InnoMedia ESBC

Enterprise Session Border Controller

Administration Guide

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1 Safety Check

1.1 Important Safety Instructions

This section contains important safety information you should know before working with the ESBC. Use the following guidelines to ensure your own personal safety and to help protect your ESBC from potential damage.

This warning symbol means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents

Only trained and qualified personnel should be allowed to install, replace, or service this equipment.

To avoid electric shock, do not connect safety extra-low voltage (SELV) circuits to telephone-network voltage (TNV) circuits. LAN ports contain SELV circuits, and WAN ports contain TNV circuits. Some LAN and WAN ports both use RJ-45 connectors. Use caution when connecting cables. Statement 1021

Before working on a system that has an on/off switch, turn OFF the power and unplug the power cord.

This unit is intended for installation in restricted access areas. A restricted access area is where access can only be gained by service personnel through the use of a special tool, lock and key, or other means of security, and is controlled by the authority responsible for the location.

This product relies on the building's installation for short-circuit (overcurrent) protection. Ensure that a fuse or circuit breaker no larger than 120 VAC, 15A U.S. (240 VAC, 10A international) is used on the phase conductors (all current-carrying conductors).

This equipment must be grounded. Never operate the equipment in the absence of a suitably installed ground conductor. Contact the appropriate electrical inspection authority or an electrician if you are uncertain that suitable grounding is available.

Do not work on the system or connect or disconnect cables during periods of lightning activity.

Before working on equipment that is connected to power lines, remove jewelry (including rings, necklaces, and watches). Metal objects will heat up when connected to power and ground and can cause serious burns or weld the metal object to the terminals.

The safety cover is an integral part of the product. Do not operate the unit without the safety cover installed. Operating the unit without the cover in place will invalidate the safety approvals and pose a risk of fire and electrical hazards.

Enclosure covers serve three important functions: they prevent exposure to hazardous voltages and currents inside the chassis; they contain electromagnetic interference (EMI) that might disrupt other equipment; and they direct the flow of cooling air through the chassis. Do not operate the system unless all covers are in place.

Ultimate disposal of this product should be handled according to all national laws and regulations.

To prevent bodily injury when mounting or servicing this unit in a rack, you must take special precautions to ensure that the system remains stable. The following guidelines are provided to ensure your safety:

- This unit should be mounted at the bottom of the rack if it is the only unit in the rack.
- When mounting this unit in a partially filled rack, load the rack from the bottom to the top with the heaviest component at the bottom of the rack.
- If the rack is provided with stabilizing devices, install the stabilizers before mounting or servicing the unit in the rack.

1.2 Safety Guidelines

To reduce the risk of bodily injury, electrical shock, fires, and damage to the equipment, observe the following precautions.

1.2.1 General Precautions

Observe the following general precautions for using and working with your system:

Opening or removing covers might expose you to electrical shock. Components inside these compartments should be serviced only by an authorized service technician.

- If any of the following conditions occur, unplug the product from the electrical outlet and replace the part or contact your authorized service provider:
    - The power cable, extension cord or plug is damaged.
    - An object has fallen into the product.
    - The product does not operate correctly when you follow the operating instructions.
    - The product has been exposed to water.
    - The product has been dropped or damaged.
    - The product does not operate correctly when you follow the operating instructions.
- Keep your system components away from radiators and heat sources. Also, do not block cooling vents.
- Do not spill food or liquids on your system components, and never operate the product in a wet environment.
- Do not push any objects into the openings of your system components. Doing so can cause fire or electric shock by shorting out interior components.
- Allow the product to cool before removing covers or touching internal components.
- Use the correct external power source. Operate the product only from the type of power source indicated on the electrical ratings label. If you are not sure of the type of power source required, consult your service representative or local power company.
- Use only approved power cables. If you have not been provided with a power cable for your ESBC or for any AC-powered option intended for your system, purchase a power cable that is approved for use in your country. The power cable must be rated for the product and for the voltage and current marked on the product's electrical ratings label. The voltage and current rating of the cable should be greater than the ratings marked on the product.
- To help prevent electric shock, plug the system components and peripheral power cables into properly grounded electrical outlets. These cables are equipped with three-prong plugs to help

ensure proper grounding. Do not use adapter plugs or remove the grounding prong from a cable. If you must use an extension cord, use a three-wire cord with properly grounded plugs.

- Observe extension cord and power strip ratings. Make sure that the total ampere rating of all products plugged into the extension cord or power strip does not exceed 80 percent of the extension cord or power strip ampere ratings limit.
- To help protect your system components from sudden, transient increases and decreases in electrical power, use a surge suppressor, line conditioner or uninterruptible power supply (UPS).
- Position cables and power cords carefully; route cables and the power cord and plug so that they cannot be stepped on or tripped over. Be sure that nothing rests on your system components' cables or power cord.
- Do not modify power cables or plugs. Consult a licensed electrician or your power company for site modifications. Always follow your local or national wiring rules.

1.2.2 Protecting Against Electrostatic Discharge

Static electricity can harm delicate components inside the equipment. To prevent static damage, discharge static electricity from your body before you touch any of your system's electronic components. You can do so by touching an unpainted metal surface on the chassis.

You can also take the following steps to prevent damage from electrostatic discharge (ESD):

- When unpacking a static-sensitive component from its shipping carton, do not remove the component from the antistatic packing material until you are ready to install the component in your system. Just before unwrapping the antistatic packaging, be sure to discharge static electricity from your body.
- When transporting a sensitive component, first place it in an antistatic container or packaging.
- Handle all sensitive components in a static-safe area. If possible, use antistatic floor pads and workbench pads.

2 Getting Started with the ESBC

2.1 ESBC Model Differentiations and Key Features

The Innomedia ESBC product family seamlessly migrates your enterprise telephony system to state-of-the-art IP-based SIP trunking or hosted voice services.

2.1.1 TDM PRI with PRI ESBC (ESBC 9x80 series)

The InnoMedia Enterprise Session Border Controllers are capable of both B2BUA and SIP ALG operation as well as having TDM PRI interfaces. These features allow broadband service providers to offer services to TDM-PBX customers today, with an easy migration path to SIP trunking or hosted services later when the customers transition from TDM to IP by adopting IP-PBX or IP Centrex services.

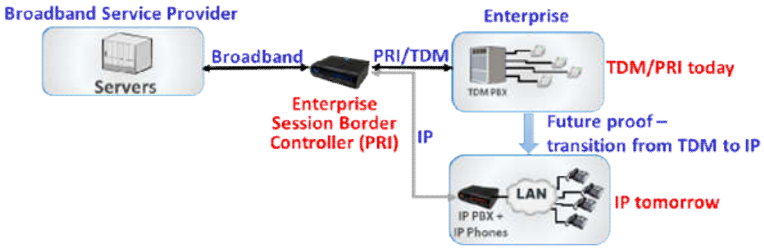


Figure 1. TDM-PRI with the PRI ESBC

2.1.2 SIP Trunking Using ESBCs with B2BUA (ESBC 8xxx and 9xxx series)

As part of InnoMedia's comprehensive business voice service solutions, InnoMedia's highly manageable Enterprise Session Border Controller (ESBC) product family provides complete B2BUA functionality for comprehensive signaling normalization/header manipulation, transcoding for codec/fax/DTMF media translation, NAT traversal, topology hiding, SHMR for in-field header manipulation, QoS management, and many other features to support the ability for a service provider to deliver a scalable and reliable SIP Trunking offering. These IMS-ready and SIPConnect-compliant ESBCs are ideal for service providers looking for seamless network migration.

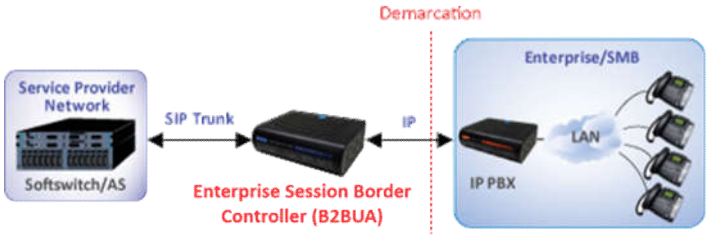


Figure 2. SIP Trunking Using ESBCs with B2BUA

2.1.3 Hosted Service Using ESBCs with SIP ALG (ESBC 8xxx and 9xxx series)

As part of InnoMedia's complete and comprehensive business voice service solutions for service providers, InnoMedia's highly manageable ESBC product family provides a SIP ALG function for topology hiding, NAT traversal, SIP Header Manipulation Rules (SHMR) for in-field header manipulation, QoS management, and many other features to support the ability for a service provider to deliver hosted voice services. Being highly integrated, InnoMedia's ESBC family is ideal for service providers looking to offer reliable and scalable hosted services.

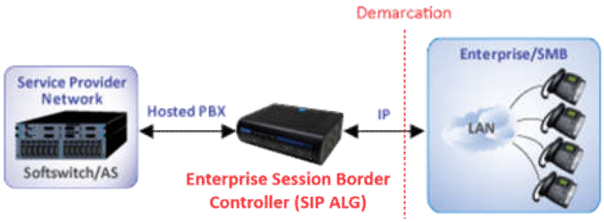


Figure 3. Hosted Service Using ESBCs with SIP ALG

2.1.4 High Capacity B2BUA and Transcoding Integrated Model: ESBC 10K –MDX series

The ESBC10K-MDX. A carrier-grade, high-capacity, high-performance, and cost effective ESBC solution with an optimum level of B2BUA and Media Transcoding integration, enables service providers to offer highly scalable SIP trunking and hosted voice services to mid and large-size enterprise customers.

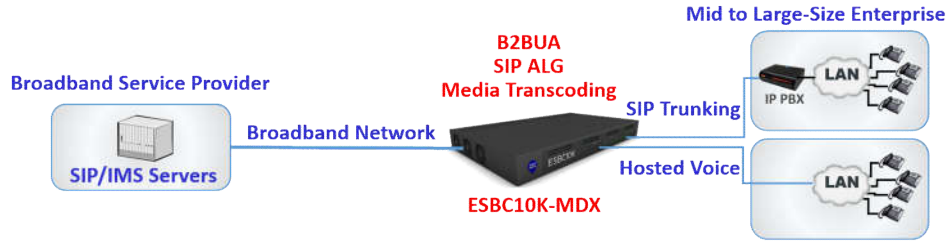


Figure 4. High Capacity B2BUA and Transcoding Integrated Model: ESBC 10K –MDX series

Model Name	WAN	B2BUA (SIP Trunk)	SIP ALG (Hosted Service)	Transcoding	T1/E1	QoS
ESBC8528-4B	DOCSIS 2.0	Yes	Yes	-	-	*Smart*-DQoS™
ESBC9528-4B	DOCSIS 3.0	Yes	Yes	-	-	*Smart*-DQoS™
ESBC9578-4B	DOCSIS 3.0	Yes	Yes	Yes	-	*Smart*-DQoS™
ESBC9580-4B	DOCSIS 3.0	Yes	Yes	-	Yes	*Smart*-DQoS™
ESBC8328-4B	10/100BT	Yes	Yes	-	-	ToS/DSCP
ESBC9328-4B	Gigabit	Yes	Yes	-	-	ToS/DSCP
ESBC9378-4B	Gigabit	Yes	Yes	Yes	-	ToS/DSCP
ESBC9380-4B	Gigabit	Yes	Yes	-	Yes	ToS/DSCP
ESBC-10K-MDX	Dual Gigabit	Yes	Yes	Yes		ToS/DSCP
ESBC-10K-MD	Dual Gigabit	Yes	Yes	No		ToS/DSCP

Table 1. The ESBC Product Summary

2.2 Capacity and License

The ESBC series platforms support software licenses so that the platform can be upgraded or downgraded in the field.

The ESBC license number essentially is equivalent to the number of concurrent calls allowed on a system. Hence, adding licenses increases the maximum number of concurrent calls handled by the device. There is no need to purchase other license type for registered SIP UAs. Check your ESBC system capacities from the following page.

Login to the ESBC Administrative web console, and navigate to **System** > **License**. (See section 2.5 for descriptions of login to the console)

Figure 5. Managing the ESBC licenses

License Control	Description
Licensed Date	The date when license string (or file) was input to the system.
B2BUA Calls	The number of concurrent calls for SIP trunk voice service.
SIP ALG Calls	The number of concurrent calls for hosted voice service.

Note: The ESBC system maximum capacities are model dependent. See section 6.11 for the maximum capacities of your ESBC system.

2.3 Installing the ESBC9xxx and 8xxx series to an enterprise network

Getting Started. Please refer to the document: InnoMedia ESBC deployment checklist for Voice Service Deployment.

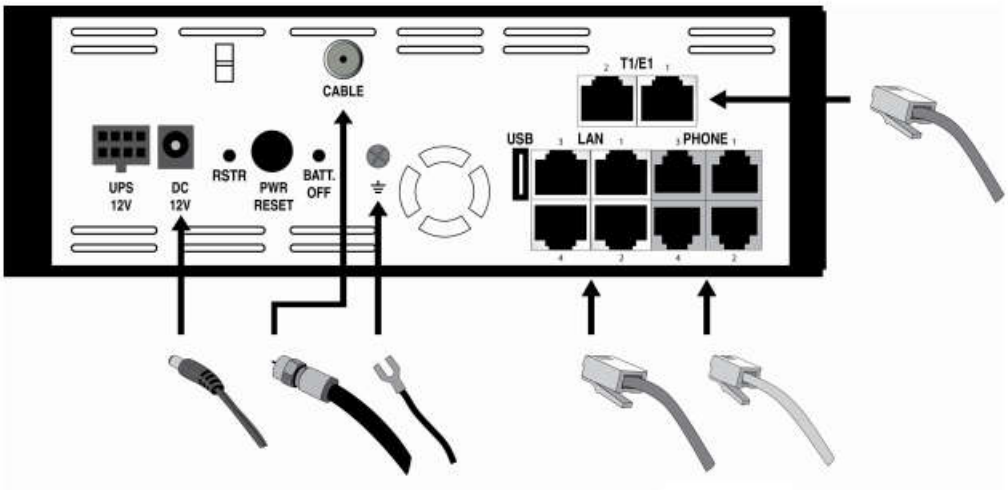


Figure 6. Hardware interface. The ESBC9580 back panel

Note 1. The ESBC93xx series WAN interface is Gigabit Ethernet; the ESBC95xx WAN interface is a DOCSIS 3.0 Cable Modem.

Note 2. The T1/E1 interfaces are applicable for ESBC9x80 models only.

Step 1–Connecting the panel ports.

1. Connect the active RF coaxial cable to the “CABLE” connector (for ESBC 8528/9528/9580) or the RJ-45 cable to the “WAN” connector (for ESBC 8328/9328/9380/9580/9378).
2. Connect the administrator PC to LAN port 1.
3. Connect LAN ports 2, 3, or 4 to the corporate LAN which resides in the same network as the IP PBX or IP phones. Skip this step for a TDM PBX with E1/T1 connections.
4. Optionally, connect T1/E1 port(s) to a corporate TDM PBX. Please ensure the cable between the interface port and the PBX is connected correctly. Do not connect to T1/E1 Port 2 unless T1/E1 Port 1 is also connected to the same TDM PBX.
5. Optionally, connect any standard analog phone or fax machine to the “PHONE” connectors, labeled 1-4.
6. Open the battery compartment and insert the optional battery.
7. Connect included AC power cable to the electrical outlet and its cable to the ESBC’s 12V DC connector.

Step 2 – Configure the administrator PC to access the ESBC

The default LAN IP address of the ESBC is 172.16.1.1 with a subnet mask of 255.255.0.0. The ESBC LAN should be placed on the same LAN network where your IP PBX and IP Phone reside. For other network placements, please refer to section 0 for detailed descriptions.

8. Configure your PC with an appropriate IP address (e.g., 172.16.0.5) within the same network as the ESBC LAN.
9. Start your web browser, and enter http://172.16.1.1 in the address field to connect to the ESBC. The login page will appear. The default user name is “admin” and the password is “123”. Click the login button to enter the ESBC main page.

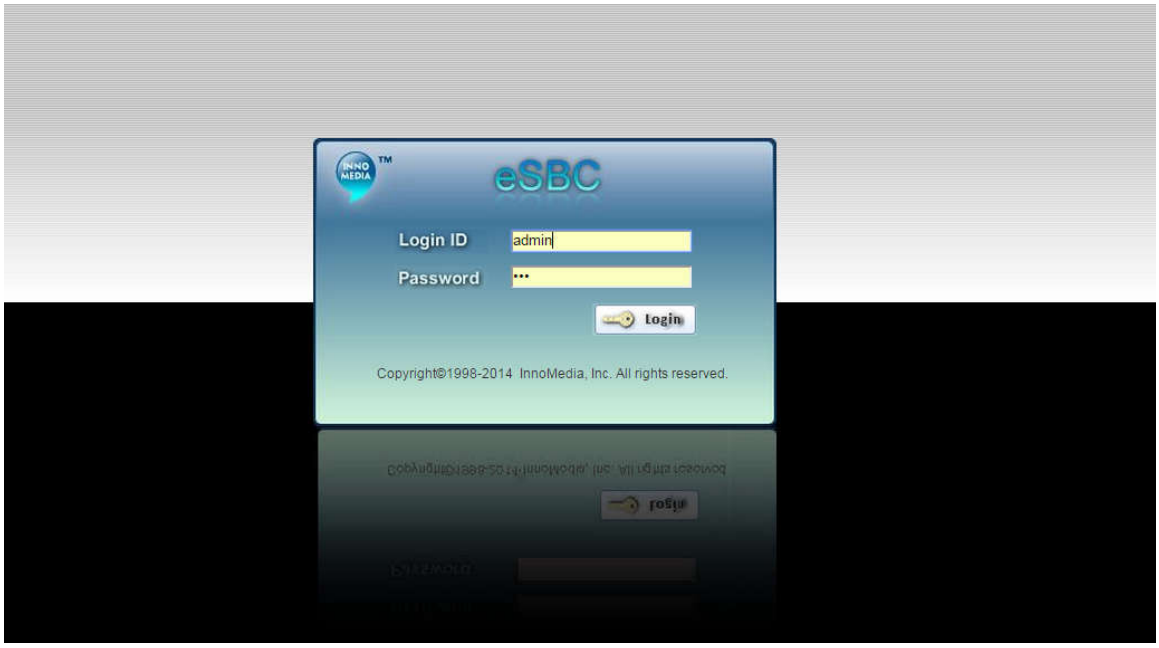


Figure 7. the ESBC login page (web console)

2.4 Installing the ESBC10K series to an enterprise network

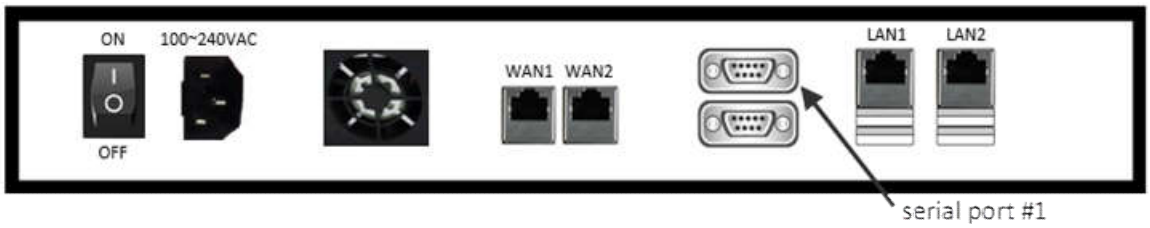


Figure 8. The ESBC 10K back panel

Note 1. The ESBC 10K series support dual WAN interfaces, which support layer 2 redundancy features.

Note 2. LAN1 by default is configured as a management port whose default IP address is 10.10.200.1/255.255.255.0. This logical port is designed for an administrator PC.

Note 3. LAN 2 is the Voice-NAT port whose default IP address is 172.16.1.1/255.255.0.0. This logical port is designed for telephony services.

Note 4. When the management port is enabled, the administrator PC can access the ESBC10K console only via LAN1. When the management port is disabled from the web console, LAN1 is disabled. The administrator PC and telephony devices and equipment connect to LAN2 for management and telephony services.

Step 1–Connecting the panel ports.

1. Connect the RJ-45 cable to either the WAN 1 or WAN 2 interfaces. Or connect two cables to WAN1 and WAN 2 respectively. Note that when both WAN ports are used, connect them to 2 different Ethernet switches.
2. Connect the administrator PC to LAN1.
3. Connect LAN 2 to the corporate LAN which resides in the same network as the IP PBX and/or IP Phones.
4. Connect included AC power cable to the electrical outlet.

Step 2 – Configure the administrator PC to access the ESBC.

5. Configure your PC with appropriate IP address (i.e., 10.10.200.5) within the same network as the ESBC management port.
6. Start your web browser, and enter http://10.10.200.1 in the address field to connect to the ESBC. The login page will appear. The default user name is “admin” and the password is “123”. Click the login button to enter the ESBC main page.

If the management port is disabled, connect your PC NIC to ESBC LAN2, and the login procedure is the same as that of the ESBC 9xxx series. See section 2.3 for further details.

Start your web browser and enter http://10.10.200.1 (or http://172.16.1.1) in the address field to connect to the ESBC. The login page will appear. The default user name is “admin” and the password is “123”. Click the login button to enter the ESBC main page. See Figure 47.

2.5 WEB based management (HTTP, HTTPs)

This administrative guide is based on the operation of WEB based management. There are other supporting management interfaces: CLI, XML, SNMP, Provisioning and EMS, which will be described briefly in the following sections.

Access the ESBC WEB management console through one the following URLs from a web browser. See section 3.2 for detailed descriptions of configuring the ESBC Ethernet interfaces.

- LAN access:
  - For ESBC 8xxx/9xxx series, enter http://NAT_VOICE_IP (or http://ESBC_LAN_IP). The default IP address is 172.16.1.1/16
  - For the EBSC 10K series, enter http://management_port_IP. The default IP address is 10.10.200.1/24.
- WAN access: http://WAN_IP:8080; or https://WAN_IP. (The default configuration of the ESBC WAN interface is DHCP client.) When the WAN Interface mode is configured as “Multiple Interfaces”, enter the ESBC IP address assigned to OAMP network.

The default credentials to logon to the WEB management console are: User ID: admin; Password: 123.

When the ESBC management port is configured and enabled, web console access from the NAT_Voice interface will be disabled. See section 3.4.3.1 for details.

Note: Multiple users are allowed to access the ESBC WEB console simultaneously.

- When an admin user who has read/write privileges has logged into the system, other admin users with different usernames can login to the system simultaneously but with only read-only privilege, or the second admin user can force the current user to logout by selecting the “override” option.
- When an admin user is currently logged in, users with “oper” or “tech” privileges may login to the system but will have “read–only” privilege.

2.5.1 The Console Home Page: System Overview

Once logged on to the ESBC WEB management console successfully, the dashboard page displays system configurations and status.

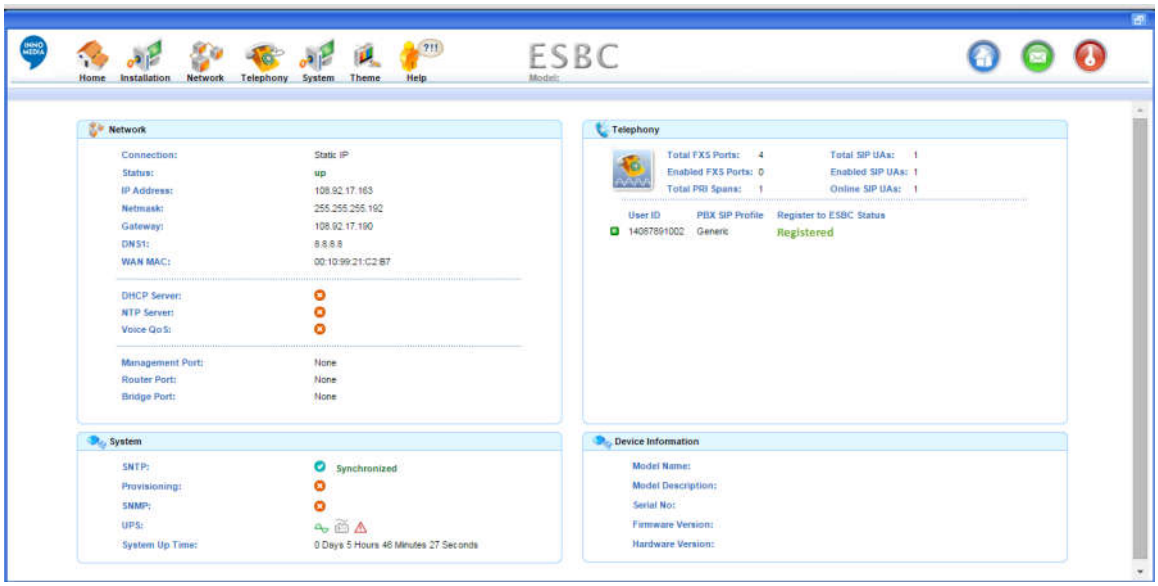


Figure 9. The ESBC Home Page

2.5.2 Real Time Activity Monitor

The ESBC provides a real time system activity monitor screen, including Network and Telephony activities.

2.5.2.1 Network Status

This Monitor page displays overall IP connection status. Navigate to **Network** > **Settings** > **Monitor**.

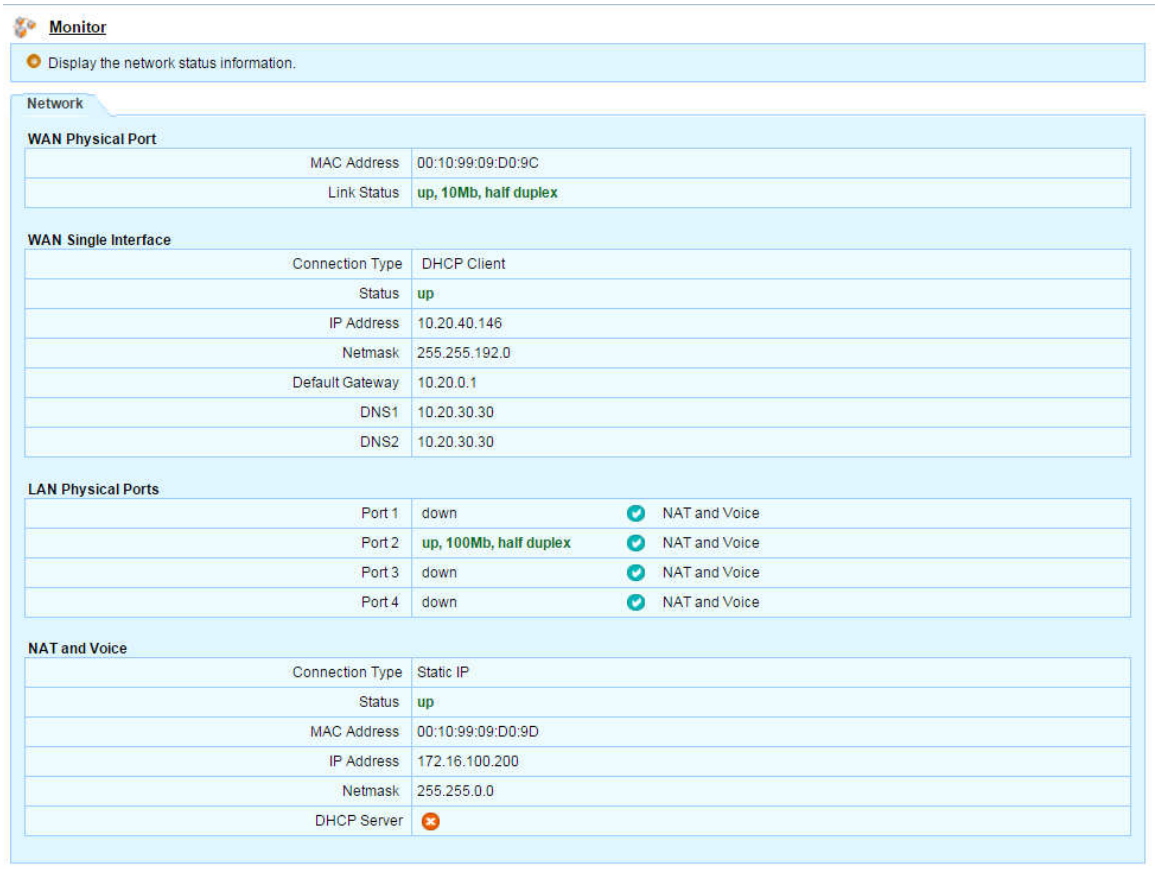


Figure 10. Network Connection Status Monitor Page (ESBC 93xx example)

Internet Connection	Description
Current Connection Type	The mechanism of IP addressing, either DHCP client, or Static IP.
Log	Displays the DHCP client connection event history.
Status	The layer 3 IP connection status of the WAN interface.
Link Status	The layer 2 (data link) connection status of the WAN interface.
MAC address	The MAC address of the ESBC Internet Ethernet interface (WAN).

LAN Connection	Description
Port 1 ~ 4	Up or Down. Link speed (10, 100, or 1000Mbps), duplex mode (full or half). Displays data link connection status for all four LAN ports, respectively. Each ESBC LAN port can be assigned a different subscribed service, i.e., NAT-Voice, Bridge, Router, and

	Management. See section 3.4.2 for details.

NAT and Voice	Description
Current Connection Type	Static IP or DHCP Client
Status	Up or Down
MAC Address	The MAC Address of the LAN NIC interface card
IP Address	The IPv4 address assigned to the NAT and Voice interface
Netmask	The netmask for the enterprise NAT and Voice network

Router	Description
IP Address	The IPv4 address assigned to the Router interface, if Router port is enabled on the ESBC.
Netmask	The netmask for the enterprise data network.

2.5.2.2 Port Mapping Table

The port mapping table assigned for remote access to hosts residing on the ESBC NAT-Voice network via the WAN interface (see section 6.10.1).

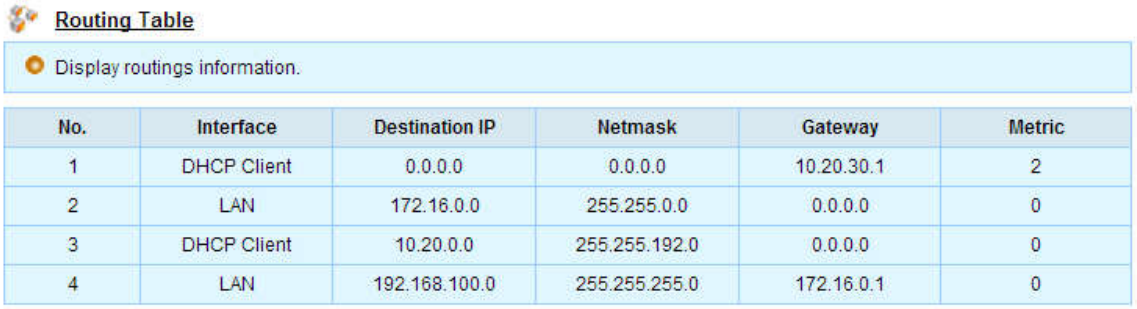
Note: Port Mapping Table is not available when the WAN interface is configured with multiple logical IP networks.

Figure 11. The Port Mapping Table

2.5.2.3 Routing Table

Click the <Routing Table > button to view the ESBC network routing information for both Internet and LAN connections.

Note: Routing Table is not available when the WAN interface is configured with multiple logical IP networks.



No.	Interface	Destination IP	Netmask	Gateway	Metric
1	DHCP Client	0.0.0.0	0.0.0.0	10.20.30.1	2
2	LAN	172.16.0.0	255.255.0.0	0.0.0.0	0
3	DHCP Client	10.20.0.0	255.255.192.0	0.0.0.0	0
4	LAN	192.168.100.0	255.255.255.0	172.16.0.1	0

Figure 12. Network Routing Table

See section 3.4 for suggestions on LAN side network topology design.

2.5.3 Telephony Activities

Navigate to **Telephony** > **TOOLS** > **Monitor**. Click the associated tab to display the real-time states of SIP Servers, Lines, and Active calls.

The ESBC admin web GUI page refreshes at configurable interval (default 3 seconds), see section 6.10.1. If necessary, click the <Refresh> button to get the latest status of the server status information.

2.5.3.1 SIP Server Redundancy

This page displays the enquiry results and status of redundant sip servers, see section 4.1.2.

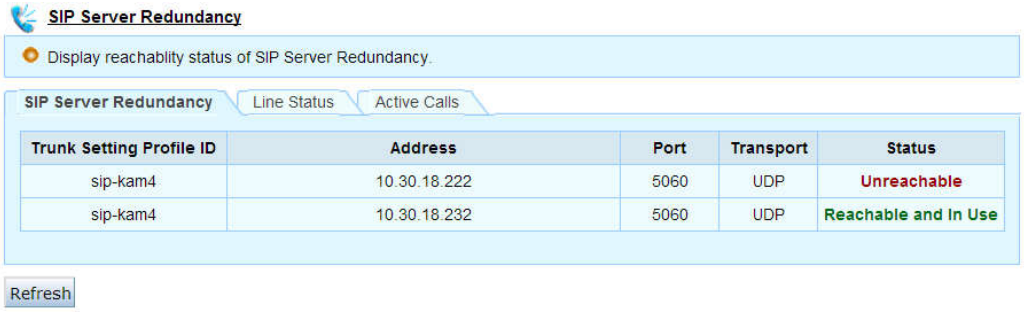


Figure 13. SIP Redundant Server List

2.5.3.2 Line Status

This page displays the current state of all user accounts configured on the ESBC, as busy or idle states. If in a busy state, call duration, call type, and peer telephone number are displayed as well.

The upper right corner shows the number of active calls at any particular moment.

