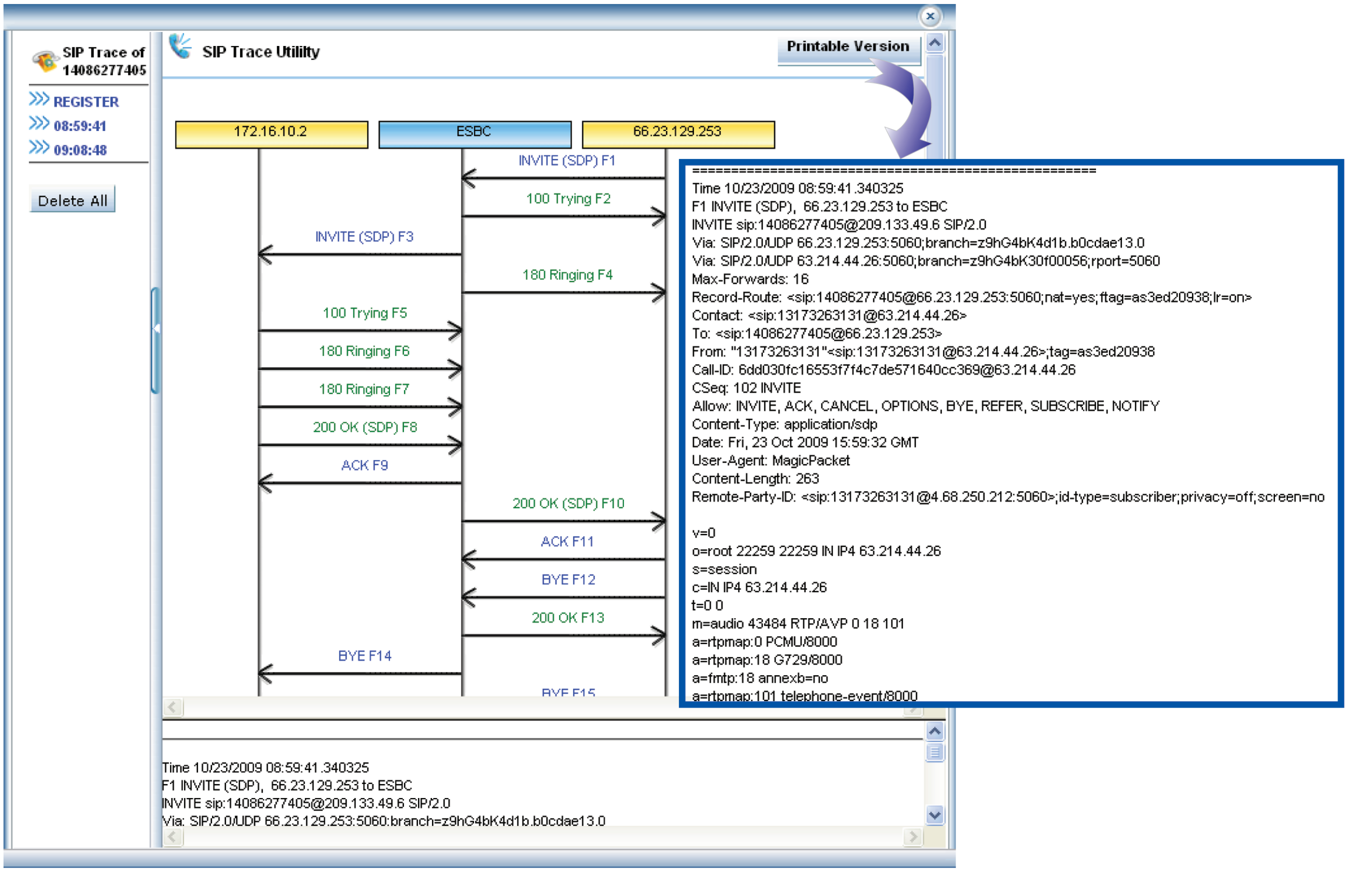
Figure 9



No.	Time	Duration	Call Type	From Number	To Number	Tel Mode	Caller MOS	Callee MOS
1	05/15/2014 16:51:45	00:00:32	Inbound	4089438138	14089074916	B2BUA	4.30	None
2	05/15/2014 16:50:02	00:00:31	Inbound	4089438138	14089074916	B2BUA	4.30	None
3	05/15/2014 16:37:34	00:00:51	Outbound	14089074916	14089438138	B2BUA	4.30	1.10
4	05/15/2014 16:33:54	00:00:57	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
5	05/15/2014 14:13:58	00:00:50	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
6	05/15/2014 14:09:30	00:01:09	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
7	05/15/2014 14:08:37	Unanswered	Outbound	14089074916	14089438138	B2BUA	None	None
8	05/15/2014 14:04:25	00:01:05	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
9	05/15/2014 13:57:57	00:01:09	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
10	05/15/2014 13:51:02	00:00:59	Outbound	14089074916	14089438138	B2BUA	4.30	4.30
11	05/13/2014 18:13:06	00:00:04	Outbound	14087891301	14087895502	B2BUA	4.30	4.30
12	05/13/2014 18:13:06	00:00:04	Inbound	14087891301	14087895502	B2BUA	4.30	4.30