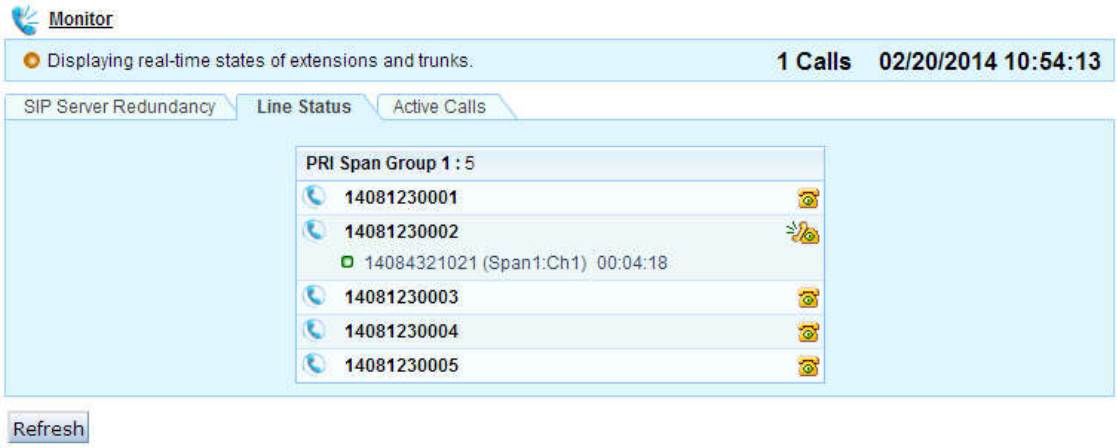


Figure 14. The current status of all user accounts (lines)

2.5.3.3 Active Calls

Click the Active Calls tab to display current active calls.

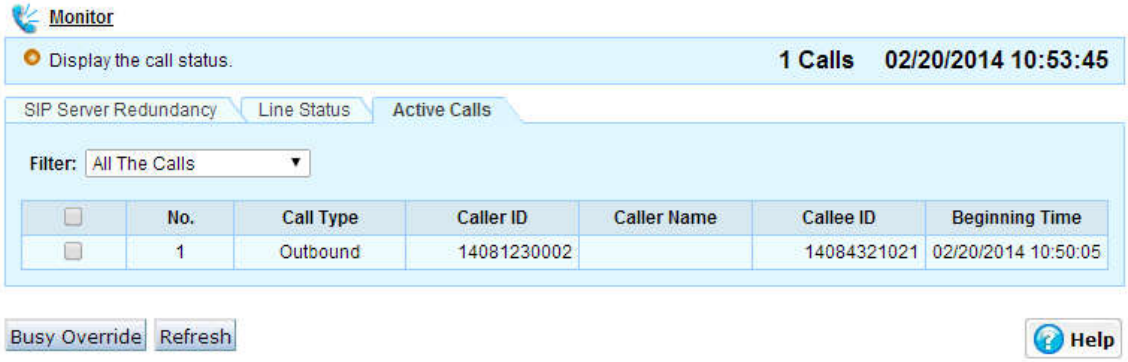


Figure 15. Displaying active calls

Active Calls	Description
Busy Override	The selected calls will be disconnected and associated parties will hear busy tones.

2.6 CLI Based Management

The ESBC supports a CLI (command line interface) based console interface to configure system parameters.

- ESBC 9x, 8x: connecting via SSH and EMS-telnet clients.
- ESBC 10K: connecting via SSH and EMS-telnet clients, and also serial console.

The login ID and password are identical to those for WEB console. Once you login to the ESBC CLI console, the ESBC's current running version and the LAN IP address are displayed.

Note: If accessing the ESBC CLI console via WAN interface, when the WAN Interface mode is configured as "Multiple Interfaces, enter the ESBC IP address assigned to OAMP network.

Serial port connection settings (applicable to ESBC-10K only). Connect the serial port (port #1) on the ESBC-10K back panel to your PC, with speed (baud rate) "115200", data bits "8", stop bits "1", Parity "None" and Flow control "XON/XOFF".

Type "?" to get help.

2.6.1 Root mode

Command	Description
passwd	Change administrator login password (changing password for both CLI and WEB consoles)
enter [net | system | root]	Enter configuration mode
pwd	Display the current mode
show version	Show running information
help | ?	Display command list
reboot	Reboot the system
quit	Exit from current configuration mode
bye	Exit from the CLI console

- The following commands can be set under any mode: reboot, quit, bye, help|?, pwd, enter root.
- Under any mode, type "?" at the end of each unfinished command followed by <cr> to display command hint.

2.6.2 net mode

```
esbc> enter net            <cr>

net>
```

2.6.2.1 LAN

Command	Description
show lan	Show lan interface running information
enable | disable lan	Enable or disable LAN access (not applicable to ESBC-10K)
set lan [?]	dhcpc : store lan interface as dhcp client domain : store lan domain hostname : store lan host name staticip : store lan staticip information. (set lan staticip ip **IP** mask **NETMASK**)

2.6.2.2 WAN

The WAN commands are applicable to "Single Interface".

Command	Description
show wan	Show wan interface running information
enable|disable wan [access | https | ssh]	Enable or disable WAN access or with the specified protocol to remote users.
set wan [?]	• access port : store WAN access open port (<1-65535>) for WEB console. • dhcpc: set wan interface as dhcp client • none : disable WAN connection • staticip : store wan staticip information. (set wan staticip ip **IP** mask **NETMASK** gateway **GATEWAY** dns1 **DNS1 [**dns2 **DNS2]**)
restore WAN-MAC	Restoring WAN MAC to its default value

2.6.3 system mode

```
esbc> enter system          <cr>
system>
```

2.6.3.1 Provisioning

Command	Description
show provisioning	Display provisioning configuration
enable|disable provisioning	Enable or disable auto-provisioning.
set provisioning [?]	• server **ADDRESS** port **PORT**: set provisioning server IP|FQDN (up to 40 characters) and port (<1-65535>) • account **USERNAME PASSWORD** : authentication user name (up to 40 characters) and password
start provisioning	Trigger provisioning process. (provisioning should be enabled)

2.6.3.2 EMS

Command	Description
show ems	Display EMS configuration
enable|disable ems	Enable or disable EMS function
set ems [?]	• device type **VALUE <0-254>** : store EMS device type (0-254) • heartbeat type **[V2|V3]** : store EMS heartbeat type. • key derivation function **VALUE (1 | 2)** : 1-InnoMedia ; 2-PBKDF2-sha1. • local port <1-65535> : default 5200 • region id **ID**: store EMS region ID. • server[2] : server **ADDRESS** port **PORT**: set EMS server IP|FQDN (up to 40 characters) and port (<1-65535>) Note: when "1-InnoMedia" is selected for key derivation function, please refer to the document for "the use of RC4_102."
restart ems	Restart EMS service

2.6.3.3 PRI (applicable to ESBC-9xxx series)

Command	Description
show pri parameters	Show adaptive jitter buffer configurations and echo cancellation status.
enable|disable pri	echo cancellation: enable or disable echo cancellation on PRI trunks.

2.6.3.4 Function ID (applicable to ESBC-9xxx series)

The function ID is to set the PCIC card for the ESBC to operate at PRI or SIP B2BUA (transcoding) mode.

Command	Description
show function id	Display function id of PCIC card.
set function id <3-15>	Configure function id for PCIC card. Note, do not change function id unless you are guided with instructions. Changing to inappropriate ID number may result in unexpected system behavior.

2.7 SNMP based management

The ESBC's embedded SNMP agent works with a standard SNMP Manager to operate, maintain and provision (OAMP) the system. It supports standard and proprietary MIBs (Management Information Base) which allow the operator to collect information and hence enable a deeper probe into the device.

The ESBC can also send unsolicited events (traps) towards the SNMP manager. **All supported MIB files are supplied for each new ESBC firmware release.**

Please refer to section 6.9.1 for all traps for alert notifications.

2.7.1 Trap host configurations

The SNMP Basic Setting page allows you to configure the SNMP trap host based on IP address. The ESBC SNMP agent accepts GET and SET requests from the configured IP address with correct community strings. SNMPv1 and SNMPv2 use the notion of communities to establish trust between managers and agents. An agent is configured with three community strings: read-only, read-write, and trap. The "SNMP Community string" is like a user id or password that allows access to the ESBC parameters. . If the community string is correct, the EBSC responds with the requested information. If the community string is incorrect, the ESBC simply discards the request and does not respond.

1. The SNMP community strings are used on SNMPv1 and SNMPv2 protocol. The SNMPv3 uses username/password authentication along with an encryption key.
2. The ESBC sends traps to two SNMP trap hosts simultaneously if both IP addresses of Trap Host 1 and Trap Host 2 are configured

Navigate to **System** > **SNMP.**

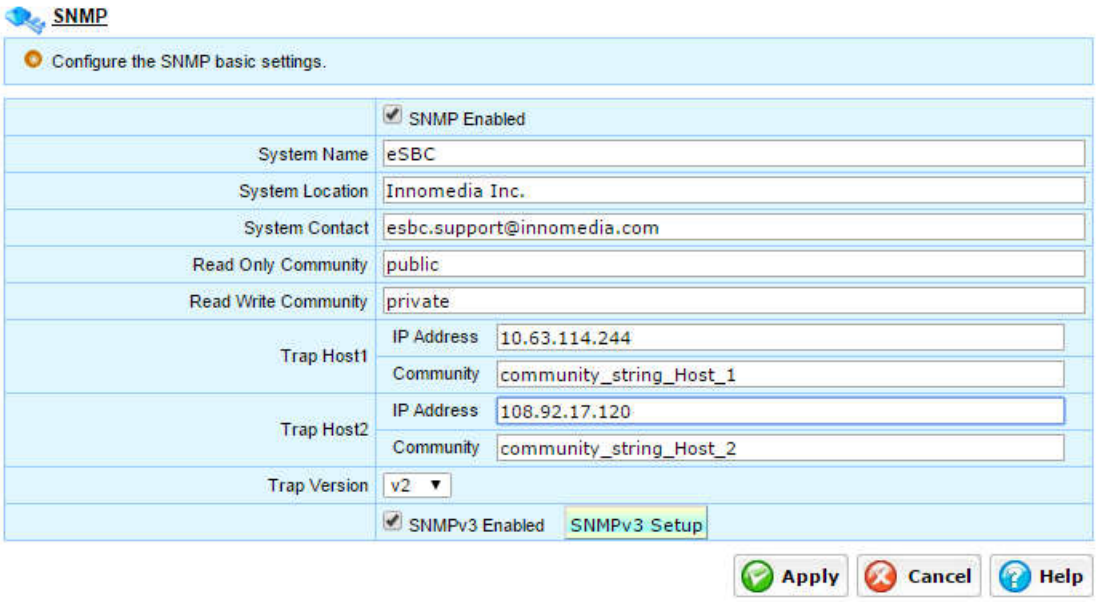


Figure 16. Configuring the SNMP Trap Host Information

SNMP Host	Description
System Name System Location System Contact	Enter the designated values for this deployed ESBC unit. The name of the ESBC system; the location of deployed premises, the contact info.
Read Only Community	Set the SNMP read only community string. Enabling a remote device to retrieve “read-only” information from the ESBC. The default string is set to "public". It is suggested that the network administrator change all the community strings so that outsiders cannot see information about the internal network.
Read Write Community	SNMP Read-Write community string. Enabling a remote device to read information from the ESBC and to modify settings. The default string is set to “private.” It is suggested that the network administrator change all the community strings so that outsiders cannot see information about the internal network.
Trap Host1 and Trap Host 2	The SNMP trap host destinations, an IPv4 address. See section 6.9.1 for Trap alarm descriptions.
Trap Host: Community	SNMP Trap community string which is used when sending SNMP Traps to SNMP Trap Host. This community string is different from the polling (read and read-write) community strings.
Trap Version	The notion of communities is applicable to SNMPv1 or SNMPv2.

2.7.2 SNMP v3 setup

SNMPv3 provides secure access to devices by a combination of authenticating and encrypting packets over the network. It includes three important services: **authentication**, **privacy** and **access control**.

You can create users, determine the protocol used for message authentication as well as determine if data transmitted between two SNMP entities is encrypted. In addition, you can restrict user privileges by defining which portions of the Management Information Bases (MIB) that a user can view. In this way, you restrict which MIBs a user can display and modify. In addition, you can restrict the types of messages, or traps, the user can send.

2.7.2.1 Security Levels in SNMPv3

The ESBC SNMPv3 Agent supports the following set of security levels as defined in the USM MIB (user security module), RFC 2574.

- **NoAuth/NoPriv** – Communication without authentication and privacy.
- **Auth/NoPriv** – Communication with authentication and without privacy. The protocols used for Authentication are MD5 and SHA (Secure Hash Algorithm).
- **AuthPriv** – Communication with authentication and privacy.

Configure authentication and privacy for SNMPv3 users as follows.

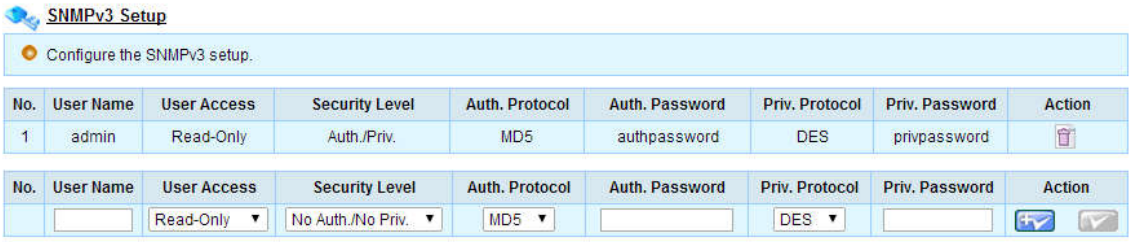


Figure 17. SNMPv3 Setup page

SNMPv3 Setup	Description
User Name	The user id.
User Access	MIB views. Read-only, or read-write.
Security Level	The SNMPv3 security level. Options available: No Authorization/No privacy, Authorization/No Privacy or Authorization/Privacy.
Auth. Protocol	The SNMPv3 (user-based security module) authorization type to use. Options available: MD5 or SHA.
Auth. Password	The SNMPv3 USM passphrase. Min string length: 8 characters.

Priv. Protocol	Privacy protocols supported currently are DES or AES.
Priv. Password	The SNMPv3 passphrase for encrypting data between two entities. The string must be at least 8 characters long. If you choose to not assign a privacy value, then SNMPv3 messages are sent in plain text format.
Action	Add, Edit, and Delete.

2.8 Email (SMTP) Based Management

The ESBC supports sending Alert Notifications via Emails with SMTP (simple mail transfer protocol). See section 6.9.2 for Email alarm descriptions.

Navigate to **System** > **SMTP** for the Email server settings.

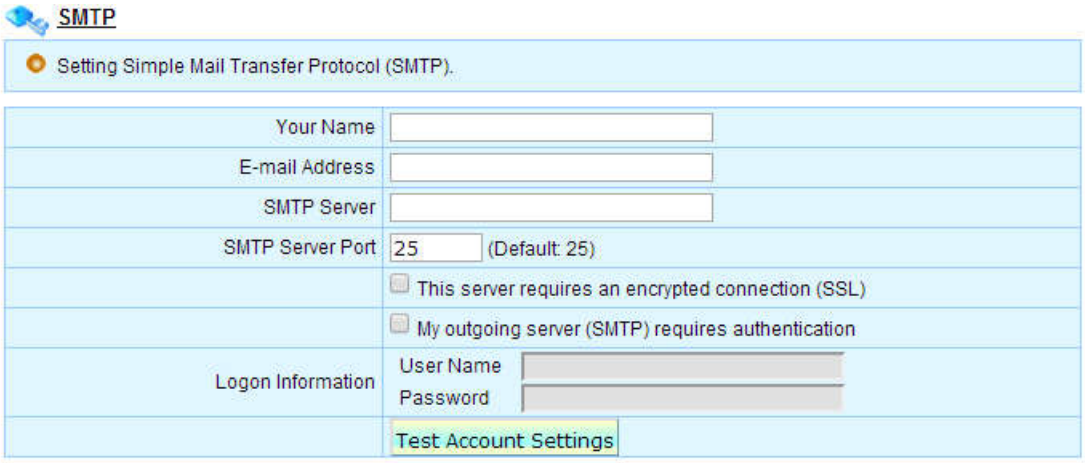


Figure 18. Email based management configuration

SNMPv3 Setup	Description
Your Name	Enter the name which you would like it to appear on the emails sent out.
E-Mail Address	An email notification will be sent to this email account.
SMTP server	Enter the SMTP server IP, or FQDN for outgoing emails.
SMTP server port	Enter the SMTP Server Port. If no SSL connection is required, the default communication port is 25. Check with your email administrator for SSL configuration requirements for outgoing emails.
Logon Information	Enter the user name and password which are associated with the E-Mail address specified above.
Test Account Settings	Click this button and the ESBC will send out a test mail to the E-mail account specified above.

2.9 XML config-file based management

Please refer to the ESBC provisioning tag document for detailed descriptions of all tags and sample configuration files.

The XML configuration file is a text-based file (which can be edited with, for example, notepad) that contains any number of provisioning tags (parameters). The XML configuration file can be imported to the ESBC via the following methods:

- Auto-provisioning. (See section 2.10 for a detailed description)
- XML config import from the WEB administrative console. (See section 6.4.3 for a detailed description.)

See section 6.4 for the ESBC configuration backup.

2.10 Auto-Provisioning based management

2.10.1 Basic Provisioning Mechanism Configurations

The ESBC supports auto-provisioning based management features which allow the provisioning of user accounts, service features, system capacity, and upgrading system firmware through auto-provisioning servers.

Provisioning Method: DHCP | TFTP | HTTP | HTTPS | SecHTTP

Supported configuration file formats: XML | INI

Please refer to the ESBC provisioning tag document for detailed descriptions of all tags and sample configuration files.

Navigate to **System** > **Provisioning.**

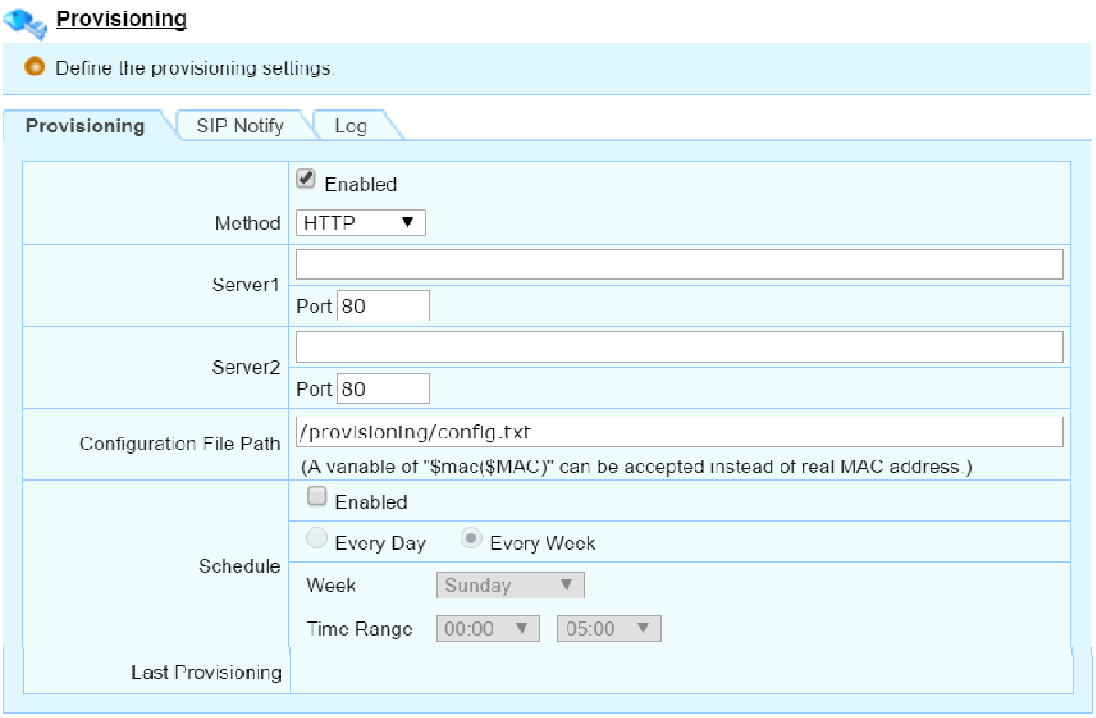


Figure 19. Auto Provisioning Management

The use of the supported provisioning methods is described in the following sections.

2.10.1.1 DHCP Provisioning Method

The ESBC supports auto provisioning mechanism by utilizing DHCP Options 66 and Option 67

- When DHCP Provisioning Method is selected, the ESBC Internet Connection has to be configured as DHCP client.
- DHCP Option 66 only: Specify provisioning server IP address (or host name) together with the complete provisioning URL, e.g., "protocol://host:port/path/prov_file_name." The protocol used can be TFTP, HTTP, or HTTPS.

  URL1 http://provision.example.org:8080/Config_Path/$MAC_Config.xml

  URL2 tftp://provision.example.org/Config_Path/$MAC_Config.ini
- DHCP Option 66 with Option 67. Use Option 66 to specify the TFTP IP address or host name, and Option 67 for the config file path and name. The protocol used is TFTP.

  DHCP Option 66 – provision.example.org

  DHCP Option 67 – config/$MAC_config.xml

2.10.1.2 HTTP / HTTPS / TFTP/ SecHTTP Provisioning Methods

Item	Description
Provisioning method	HTTP / HTTPS /TFTP/SecHTTP
Server 1 & Server 2	Provisioning server IP Address or FQDN The ESBC supports redundant provisioning servers. When Server 1 is not available, the ESBC will initiate the provisioning process with Server 2.
Port	The port number used for selected provisioning method. Default Port: 80(HTTP), 443(HTTPS), 69(TFTP)
Configuration File Path	Enter the path and file name for the configuration file location on the server. MACRO commands (such as $MAC) can be used. This item is not applicable to SecHTTP method.
Schedule	Allows the ESBC to perform an automatic scheduled provisioning process. After a successful provisioning process, the ESBC will need to reboot in order to activate the new settings. The reboot will be triggered immediately. If there are active calls, the reboot will be delayed until 5 seconds after the last active call ends in the time window. Settings: • Disable or enable scheduled provisioning • Frequency: (every day | every week), during the configured one hour window.
User Name and Password for HTTP/HTTPS/SecHTTP Methods	To configure User Name and Password requires the user to login to the Command Line Interface (CLI) console. See section 2.6 and CLI command reference document for the details.

	Note. SecHTTP is a proprietary provisioning protocol which is used for communicating with the InnoMedia EMS server. If this provisioning method is selected, it is necessary to use the rc4-102 encryption utility (InnoMedia proprietary) to encrypt the config file. Please refer to the document “the use of rc4-102 utility.”

2.10.2 Server Initiated Provisioning: SIP NOTIFY

The ESBC supports an unsolicited SIP NOTIFY to perform requested operations from the SIP server.

Event: reboot | resync | report

- reboot: the ESBC reboots itself and re-fetches the config file from the provisioning server.
- resync: the ESBC re-fetches the config file from the provisioning server without rebooting.
- report: the ESBC sends its profile to the specified FTP server as configured in Figure 20.

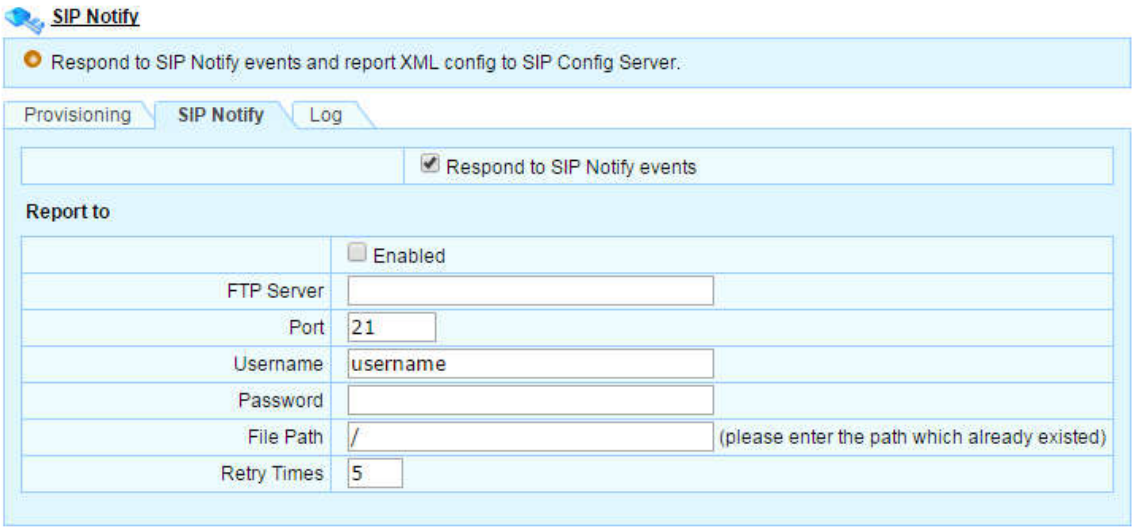


Figure 20. The SIP Notify Configuration—server initiated provisioning

2.10.3 Log

“Log” tab to view the image/configuration file changes of this ESBC unit.

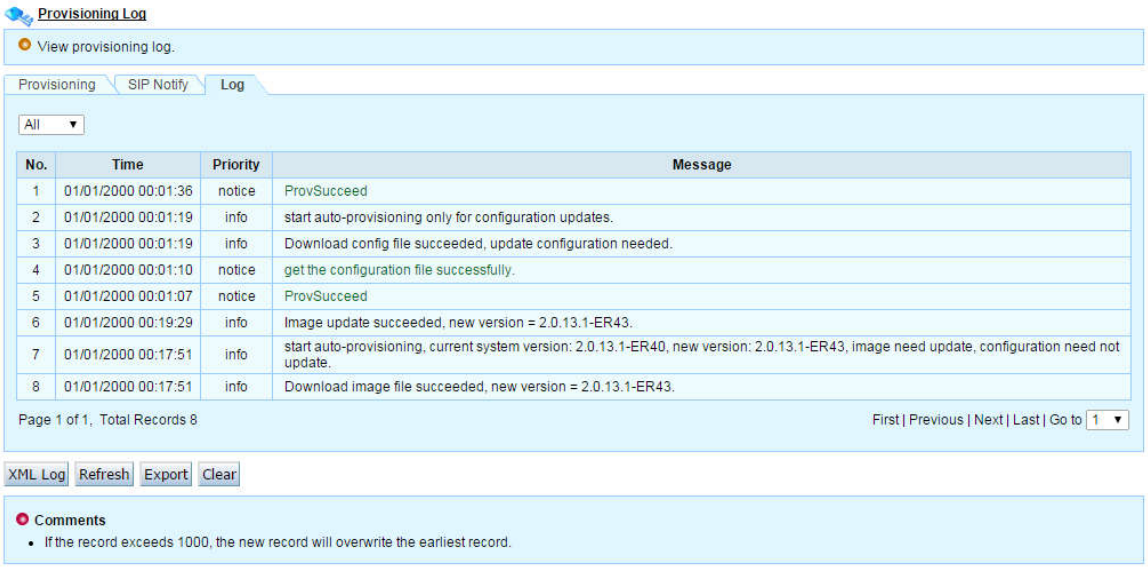


Figure 21. Provisioning Log

Click <XML Log> button to view tag-parameter updates for the latest provisioning event.

Click <Export> to export the current page view to text file.

Click <Refresh> to refresh the page view, and click <Clear> button to clear all records.

2.10.4 EMS based management

InnoMedia EMS (element management system) is a scalable and fully redundant solution covering OAM&P features such as device auto-provisioning and device management functions (via SNMP).

- Auto-Provisioning protocols supported on the EMS: HTTP | SecHTTP | TFTP
- EMS generates device-dependent configuration files on the fly, providing maximum flexibility for the provisioned device with per device parameters.

Device Management: the EMS allows a service provider's customer service and maintenance personnel to provide effective device management, trouble-shooting and statistics collection, all from an easy-to-use and secure browser interface. It is possible to access the web console of a particular ESBC unit through the EMS administrative console. The EMS system is able to manage and communicate with devices even if they are behind a NAT router.

Note that the EMS voice loopback test cannot be performed when the ESBC WAN interface is configured with multiple IP networks for both voice and OAMP services and EMS is in the OAMP service network,

Navigate to **System > EMS**.

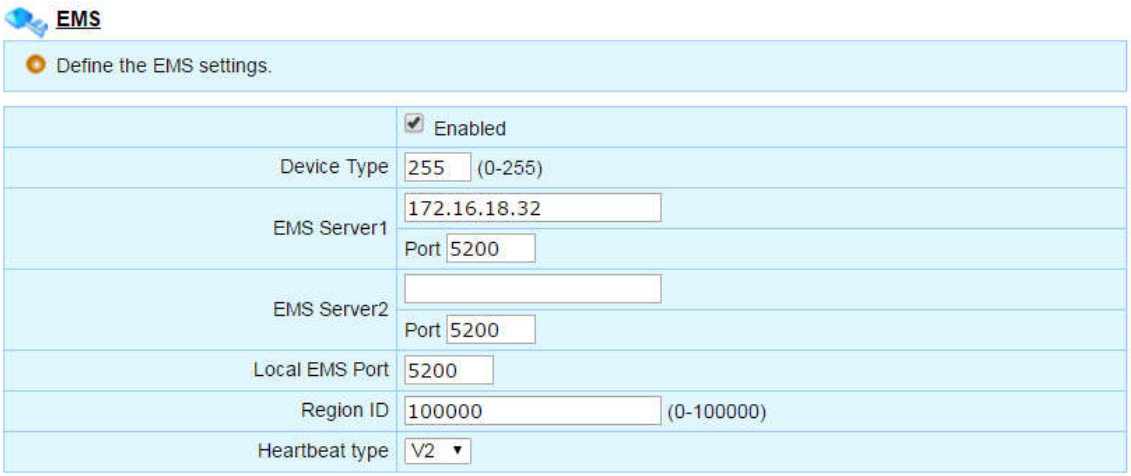


Figure 22. InnoMedia EMS server configuration

EMS Server settings	Description
Enabled	Check this box to enable management via an EMS server.
Device Type	The device type ID defined in the EMS server to categorize connected devices by models. Check with the EMS administrator to input the desired value for your deployed CPE units. When the value of Device Type is not available at the initial setup

	stage, set the value to **255** which is the factory default value. The ESBC will communicate with the EMS server by sending its model related system information to the EMS server through a heartbeat message, and obtain its configured device type automatically after registering to the EMS successfully. This requires the EMS to also have this feature enabled.
EMS Server 1 EMS Server 2	The ESBC supports geographically redundant EMS servers. Server 1 is the active server, and server 2 is the backup. Enter EMS IP (or FQDN), and port information. The communication port for EMS is 5200 by default. If the active EMS server is down, the ESBC automatically switches to the backup server.
Local EMS port	The communication port for EMS is 5200 by default. Check with your service provider for any different configurations.
Region ID	The deployed region ID defined in the EMS server to categorize connected devices by regions. Check with the EMS administrator to input the desired value for your deployed CPE units. When the value of Region ID is not available at the initial setup stage, set the value to **100000** which is the factory default value. The ESBC will communicate with the EMS server by sending its model related system information to the EMS server through a heartbeat message, and obtain its configured Region ID automatically after registering to the EMS successfully. This requires the EMS to also have this feature enabled.
Heartbeat Type	The InnoMedia proprietary keep-alive protocol communicating between CPEs and the EMS.

3 The ESBC Network requirements and configurations

Login to the ESBC web console via the LAN VoIP-NAT port interface (see section 2.5) to configure the ESBC WAN interface. This is required to access the operator's network (and vice versa). Note that updating the WAN IP address can only be performed via LAN interface.

The WAN interface is used to route traffics generated by the ESBC itself and from the LAN networks with the service provider networks. Traffic types include: voice signal and media, OAMP data, and LAN network data in general.

3.1 Determining the network requirements for voice services

3.1.1 Understand the network factors which affect quality of service

Identifying the network connectivity requirements is the key to the success of voice service deployment. It is necessary for the IP WAN and LAN to provide networks that meet the requirements for toll-quality service. It is important to identify the following factors which affect the quality of services and service level agreements.

- Bandwidth
- Latency
- Jitter
- Packet Loss

3.1.1.1 Bandwidth Requirement

The amount of bandwidth for voice calls depends on these factors:

Number of concurrent calls |The codec used for voice communications |Signaling overheads

These protocol header assumptions are used for calculations:

- Headers: 40 Bytes overall consisting of IPv4 (20 Bytes)/ UDP (8 Bytes) / RTP (12 Bytes)
- 38 Bytes for fixed Ethernet headers

Voice codecs	G.711	G.729
Sampling rate	8 kHz	8 kHz
Effective sample size	8 bits	1 bit
Data rate	64 kbps	8 kbps

Bandwidth consumption for one way voice

Codec	Bit rate	Packetization	Payload size	Ethernet

		period (ptime)		bandwidth
G.711	64 kbps	20 ms	1280 bits	95.2 Kbps
G.729	8K bps	20 ms	160 bits	39.2 Kbps

- PPS (packet per sec) = (Sampling rate) / (sample period)
- Voice payload size = Data rate / PPS
- Total packet size = (IP/UDP/RTP header) + (voice payload size) + (fixed Ethernet overhead)
- Bandwidth consumption for one way voice= (Total packet size) * PPS

3.1.1.2 Latency

Latency is one way delay from “mouth to ear”. It comprises the following processes:

- Time required to sampledigitize (or encode) and packetize the sender’s voice
- Time required to send the packet over the IP network
- De-packetization, decoding and relaying the speech to the receiving party

3.1.1.3 Jitter

Jitter is the variation of latency across the network and the variation in the timing of packet processing inside the devices. To compensate for jitter, modern devices usually utilize an adaptive jitter buffer. However, high levels of jitter can cause packets to be discarded by the jitter buffer at the receiver, and also increase latency as the jitter buffer adapts.

3.1.1.4 Packet Loss

The human ear is very good at handling the short gaps that are typical of packet loss. So it may take a significant amount of packet loss for the user to be significantly affected by packet loss to report it. On the other hand, fax and modem calls are particularly sensitive to packet loss, almost demanding 0% packet loss to avoid problems with fax/modem transmission.

There are two types of packet loss in a VoIP system: **received packet loss**, and **received packet discard**. Received packet loss is where a packet is never delivered to the receiving system; while receiving packet discard is where a packet is received after a time when it is not useable for generating audio playback.

Packets could be dropped somewhere in the network causing received packet loss, or packets could be delayed somewhere in the network causing received packet discard. Network issues could include:

- A poor link causing packet errors which may vary by time of day or load.
- Network congestion causing the router or switch buffer to overflow or produce high jitter.

- A transient network problem. Packets will get dropped if a valid alternate path is not immediately available.

3.2 Dual WAN Redundancy

The InnoMedia ESBC offers a redundant WAN interface to allow a backup network to take over the Voice and Data Internet connection in the event of a failure on the primary interface. Two interfaces can be configured to act as a redundant pair, i.e., the primary and secondary interfaces.

The ESBC WAN redundancy feature utilizes two separate interfaces which connect to two different physical routes (networks) respectively. The ESBC WAN physical interface, either through the WAN Ethernet port (ESBC93xx) or the embedded cable modem (ESBC95xx) connects to the primary network. The backup WAN interface, if enabled, is selected from one of the LAN Ethernet ports, and connects to the standby network. The ESBC constantly monitors the availability of the primary network, and automatically switches to the secondary interface if the primary is determined unreachable. Once the primary network comes back online, the ESBC can be configured to automatically revert the service back to the primary (under Revertive mode), or can be switched back under a Manual Switchover action being triggered by the administrator.

Applicable models: ESBC9380, ESBC9378, ESBC9328, ESBC9580, ESBC9528, and ESBC9578.
Note: The WAN redundancy feature is not available when the WAN interface is configured in “Multiple Interfaces” mode. See section 3.3 for more details.

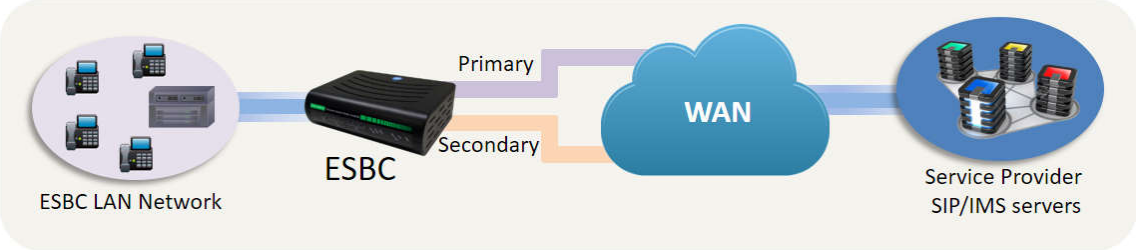


Figure 23. The ESBC WAN Redundancy Topology

3.2.1 The Configuration of Redundant WAN

3.2.1.1 Enable the Backup WAN port

To enable the WAN Redundancy feature, navigate to the **LAN Interface** page and select one Ethernet LAN port as the WAN Backup port.

Network > Advanced > LAN Interfaces.

After selecting the WAN Backup port, click the <Apply> button. A warning message pops up as a reminder that the ESBC needs to reboot in order for this configuration to take effect.

Note.

1. The Multiple Subnet feature for the ESBC WAN interface is not supported when the redundant WAN interface feature is enabled.
2. Only one LAN interface can be configured as a WAN Backup port. Trying to configure another port as WAN Backup when one already exists will produce a warning message and the WAN Backup will be switched from the old to the new LAN port.

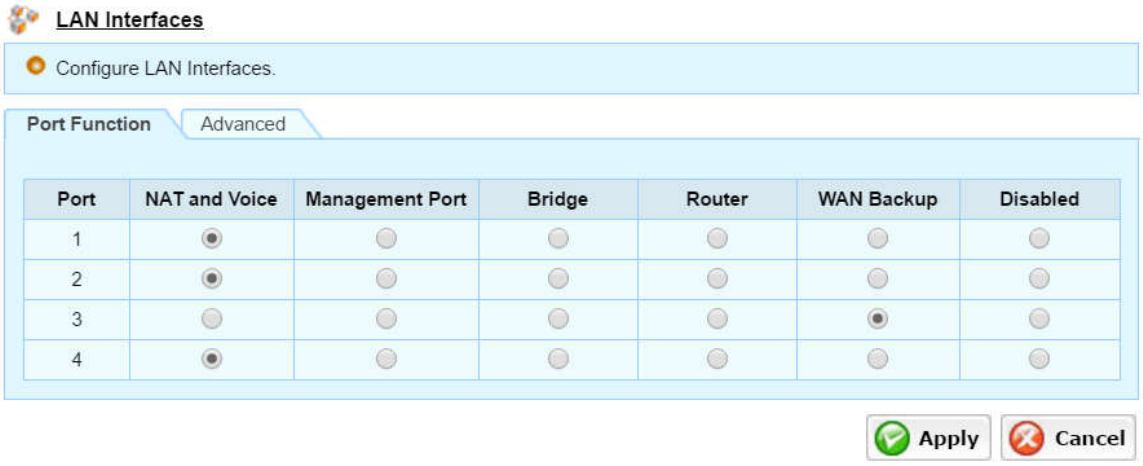


Figure 24. Selection of one Ethernet LAN port as the WAN Backup port

3.2.1.2 Configuring the Backup WAN Interface

Navigate to the **WAN Interfaces** page to configure the attributes of the backup WAN interface, i.e., the secondary interface IPv4 configuration.

Network > Settings > WAN > Network Interfaces (tab)

Three connection types are available to configure the IP address of this physical Ethernet port. Note that the Secondary WAN interface should be connected to a different network from that of the Primary interface.

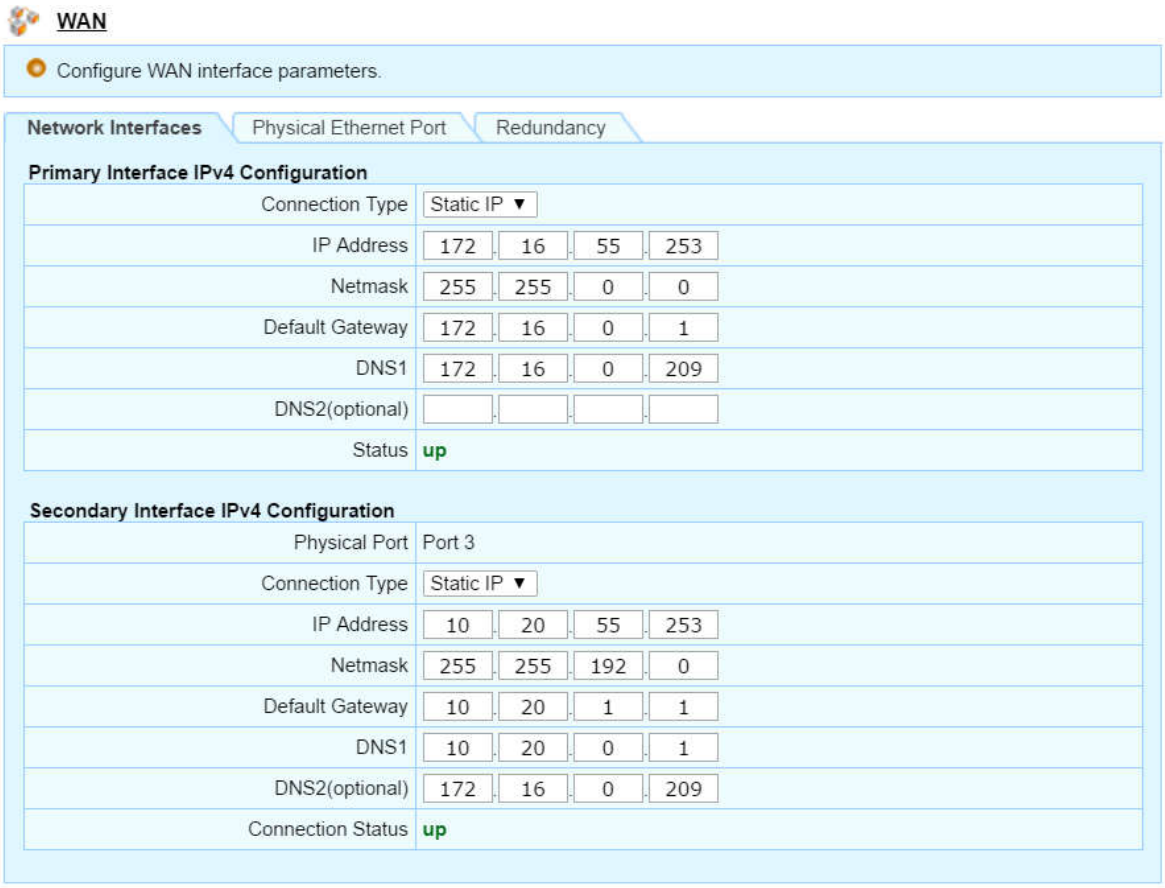


Figure 25. Configuring the IP settings

Connection Type	Description
None	This port is not configured, and is virtually disabled at this point.
DHCP	The ESBC DHCP Client uses DHCP to setup IP address details.
Static IP	Use the static IP provided by your ISP to setup the IP address details. Enter the IP address, Netmask, Default Gateway, DNS server(s). Note that when **Static IP** is selected, the IP address of this interface is always available, i.e., no loss of IP address event will be detected by the ESBC. See section 3.2.1.3 on the types of events that will trigger a WAN Redundancy failover.

3.2.1.3 WAN Redundancy Settings

The ESBC WAN Redundancy feature utilizes two separate interfaces which connect to two different **networks (routes)**. If a communication failure is detected on the primary network, the ESBC automatically switches to the secondary network to ensure reliable voice services.

The ESBC detects network availability according to three factors:

- The physical Ethernet link status (for ESBC93xx), or the connection status between the embedded CM and CMTS (for ESBC95xx).
- The availability of the WAN IP address
- The network or route accessibility through a ping detection capability

If the ESBC is connected to the secondary network, there are two options for the ESBC to failback to the primary network once it detects the primary network comes back online:

- Automatic failback by enabling **Revertive Mode**, and
- **Manual Switchover**.

These two options are described in sections 3.2.1.4 and 3.2.1.5.

3.2.1.4 Monitor WAN availability

Navigate to the Monitor screen once the Primary and Secondary interfaces are configured and connected to their respective networks.

Navigate to Network > Settings > Monitor.

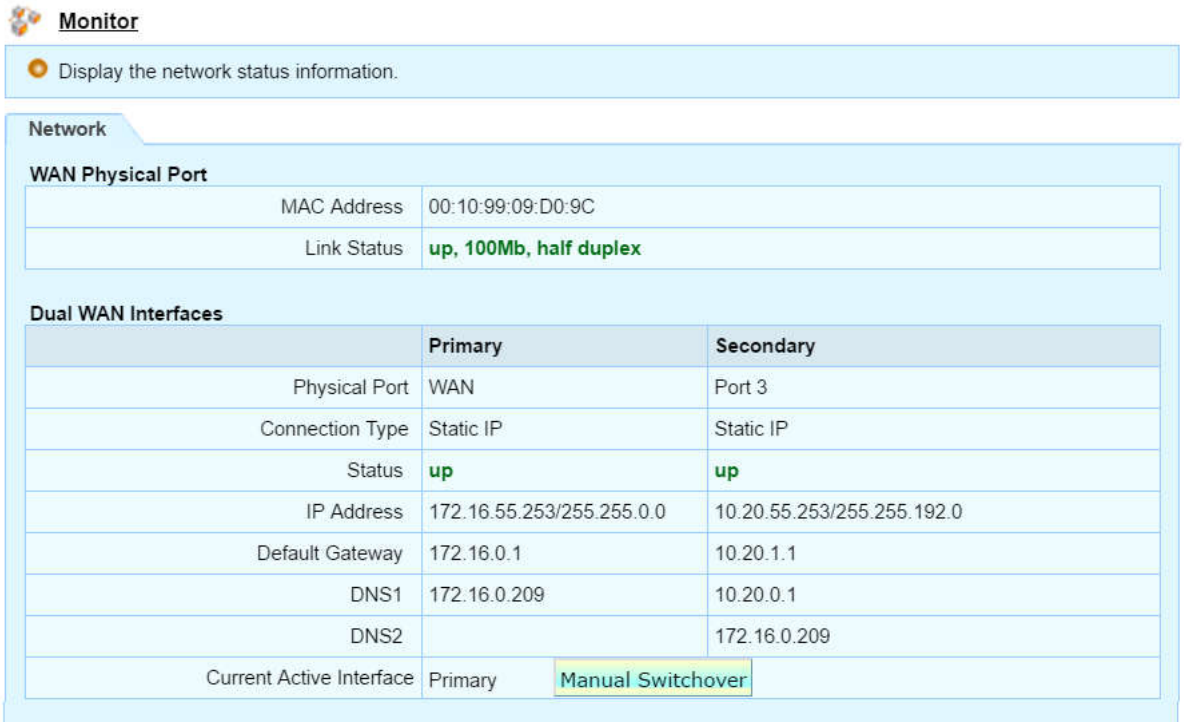
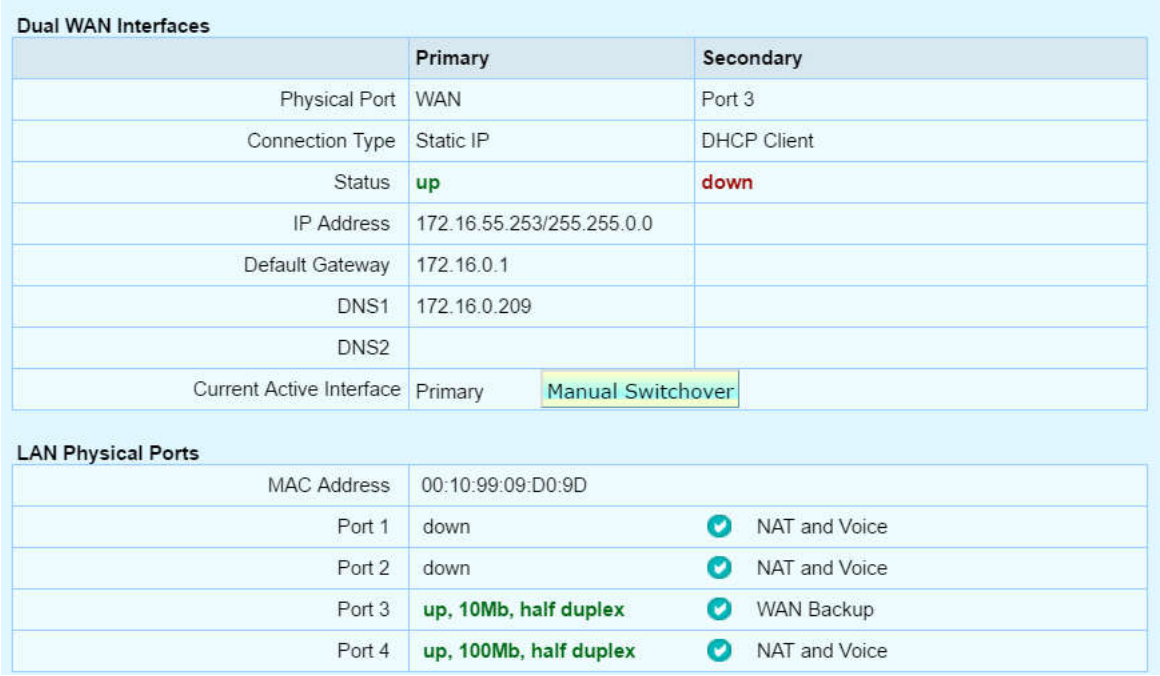


Figure 26. The dual WAN interface real time status monitor

WAN Physical Port	Description
MAC Address	The MAC address of the physical WAN interface.
Link Status	Only applicable to the ESBC93xx models where the WAN interface is an Ethernet link. The physical connection status between the ESBC WAN port and its directly connected switch port. • Link up/down status • Speed • Duplex mode

Dual WAN Interfaces	Description Primary and Secondary Networks
Physical Port	Primary: WAN Secondary: the selected port number from one of the four LAN Interfaces.
Connection Type	DHCP Client | Static IP | None
Status	**up** | **down** Connection Type – **DHCP Client**: Status shows "up" as long as the physical link connection is good to the directly connected switch port AND the ESBC is able to obtain an IP address from the DHCP server. Shows "down" otherwise. Connection Type – **Static IP**: Status shows "up" as long as the physical link connection is good to the directly connected switch port. Shows "down" otherwise. Connection Type – **None**: Status always shows "down".
IP Address | Default Gateway | DNS1 | DNS2	The IP address and network information for this particular interface.
Current Active Interface	Primary | Secondary. The current active interface providing voice & data services. Primary: When the primary network is up and running. Secondary: When the Primary network is not reachable, the ESBC uses the Secondary interface.
Manual Switch Over	Trigger the action of manually switching the current active interface to another. Refer to section 3.2.1.5 for the 'Revertive Mode' setting which allows automatic failback.



Dual WAN Interfaces

	Primary	Secondary
Physical Port	WAN	Port 3
Connection Type	Static IP	DHCP Client
Status	up	down
IP Address	172.16.55.253/255.255.0.0	
Default Gateway	172.16.0.1	
DNS1	172.16.0.209	
DNS2		
Current Active Interface	Primary Manual Switchover	

LAN Physical Ports

MAC Address	00:10:99:09:D0:9D	
Port 1	down	NAT and Voice
Port 2	down	NAT and Voice
Port 3	up, 10Mb, half duplex	WAN Backup
Port 4	up, 100Mb, half duplex	NAT and Voice

Figure 27. Status ‘down’ vs. physical link ‘up’

In the particular example shown in Figure 27, LAN Port 3 is selected as the WAN Backup port. This backup WAN interface is configured as a DHCP client. When the secondary network has issues and the ESBC cannot obtain an IP address from the network DHCP server, the Status of the Secondary Interface shows ‘down,’ even though the physical LAN port 3 is still up, i.e., the physical wire connection is still good.

3.2.1.5 Configuring the Redundancy and Failover Settings

Navigate to the Redundancy page to configure the physical link detection and layer 3 ping detection to determine the availability of the network connection for the primary and secondary interfaces.

Navigate to Network > Settings > WAN > Redundancy (tab)

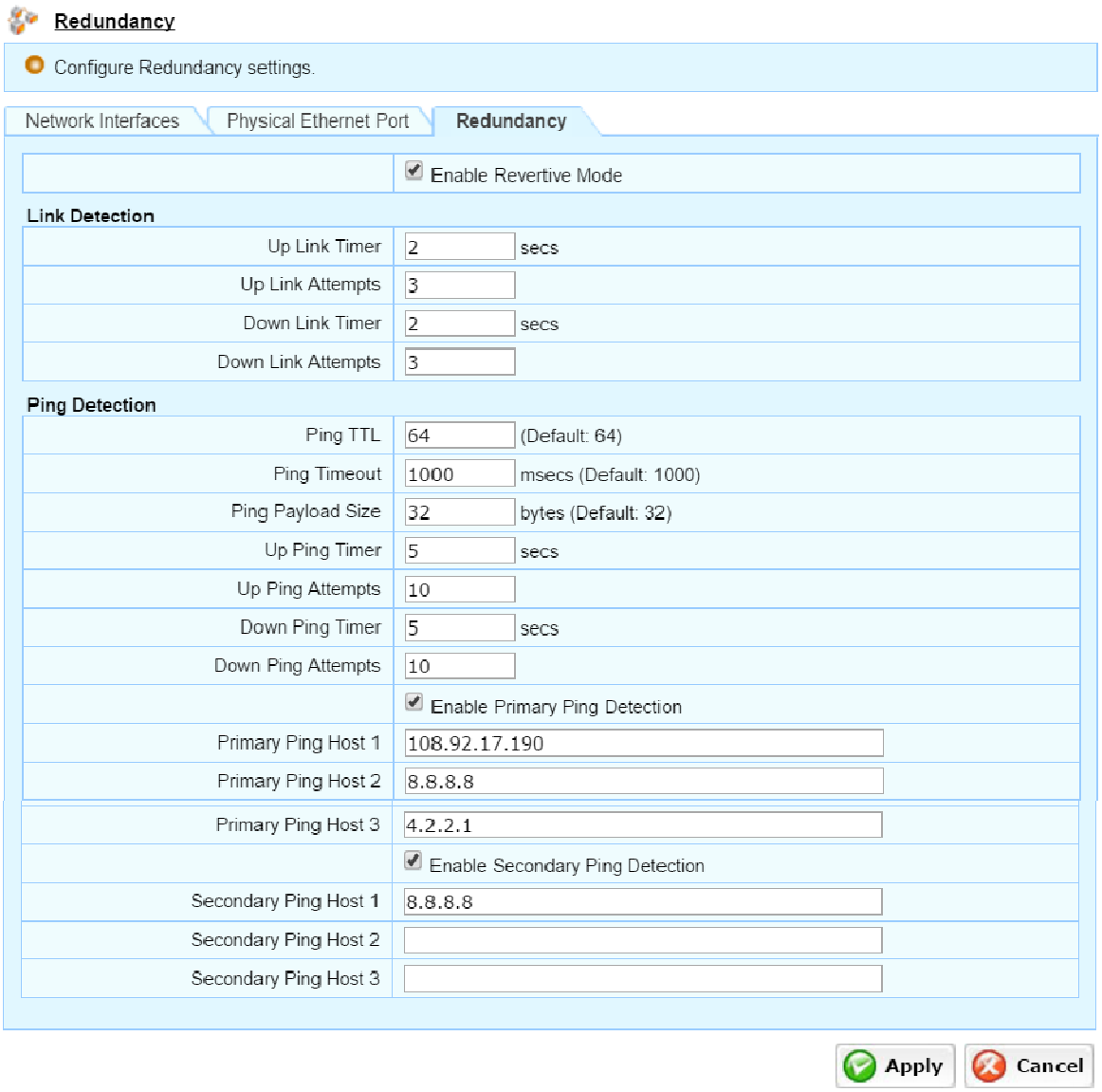


Figure 28. Configuring redundancy and failover settings

Redundancy	Description
Enable Revertive Mode	• Enable Revertive Mode. If the ESBC is using the secondary interface, this setting allows the ESBC to revert back to the primary interface automatically, as soon as the ESBC detects the primary interface connection is back online, i.e., network reachability for the primary interface changes from unavailable to available. Note: If the operator manually switches the active interface to the

	secondary interface, the ESBC does not switch back to the primary (regardless of whether the primary interface is available or not) until the Manual Switchover action is triggered again or the primary interface goes through a down-and-up status change. • Disable Revertive Mode. If this box is disabled, once the ESBC detects the primary interface is down and fails over to the secondary interface, the ESBC will continue to use the secondary interface until the Manual Switchover action is triggered. See section 3.2.1.4 for the description of 'Manual Switchover.'

Link Detection	Description
Up Link Timer	Seconds. When the physical link state is down, the intervals at which the ESBC checks the physical interface to detect the link is up.
Up Link Attempts	The number of consecutive successful physical interface checks before the ESBC determines the link is up
Down Link Timer	Seconds. When the physical link state is up, the intervals at which the ESBC checks the physical interface to detect the link is down.
Down Link Attempts	The number of consecutive failed physical interface checks before the ESBC determines the link is down.

Ping Detection	Description
Ping TTL	The time-to-live value, i.e., the number of network hops (TTL) that a ping packet traverses.
Ping Timeout	msecs. The ESBC should receive the ICMP ping response from a host within this predetermined time for the ping detection to be successful.
Ping Payload Size	Bytes. The number of ICMP data bytes, excluding other overhead.
Up Ping Timer	Seconds. The interval between the ICMP packets sent by the ESBC to detect the network has come back up.
Up Ping Attempts	The number of consecutive ICMP packets before the ESBC determines the network has come back up.
Down Ping Timer	Seconds. The interval between the ICMP packets sent by the ESBC to determine the network is unreachable.
Down Ping Attempts	The number of consecutive ICMP packets before the ESBC determines the network is unreachable.
Enable **Primary | Secondary**	Enable Primary | Secondary Ping Detection checkbox to allow the ESBC periodically to send ICMP ping packets to monitor the status of

Ping Detection	the associated interface. At most three hosts (IP addresses or FQDNs) can be configured for each interface. It is recommended that hosts are selected which reside in remote networks, i.e., not in the network directly connected to the WAN interfaces. As long as the ESBC receives ping responses from one host, the ESBC will determine that the associated interface is in an up state.

Overall, the ESBC will make a determination on whether an interface is “down” and switch to the other interface if:

1. Physical link is detected to be down OR
2. No WAN IP address OR
3. The ping detection status is “down”

3.2.2 Changes to ESBC services when WAN redundancy mode is enabled

Once the redundant WAN port is enabled, this will result in some changes to the availability of certain features:

- WAN port VLAN feature is not supported
- Multiple logical IP network service model is not supported
- Voice-NAT & Management ports are supported
- Bridge & Router ports are supported and they are mapped to the primary interface only.
- The following services are available on **both the primary and secondary interfaces simultaneously**:
  - ACL
  - WAN WEB console access
  - WAN SSH console access
  - Allow Ping to the ESBC WAN Interface
  - SNMP (walk|get|set)
- The following services are available on the **current active interface**:
  - Network diagnostics
  - PPTP server
  - Syslog
  - SNMP Trap

  - Provisioning
  - SNTP
  - Auto Back
  - EMS
- The following services are only applicable on the **primary interface**.
  - Bridge port
  - Router port
  - Access Control for LAB host data services
  - DNS proxy
  - Port forwarding
  - DMZ

3.3 Single and Multiple logical IP network service models

The ESBC supports the following two deployment architectural models.

- Single IP network interface. With this mode, configuring a single IP address to the ESBC WAN port for routing all types of traffics which are generated by the ESBC and from the LAN networks, to/from the service provider networks. (see section 3.3.1.1)
- Multiple logical IP network interfaces. With this mode, configuring multiple IP networks to the ESBC WAN port and route separately OAMP and Voice (signaling and media) traffics to different network segments for security, administration, and QoS quests. This can be achieved by adopting both VLAN (layer 2) and IP subnet (layer 3) logics. (see 3.3.1.4) This feature is available on the ESBC93xx models.

OAMP traffics include operational data from administrative consoles (WEB/CLI), EMS, provisioning, syslog, SNMP, and etc. **Voice Signaling + Media** traffics includes SIP, and RTP/RTCP packets.

3.3.1.1 Single IP Network Interface

A single IP address is configured for the ESBC WAN port for routing all types of traffic between the service provider network and the ESBC itself and its LAN ports.

When a single interface is selected, “Connection Type” and “VLAN” tagging features are available for separate traffic types.

The ESBC assigns VLAN tags to WAN traffic, which provides application-level control over VLAN connections.

Navigate to **Network** > **Settings** > **WAN.** Choose **Single Interface** mode.

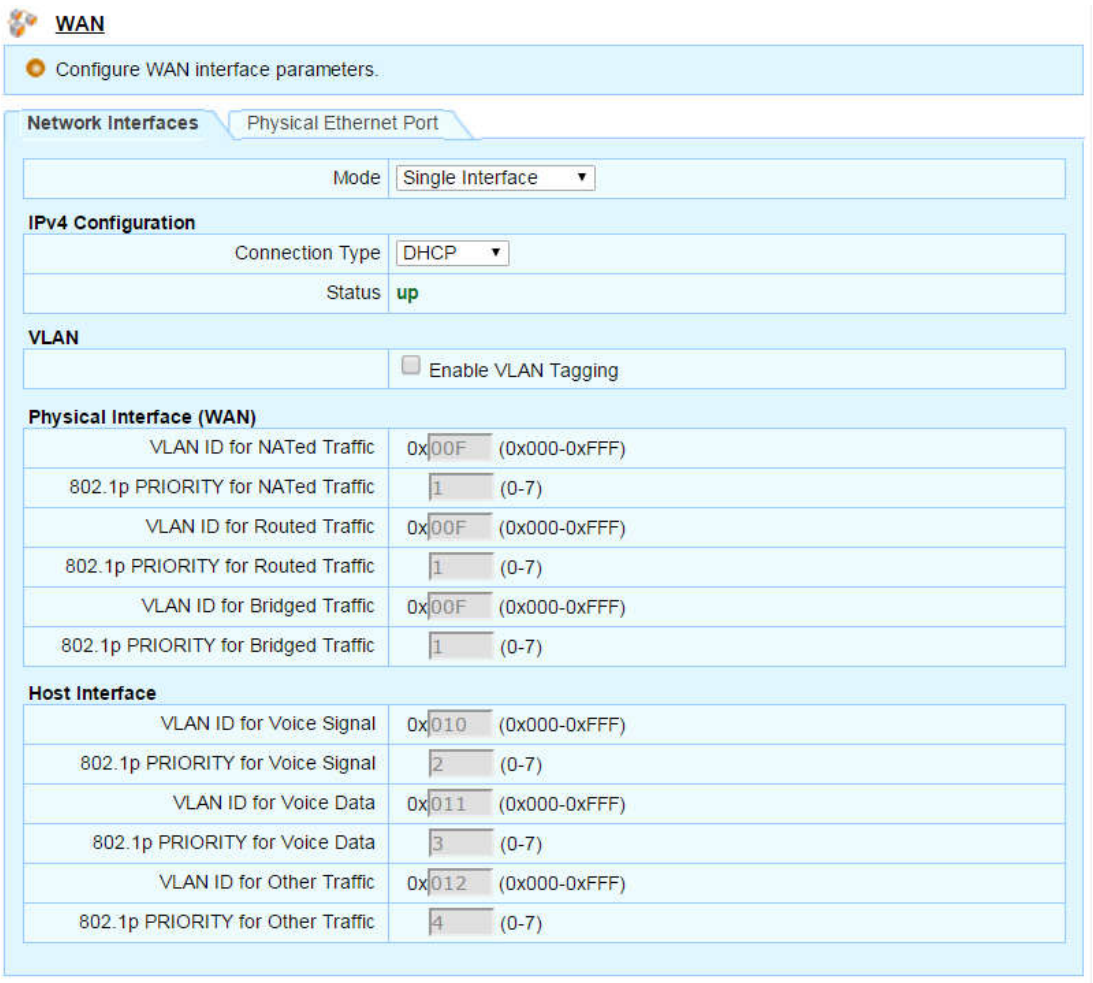


Figure 29. IP address and VLAN Tag configuration: single interface mode

The ESBC LAN ports can be configured for different types of services as described in section 3.4.2, including router port, bridge port, and NAT-voice port.

- Physical Interface (WAN): ESBC tags traffic from these different types of ports, e.g., Router, Bridge, and Voice-NAT, with associated VLAN IDs and sends it to the WAN/Service provider network, mainly for data services. (The management port is designed for console access from the LAN interface, and hence no VLAN tagging is provided for it toward the WAN interface.)
- Host Interface: Traffic generated by the ESBC (e.g., SIP signaling, voice data and other management traffic such as HTTP, SNMP, DNS, etc.) are tagged with associated VLAN IDs to communicate with the WAN/Service provider network, mainly for telephony services.
- When "Enable VLAN Tagging" is enabled, hosts accessing the ESBC's WAN interface using GUI or CLI console should be configured with the same VLAN ID as the NATed Traffic.

WAN Interface Setting	Description
Mode	Single Interface | Multiple Interfaces
IPV4 configuration	DHCP | Static IP DHCP Client (the default setting): use DHCP service to setup internet connection. Static IP: Use the static IP provided by your ISP to setup internet connection.
IP Address	When Static IP is selected as the Connection Type, enter the IP address.
Netmask	When Static IP is selected as the Connection Type, enter the netmask associated to the IP address.
Default Gateway	When Static IP is selected as the Connection Type, enter the default gateway of this IP network.
DNS1 (and DNS2)	When Static IP is selected as the Connection Type, enter the DSN server(s) used with this IP network interface.

3.3.1.2 VLAN settings for Multi-Service Capabilities to the WAN Logical Interface

Service providers can utilize VLAN tagging with operations backbone infrastructure technologies with advantages such as (1) traffic engineering and (2) multi-service networks. With the ESBC VLAN traffic segregation function, Service Providers do not require enterprises to invest in further VLAN switches in order to deploy multi-service capabilities to customers.

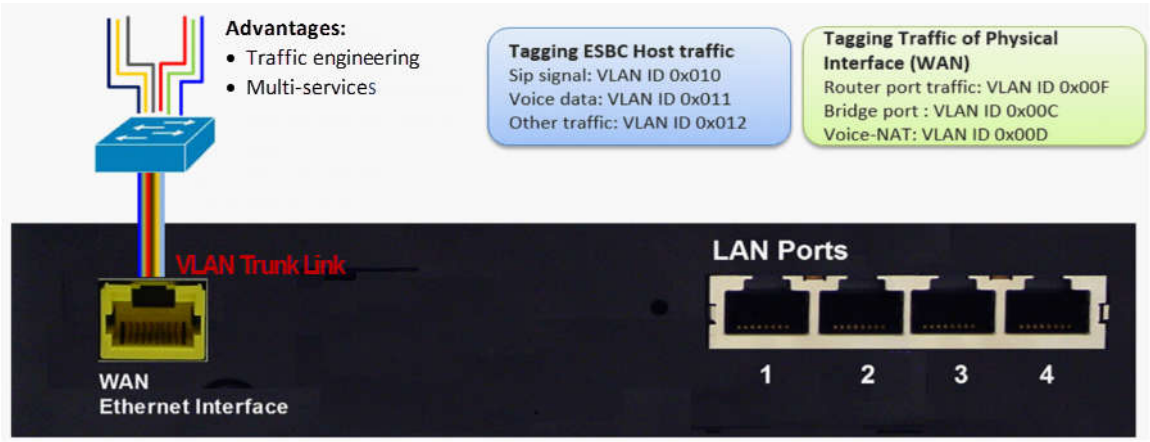


Figure 30. ESBC VLAN support for prioritizing services

3.3.1.3 The Physical Ethernet Port Configurations

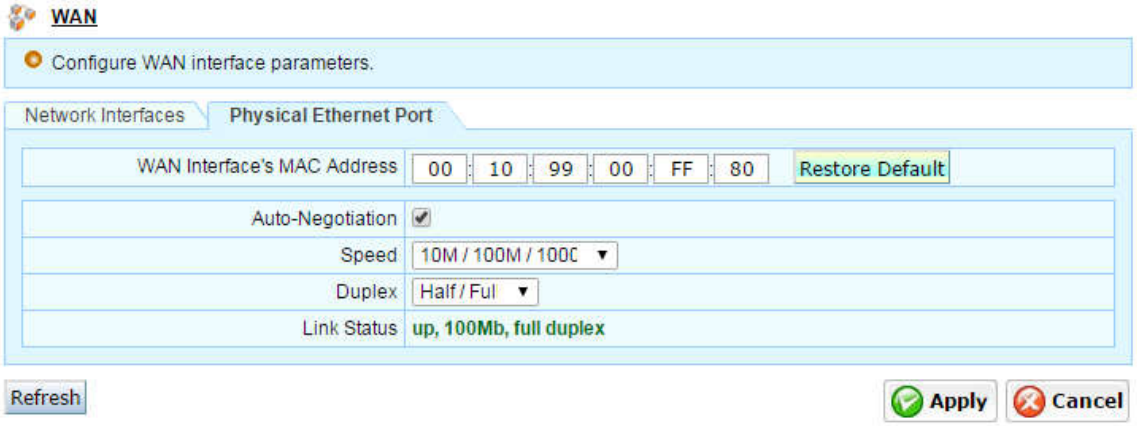


Figure 31. Configuring Physical Ethernet Port of WAN Interface

Physical Ethernet Port	Description
WAN Interface's MAC Address	The MAC address assigned to the ESBC WAN port interface. It is possible to clone a MAC address to this interface. Click <Restore Default> button to use the factory default MAC Address.
Auto-Negotiation	Ethernet connection configurations. Checked > Auto-negotiation mode; unchecked > Manual mode. Speed: 10M/100M/1000M Duplex: Full/Half (1000M is not applicable to Manual mode) See section 3.4.4 for detailed descriptions
Link Status	The connection status of the data link layer.