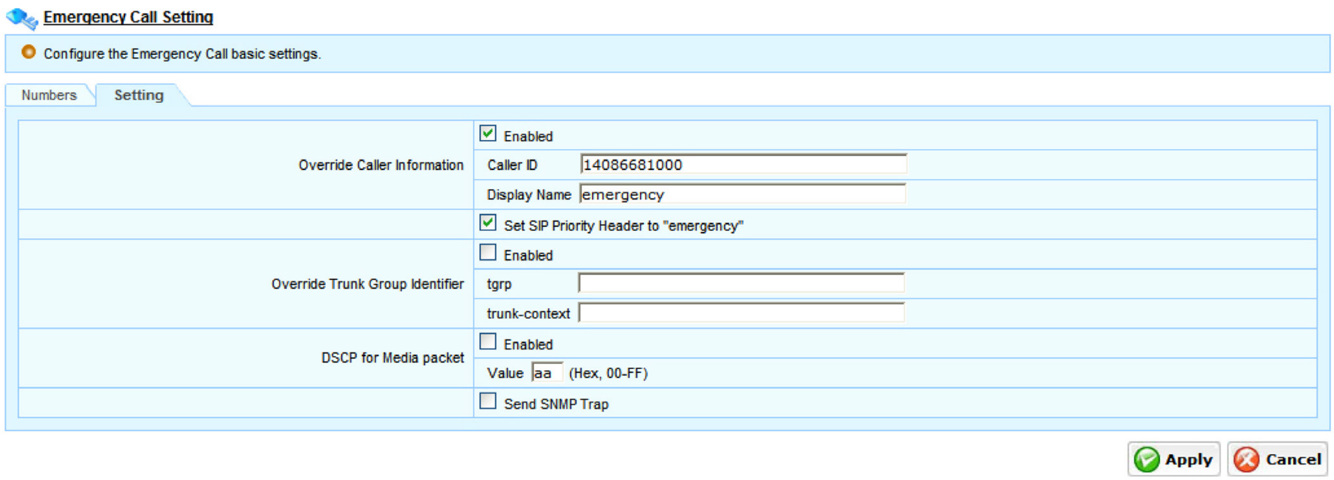
CALL TRACE GUI

Figure 10



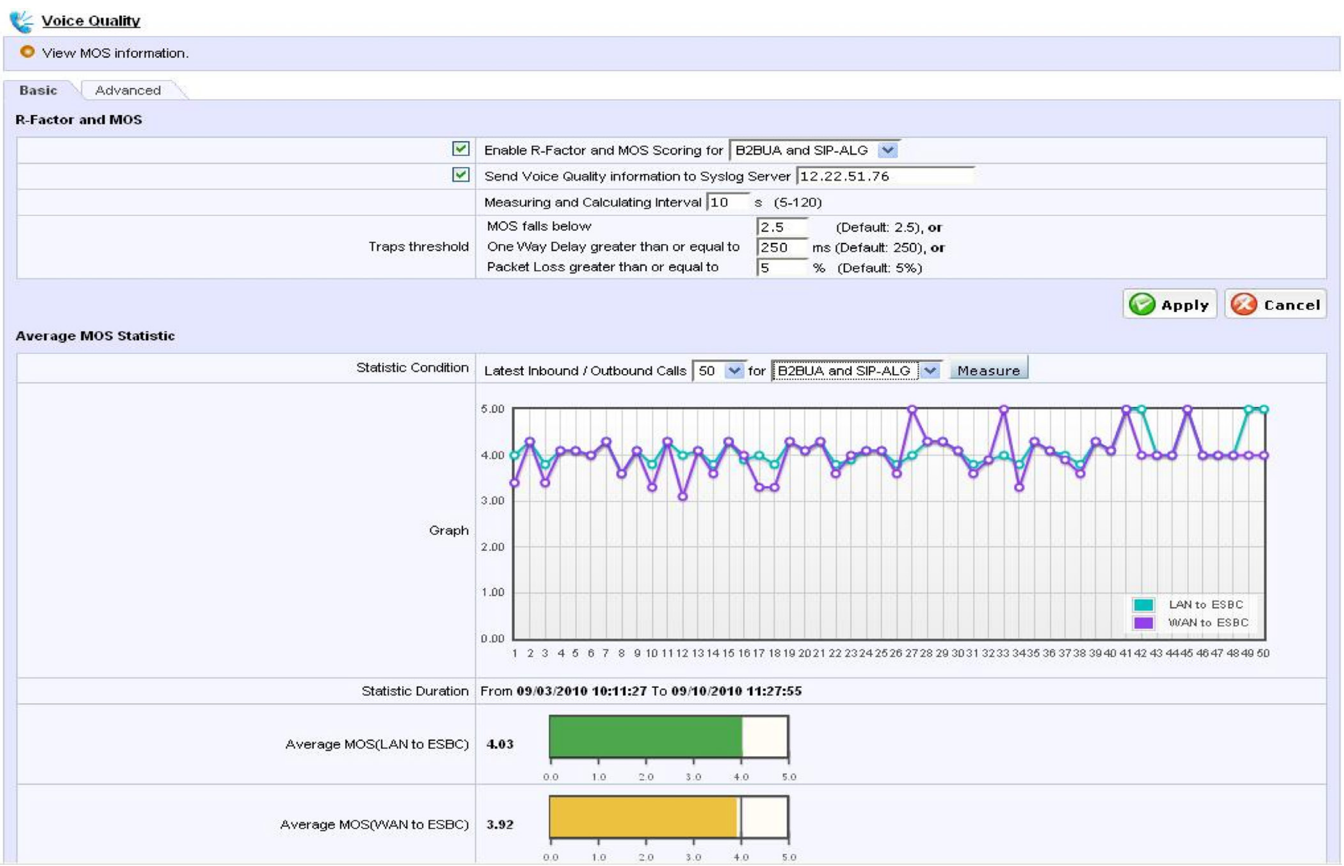
EMERGENCY CALL HANDLING

Figure 11



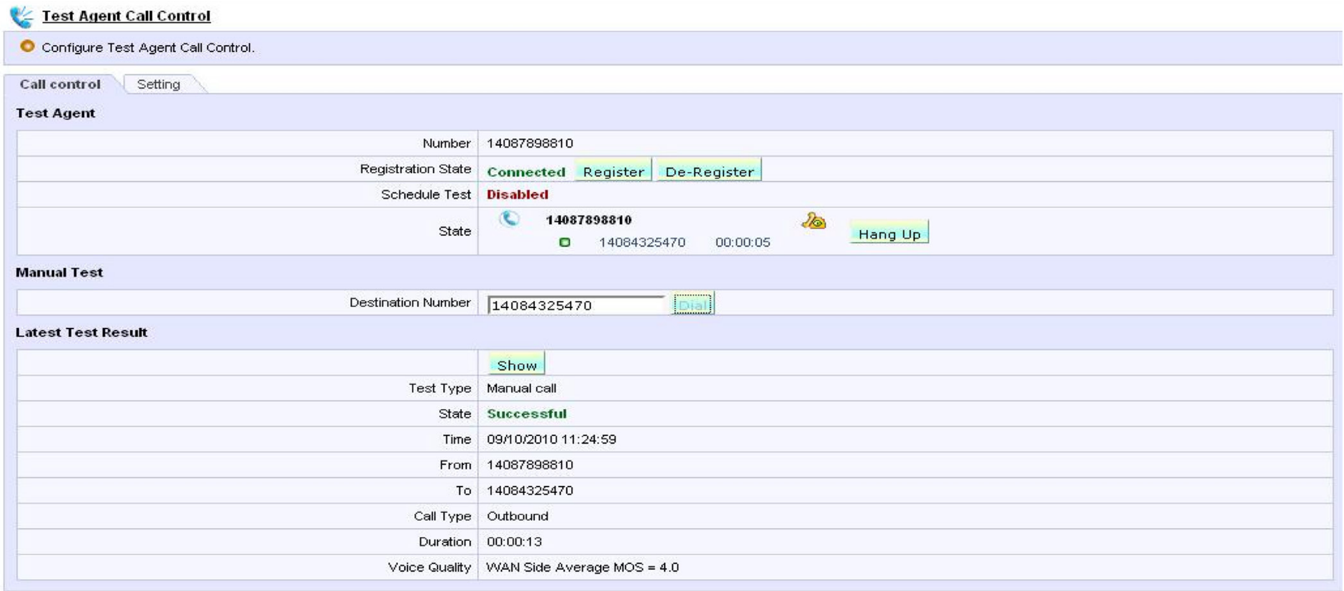