3.3.1.4 Multiple logical IP networks (available on the ESBC 93xx models)

The InnoMedia ESBC 93xx model series supports one physical WAN interface to accommodate either SINGLE or MULTIPLE logical network configurations.

Multiple logical IP network interfaces. With this mode, multiple IP addresses can be configured for different subnets on the ESBC WAN port, separating OAMP and Voice (signaling and media) traffic onto different network segments. This can be achieved by employing layer-2 VLANs together with layer-3 broadcast network domains. The separation of traffic types at both layer-2 and layer-3 facilitates policy control with regard to service security requirements and QoS management.

1. **Security requirements.** Voice and OAMP traffic on the WAN interface can be segregated using both Layer 2 VLANs and Layer 3 broadcast network domains. Therefore, voice traffic is routed directly to/from the service provider's voice network to the "ESBC WAN-Voice interface" on a particular network interface which is managed to ensure no external access. On the other hand, OAMP traffic may be from the open Internet or from a separate managed network. OAMP traffic (which may include web GUI access, provisioning, SNMP, etc.) is routed via a different WAN-OAMP interface. Therefore, in this case, it may be easier to avoid SIP attacks on the ESBC voice-WAN interface. This may then allow security configurations (such as SIP firewall rules and access control lists) to be greatly simplified in attempting to mitigate the effects of sip attacks.
2. **QoS management.** QoS parameters and routing policy are often very different for Voice and OAMP traffic. A more flexible network topology and traffic management methodology can, therefore, often be achieved through the use of multiple network interfaces.

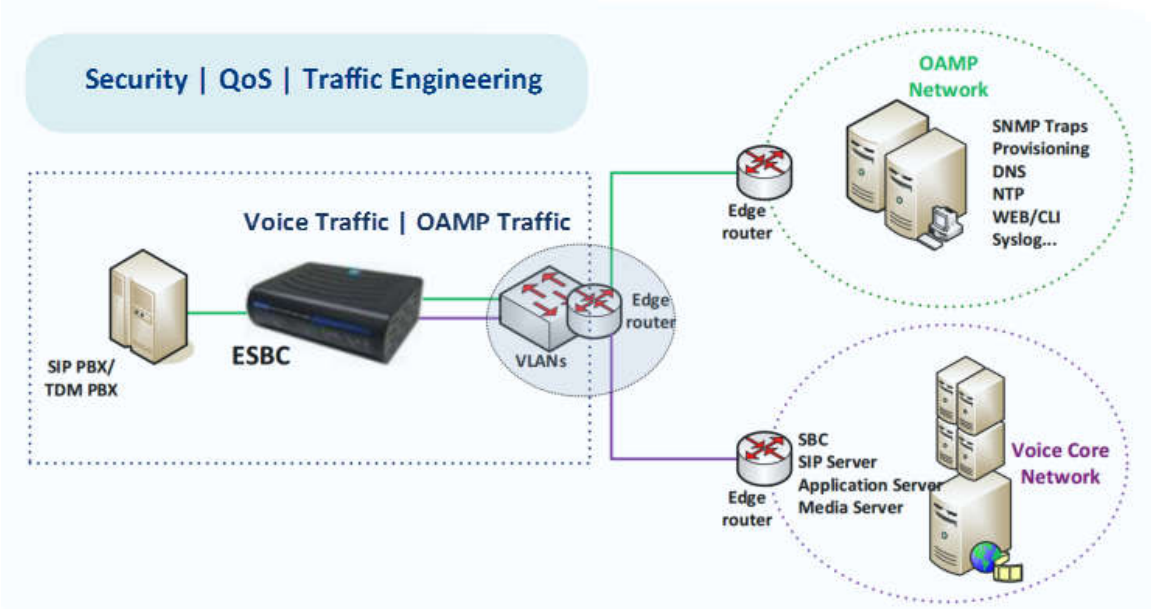


Figure 32. Multiple subnet architecture example for SIP Trunk voice service

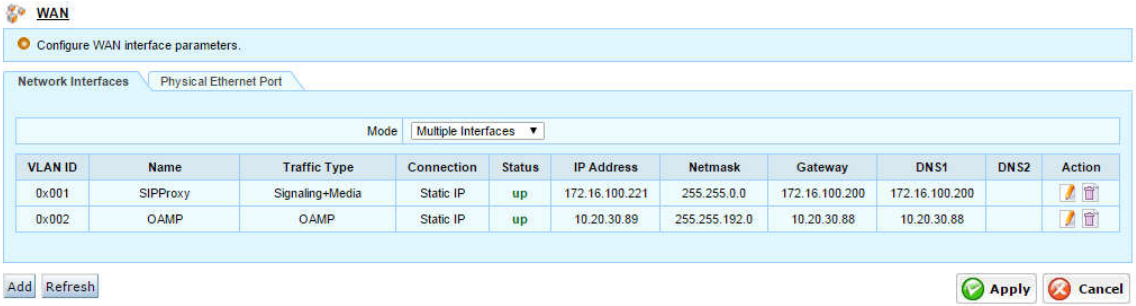


Figure 33. Configuring multiple logical IP addresses for the ESBC WAN port

The multiple interface table allows the configuration of logical IP network attributes assigned to OAMP and Voice networks.

- OAMP traffic includes service and control packets such as SNMP/EMS, provisioning, DNS, NTP, WEB/CLI access, syslog etc.
- Voice traffic, "Signaling+Media", includes SIP signaling call control messages, RTCP and RTP media packets.

Figure 32 illustrates a typical deployment where "multiple logical network interfaces" are used to separate management and voice traffic. The ESBC WAN interface is connected to a VLAN switch which connects the ESBC to two layer-3 broadcast domains using VLAN tags. For this type of configuration, Figure 33 represents a sample ESBC Multiple Interface Table.

Configuring Multiple Interfaces

The Multiple Interface Table page allows the definition of a logical interface with the following attributes:

- IP address and subnet mask (or DHCP)
- VLAN ID
- Default Gateway
- DNS

Both the OAMP and "Signaling+Media" networks may have DNS and DHCP servers configured individually.

Navigate to **Network** > **Settings** > **WAN**. Choose Multiple Interfaces mode, and click the <Add> button to add a (1) Signaling + Media network and a (2) OAMP network separately.

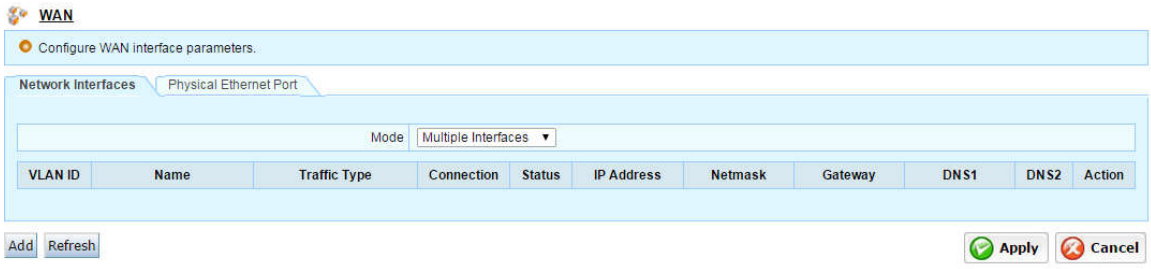


Figure 34. Configuring Multiple Network Interfaces

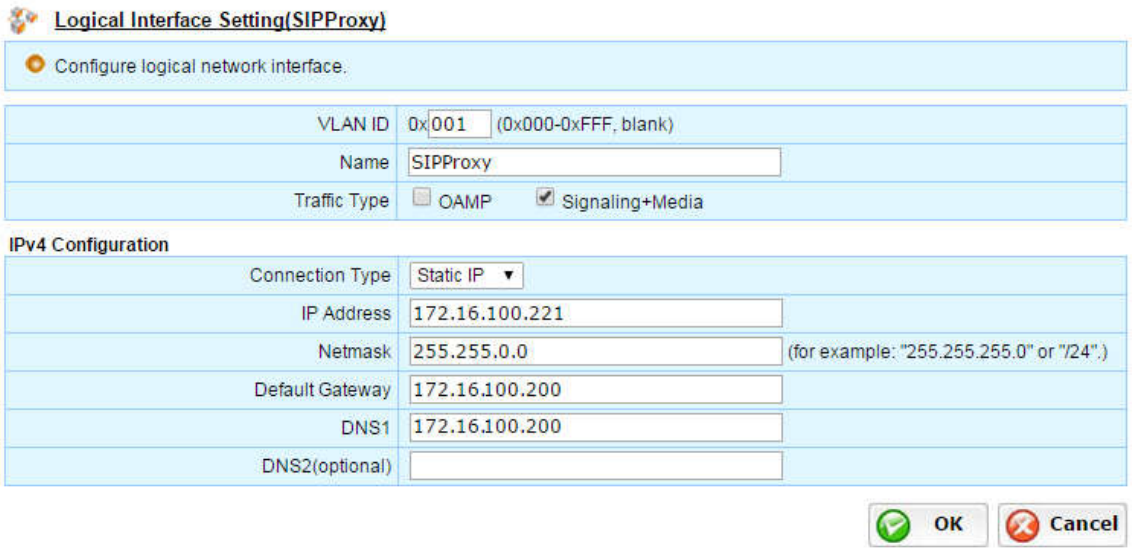


Figure 35. Configuring the “Signaling + Media” logical network interface

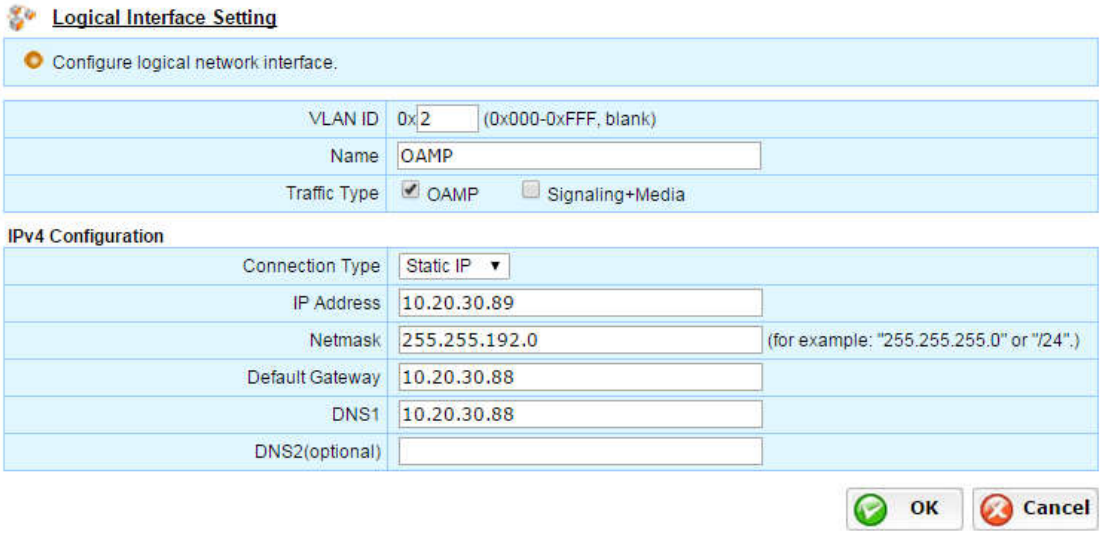


Figure 36. Configuring the “OAMP” logical network interface

Logical Interface Setting	Description
VLAN ID	Enter the VLAN ID assigned to this IP network.
Name	Enter the name of this logical network interface. This is a text string that can be assigned by the user to this interface for ease of reference.

Traffic Type	Configure “Signaling+Media” and “OAMP” separately with different networks and VLAN IDs. Note: Do not configure “Signaling+Media” and “OAMP” with the same VLAN ID and IP address for the mode of “Multiple Interfaces.” If “Signaling+Media” and “OAMP” traffic flows through one logical network, choose the mode of “Single Interface.”
Connection Type	DHCP Client (the default setting): use DHCP service to setup network connection. Static IP: Use the static IP provided by the network provider to setup the network connection.
IP Address	When Static IP is selected as the Connection Type, enter the IP address.
Netmask	When Static IP is selected as the Connection Type, enter the netmask associated with the IP address.
Default Gateway	When Static IP is selected as the Connection Type, enter the default gateway of this IP network.
DNS1 (and DNS2)	When Static IP is selected as the Connection Type, enter the DNS server(s) used with this IP network interface.

Note:

1. WEB/CLI console access through the WAN interface should connect to the **OAMP** network IP if it is configured differently from the Signaling+Media interface.
2. The **WAN Backup**, **Bridge** and **Router** port features for the ESBC LAN interfaces are not available when the WAN interface is configured in “Multiple Interfaces” mode. See section 3.3 for more details.

3.3.2 Cable modem embedded ESBC models

3.3.2.1 Logical Network Interface

Navigate to **Network** > **Settings** > **WAN**

The cable modem embedded models can be configured in “Single Interface” mode. Please refer to the descriptions in section 3.3.1.1 for details. VLAN settings are not applicable.

The ESBC configured as a DHCP client

The ESBC95xx series sends a TLV string in DISCOVER messages in Option 60 to allow the DHCP server to provide an associated IP address from the correct pool. (The TLV string format is defined in the PacketCable 1.5 specifications).

“pktc1.5:
0531010101020117 0B0206090C01010D0101100109120200041301011401011501011601011706020001
020100180100190101”

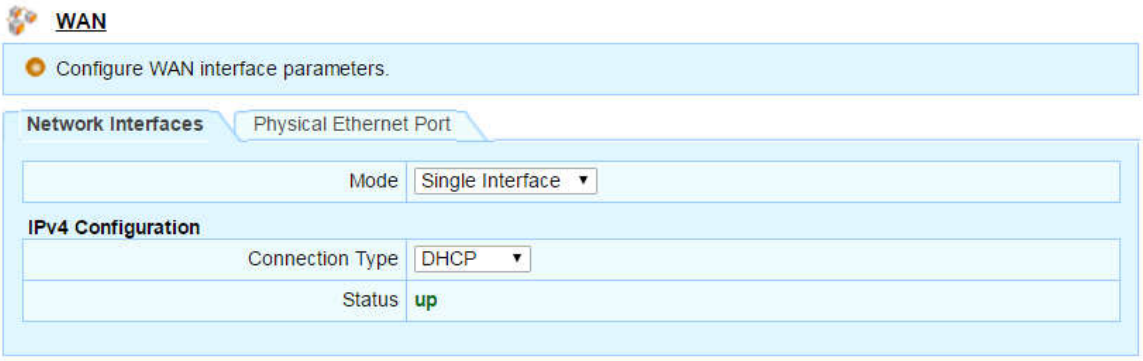


Figure 37. The ESBC is configured as DHCP Client

The ESBC configured with Static IP Address

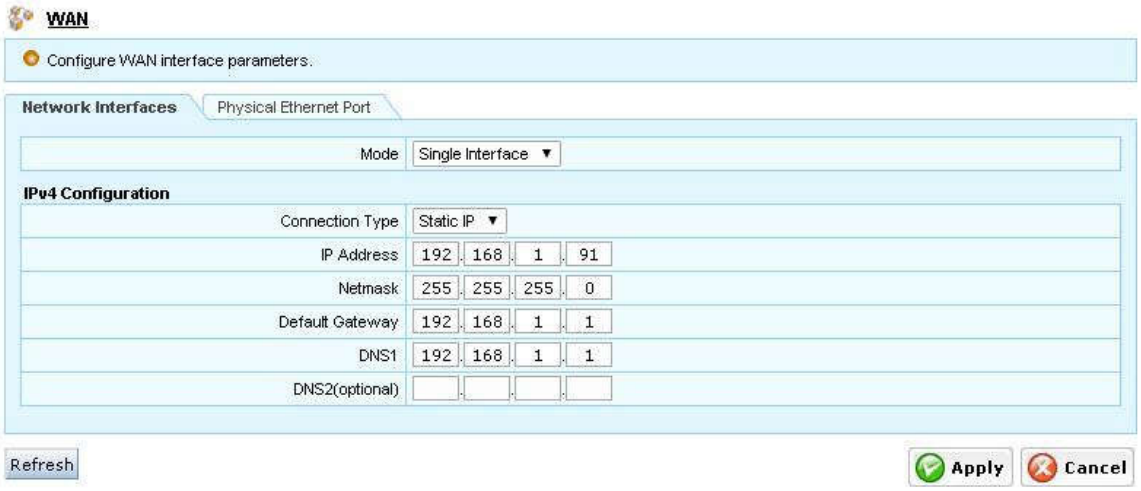


Figure 38. Configuring WAN IP address for cable modem embedded ESBC models

3.3.2.2 Physical Ethernet Port

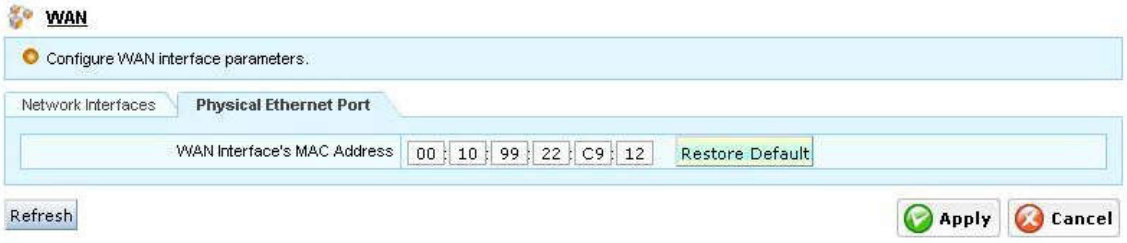


Figure 39 Configuring MAC address of the ESBC WAN interface

Please refer to section 3.3.1.3 for descriptions of configuring MAC address for the ESBC WAN interface.

3.4 LAN interface configurations

3.4.1 The LAN interface configurations for voice services

The default configurations of the LAN interfaces provide NAT-and-Voice services (applicable to ESBC-9xxx/8xxx series models). The four switch ports connect to the enterprise telephony network for SIP PBX, IP phones, and PCs to access the ESBC administrative consoles via the LAN interfaces, including WEB GUI and SSH CLI.

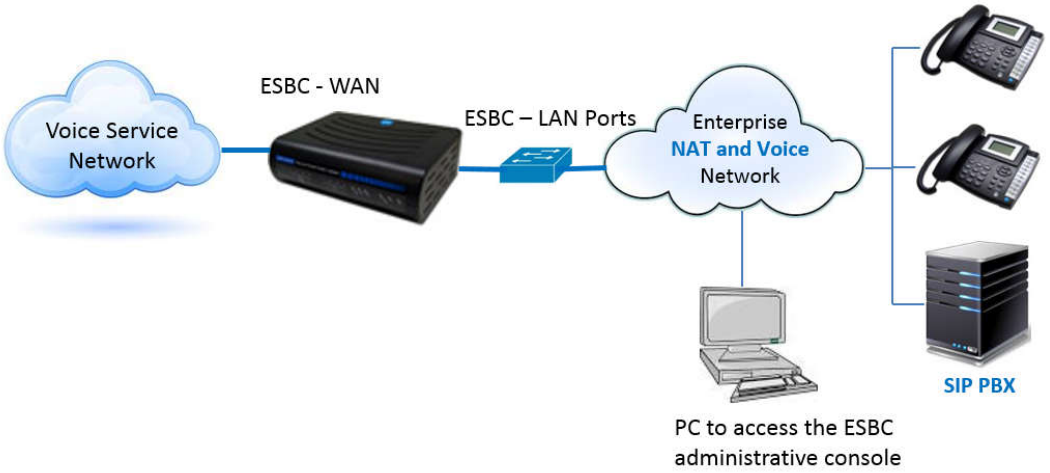


Figure 40. Default configuration: LAN ports serve enterprise telephony services

Navigate to **Network** > **Settings** > **LAN** to configure the IP address. By default, all the LAN ports serve **NAT-and-Voice** services which are switch ports and share one IP address.

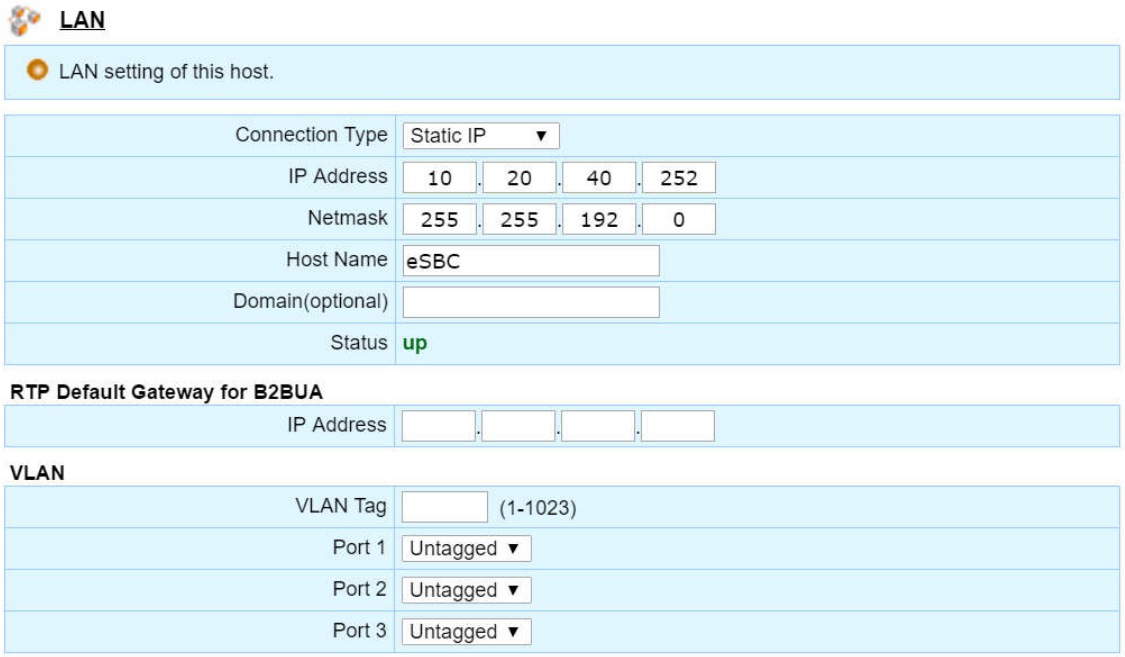


Figure 41. Configuring the IP address of the ESBC LAN for NAT-and-Voice services

LAN Settings of this host	Description
Connection Type	Static IP, or DHCP Client. When DHCP client is selected for the ESBC LAN interface, it is recommended that the ESBC-LAN MAC address is bound with an IP address from the DHCP server.
IP Address, Netmask	When “Static IP” is selected as the Connection Type, enter the IP address and its associated netmask value.
Host Name	The host name designated for the ESBC.
Domain (optional)	The network domain name, if the network administrator defines a domain name and nominates servers to control security and permissions for the ESBC telephony network.
Status	The connection status of the data link layer.
RTP Default Gateway for B2BUA	The IP address of the router which connects two corporate VoIP networks. See section 3.4.1.1 for more details.

VLAN	Description
VLAN Tag	When the ESBC LAN interface needs to configured as a tagged VLAN port (Trunk port), appropriate tagging information is needed. If this field contains a VLAN-ID value, traffic from devices connecting to Voice-NAT port(s) must be tagged with the same

	VLAN-ID.
Port N	Assign the property Tagged or Untagged VLAN, for any port configured as “Voice/NAT” mode. • Tagged or “trunk port” (tagged with 802.1q tag). Connect to another trunk port of the same VLAN ID. • Untagged or “access port”. Connect to a non-VLAN tagged port.

3.4.1.1 LAN side Topology: RTP Default Gateway for SIP Trunk (B2BUA) voice services

The use of RTP default gateway. In a typical deployment scenario, the SIP PBX registers to the ESBC LAN interface, and the SIP IP Phones (SIP PBX clients) register to the SIP PBX. Some SIP PBXs control SIP signaling from the SIP Phones but do not route RTP packets. The RTP packets travelling through the corporate data network between the ESBC and the SIP Phones do not necessarily go through the SIP PBX. When SIP Phones and SIP PBX are not located in the same network, one-way communication or no voice is possible if no static routes nor RTP Default Gateway are configured on the ESBC.

The ESBC RTP Default Gateway feature has been designed to make the communication between SIP user agents in different corporate networks possible, allowing scalable distributed SIP VoIP networks. When direct end-to-end media communication is not possible, the media (RTP) streams have to be relayed through another host (i.e., the corporate router acting as the RTP default gateway) to play the role of proxying RTP streams for SIP phones. As the corporate network grows, the ESBC is adaptive to new corporate network configurations without the need to manually update its static routing rules. The RTP Default Gateway feature can also be used in combination with configuring static routing rules (see section 3.4.1.2) to the ESBC to build complex VoIP networks.

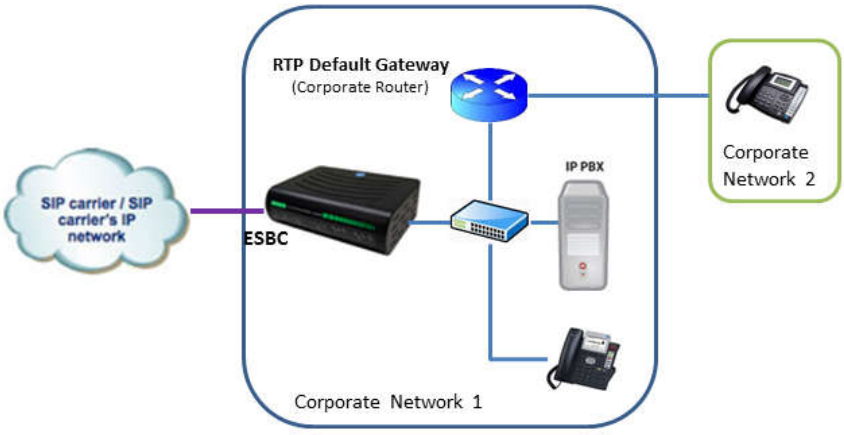


Figure 42. The enterprise Router acting as the RTP default gateway for SIP devices

With the example illustrated in Figure 42, the SIP Phones in Corporate Network 2 register to the IP PBX in Network 1. With the RTP default gateway (the corporate router's IP address) configured on the ESBC, the RTP (media) streams can be handled by the ESBC and communicate with the service provider network. Note that when RTP default gateway is configured, the static routing rules may not be necessary.

The IP PBX has to be located in the same network as that of the ESBC LAN interface if there is no static route rule configured (see section 3.4.1.2).

3.4.1.2 LAN side Topology Design: Static Routing Configurations

For enterprises with multiple voice networks, static routing rules are needed if one of the following conditions is true.

- SIP Trunk voice service (B2BUA mode): IP PBX (or any SIP User Agents) which register to the ESBC are located in a network other than that of the ESBC NAT-Voice interface.
- Hosted voice service (SIP-ALG mode): the IP phones are not located in the ESBC NAT-Voice network.

Note that when static routing rules are configured, the RTP default gateway may or may not be necessary. To configure Static Routing rules, navigate to **Network** > **Advanced** > **Static Routing**.

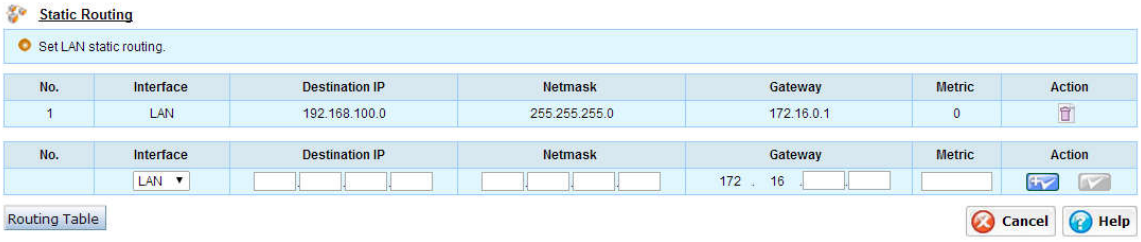


Figure 43. Configuring network static routing rules for the ESBC LAN interface

Static Routing	Description
Number	Record number.
Interface	The interface to which static routing rules apply (LAN only).
Destination IP	The destination network address which this route reaches.
Netmask	The destination network netmask
Gateway	IP address of the router to which the ESBC can reach to route packets for this particular network.
Metrics	Metrics count. If not configured, the default value is 1. (Note: Metric is the network administrative distance. The default value for connected interface is 0, and static route is 1. Lower numbers take priority over higher numbers.)

Click <Routing Table> button to display the ESBC network routing information. (<Routing Table> display is available only when the WAN Interface mode is configured as Single Interface.)

3.4.2 DHCP Server

The ESBC DHCP server allows clients which connect to the Voice-and-NAT ports to obtain dynamic IP addresses. The administrator can also view a DHCP client list and MAC bindings.

Navigate to **Network** > **Settings** > **DHCP Server**.

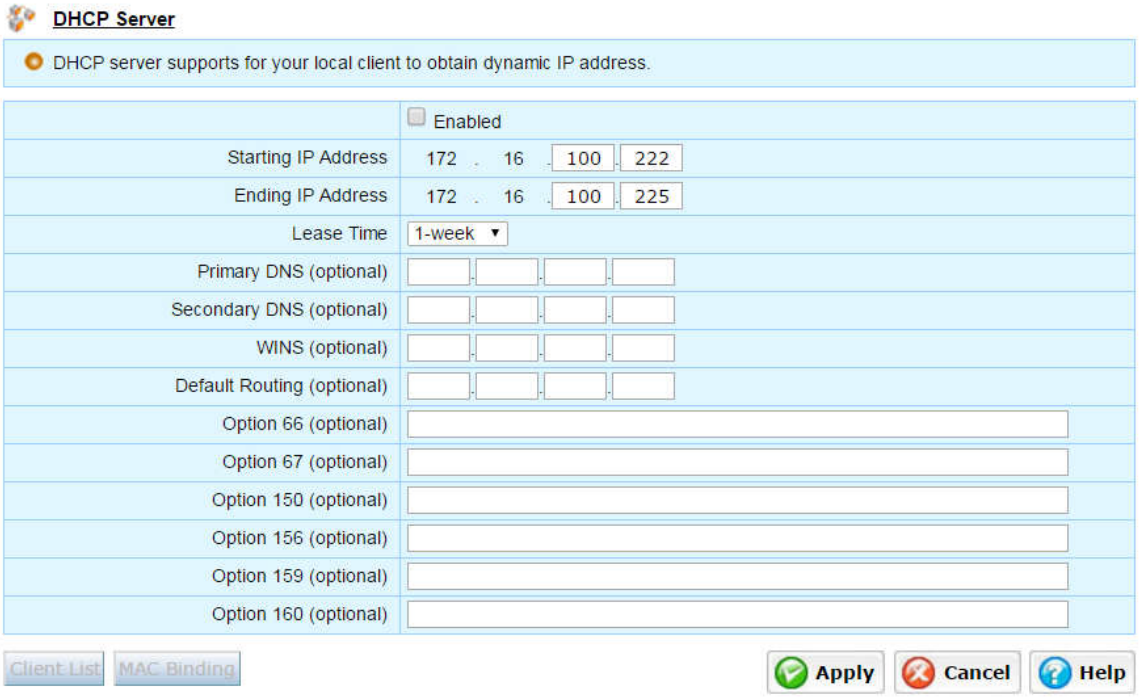


Figure 44. The DHCP Server for devices in the Voice-and-NAT network

Static Routing	Description
Enabled	Check this box to enable DHCP server service on the ESBC. The default configuration is disabled.
Starting IP Address Ending IP Address	The range of IP addresses assigned to the LAN clients. Note that this range should not include the broadcast address, e.g., network 172.16.0.0/16 has a broadcast address of 172.16.255.255.
Lease Time	The time period for which the IP address assigned to DHCP clients is valid.
Primary DNS (optional) Secondary DNS (optional)	The DNS server(s) specified provides name service for DHCP clients.
WINS (optional)	Windows Internet Naming Service, for NetBIOS names.

Default Routing (optional)	
Option 66 (optional)	Obtain the host name or IP address of the provisioning server for DHCP clients, i.e., allows the connected SIP devices to obtain a provisioning server address. A text string.
Option 67 (optional)	Boot file name. A text string.
Option 150 (optional)	List of TFTP Server IP address(es) for IP phone image download.
Option 156 (optional)	FTP server details. A text string
Option 159 (optional)	Specifies a text string in the form of a FQDN. It can be used to point the IP phones to the domain name of a TFTP server using HTTP.
Option 160 (optional)	Specifies a text string in the form of a FQDN. It can be used to point the IP phones to the domain name of a TFTP server using HTTPS.

3.4.2.1 Client List

This client list table include IP address, MAC address, obtained time and expired time of all DHCP clients.

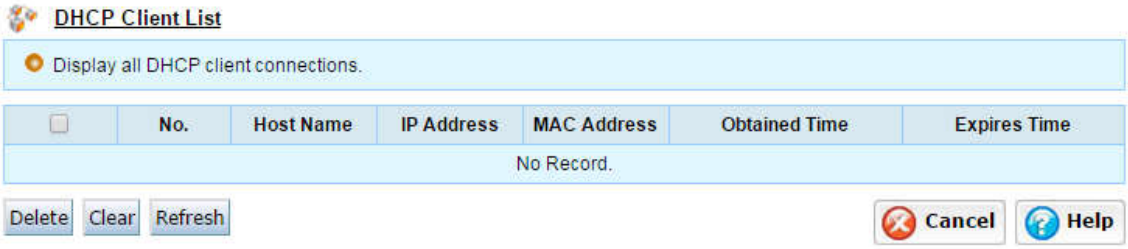


Figure 45. The ESBC DHCP server – client list.

3.4.2.2 MAC Binding

The DHCP MAC Binding feature allows the system administrator to bind an IP address to a client's MAC address, so the ESBC will always assign a fixed IP address to this client. Click the <MAC Binding> button.

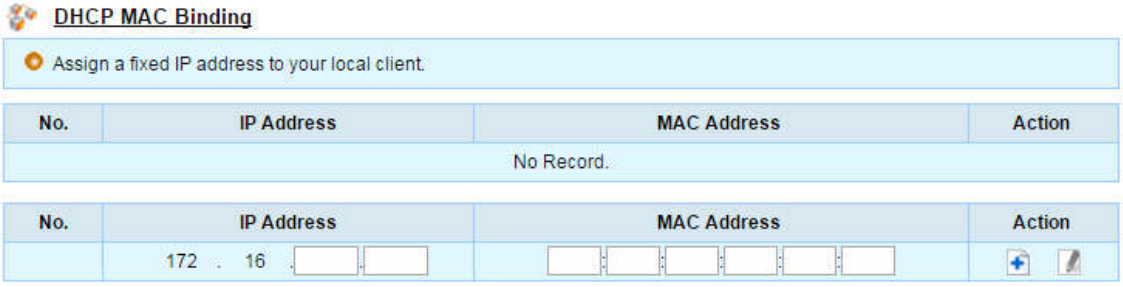


Figure 46. The ESBC DHCP server – MAC binding.

3.4.3 Advanced Configurations for Voice and Data Featured Services

The ESBC 9xxx and 8xxx series are equipped with four Ethernet switch ports which can be configured individually to fulfill various featured services for field deployments. These featured services can be all or partially activated on one ESBC unit. They are

- NAT-and-Voice port
- Management port
- Router port (applicable only when the WAN Interface mode is configured as Single Interface)
- Bridge port (applicable only when the WAN Interface mode is configured as Single Interface)

Figure 47. The ESBC 9xxx interface for network access (conceptual view)

To configure the ESBC featured service, navigate to **Network** > **Advanced** > **LAN Interfaces > Port Function**.

Note that Bridge port and Router port are available when the WAN Interface mode is configured as Single Interface.

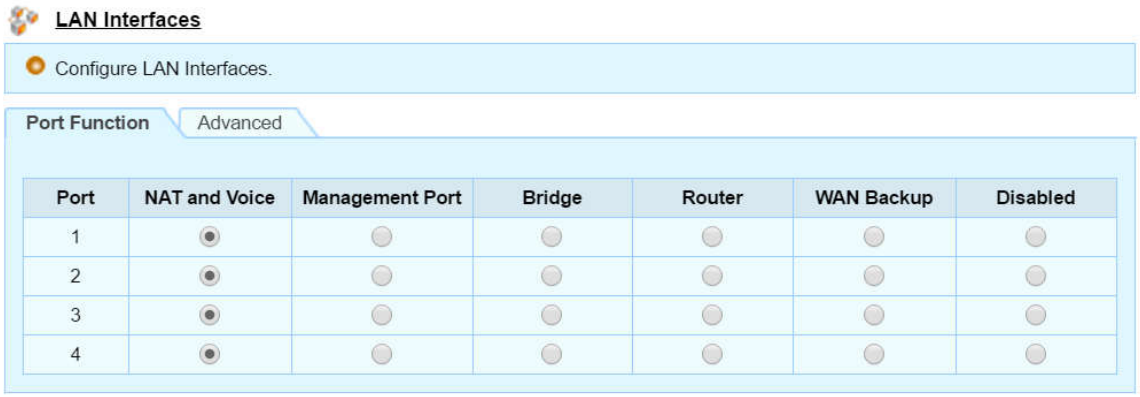


Figure 48. The ESBC 9xxx LAN interfaces. Default Settings (configuration view)

3.4.3.1 Enabling the Management Port

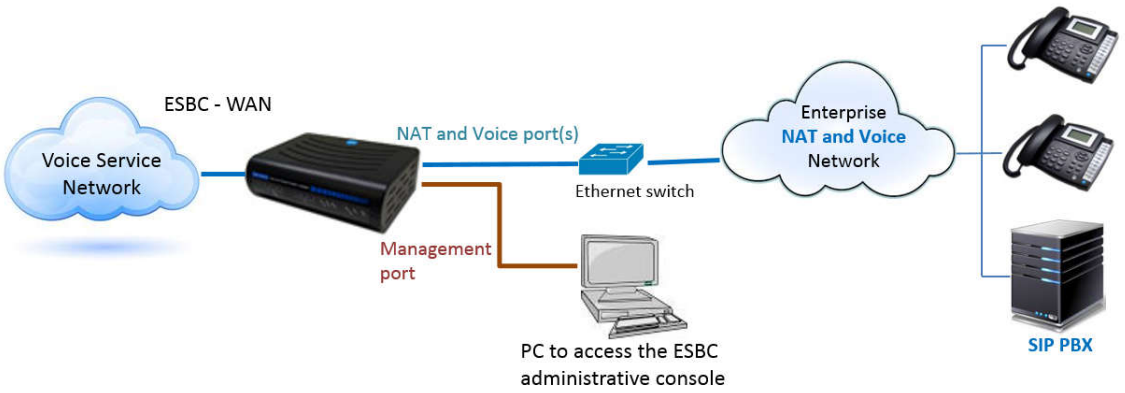


Figure 49. Enabling the ESBC management port

The management port is used by the PC to access the ESBC administrative console (WEB or CLI) through the LAN interface. When the management port feature is enabled, the LAN NAT-and-Voice ports only serve voice traffic, and do not allow PCs to access the administrative console. Access via the ESBC WAN interface remains unchanged.

Navigate to **Network** > **Advanced** > **Management Port.**

The ESBC Management port can serve a subnet. If the associated DHCP function is enabled, its netmask and IP address range have to be properly managed.

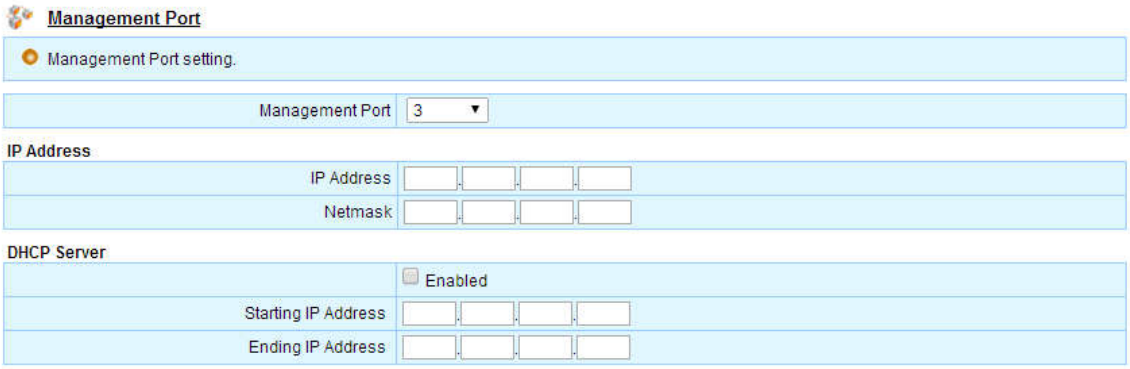


Figure 50. Configuring the ESBC Management Port

Note: If the management port is enabled, it is possible to access the administrative console through management port only, or the WAN port.

The ESBC 10K series models are equipped with two LAN ports. Port #1 is assigned as a management port, and port #2 is a NAT-and-Voice port.

3.4.3.2 Enabling the Bridge Port

Bridge port is applicable when the WAN Interface mode is configured as Single Interface.

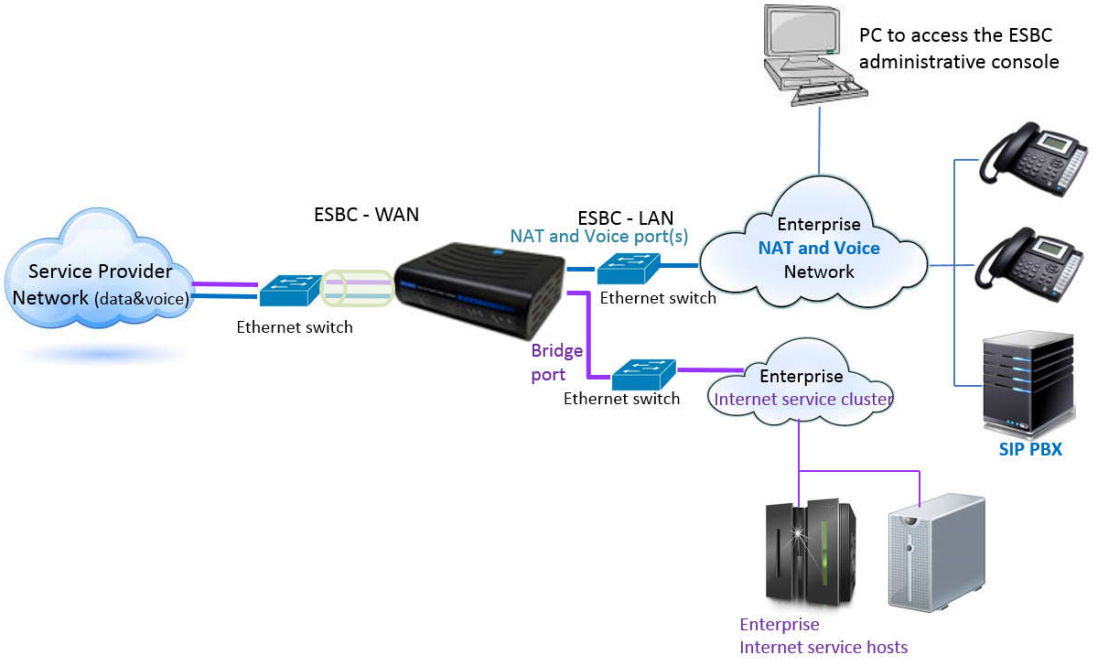


Figure 51. Enabling the ESBC Bridge Port for Enterprise Internet Service Hosts

The bridge port is transparent to the ESBC WAN interface. Typical applications of the bridge port are as follows:

- To serve enterprise Internet service hosts
- ESBC installers or technicians may make use of it to access an OSS in the service provider's core network, especially for cable modem embedded ESBC models.

3.4.3.3 Enabling the Router Port for data services

Router port is applicable when the WAN Interface mode is configured as Single Interface.

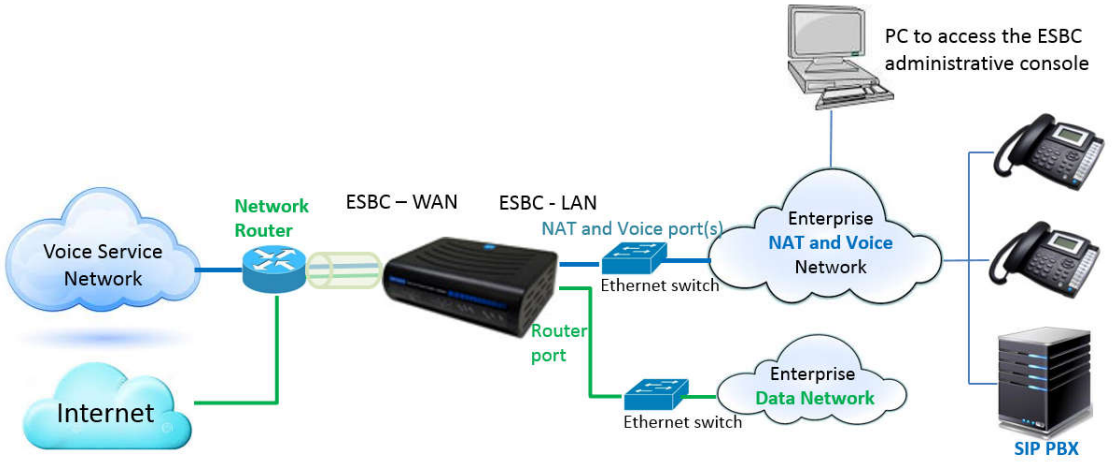


Figure 52. Enabling the ESBC Router Port for Data Service

When the router port is enabled, the ESBC router port performs data network router functionality, which allows service providers to offer data services to enterprise customers. As Figure 52 illustrates, hosts behind the ESBC router port may use public IP addresses which are routed by the ESBC to the service provider network. The ESBC dynamically updates its routing table with the Network Router in the service provider's core network.

Navigate to **Network** > **Advanced** > **Router Mode** to configure router and protocol security parameters.

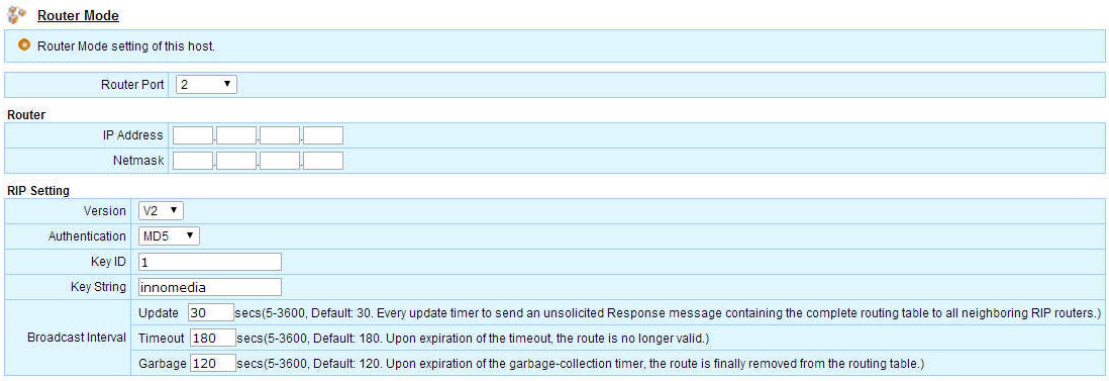


Figure 53. Configuring the Router Mode Port

Router Mode	Description
Router Port	Select the appropriate port number as the router port.
Router- IP address Netmask	Enter the public IP address and netmask that the service provider designated for this enterprise Router which serves as the default gateway for hosts on the enterprise data network.
RIP Setting	
Version	V2: standard routing protocol with associated authentication features. V1: standard routing protocol.
Authentication	MD5 | Text | None MD5 is the default mode if RIPV2 is selected. Plain text authentication should not be used when security is an issue. Select appropriate method which interoperates with the core router in the service provider's network.
Key ID	Applicable to MD5. Give the RIPV2 key chain name. This need not to be identical with that of the remote router.
Key String	Applicable to MD5. The actual password. It needs to be identical to the key string on the remote router.
Broadcast Interval	Update. The interval after which the EBSC sends an unsolicited response message of the complete routing table to all neighboring RIP routers. Timeout. (Upon this timeout threshold being reached, the particular route is no longer valid. Garbage. Upon this timer expiring, this particular route is removed from the routing table.

Note: RIP route authentication is configured on a per-interface basis. All RIP neighbors on interfaces configured for RIP message authentication must be configured with the same authentication mode and key for adjacencies to be established.

3.4.4 Ethernet Advanced Configurations for LAN Interfaces

See Application note: ESBC Application Notes-Ethernet Control.doc

Ethernet interfaces. Navigate to **Network** > **Advanced** > **LAN Interfaces** > **Advanced.**

The ESBC's Ethernet connection can be configured for one of the following modes:

Manual	
Speed	10M, 100M
Duplex	Full, Half
Auto Negotiation	
Speed/Duplex	Automatically choose common transmission parameters. Speed: 10M/100M/1000M Duplex: Full/Half

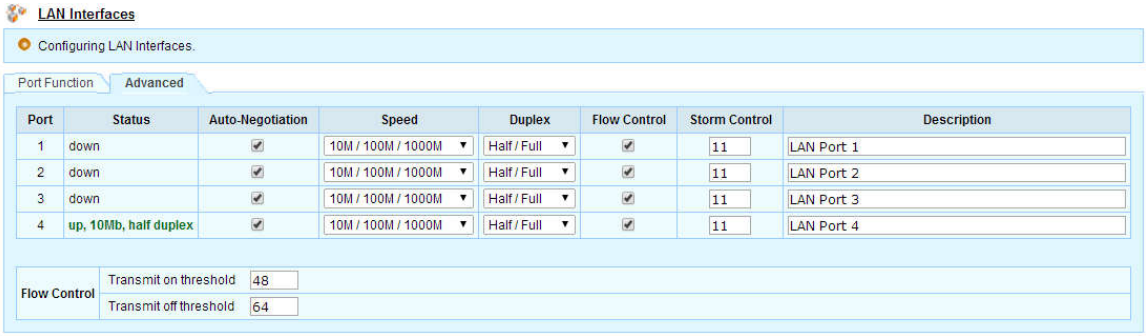


Figure 54. Ethernet Interface Controls for the LAN Interfaces of ESBC 8xxx and 9xxx series models

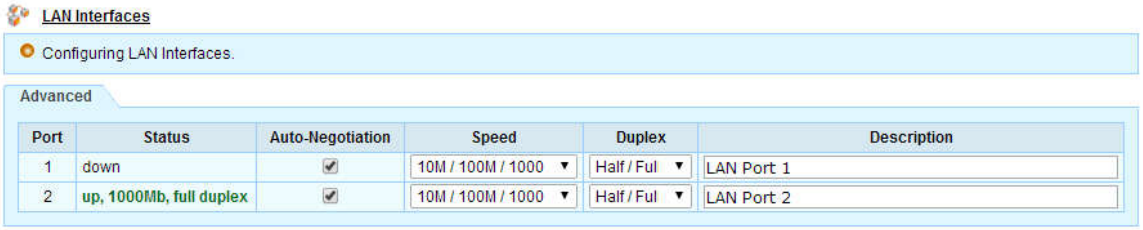


Figure 55. Ethernet Interface Controls for the LAN Interfaces of ESBC 10K series models

Note: If the management port is enabled, it is possible to access the administrative console through the management port only or the WAN port. The ESBC 9xxx and 8xxx series models support four LAN ports which are configurable. The ESBC 10K series models are equipped with two LAN ports. Port #1 is assigned as a management port, and port #2 is a NAT-and-Voice port.

- If the connected devices support auto-negotiation, it is highly recommended that auto-negotiation is used. The remote side port must also operate in auto-negotiation mode.
- When configuring the device port running in manual mode, the same mode (i.e., duplex and speed) must be configured on the remote port manually.

WARNING: An Ethernet port that does not match the settings of the connected device can lose connectivity.

Link status of Auto-Negotiation. The correct behavior of link status with auto-negotiation in accordance with IEEE Std 802.3z-1998 should be as follows:

- If A is enabled and B is enabled, then the link status should be reported on both devices as link up.
- If A is disabled and B is enabled, then A should report link up and B should report link down.
- If A is enabled and B is disabled, then A should report link down and B should report link up.

Flow Control. ESBC Ethernet flow control is used to regulate the amount of traffic sent out of the interface. There is a PAUSE functionality built into the ESBC Ethernet LAN interfaces to avoid dropping of packets. Flow control **must** be negotiated, and hence both devices must be configured for full duplex operation to send PAUSE frames.

- Threshold of **transmit on** (48 buffers by default, ranges from 16-256).
- Threshold of **transmit off** (64 buffers by default, ranges from 16-256, must be greater than threshold of transmit on).

Storm Control. The ESBC storm control is used to for network broadcast and multicast packets. It prevents network outage on the LAN interfaces, and rate limit broadcast and multicast traffic at a specified level and to drop packets when the specified traffic level is exceeded.

When the broadcast and multicast storm control is enabled, all broadcast and multicast packets beyond the thresholds are discarded. The threshold values are as follows:

Threshold of Storm Rate: unit in frames, Rn, n ranges from 1 to 11 (1M fps by default).

3.4.5 Remote access to the ESBC LAN Interfaces and LAN hosts

3.4.5.1 Through VPN

The ESBC allows remote computers (Windows or any hosts supporting PPTP) to access the ESBC LAN interfaces for administrative tasks or other LAN hosts via a PPTP VPN secured tunnel. Note that when it is used to reach other LAN hosts, an additional LAN router needs to be configured to route traffic from the LAN hosts to the ESBC VPN clients.

Navigate to **Network** > **VPN** > **PPTP Server**.

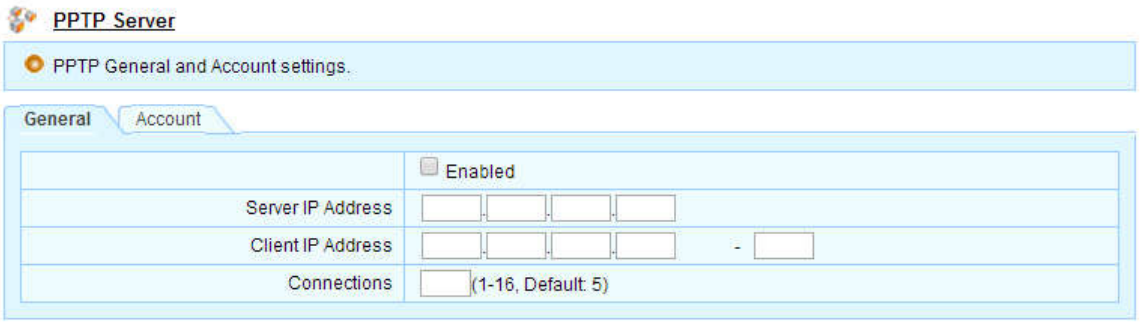


Figure 56. Configuring the VPN client IP range

PPTP-Server (General)	Description
Enabled	Enable the PPTP VPN server. The default for the VPN feature is disabled.
Server IP Address	Create and enter the VPN server IP address within this VPN network range. This virtual network should not conflict with the networks configured on both the ESBC WAN and LAN interfaces.
Client IP Address – range	The VPN server acts as a DHCP server and grants an IP address to clients when they pass the security checks.
Connections	The number of VPN concurrent connections.

Enter the ESBC WAN IP address when configuring the “Internet Address” on the VPN client running on the host operating system, such as Windows. When the ESBC WAN Interface mode is configured as “Multiple Interfaces”, enter the IP assigned to the OAMP network.

Figure 57. Configuring the VPN client ID and password for connection

Use the configured IDs and passwords for the remote VPN clients to access to the VPN server, the ESBC LAN and the LAN hosts.

3.4.5.2 Through Port Forwarding

Port forwarding is applicable when the WAN Interface mode is configured as Single Interface.

Software ports are numbered connections that a computer uses to sort different types of network traffic. The ESBC supports port forwarding features which allow remote computers to access services offered by other LAN hosts. A few standard services are listed on the Comments section of the ESBC port forwarding page. By default, the ESBC closes all software ports to the Internet unless they are configured in the port forwarding list, and SIP/RTP ports which are dynamically opened for communication purposes.

Navigate to **Network** > **Advanced** > **Port Forwarding.**

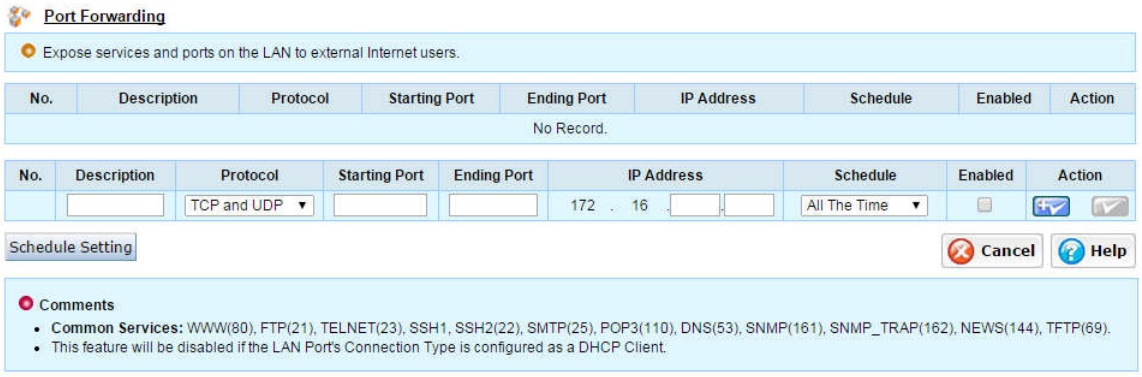


Figure 58. The port forwarding feature for remote access to LAN services

Port Forwarding	Description
Description	Enter the purpose of forwarding the specified port number range to access software services provided by LAN hosts.
Protocol	Enter the transportation protocol(s) used by the software service.
Starting port – Ending port	Enter the port range used by the software service.
IP Address	Enter the IP address of the LAN host which offers the software service.
Schedule	All the time/Working Time. Click the <Schedule Setting> button to configure the working time slots.

3.4.6 Enabling Data Service Access for the ESBC LAN hosts

3.4.6.1 DNS Proxy

The DNS Proxy feature is applicable to the ESBC-93xx and 83xx serial models.

Because DNS is used by virtually every device connected to the Internet, it is a common target for hacker attacks. For security and performance considerations, the ESBC DNS proxy feature includes rule sets to control outgoing DNS requests from its trusted hosts. A typical DNS proxy processes DNS queries by issuing a new DNS resolution query to each name server in the list until the hostname is resolved. A DNS proxy improves domain lookup performance by caching previous lookups. When a DNS query is resolved by a DNS proxy, the result is stored in the device's DNS cache. This stored cache helps the devices to resolve subsequent queries from the same domain and avoid network latency.

Navigate to **Network** > **Advanced** > **DNS Proxy**.

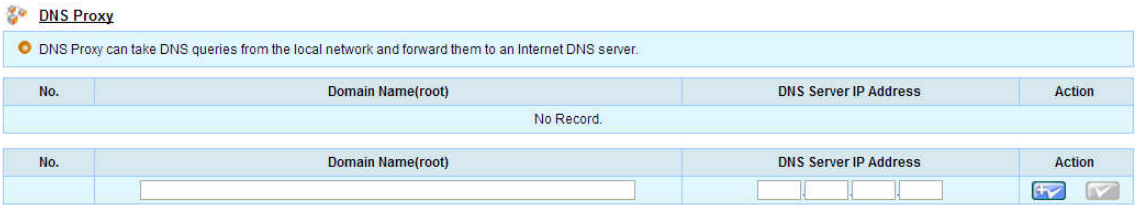


Figure 59. Configuring the DNS Proxy service for trusted LAN hosts

DNS Proxy	Description
Domain Name (root)	Domain Name suffixes, e.g. the domain name root of ftp.abc.com is abc.com.
DNS Server IP	Enter the IP address of the DNS server used to query this particular domain name.

3.4.6.2 Access Control

The Access Control feature is applicable when the WAN Interface mode is configured as a Single Interface. The Access Control feature provides the basic traffic filtering capabilities which enables ESBC LAN hosts, i.e., clients connecting to the ESBC Voice-and-NAT ports, to access Internet data services other than voice services (SIP Trunk or Hosted voice services).

Navigate to **Network** > **Advanced** > **Access Control**.

Access Control – LAN

The Access Control feature provides basic traffic filtering capabilities which enable ESBC LAN hosts, i.e., clients connecting to the ESBC Voice-and-NAT ports, to access non-voice traffic.

Enable legitimate ESBC LAN hosts to access Internet data services (such as provisioning), by specifying:

- hosts within particular IP address ranges
- hosts within particular subnets
- hosts with specified MAC addresses
- ports employed by applications

In cases where the SIP devices connected to the ESBC Voice-and-NAT port(s) need to access the provisioning server to update configuration files or an image, it is necessary to enable LAN devices to access the Internet using access control.

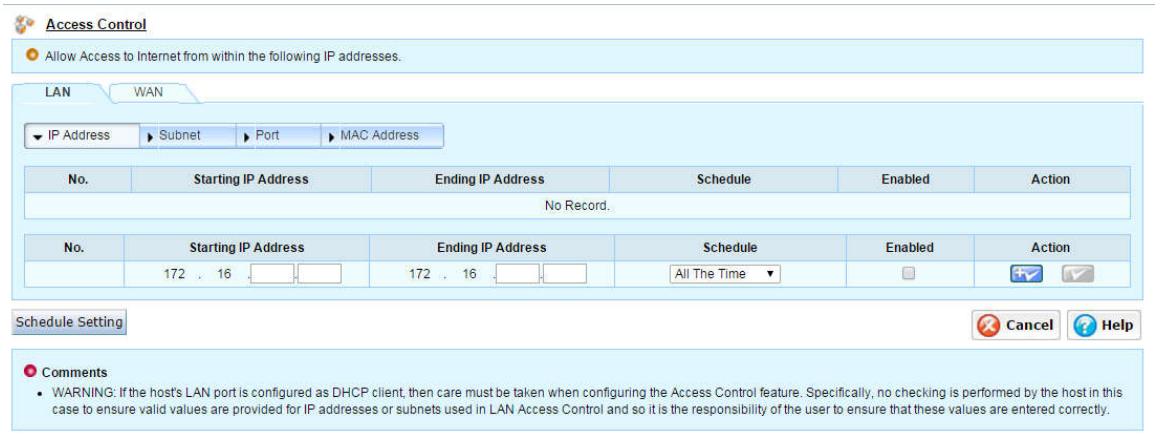


Figure 60. To enable internet data access to the ESBC LAN hosts or applications

Note that when the ESBC LAN interface connection type is configured as DHCP client (see section 3.4 for details), the ESBC does not check the validness of access control configurations.

Click <Schedule Setting> to define the time periods during which the legitimate hosts/ports are allowed to access Internet data services.

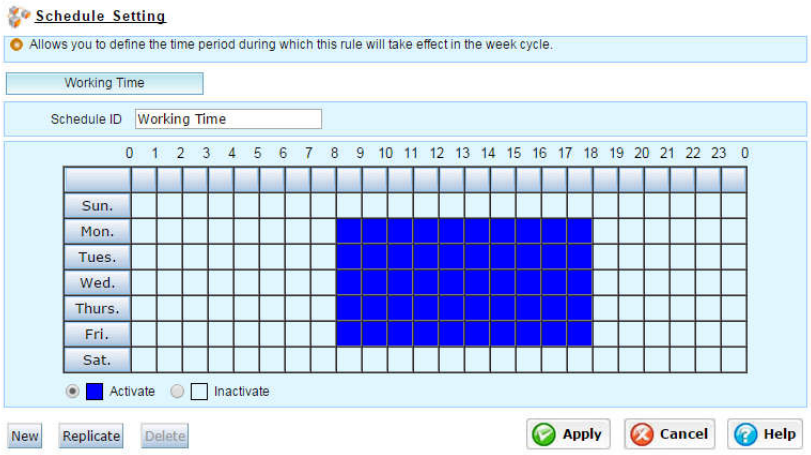


Figure 61. Schedule setting for LAN hosts and application to access Internet services

Access Control – WAN

The ESBC may impose restrictions on Internet resources to which the LAN clients can access, by specifying

- particular IP address ranges
- particular Subnets,

- ports employed by applications
- domains

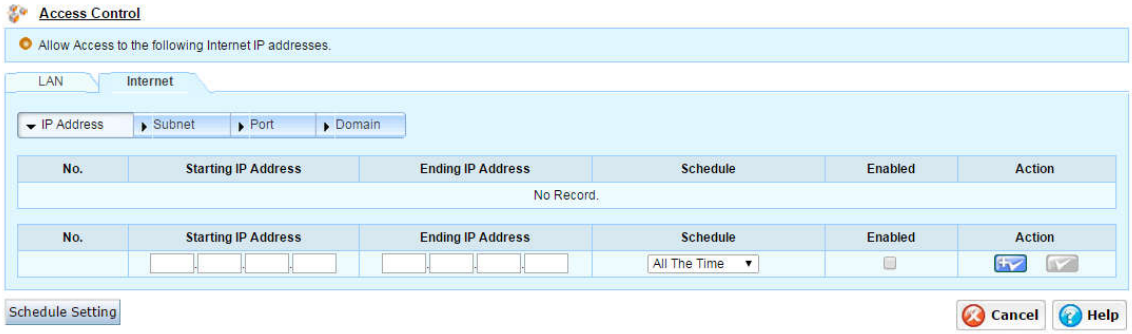


Figure 62 Legitimate Internet resources

Click <Schedule Setting> to define the time periods during which the legitimate Internet resources are accessible by the ESBC LAN hosts.

3.4.6.3 UPnP

The UPnP feature is applicable to the ESBC-93xx and 83xx serial models, and is applicable when the WAN Interface mode is configured as Single Interface.

The ESBC supports UPnP, and hence it can auto discover and control Internet connections at any place in a small office environment, provided its north bound switch/router supports UPnP.

Navigate to **Network** > **Advanced** > **UPnP**.

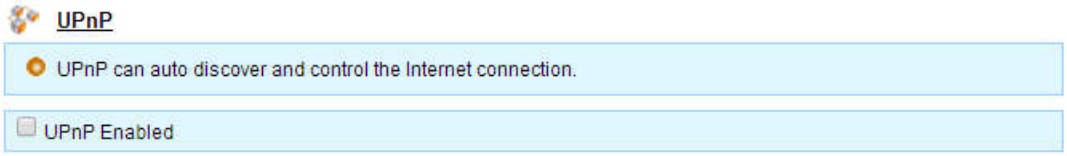


Figure 63. Enabling the UPnP network feature

Enable the UPnP feature by checking the box if the office network switch/router is equipped with this capability.

3.4.6.4 DMZ (De-militarized Zone)

The DMZ feature is applicable to the ESBC-93xx and 83xx serial models, and is applicable when the WAN Interface mode is configured as Single Interface.