ESBC VOICE QUALITY MONITORING AND SNMP TRAP THRESHOLD SETTING

Figure 12. 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 13



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	14087898810 14084325470 00:00:05 Hang Up

Manual Test

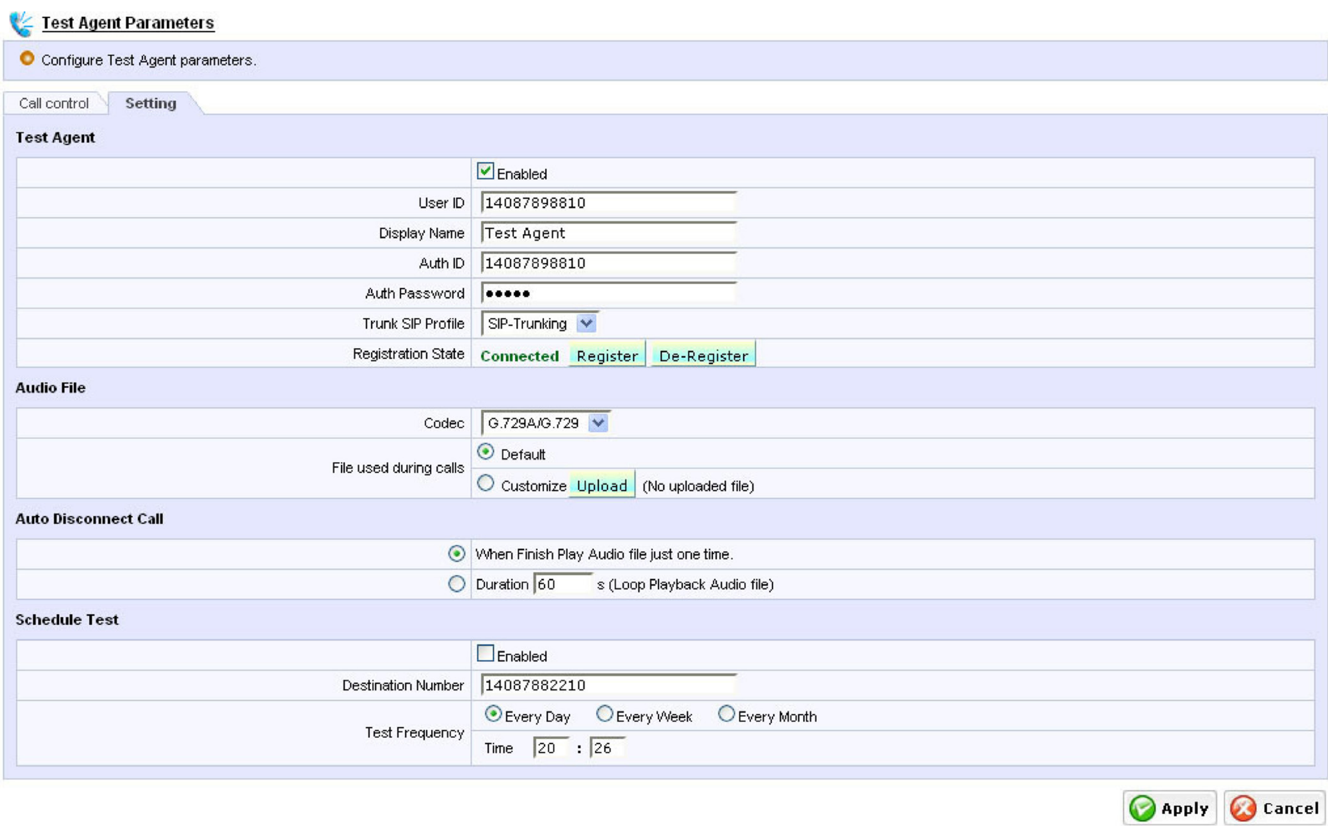
Destination Number	14084325470 Dial

Latest Test Result

	Show
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 14



Test Agent Parameters

Configure Test Agent parameters.