In the ESBC setup, most devices on the LAN run with ESBC firewall protection to communicate with the Internet/service provider network. The DMZ is a device inserted in the “neutral zone” between the ESBC private network and the outside public network. It prevents outside users from getting direct access to an ESBC LAN host. A DMZ is optional, and acts as a proxy server as well.

Navigate to **Network** > **Advanced** > **DMZ**.

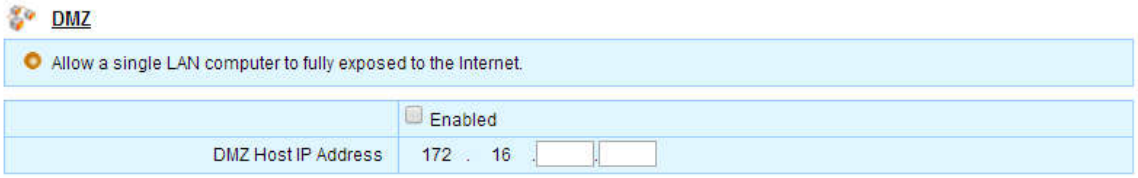


Figure 64. Enabling the DMZ host on the ESBC LAN network

Enter the IP address which is assigned to the DMZ host.

3.4.6.5 Miscellaneous

Navigate to **Network** > **Advanced** > **Miscellaneous.**

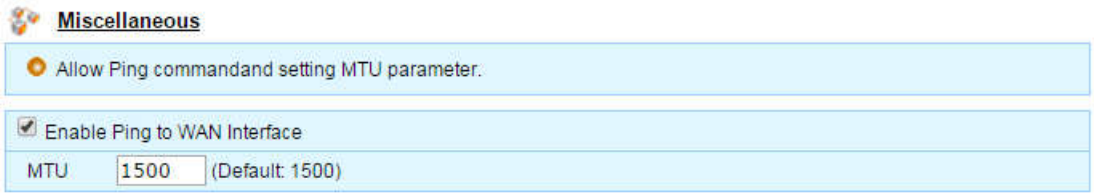


Figure 65. Miscellaneous configurations of network attributes

Miscellaneous	Description
Enable Ping to WAN Interface	For security purpose, disable “Ping” to WAN interface, i.e., not to respond to ICMP messages, by unchecking this box. When the WAN Interface mode is configured as Multiple Interfaces, “ping” command is applicable to the IP assigned for the OAMP network.
MTU (maximum transmission unit)	The default value 1500 (byte per packet), the largest packet size allowed by Ethernet. A larger MCU brings greater efficiency. Reduce MTU value if any network device cannot support the specified MTU size.

3.5 Using an NTP server to offer time information to ESBC LAN devices

The ESBC can be configured to act as an NTP server to offer time information to ESBC LAN devices.

Navigate to **Network** > **Settings** > **NTP Server.**

Figure 66. Enabling or disabling the ESBC NTP feature

The process to configure the time zone and the Internet server with which the ESBC synchronizes is described in section 6.1.2.

When the ESBC LAN devices need to utilize the ESBC to synchronize system time, they need to be “SNTP client” compliant and point their SNTP server to the ESBC LAN IP address (NAT-Voice port).

3.6 QoS Control

3.6.1 QoS Settings for Voice and Data Traffic

The ESBC manages traffic from different interfaces, and prioritizes voice traffic.

- Uplink "Data Bandwidth Control". When this feature is enabled, the ESBC dynamically assigns voice and data packets to prioritized queues transmitted toward the WAN interface. The priorities of different traffic queues are as follows:

  Voice packets > data packets generated by the ESBC or data generated from the NAT-Voice or the RIPv2 router port.

- ToS (type of service) labels traffic toward the hosts of the LAN and WAN, and distinguishes signaling and voice traffic from others.

Note:

1. By default, the ESBC disables data service for hosts of the ESBC NAT-Voice network. (Please refer to section 3.4.6.2 "Access Control" to enable data service for hosts on the NAT-Voice network.)
2. The ESBC does not control data traffic through the bridge port network. When heavy traffic transmitting through the bridge port, the bandwidth for voice is not guaranteed.

Navigate to **Network** > **QoS Control** > **Voice QoS.**

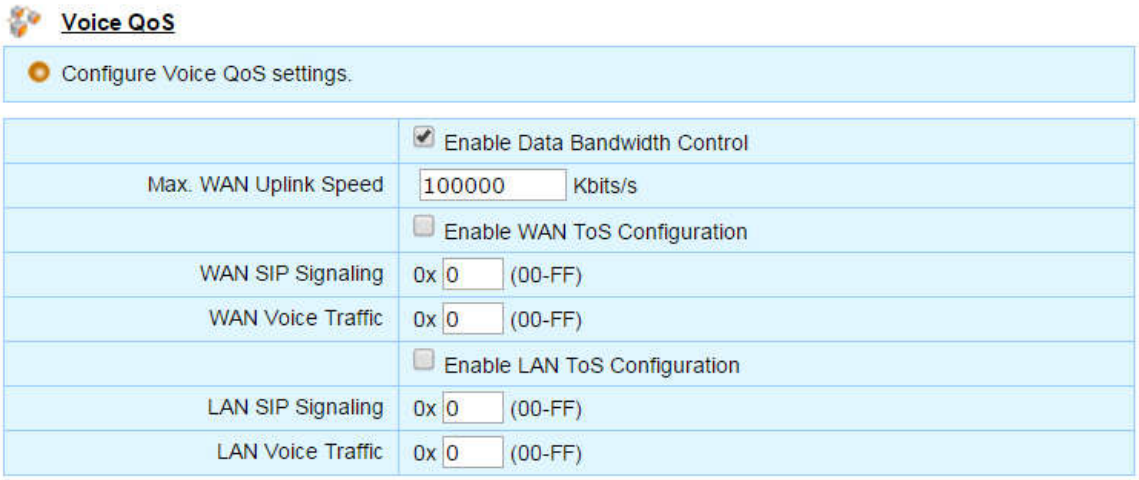


Figure 67 QoS Settings

Voice QoS Settings	Description
Enable Data bandwidth control	Check to enable data bandwidth control.
Max WAN Uplink speed	Enter the bandwidth (bit rates) that are allocated to the ESBC WAN interface for uplink traffic direction.
WAN ToS Configuration LAN ToS configuration	Labeling ToS (Type of Service) to packets toward the designated interface directions. The IP header contains an 8-bit field for QoS control (RFC2474). Values range from 00 to FF (in hex).

3.6.2 Cable Modem Embedded Models: ESBC 95xx and 85xx

The ESBC embedded cable modem models, ESBC95xx and ESBC85xx, support industry leading patent pending “Smart DQoS” technology. It relies on an intelligent edge device (the ESBC) with an embedded DOCSIS cable modem initiating DOCSIS UGS service flow requests based on user or signaling events. With Smart DQoS, the addition of policy servers and complex interactions among various network server components can be avoided.

3.6.2.1 DQoS service flow settings

Navigate to **Telephony** > **SIP TRUNKS** > **DQoS Settings**.

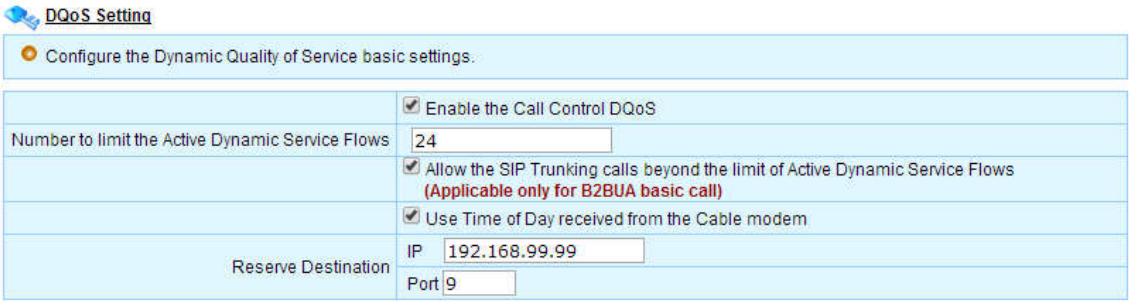


Figure 68 DQoS Call Control – QoS Settings on DOCSIS networks

DQoS Setting	Description
Enable the Call Control DQoS	Enabling this option allows a guaranteed service flow mechanism between the CMTS and ECMM/ESBC for voice connections. Disabling this feature results in all voice calls being processed with best effort.
Number to limit the Active Dynamic Service Flows (DSF)	Enter the number of service flows reserved for the ESBC. The maximum guaranteed value on DOCSIS 3.0 network is 24.
Allow the SIP Trunking Calls beyond the limit of Active Dynamic Service Flows	When the number of calls exceeds the DSF mentioned above, allowing additional calls to be connected with best effort. This configuration is applicable to SIP Trunking voice calls (B2BUA mode), not for hosted voice services (SIP ALG mode).
Use Time of Day received from	The ESBC may use “Time of Day” information from the Cable

the Cable Modem	Modem (and CMTS) for system time. When this feature is enabled, the ESBC SNTP client feature which retrieves time information from the SNTP server will be disabled. (see section 6.1.2 for details).
Reserved Destination	The IP and Port is reserved for intra component communications between the ESBC and embedded ECMM. The network address used here should not conflict with other networks configured or routed by the ESBC. Specify an IPV4 IP address and communication port. Default is 192.168.99.99:9.

4 SIP Trunk Voice Service Configurations

4.1 Routing Calls between the ESBC and SIP Trunk (service provider)

In order for the ESBC to route calls between corporate PBX users and the SIP trunk service for PSTN calls, the items below need to configured properly:

- Trunk Settings: configure the SIP Trunk Server profile(s)
- User Account Configurations: configure SIP UAs on the ESBC
- SIP Trunk Signaling Tuning for Interworking: Fine tune the Trunk SIP Profile

4.1.1 Trunk Settings: SIP Server

A Trunk denotes a subscribed SIP trunk service. Trunk Setting is used to configure the required parameters for the SIP server. The ESBC supports multiple trunk SIP proxy profiles. This flexibility allows the system administrator to configure up to 8 profiles for different subscribed services. To configure a trunk profile, navigate to **Telephony > SIP TRUNKS > Trunks Setting**,

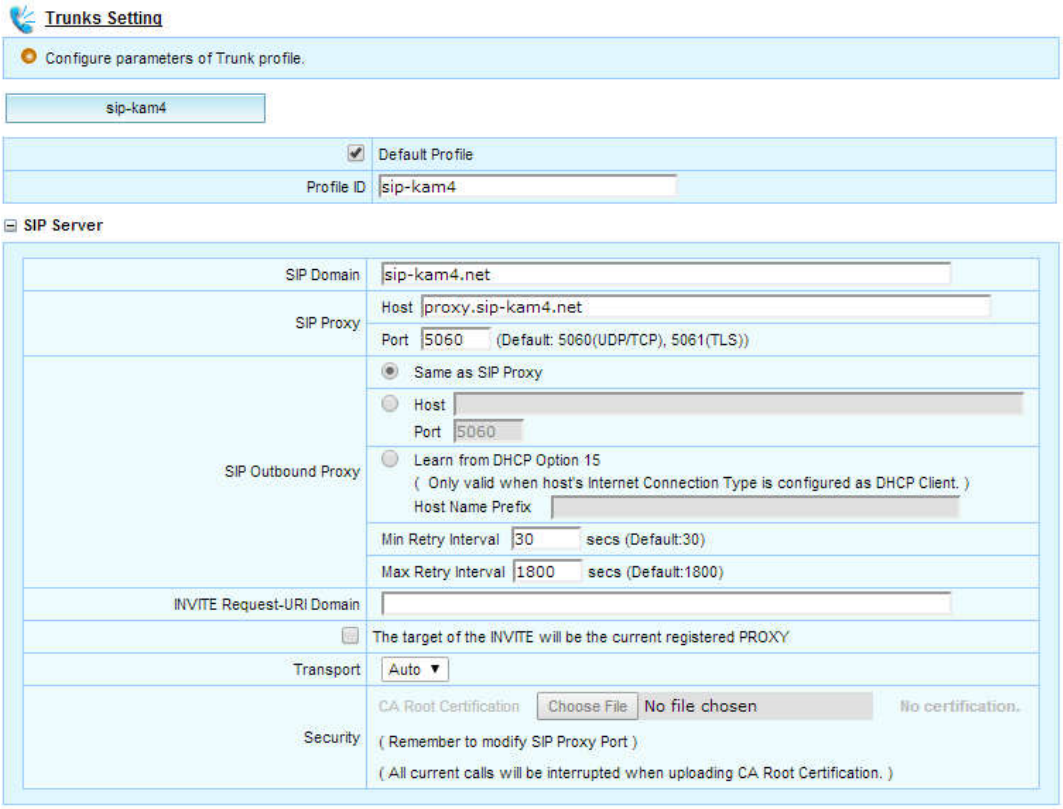


Figure 69. ESBC Trunk Settings

SIP Server	Description
Default Profile	Check the option box if you want to set this profile as the default profile. Any SIP UAs will be associated to this profile automatically.
Profile ID	Enter a unique profile ID for this profile. Enter any name which can be recognized easily.
SIP Domain	The SIP Domain name provided by your service provider for service query purposes.
SIP Proxy	The IP address or FQDN of the target SIP server (or SBC) which processes SIP requests/messages sent from the ESBC.
Port	The default SIP communication port is 5060. Change to the port number which the SIP trunk service provider specifies.
SIP Outbound Proxy	The IP address or FQDN of the target SBC (or SIP server) to which the ESBC sends SIP messages/requests. The ESBC supports DHCP Option 15 to obtain a connection-specific DNS domain suffix. (DHCP Option is applicable only when the ESBC's internet connection is configured as a DHCP client.)
INVITE Request-URI Domain	Enter the SIP domain name for the Request-URI for INVITE messages. (If it is different from the SIP Domain name of SIP REGISTER messages.)
Min/Max Retry Interval	Registration fail retry timers. When the first REGISTER attempt fails, the ESBC attempts the next REGISTER with a back-off mechanism which defines the "Min" and "Max" timers.
Transport	Specify the transport protocol for SIP signals interconnecting the ESBC and the SIP server. • Auto: the default configuration. The ESBC uses the protocol after negotiating with the SIP server. • TCP: use TCP to transport SIP signals only. • UDP: use UDP to transport SIP signals only. • TLS: use secured TLS tunnel to transport SIP signals. Be sure to set the SIP Port number to 5061.
Security- CA Root Certification	When TLS secured connection is used, it is possible to load the 'CA root certificate' issued by your service provider, and the ESBC then authenticates the Server to ensure secured and trusted connections.

4.1.2 Trunk Setting: Sip server redundancy

When the service provider offers a SIP server redundancy feature to ensure high availability of voice services, the ESBC supports real time switch over to a redundant sip server and fail back.

The ESBC obtains the list of redundant SIP servers by dynamic or static methods, and continuously monitors the availability of all sip servers from its list. When the ESBC detects that the primary server has gone down, the ESBC switches to the next reachable server. If none of the sip servers are reachable, the ESBC process all calls as internal calls. When the primary server comes back into service after the ESBC switches to the backup server, the ESBC will automatically switch back to primary server with no interruption of on-going calls.

See “the ESBC SIP Server Redundancy Application Notes” for details.

To configure SIP Redundancy, navigate to **Telephony > SIP TRUNKS> Trunks Setting**.

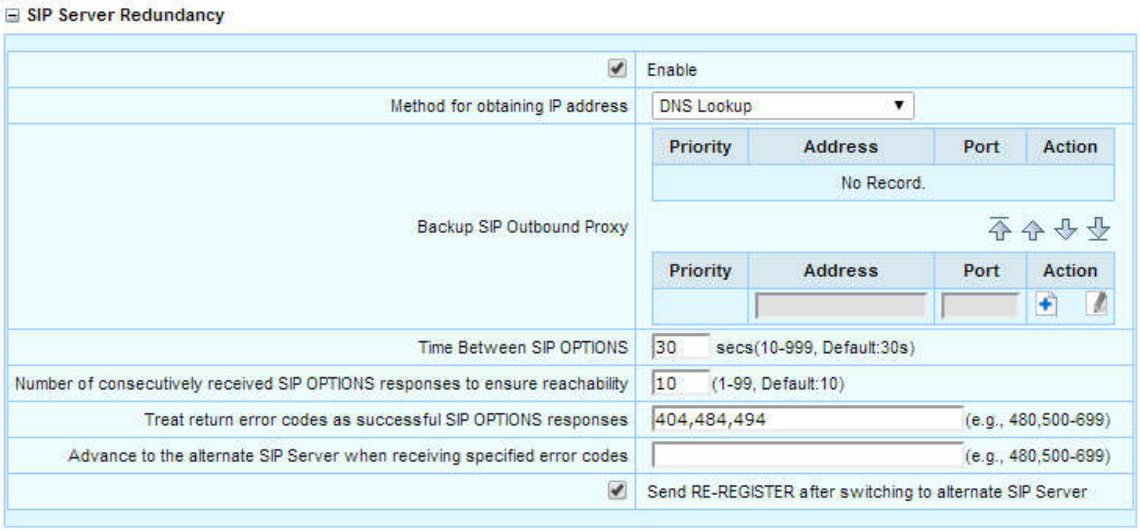


Figure 70. Configuring sip server redundancy features

SIP Server Redundancy	Description
Method for obtaining IP address	Choose “DNS lookup” for dynamic method; or “Input Backup Outbound Proxy” for “static method.”
Backup SIP Outbound Proxy	Applicable to “static method” only.
Time Between SIP OPTIONS	The interval for the ESBC to send out “SIP OPTIONS” ping messages to all SIP servers in the server list.
Number of consecutively received SIP Options	The threshold value for the ESBC to determine the availability of sip servers.
Treat return error codes as successful SIP OPTIONS	SIP servers may reply with a sip response error code when receiving SIP OPTIONS messages from the ESBC. Configure those

responses	codes which are treated as successful SIP OPTIONS responses.
Advance to alternate SIP Server when receiving specified error codes	When the ESBC receives the configured sip response error code(s) from the server, the ESBC should treat this currently connected server as not available for service and move to the next reachable sip server.
Send RE-REGISTER after switching to the alternate server	Enable this option if the databases of redundant sip servers are not synchronized. If sip servers implement HA (high availability) features, it is possible to disable this feature to save processing bandwidth and improve switching efficiency.

4.1.2.1 Dynamic Query for Redundant SIP Servers

When “DNS Lookup” is chosen from the menu “Method for obtaining IP address” (see Figure 70), the ESBC obtains the list of redundant sip servers by doing a DNS SRV/NAPTR/A Record lookup with the configured “SIP Outbound Proxy” (see Figure 69). If both “SIP Outbound Proxy” and “SIP Proxy” are not configured, the SIP domain setting is used. Optionally, the domain may be learnt from a DHCP Option 15 response.

To view the queried server, navigate to **Telephony** > **TOOLS** > **Monitor** > **SIP Server Redundancy**

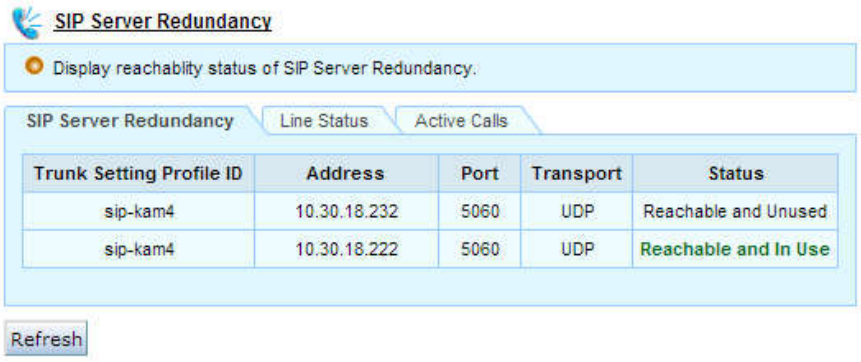


Figure 71. Redundant sip servers: queried results

4.1.2.2 Static Input for Redundant SIP Servers

When “Input Backup Outbound Proxy” is chosen from the menu “Method for obtaining IP address” (see Figure 70), the ESBC obtains redundant sip servers using user input records. <Arrow> keys are used to adjust their priorities.

4.1.3 Trunk Setting: Codec Filter

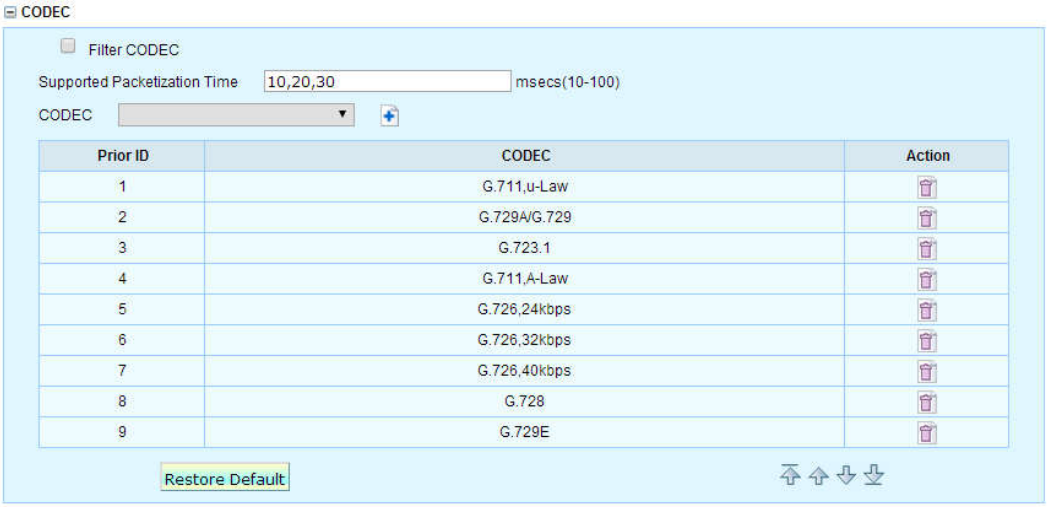


Figure 72. Codec Filter Table

Codec	Description
Filter Codec	Enable this to filter and only use selected CODECs in SIP/SDP messages. When the ESBC composes SIP messages to the SIP Proxy or to the SIP PBX, it will only use CODECs from the list. To change the priority level of CODECs, select the desired CODEC name and click the up or down arrow. To remove a CODEC, click the <Delete> button. To restore to the default, click the <Restore Default> button.
Supported Packetization Time	Specify the supported ptime values in SDP media attribute descriptions.

Note that the selection of codec filters takes higher priority than the extended codecs in the transcoding profile (see section 4.8).

4.2 Adding and Configuring User Accounts on the ESBC

4.2.1 SIP UA Setting

To configure User Accounts (UAs), navigate to **Telephony > SIP UA Setting > SIP User Accounts.** Click the <Batch Config> button to configure SIP User Accounts in bulk mode, or click the <Setting> icon to configure User Accounts individually.

To Search user accounts, select the target searching criteria from the "Search" drop-down menu, and then click the <Search> button. The public identities are user accounts which the ESBC uses to register to the SIP server. Public identities of a subscribed sip trunk service include

- One main public identity
- One or more alternate public identities

Nominating a main public identity is not mandatory for an ESBC configuration. However, if implicit registration is required by the service provider, it is necessary to nominate the main public identity and configure it as the registration agent.

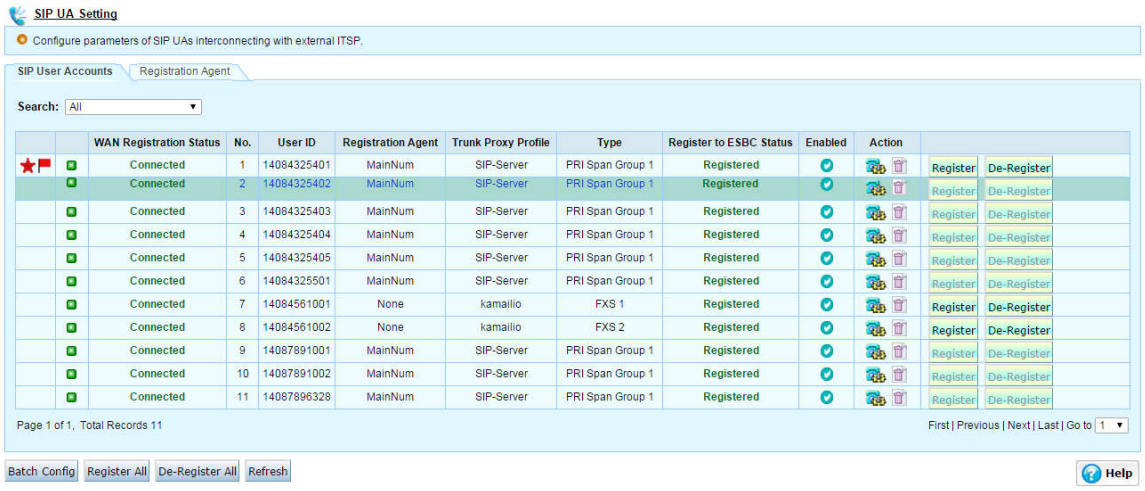


Figure 73. SIP UA Registration Status

SIP User Accounts	Description
Main Public Identity	Also known as pilot number, or main trunk number. This user account is selected/configured as a registration agent for implicit registration. The Main Public Identity registers to the SIP server on behalf all other alternate ESBC user accounts (of the same subscribed sip trunk service).
Default Route	Default route. When multiple user accounts (DIDs) have the same destination (e.g., the same connected TDM-PBX or SIP-PBX),

	the default route account can route calls on behalf of all other user accounts associated with the same destination. The ESBC directs all inbound calls whose called number is not configured on the ESBC database to the destination of the Default Route.
Status	Connected. Successful REGISTER. Not Connected. Authentication Failure. Credential information is not correct. Registration Error. Account disabled on the ESBC Static Registration (Static Operation Mode)
No.	The nth SIP Trunk user account. This number is used for identifying the user account for provisioning tags.
User ID	SIP account user name, usually the DID telephony number.
Registration Agent	The assigned Registration Agent for this trunk (used with IMS implicit registration settings).
Trunk Proxy Profile	Associate SIP UAs to a SIP Trunk Proxy Profile described in Section 4.1.1.
Type	The target type of this selected UA. There are three types of targets: PRI, FXS and SIP.
Enabled	Status of this SIP UA
Action: Setting	Configure parameters of this SIP account.
Action: Delete	Delete this SIP account.
Batch Config	Add/Configure SIP account(s) in bulk mode.
Refresh	Pages refresh automatically based on a time interval. This time interval can be set on the "Auto Refresh" page under "System". Click "Refresh" and the page will be refreshed immediately.
Register De-Register	Click this button to Register or De-Register SIP UAs with the service provider sip server.
Register All De-Register All	Click this button to perform Registration/De-Registration for all SIP UAs configured in the ESBC database.

4.2.1.1 Public identity: Batch Add

Click the <Batch Config> button to add/configure user accounts in bulk mode.

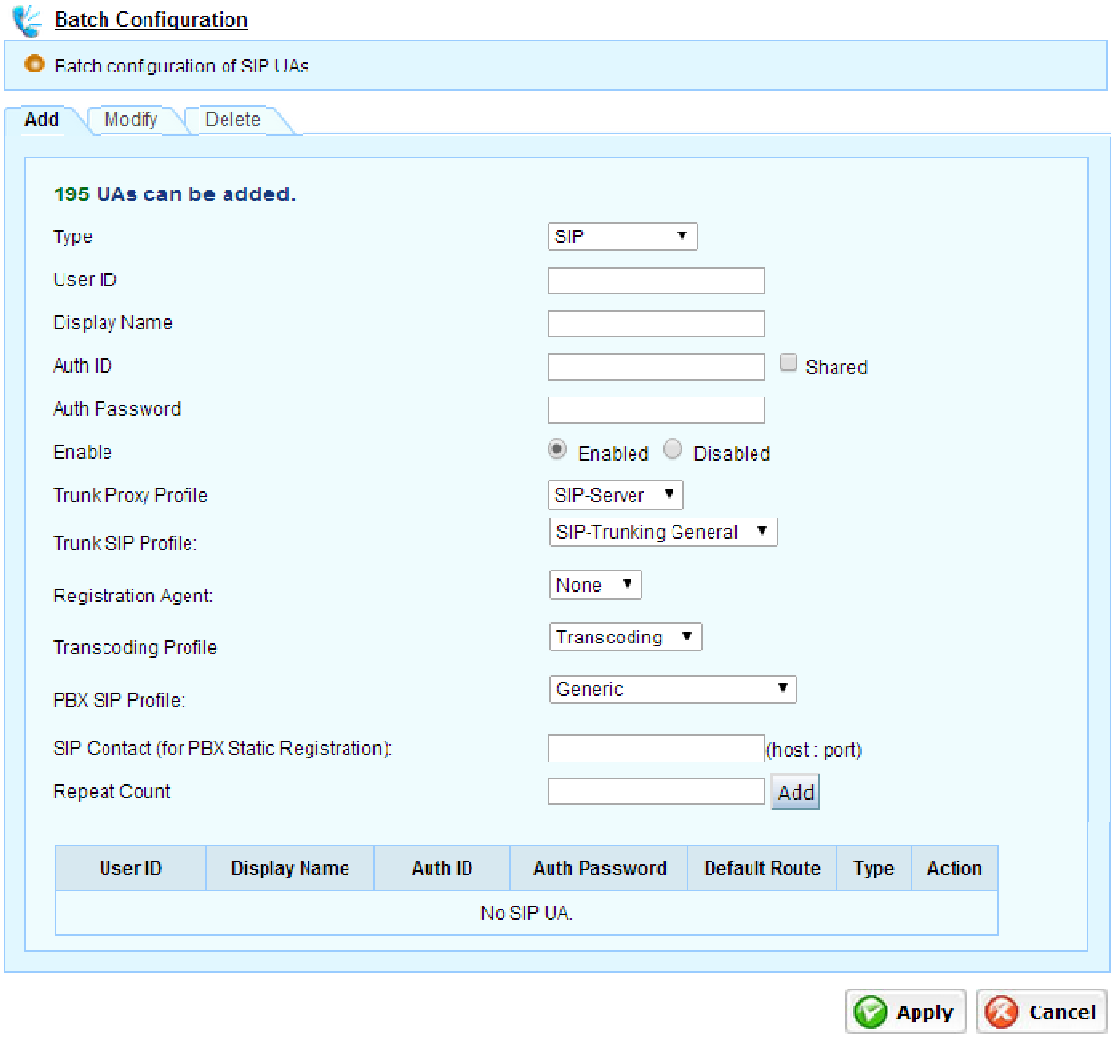


Figure 74. SIP UA Batch Configuration: Add

Enter the User ID (usually the DID number). In the Repeat Count field, enter the number of UAs you would like to add to the system. The User IDs will be generated as consecutive numbers.

Add User Accounts	Description
Type	Select appropriate attribute for the user agents to be configured. The available options are as follows: • SIP: applicable to sip devices or sip-pbx connecting to the ESBC "NAT and Voice" LAN interface. • PRI (Group n): applicable to a TDM-PBX • FXS 1~4: applicable to one of the ESBC's four FXS ports
User ID	A SIP user account to REGISTER with the SIP server. It can be a DID (TN) number or a user name.

Display Name	The caller name shown on the callee's phone device for outbound calls to the PSTN.
Auth ID	SIP ID for authentication purposes.
Shared	When this Auth ID is shared among all SIP UAs in the same batch config operation, click this check box. Otherwise, the Auth ID will increment by 1 with every new account created in this batch operation.
Auth Password	SIP Authentication password for the UAs to register to the service provider network. The Auth password will be applied to all SIP UAs created in this batch operation.
Enable	Enable or disable the SIP UAs created in this batch operation.
Trunk Proxy Profile	The Trunk Proxy Profile used for SIP UAs created in this batch operation.
Trunk SIP Profile	The Trunk SIP profile used for SIP UAs created in this batch operation. (See section 4.2.3)
Registration Agent	Registration Agent of this trunk, usually the main identity number of this trunk. (See section 4.2.2)
Transcoding profile	The profile name for the transcoding parameters applied to SIP UAs created in this batch operation. (See section 4.8) This feature is applicable to the ESBC9378 series models.
PBX SIP Profile	Choose the SIP PBX profile applied to SIP UAs created in this batch operation. If the target SIP PBX name is not in the list, choose "Generic." (See section 4.4)
SIP Contact (for PBX Static Registration)	The IP:port information for SIP UAs to which the ESBC relays sip messages. This parameter is used when the sip client uses static registration mode to connect to the ESBC. This parameter is not applicable to REGISTER operation mode.
Repeat Count	Enter the number of SIP UAs to be created in this batch operation. Note that when the User ID is in TN format, the User ID will increment by 1 with every new account created in this batch operation. The repeat count is not applicable to a text User ID.

4.2.1.2 Public identity: Batch Modify/Delete

Click the <Modify> or <Delete> tab, selecting the target User Accounts and the desired operations. Click the <Apply> button to complete. Please refer to section 4.2.1.1 for descriptions of parameters.

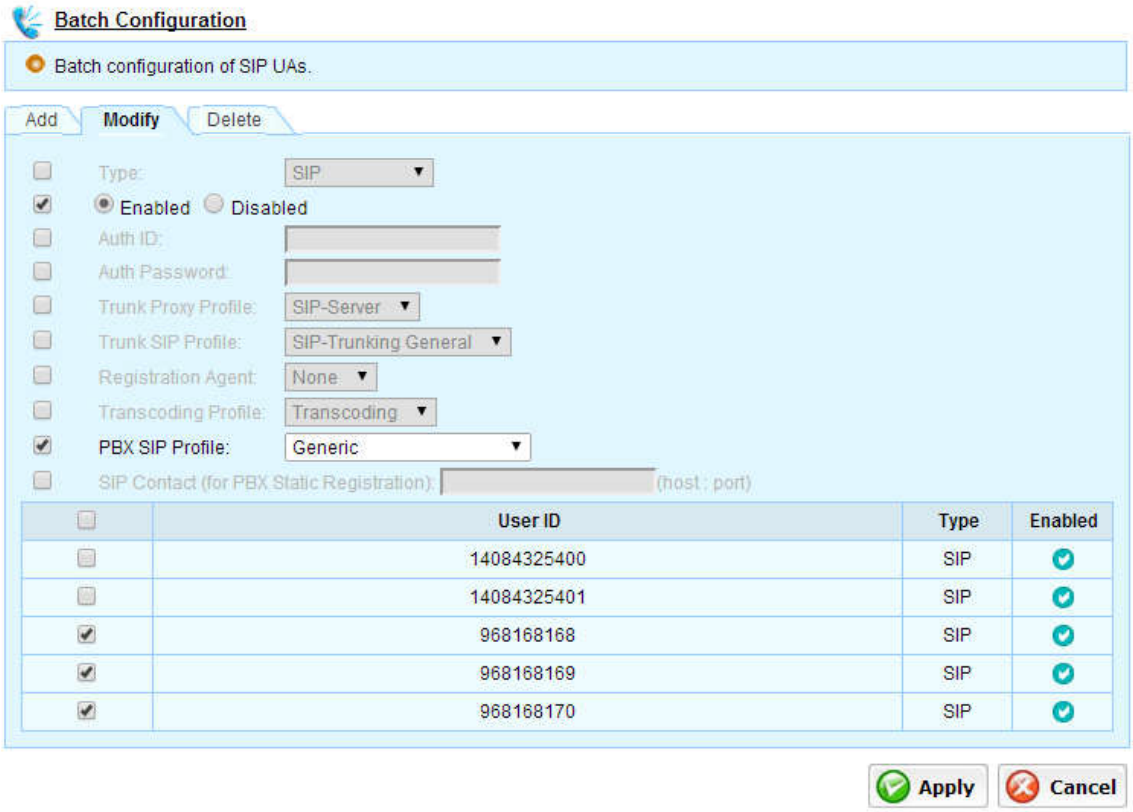


Figure 75. SIP UA Batch Configuration: Modify

4.2.1.3 Public identity: Individual Settings and Authentication

To configure a selected public identity (SIP UA) individually, click the <Action> button in Figure 73.

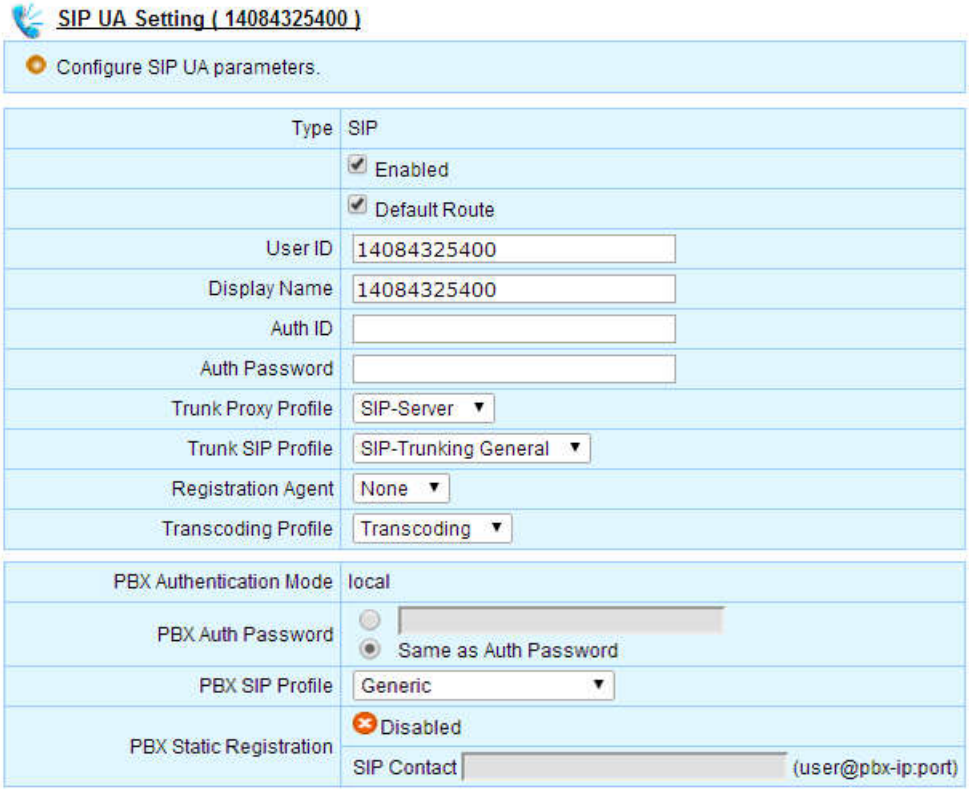


Figure 76. Individual SIP UA Settings

Most of the parameters should already be entered in a Batch-Add operation (see section 4.2.1.1). This page allows you to configure/update parameters for an individual account.

Three options are available for the PBX Authentication Mode, they are “None”, “Local” and “RADIUS”. The configurations apply to SIP PBX, SIP clients and analog FXS ports. Please see section 4.4.2 for details.

If “Local” mode is chosen, the Auth password configuration is described as follows.

Local Authentication	Description
Same as Auth Password	The ESBC applies the same “Auth Password” of the SIP Trunk to authenticate the sip request attempts from the SIP PBX (or SIP clients). The Auth Password has to be configured on the SIP PBX accordingly.
	If a different Auth Password is needed, choose the other option and enter the password accordingly.

4.2.2 Implicit registration: Registration Agent

Implicit registration is completed by the Registration Agent (RA) which is usually the pilot number. The RA registers to the SIP server on behalf all user accounts (alternate public identities or DIDs) of the subscribed service. To configure an RA, navigate to **Telephony** > **SIP ACCOUNTS** > **SIP UA Setting** > **Registration Agent**. Click the <Add> button to add an RA.

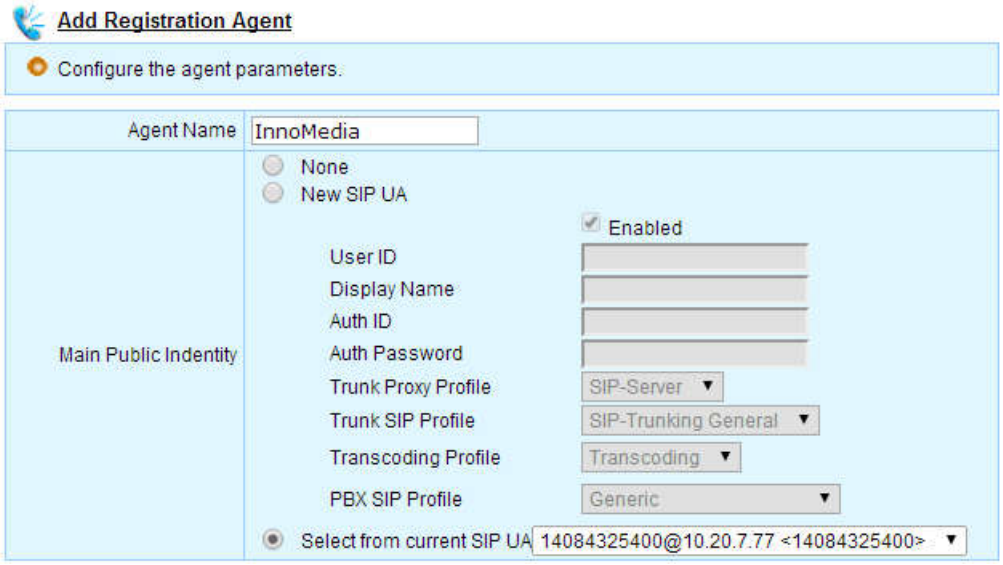


Figure 77. Add Registration Agent

Add Registration Agent	Description
Agent Name	Name of the registration Agent
None	Select none if not using implicit registration.
New SIP UA	If the RA account is not configured on the ESBC, choose this option to add a new user account. Please refer to 4.2.1.1 for parameter descriptions.
Select from current SIP UA	For the use of implicit registration, select an existing SIP UA from the drop down menu.

4.2.2.1 Bulk Assigning

The ESBC's web console provides the ability to assign Bulk SIP UAs to the FXS ports, SIP PBX, or TDM PBX. The Bulk Assigning feature is convenient when assigning numbers to different types of clients is needed.

Navigate to **Telephony** > **SIP ACCOUNTS** > **Bulk Assigning**.

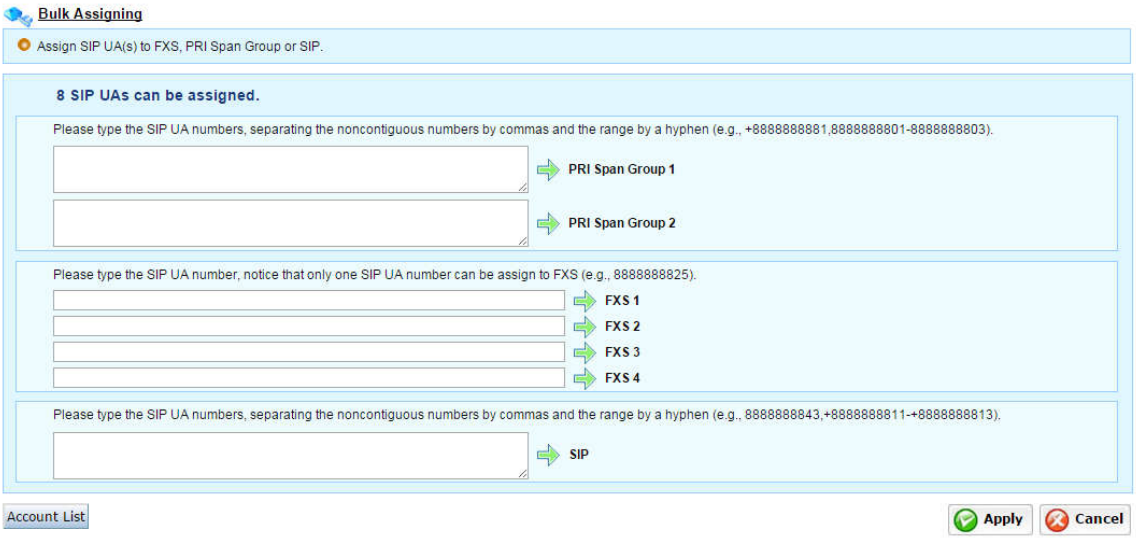


Figure 78. Bulk Assigning VoIP numbers to the ESBC clients

4.2.3 SIP Trunk Parameter Configurations

The ESBC normalizes sip signals from the enterprise network to interwork with servers in the service provider network. To configure sip signals interfacing to the server side, navigate to **Telephony** > **SIP TRUNKS** > **Trunk SIP Profile**.

Two SIP profiles are provided by the ESBC for general SIP interworking purposes, SIPConnect 1.1, and SIP-Trunk General. To create a new profile, click the <Add> button, or the <Setting> icon to edit an existing profile.

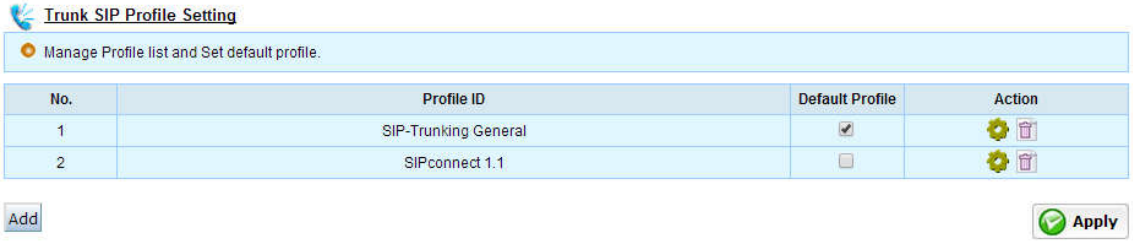


Figure 79. Trunk SIP Profile Settings

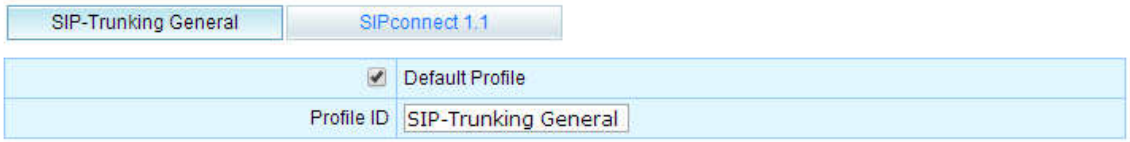


Figure 80. Editing a Trunk SIP Profile

Parameters	Description
Default Profile	Enable this option to assign all SIP user accounts to this profile if they are not configured otherwise.
Profile ID	Enter a unique name for this profile

4.2.3.1 SIP Profile Configuration: SIP Parameters

Click the 'Setting' icon on an existing profile.

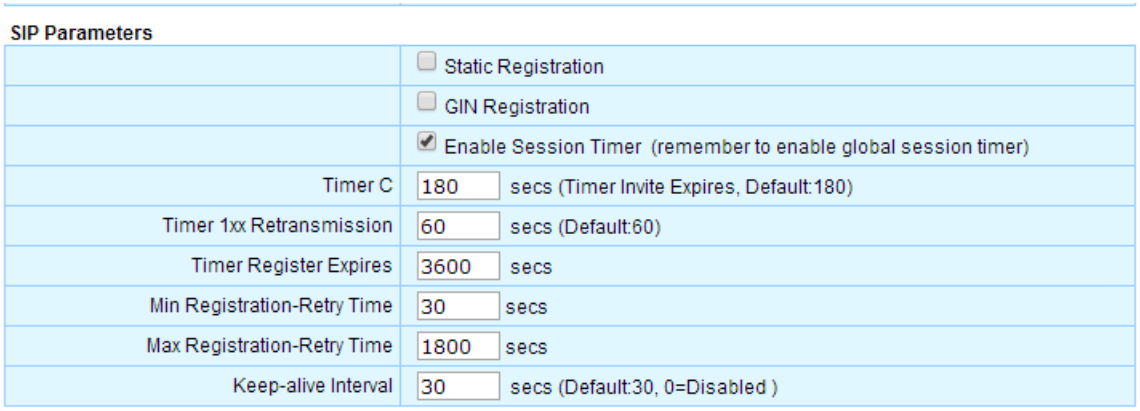


Figure 81 Configuring a Trunk SIP Profile—SIP Parameters

SIP Parameters	Description
Static Registration	If selected, the service provider network treats the ESBC as a peer network, but not a registering device.
GIN Registration	Globally Identifiable Number (GIN) Registration (RFC 6140) is used widely for implicit registrations. The ESBC supports GIN to construct and distribute a URI that can be used universally. This mechanism requires the RA to perform implicit registration to the service provider's network (see section 4.2.2).
Enable Session Timer	The SIP session timer specifies a keep-alive mechanism for SIP sessions, which limits the time period over which a stateful proxy must maintain state information without a refresh re-INVITE. (The Session Timer parameter in the SIP Parameter Page (See section 4.5) needs to be enabled in order for this setting to take effect.
Timer C, Timer 1xx Retransmission, Timer Register Expires, Min Registration-Retry Time, Max-Registration Retry Time	Standard SIP timers defined in RFC 3261
Keep-alive Interval	Specifies the interval for sending keep-alive messages for active SIP sessions

4.2.3.2 SIP Profile Configuration: Interoperability

See the application note: The ESBC Caller ID (TN) screening mechanism.

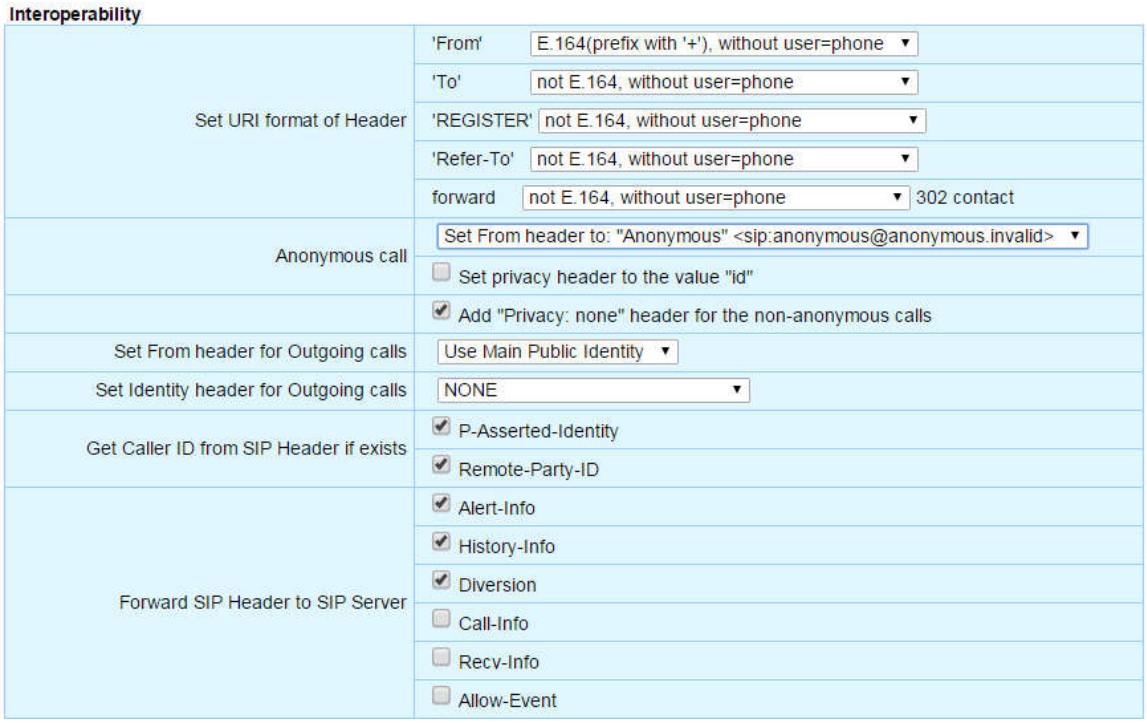


Figure 82. Trunk SIP Profile Configuration: Interoperability -1

SIP Parameters	Description
Set URI format of sip headers: TO, FROM, REGISTER, Refer-To, forward.	Depending on the sip server configuration, the ESBC provides four combinations to generate sip headers: not E.164, without user=phone not E.164, with user=phone E.164 (prefixed with '+'), without user=phone E.164 (prefixed with '+'), with user=phone Examples: sip:15616261234@example.com sip:15616261234@example.com; user=phone sip: +15616261234@example.com sip: +15616261234@example.com; user=phone
Anonymous call	Selection of **From** Header format for anonymous calls. If outbound

	calls from the PBX specify the blocking of CID, the ESBC offers various From Header formats that may be sent to the SIP Server. Privacy Header. The presence of certain values of the privacy type in a Privacy header field indicates that the user would like to keep their identity private with respect to SIP entities outside the Trust Domain with which the user is authenticated. • Set privacy header to the value 'id': **Privacy: id**. Identification must be hidden (defined in RFC3325) • Set privacy header to the value 'none': **Privacy: none** for non-anonymous calls. Identity is not hidden. Note: Outbound calls from the PBX that specify the blocking of CID are specified as follows: (1) SIP PBX: the From header specifies the blocking of CID. (2) PRI PBX: Presentation flag set to "restricted" in SETUP messages.
Set From header for outgoing calls	The From header can be used to transport caller information, such as caller ID and name displayed at the called party side. Four options are available: • Use Alternate Identity: Use the individual AOR configured in the ESBC database. • Use Main Public Identity: Use the pilot number (registration agent account) configured in the ESBC database. • Use the original caller: Use the information obtained from the PBX user. • Use a configured phone number: Enter a preferred caller ID string to override the setting.
Set Identity header for outgoing calls	Depending on the sip server privacy configuration, the ESBC may add one of the following three headers as the caller identity header (or none). • P-Asserted-Identity: defined in RFC 3325. This sip header is used among trusted sip entities to carry the identification of the user sending a sip message as it was verified by authentication. • P-Preferred-Identity: defined in RFC 3325. This sip header is used from a user agent to a trusted proxy to carry the identification the user sending the sip message wishes to be used for the P-Asserted-Header field value that the trusted entity will insert.

	• Remote-Party-ID: This sip header provides information about the remote party. See Appendix 10.2 for SIP Remote-Party-ID header parameter mapping with the PRI SETUP message.
Get Caller ID from SIP Header if exists	Choose from one among the following three options to transport information on Caller ID to the SIP PBX • P-Asserted-Identity, • Remote-Party-ID
Forward SIP Header to SIP Server	The ESBC allows the forwarding of the following SIP headers sent from the LAN SIP UAs to the Service Provider's SIP server. Alert-Info | History-Info | Diversion | Call-Info | Recv-Info | Allow-Event

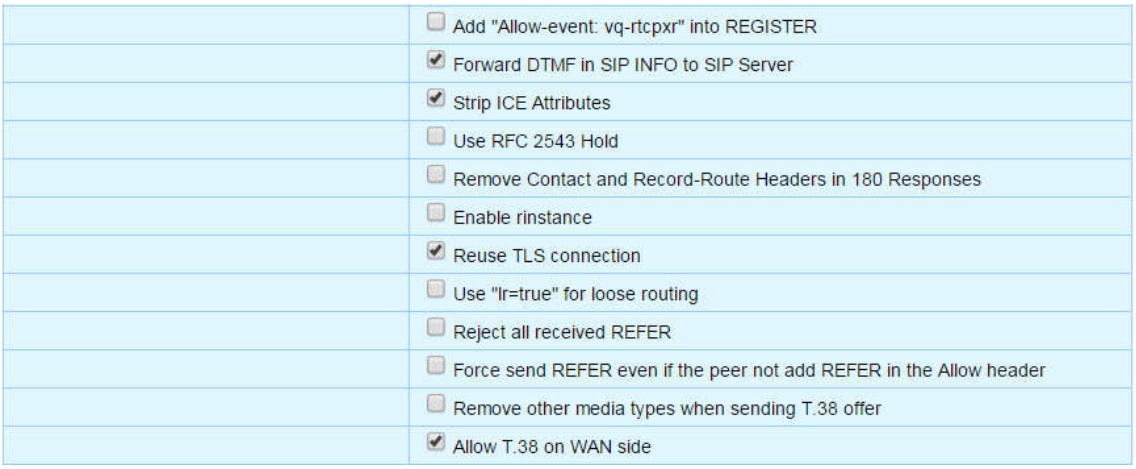


Figure 83. Trunk SIP Profile Configuration: Interoperability -2

SIP Parameters	Description
Add "Allow-event: vq-rtcpxr" into REGISTER	To support VQM (voice quality measurement) requirements of RFC6035.
Forward DTMF in SIP INFO to SIP server	When this feature is enabled, the ESBC forwards SIP INFO messages if the registered SIP UAs send DTMF tones with SIP INFO method.
Strip ICE Attribute	ICE attributes are used for NAT traversal purposes. Enable this feature to allow the ESBC to strip all ICE related parameters in SDP messages for messages sent to the SIP server. ICE related attributes in SDP include • a=candidate(.*) • a=ice(.*)
Use RFC2543 Hold	RFC2543 is obsoleted by RFC3261. For backward compatibility, the ESBC can allow the use of "c" destination addresses set to all

	zeroes (0.0.0.0) for call hold operations.
Remove Contact and Record-Route Headers in 180 responses	For SIP Server interoperability purposes, check this item when necessary.
Enable rinstance	The parameter “rinstance” (for the Contact header in REGISTER) is used when sip devices support multiple lines. It is not defined in an RFC but is an opaque URI parameter used to differentiate different lines. Checking this item will allow the ESBC to add the “rinstance” parameter to support remote SIP device with multiple line features that support this parameter.
Reuse TLS connection	Enable this feature to allow the ESBC to use the actual source port of a TLS connection in addition to the default port 5061.
Use “lr=true” for loose routing	Depending on the sip server configuration, the ESBC adds “lr=true” parameter for loose routing. In loose routing, as specified in RFC3261, the Request-URI always contains the URI of the destination user agent. As opposed to “strict routing,” where the request-URI always contains the URI of the next hop.
Reject all received REFER	When enabled, the ESBC rejects all REFER messages for Call transfer operations.
Force send REFER even if the peer not add REFER in the Allow header	The ESBC adds “REFER” to the “Allow” header.
Remove other media types when sending T.38 offer	When this parameter is selected, "m=" lines other than t38 are all removed from SDP messages.
Allow T.38 on WAN side	If this item is disabled, the ESBC rejects all T.38 transmission attempts from remote devices.

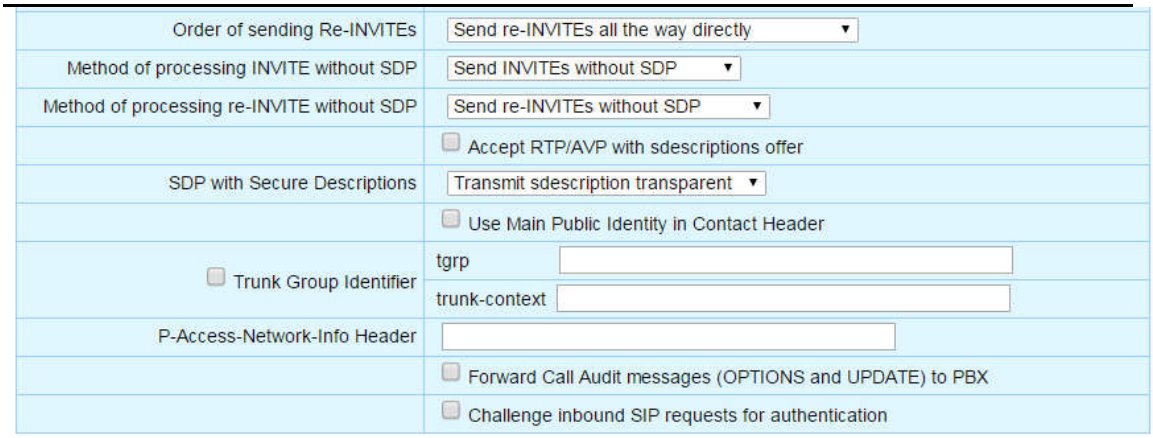


Figure 84. Trunk SIP Profile Configuration: Interoperability -3

SIP Parameters	Description
Order of sending Re-INVITEs	Some particular SIP UAs do not proceed with new sessions with reINVITEs for the current dialog unless the current session is concluded with a response code, such as 200 OK. Leave the default setting unless necessary. Scenario: A – ESBC – B. After call setup, the ESBC receives reINVITE from A. • Send re-INVITEs all the way directly. The ESBC sends reINVITE to B, and after receives 200 OK from B, then the ESBC sends 200 OK to A. • Send response before forwarding re-INVITEs. The ESBC replies 200 OK to A with old SDP, and then sends reINVITE to B. After the ESBC receiving 200 OK from B, it will not send 200 OK to A.
Method of processing INVITE without SDP	• Scenario: A – ESBC – B. A calls B, the INVITE message has no SDP. • Send INVITEs without SDP. The ESBC sends INVITE to B without SDP. • Send INVITEs with a fake SDP. The ESBC sends INVITE to B with a fake SDP (g.711). After call setup, the ESBC re-negotiates SDP by sending a reINVITE.
Method of processing re-INVITE without SDP	Scenario: A – ESBC – B. A calls B, the INVITE message has no SDP. • Send reINVITEs without SDP. The ESBC sends INVITE to B without SDP. • Sends reINVITEs with the old SDP. The ESBC sends reINVITE to B with old SDP. After call setup, the ESBC re-negotiates SDP by sending a reINVITE.
Accept RTP/AVP with sdescriptions offer	Sdescription is short for security descriptions for media streams. Some clients choose to code them as "RTP/AVP" to make clients accept the SDP as an offer. Select here if the ESBC should accept incoming offers where sdescriptions are presented as "RTP/AVP" offers.
SDP with Secure Description	The RTP/SAVP profile is defined for security services for RTP media and is signaled by use of RTP transport, i.e., SDP media line with the value: *m=RTP/SAVP*, together with the “crypto” SDP attribute. Select the appropriate item according to the SIP server’s capability to process secured RTP streams. Leave the default selection unchanged unless necessary. • Transmit sdescription transparent • Transmit all sdescription in SAVP • Transmit all sdescription in AVP

Use Main Public Identity in Contact Header	Check this box when the Contact header should include the Main Public Identity, i.e., pilot number. By default, it is unchecked, and the ESBC uses the “Alternate Identity” in the Contact header.
Trunk Group Identifier	Identifiers for a trunk group that allow a unique identity to be provisioned and allow unscreened calls from the ESBC. Defined in RFC4904, trunk groups are identified by two parameters: “**tgrp**” and “**trunk-context**”; both parameters must be present in a URI to identify a trunk group. The "trunk-context" parameter imposes a namespace on the trunk group by specifying a domain name. For example, Trunk group in a local number, with a phone-context parameter (line breaks added for readability): tel:5550100;phone-context=+1-630;tgrp=TG-1; trunk-context=example.com
P-Access-Network-Info Header	P-Access-Network-Info header. This header is useful in SIP-based networks that also provide layer 2/layer 3 connectivity through different access technologies. SIP User Agents may use this header to relay information about the access technology to proxies that are providing services.
Forward Call Audit messages to PBX (OPTIONS and UPDATE)	The ESBC may forward call audit messages (OPTIONS and UPDATE) to the SIP PBX, and the ESBC does not autonomously respond 200 OK to the SIP server. If this item is unchecked, the ESBC responds autonomously to related call audit messages on behalf of the SIP-PBX and does not pass on these messages to the PBX. This setting for the trunk sip profile should be consistent with the similar parameter in section 4.4.1.2 for the target SIP-PBX profile.
Challenge inbound SIP requests for authentication	Enables the ESBC to challenge inbound SIP request messages from the service provider network. Refer to section 10.4 “ ESBC SIP Authentication Flow” for detailed description.

4.2.3.3 SIP Profile Configuration: Security

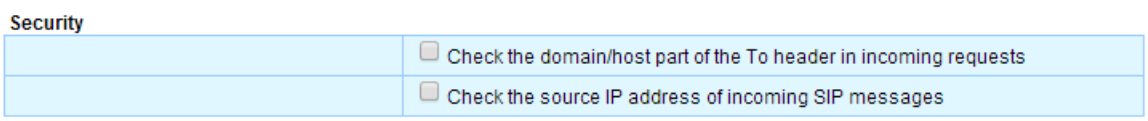


Figure 85. Configuring SIP security features

SIP Parameters	Description

Check the domain/host part of the To header in incoming requests	If the domain/host part of the To header in incoming requests is different to the ESBC configuration for the SIP domain field, the ESBC rejects these incoming SIP messages.
Check the source IP address of incoming SIP messages	If the source IP address of incoming SIP messages is different to that used when registering SIP UA accounts, the ESBC rejects these incoming SIP messages.

4.2.3.4 SIP Profile Configuration: Features

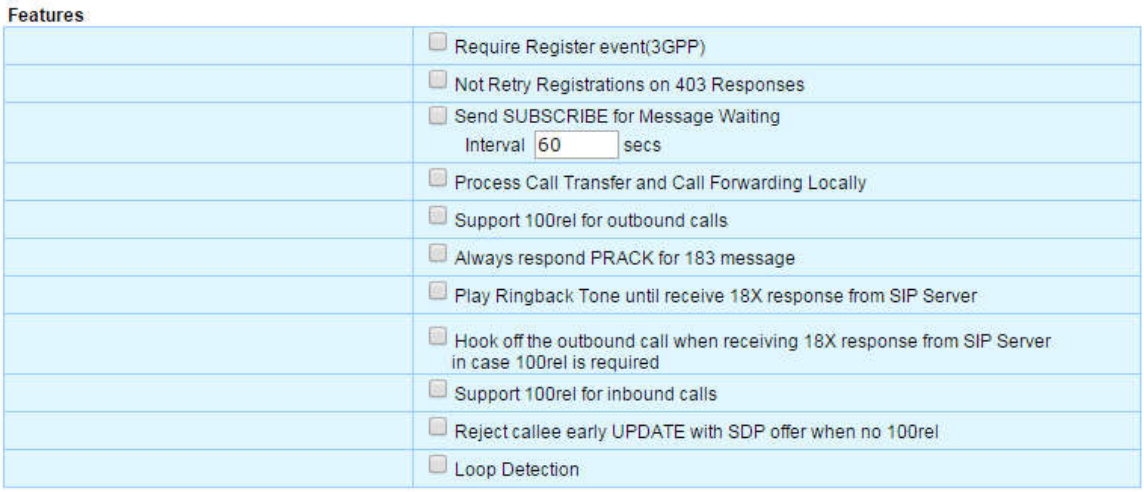


Figure 86. Trunk SIP Profile – Features

SIP Parameters	Description
Require Register event (3GPP)	Defined in RFC 3680. The ESBC subscribes to the registration event for an AOR, resulting in notifications of registration state changes. For example, when the administrator shortens the registration (e.g., when fraud is suspected), the registration server sends a notification to the ESBC which can re-register and re-authenticate.
Not Retry Registration on 403 Responses	The ESBC, upon receiving a response code 403 from the registration server, will not perform any more REGISTER attempts.
Send SUBSCRIBE For Message Waiting Interval	The ESBC sends SUBSCRIBE messages for subscribing to voice mail services (i.e., VMWI) from the server.
Process Call Transfer and Call Forwarding Locally	When this feature is enabled, the ESBC performs requests such as the following: • Process “call transfer” with re-INVITE method instead of sending REFER to the sip server. • Process “call transfer” with re-INVITE method instead of

	sending sip response code 302 (moved temporarily) to the sip server.
Support 100rel for outbound calls	When 100rel is enabled, the ESBC will include the 100rel tag in the Supported header and the PRACK method in the Allow header in outgoing INVITE messages. When the called party sends reliable provisional responses, the ESBC will send a PRACK request to acknowledge the response.
Always respond PRACK for 183 message	The ESBC always responds with PRACK with 100rel tag when receiving 183 responses for outbound calls.
Play ringback tone until receive 18X from SIP Server	When this feature is enabled, the ESBC will start to play ring back tone for outgoing calls when it receives a 18x response from the SIP server. When this feature is disabled, the ESBC plays ringback tone immediately after the INVITE messages are sent out to the LAN SIP devices, and updates the ring back tone after receiving 18x (or others) from the SIP server.
Hook off the outbound call when receiving 18X response from SIP server	When this parameter is set to disabled, the ESBC does not go into talking state when it receives 183, but instead goes into talking state when it gets 200 OK. By default, this parameter is set to disabled.
Support 100rel for inbound calls	The ESBC supports 100rel/PRACK as the UAS. This setting is used for inbound calls (ESBC in UAS mode). If enabled and the server sends INVITE with no “Supported/Require:100rel” to ESBC, ESBC will respond without 100rel; and if server sends INVITE with “Supported/Require:100rel”, ESBC will respond with 100rel.
Reject callee early UPDATE with SDP offer when no 100rel	This item is used for outbound calls. When 100rel/PRACK is not enabled on the ESBC and an UPDATE message is received from the SIP server in early state (non-RFC-compliant SIP server behavior), • If enabled, the ESBC rejects the ‘UPDATE.’ • If disabled, the ESBC sends 200 OK with the old SDP to the SIP server in response to the UPDATE Note: RFC3311 defines that the “UPDATE” method is used to modify the characteristics of a session before the call is answered, i.e., in early state. UPDATE can normally only be sent in the early state when 100rel/PRACK are used.
Loop Detection	Enable this function to prevent from a sip signaling loops from occurring. A loop is a situation where a request that arrives at the ESBC is forwarded, and later arrives back. The ESBC handles loops with the Max-Forwards header to limit the number of hops a request can transit on the way to its destination. The default initial

	value for Max-Forward is 70. If the Max-Forwards value reaches 0 before the request reaches its destination, it will be rejected with a 483 (too many hops) error response.