Call control | Setting

Test Agent

	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	Default Customize Upload (No uploaded file)

Auto Disconnect Call

	When Finish Play Audio file just one time.
	Duration 60 s (Loop Playback Audio file)

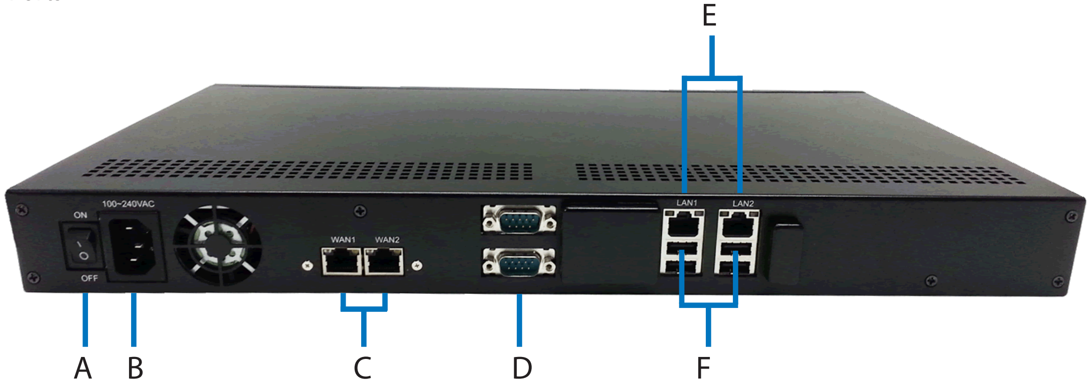
Schedule Test

	Enabled
Destination Number	14087882210
Test Frequency	Every Day Every Week Every Month Time 20 : 26

Apply Cancel

ESBC INTERFACE

A. Power ON/OFF Switch
B. Power Input Connector
C. WAN Ports
D. Serial I/F
E. LAN Ports
F. USB Ports



Product Interfaces

Category	Specification
Service Provider Interface	Dual Gigabit Ethernet RJ45 Connectors
User Data Interface	2 10/100/1000 BaseT Ethernet (RJ-45)

SPECIFICATIONS

Hardware Specification

Category	Specification
M/B	AIMB-281 Intel® Xeon E3/ Core™ i7/i5/i3/Celeron LGA1155 Mini-ITX
CPU	Intel® Celeron® Processor G1620 (2M Cache, 2.70 GHz)
BIOS	AMI 32 Mb SPI BIOS
Chipset	Intel® H61 Chipset
Memory	2G DDR3 1600 MHz SDRAM
HDD	4G SSD
Serials port	2 ports with RS-232
USB 2.0	4 ports

SPECIFICATIONS cont.

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
Media Transcoding	1. 200 maximum DTMF/CODEC/FAX Transcoding sessions 2. When "Allow calls even when transcoding resources are exhausted" is checked and the 200 FAX transcoding sessions are fully utilized, further FAX calls can still be made but without T.38-G.711 transcoding capability 3. Simultaneous CODEC Transcoding sessions and FAX Transcoding sessions will reduce the maximum transcoding capability per device
Networking Features	NAT Capabilities for Simultaneous SIP User Accounts Static IP Routing 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
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

SPECIFICATIONS cont.

Category	Specification
Signal Processing	Echo Cancellation Loop Back FAX (T.38 and G.711 fall-back) Caller ID FSK Signal Regeneration
Tones	Ring back tone Busy tone Recorder tone 5 distinct rings Dial tone Confirmation tone Ring splash Stutter tone Message waiting indicator (MWI) Off hook warning tone Caller ID generation & call waiting tone Configurable ring frequency
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)
DTMF Tone	DTMF tone detection and generation
Announcements	Play out any voice stream sent by Call Agent controlled announcement server

Physical Specifications

Category	Specification
Dimensions	1.75 in (H) x 17.25 in (W) x 10.0 in (D) / 44.45 mm (H) x 438 mm (W) x 254 mm (D)
Power Input	100 - 240 VAC
Power Consumption	Idle: 44.73W Load: 50.58W (In transcoding mode with - 100 Voice, 50 DTMF, and 8 Fax calls)
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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