4.2.4 Analog interface FXS Configuration: FAX and Modem Calls

Once the user accounts have been configured on the ESBC database (see section 4.2.1), you may select accounts (numbers) and assign them to the ESBC FXS ports to which analog devices are connected, such as FAX machines, point of sale station modems, or POTS phones.

Navigate to **Telephony** > **FXS** > **FXS Port Setting**.

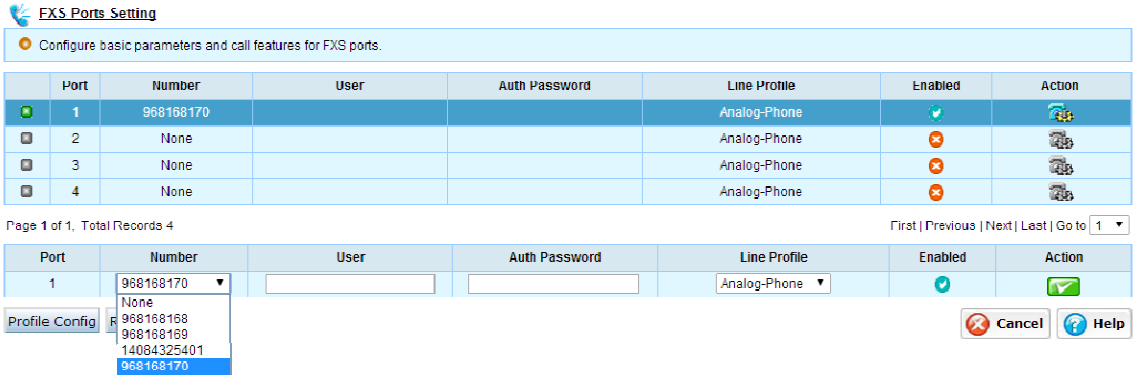


Figure 87. FXS port settings

Click the drop down menu of the Number column, choose a number for this particular FXS port, and enter the User name and Auth Password if needed (see section 4.2.1.3).

Screen Display	Description
Status	Displays the connected status of analog ports Normal H/W Fault Authentication Failed Disabled
Port	The connected port ID, as on the label on the back panel for each port interface.
Number	Selected SIP Trunk User Account.
User	The user name assigned to this port.
Auth Password	Enter the password if needed (see section 4.2.1.3).
Line Profile	The media parameters defined through the <Profile Config> operation (see section 4.2.4.1)
Action	Configure call features for this particular analog port (see section 4.2.4.2)

4.2.4.1 Media Parameter Configuration for Analog Ports

Click <Profile Config> button to configure media parameters for analog ports.

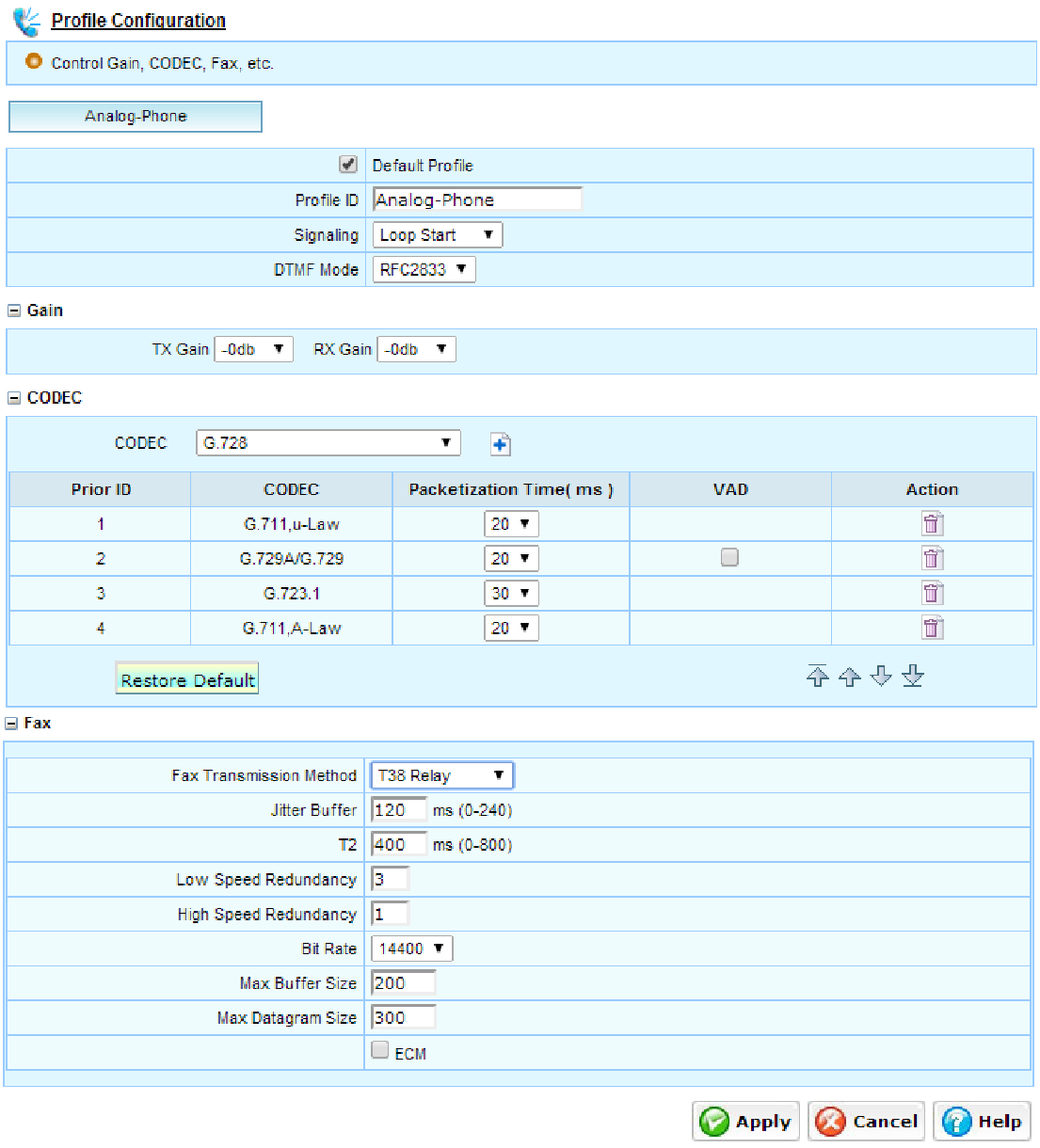


Figure 88. Configuring media parameters for analog ports

FXS port media parameters	Description
Profile ID	Name of this profile
Signaling	"Loop Start" and "Ground Start" are supported. Choose the correct signaling type according to the analog equipment to which

	the FXS port is connected.
DTMF mode	"RFC2833" out-of-band and "In-band" methods are both supported. If RFC 2833 is chosen, the ESBC supports auto negotiation with the SIP servers (or SIP devices) at the SIP Trunk side. Choose the correct mode according to the specification of the service provider, or leave the default settings (2833) unchanged.
Gain	Controls telephone speaker and earpiece volume
CODEC	Control SIP media negotiation. To change the priority level of a CODEC, highlight it and click the arrow keys to adjust its position. To remove a CODEC, click the <Delete> button under the Action column.
Fax	The ESBC supports both "T.38 Relay" and Pass_Through modes for fax transmission over an IP network. Parameters for Pass_Through FAX calls transmitted in Pass_Through mode are treated as voice calls and transmitted using the G.711 codec only. Depending on the region where the ESBC is deployed, choose either G.711 u-Law, or G.711 A-Law for transmitting FAX calls. Parameters for T.38 Relay (Note: If the peer gateway does not support T.38 relay, FAX calls will fall back to Pass_Through mode) • Jitter Buffer. Default value is 120 ms. Do not change the default value unless necessary. • T2. Timeout timer for receiving packets. Default value is 400 ms. Do not change the default value unless necessary. • Low Speed Redundancy. Number of redundant T.38 fax packets to be sent for the low speed V.21-based T.30 fax machine protocol. Default value is 3. Do not change the default value unless necessary. • High Speed Redundancy. Number of redundant T.38 fax packets to be sent for high-speed V.17, V.27, and V.29 fax machine image data. Default value is 1. Do not change the default value unless necessary. NOTE: Increasing the High Speed Redundancy parameter may cause a significant impact in the network bandwidth consumed by fax calls. • Bit Rate. Choose a fax transmission speed to be attempted: 2400, 4800, 9600, or 14400. By choosing 14400, the ESBC can

	automatically adjust/lower the speed during the transmission training process. The ESBC supports G3 Fax. • Max Buffer Size. This option indicates the maximum number of octets that can be stored on the remote device before an overflow condition occurs. Default value is 200. Do not change the default settings unless necessary. • Max Datagram Size. Maximum datagram size. This option indicates the maximum size of a UDPTL packet that can be accepted by the remote device. Default value is 300. Do not change the default settings unless necessary. • ECM. Enable Error Correction Mode (ECM) for the gateway.

4.2.4.2 Call Feature Configuration for Analog Ports

The ESBC analog ports can be to provide voice communications to enterprise users for backup purposes. Click the <Action> icon in Figure 87 and configure the parameters on this page to enable or disable certain call features for the selected port. Choose the <Feature Setting> tag.

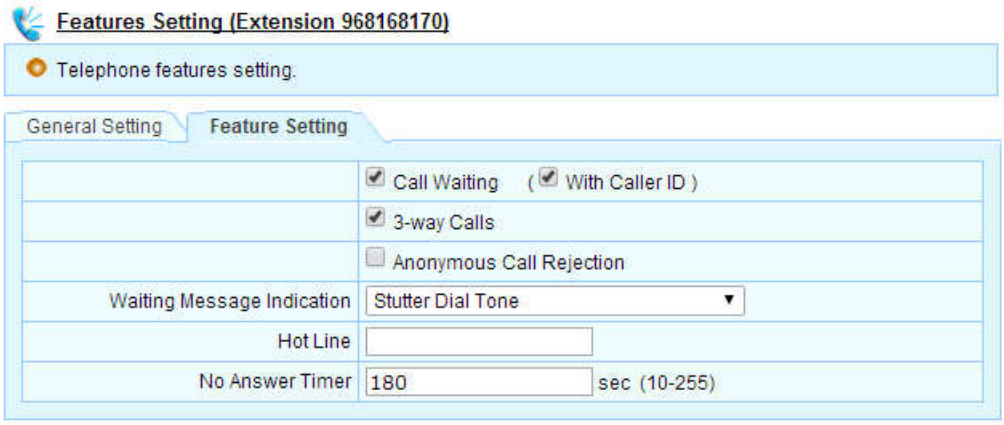


Figure 89. Call features for analog ports

Analog port call features	Description
Call Waiting	If the line is busy, inform the user of an incoming call. Display caller ID of the incoming caller.
3-way Calls	The 3-way calls feature allows this analog port to mix media streams for two different parties.
Anonymous Call Rejection	Block calls from parties who have their caller ID blocked.
Waiting Message Indication	Select one of various types of VMWI signals to use as notification

	of new voicemails.
Hot Line	With the hotline number entered, the telephone automatically connects to this destination number as soon as the user lifts the handset.
No Answer Timer	No answer timeout for incoming call.

4.3 Verifying Calls between the ESBC and SIP Trunk: Test Agent

4.3.1 The Test Agent Setting

Click the <Setting tab> to configure the Test Agent parameters. The ESBC Test Agent provides two options to use a SIP UA to act as the Test Agent for test calls: Use a standalone SIP UA, or Use current SIP UA.

- Option 1**. Use a standard SIP UA**. The ESBC uses a SIP UA account which is dedicated to perform test call operations.

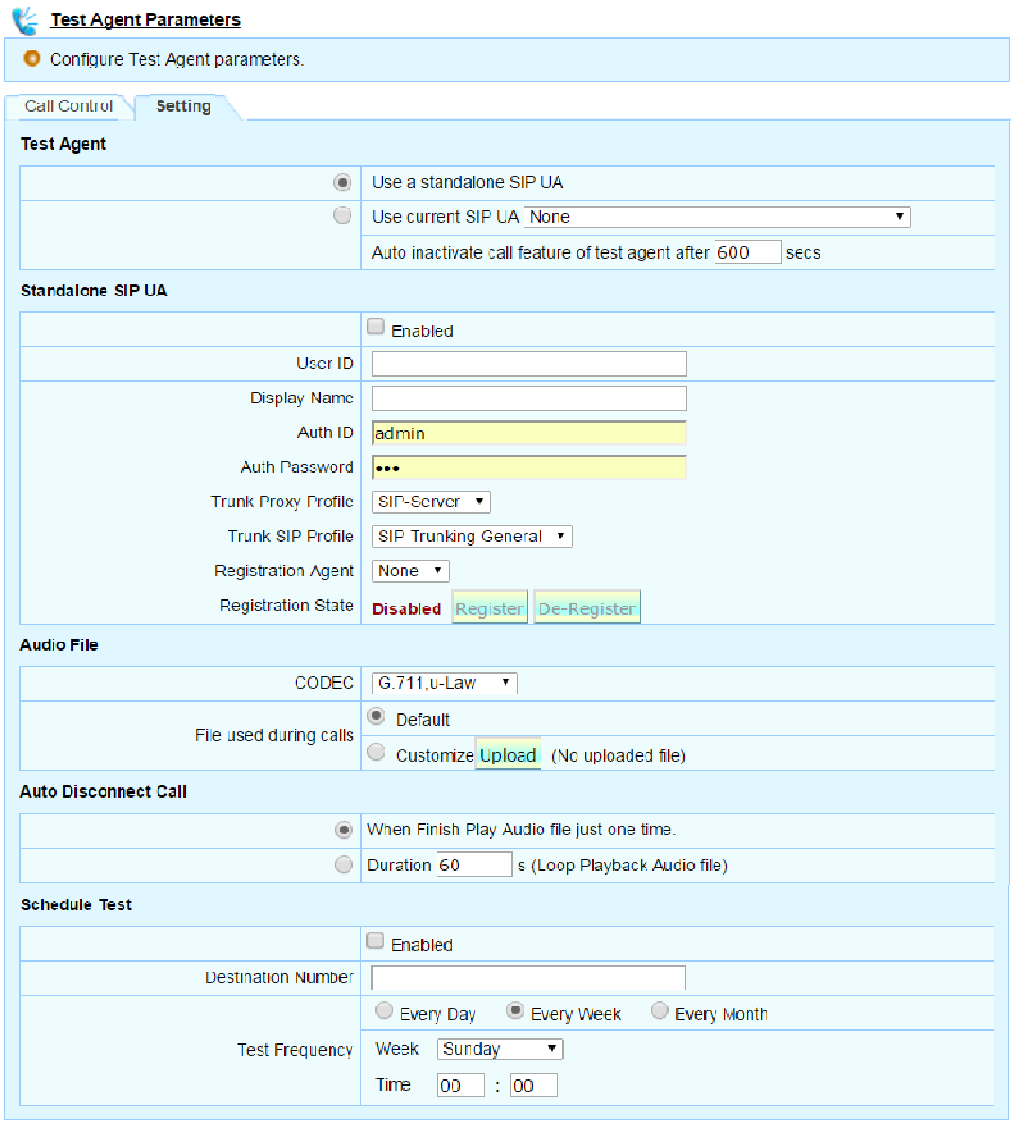


Figure 90. Test Agent Settings-Option 1. Use a standalone SIP UA

Test Agent	Description
Enabled	Check the option box to enable the feature; uncheck to disable it.
User ID, Display Name, Auth ID, Auth Password, Trunk SIP Profile	See section 4.2.1.1 for detailed description.
Registration State	Shows the registration status of the test agent (i.e. connected or disabled). You can click the Register button to register your test agent or click the De-register button to disable it.

Audio File	Description
CODEC	Select the voice CODEC to be used for voice quality tests.
File used during calls	Choose to use the default audio file, or upload your own. This is the audio sound that will play when the test call is established.
Auto Disconnect Call	Configure the test agent to automatically disconnect the call when finished playing the audio file once, or disconnect after defined Duration.

Schedule Test	Description
Enabled	Check the option box to enable the Scheduled Test feature.
Destination Number	Enter the destination number that the test agent calls for testing.
Test Frequency	Define the test frequency in the fields.

- **Option 2. Use current SIP UA**. The ESBC allows the temporary use of a chosen SIP UA from a regular user account as the test agent. After finishing the test call, this SIP account will be released from Test Call Agent duty within the configured timer. (Auto inactivate call feature of test agent after **xxx** seconds)

  The automatic scheduled test is not available for this option.

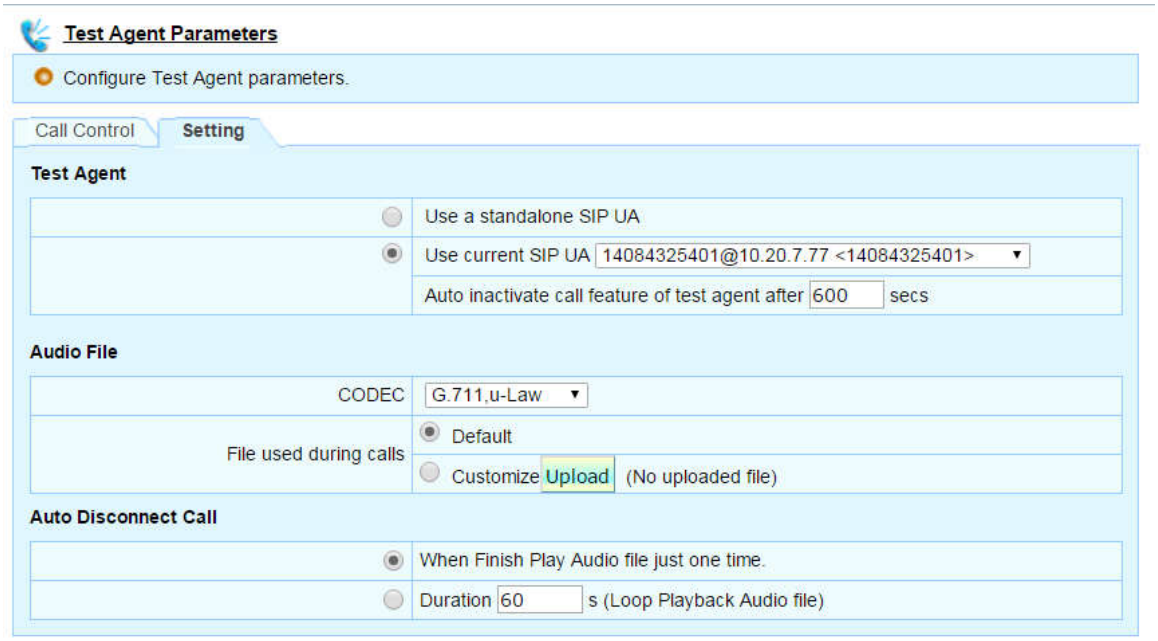


Figure 91. Test Agent Settings-Option 2. Use current SIP UA

4.3.2 The Usage of Test Agent

After the test call a built-in SIP device, the Test Agent (TA), to verify connectivity and voice quality with the service provider network. In addition, automatic scheduled voice quality testing allows service provider's the ability to view variations of call quality over time. (The TA also supports both media loopback and RTP loopback tests with the InnoMedia EMS server.)

Navigate to **Telephony**>**TOOLS**>**Test Agent**.

Enter the target auto-answer phone number in the "Destination Number" field. Click <Dial>, and the TA automatically connects the call and hangs up the call after 90 seconds, you may press <Show> button to display the latest test call result. Please see section 6.7.3 for detailed explanations of voice quality parameters.

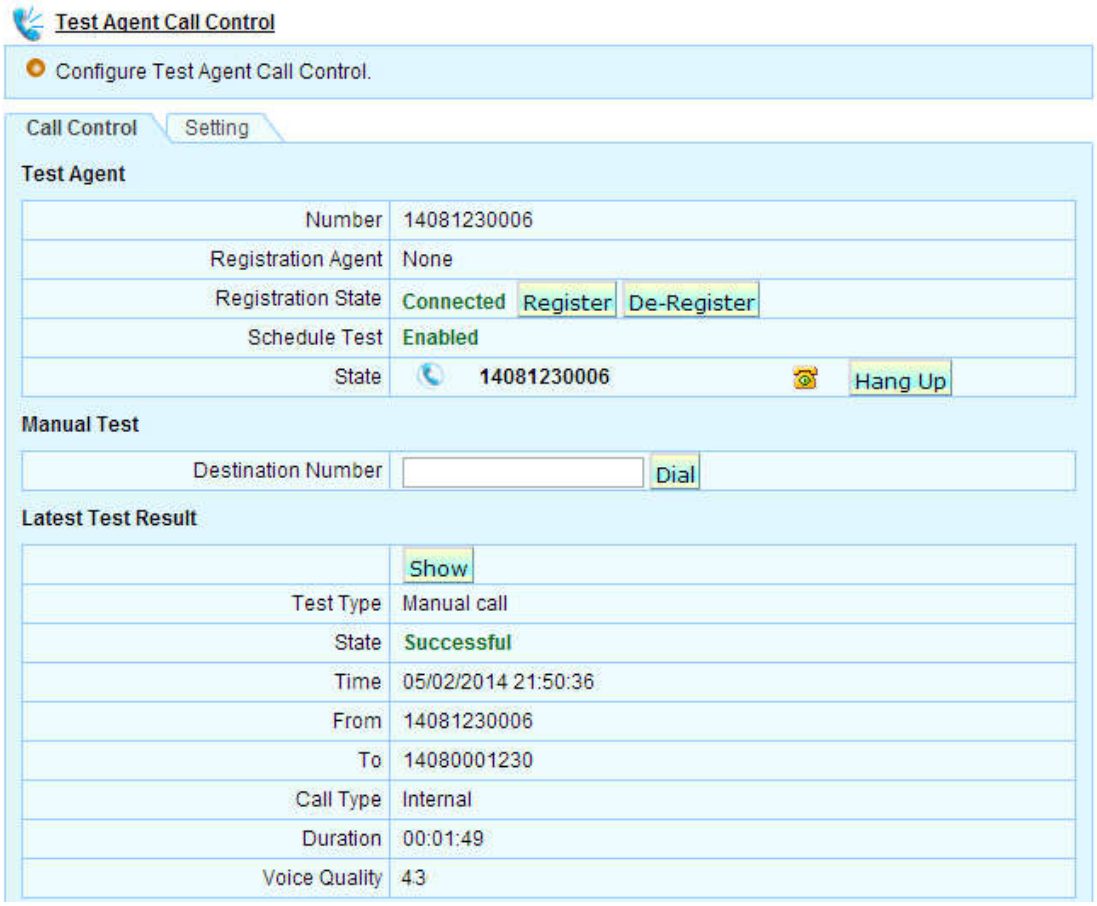


Figure 92. Test Agent Call Control

4.4 Routing Calls: ESBC with a SIP-PBX

Please refer to Section 3.4.1 to make sure the arrangement of appropriate enterprise voice network for the SIP PBX connecting to the ESBC Voice and NAT interface.

This section addresses the configurations of sip telephony signaling and media interconnecting with the SIP PBX.

4.4.1 SIP PBX Profile

SIP PBX models used in deployments, though conforming generally to SIP requirements, may also be designed with some deviations for specific needs. The ESBC normalizes the sip signals for the target SIP PBX to allow interconnection with the SIP server. Choose the target SIP PBX model from the profile list; otherwise choose the "Generic" profile and make the associated configuration changes according to the specific requirements of the PBX.

Navigate to **Telephony** > **SIP-PBX** > **PBX SIP Profile**.

Choose the target SIP PBX model from the profile list as the "Default Profile." Click the <Apply> button. The ESBC support multiple SIP PBX models (profiles) to connect with. The configurations have to match with the assigned SIP Accounts. Please also see section 4.2.1.1 for configuring SIP Accounts.

When there is a need to adjust the parameters of any target PBX profile, click the <Setting> icon. The configurations can be exported (click <Export>) for backup purpose, and restored to the ESBC system (click <Import>).

The definitions and usage for most of the parameters are the same as those of the Trunk SIP Profile. They are briefly described in this section.

4.4.1.1 Basic SIP Parameters

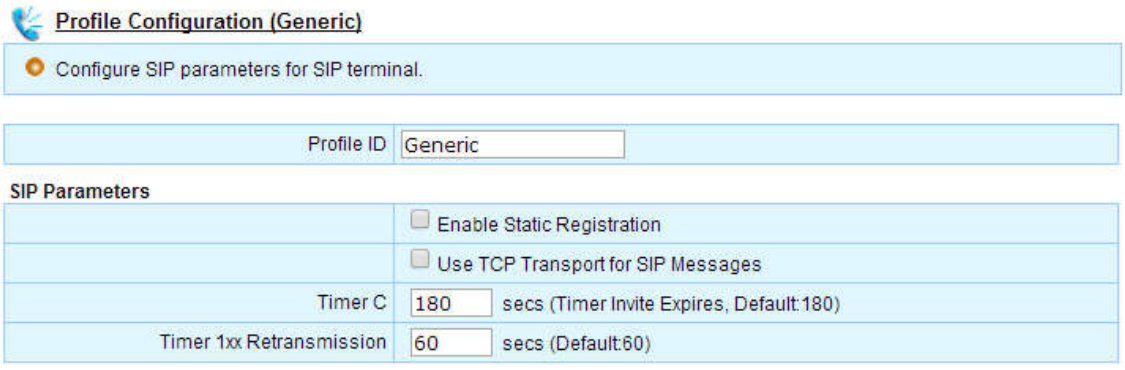


Figure 93 Basic sip parameters in connecting to the SIP PBX

SIP PBX Profile	Description
Profile ID	Enter a unique name for this profile. Usually enter the SIP PBX model name.
Enable Static Registration	Check this option when the target SIP PBX uses static registration mode to connect to its north bound sip server, i.e., the ESBC. Please refer to the SIP-PBX requirements for configuring this option.
Use TCP Transport for SIP Messages	Check this option when the target SIP PBX uses only the TCP transport protocol for SIP messages (but not UDP). Please refer to the SIP-PBX requirements for configuring this option.
Timer C, Timer 1xx Retransmission	Standard SIP timers defined in RFC 3261

4.4.1.2 Interoperability

See the application note: The ESBC Caller ID screening mechanism.

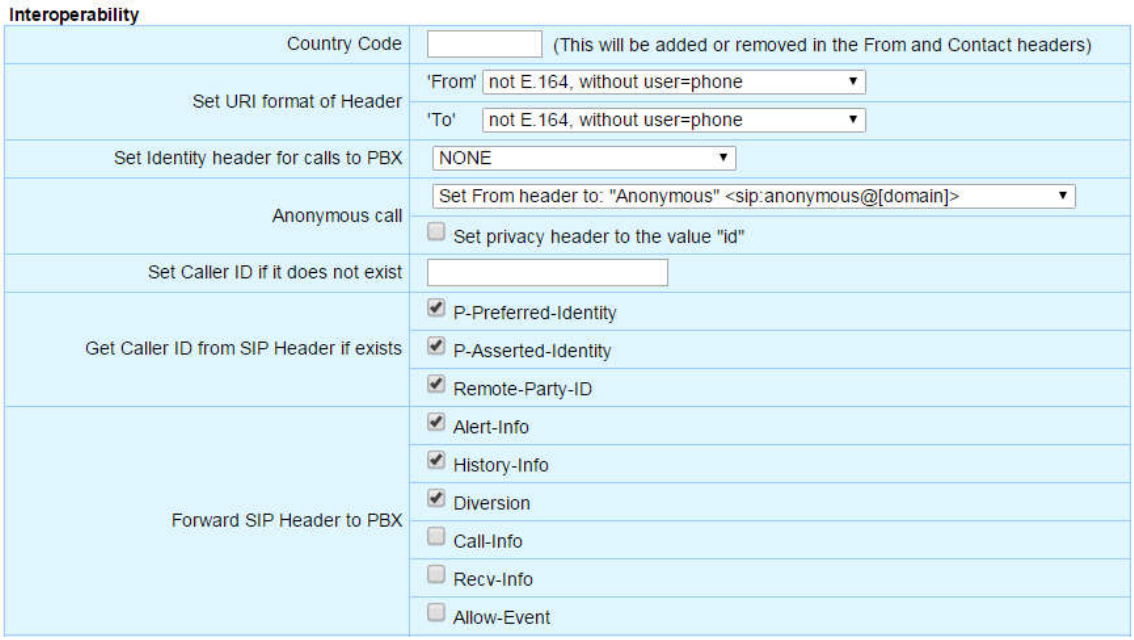


Figure 94 SIP-PBX Interoperability I

SIP PBX Interoperability	Description
Country Code	The DIDs or user accounts configured on the ESBC and SIP server usually do not include country code information. When the PBX configured numbers are composed of “country code” + “DID”,

	input the “country code” value in this field, and hence the ESBC will translate the calling/called numbers with or without country code, and connect calls.
Set URI format of Header: From, To.	Depending on the SIP PBX configuration, the ESBC generates the URI format and sends to the SIP PBX accordingly. 1. not E.164, without user=phone 2. not E.164, with user=phone 3. E.164 (prefixed with ‘+’), without user=phone 4. E.164 (prefixed with ‘+’), with user=phone
Set Identity header for calls to SIP terminal	Depending on the SIP PBX privacy configuration, the ESBC may add one of the following headers as the caller identity header for privacy purposes and forward to the SIP PBX. • P-Asserted-Identity: defined in RFC 3325. This sip header is used among trusted sip entities to carry the identification of the user sending a sip message as it was verified by authentication. • Remote-party-id: is defined in a sip draft. This sip header provides information about the remote party.
Anonymous Call	Depending on the SIP PBX configuration, the ESBC generates the sip From header in four different formats. Select one of these from the drop down menu.
Anonymous call- Set privacy header to the value “id”	The ESBC is able to assert an identity and forward to a trusted SIP PBX by inserting “P-Asserted-Identity” and a “Privacy: id” header (RFC3325)
Set Caller ID if it does not exist	When the inbound call to the SIP PBX does not include a Caller ID, the ESBC may insert a specified caller ID.
Get Caller ID from SIP Header if exists	Choose desired caller ID source(s) among the following three options to transport the SIP PBX P-Asserted-Identity | Remote-Party-ID | P-Preferred-Identity
Forward SIP Header to PBX	Check these items to allow the ESBC to forward the following SIP headers from the service provider network to the SIP PBX. The ESBC itself does not generate the following headers, it simply forwards. Alert-Info | History-Info | Diversion | Call-Info | Recv-Info | Allow-Event

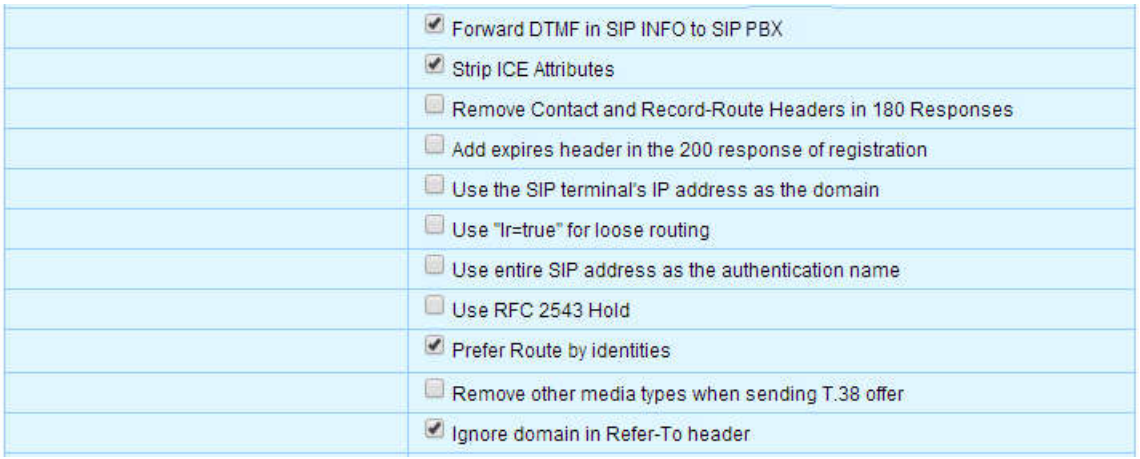


Figure 95. SIP-PBX Interoperability II

Please refer to the application notes: ESBC Application Notes- Configurations for SIP PBX Call Transfer-REFER.

Please refer to the application notes: The ESBC Caller ID screening mechanism.

SIP Interoperability	Description
Forward DTMF in SIP INFO to SIP Server	When this feature is enabled, the ESBC forwards SIP INFO if the registered SIP UAs send DTMF tones using SIP INFO method.
Strip ICE Attribute	ICE attributes are used for NAT traversal purposes. Enable this feature to allow the ESBC to strip all ICE related parameters in SDP messages for messages sent to the SIP PBX. ICE related attributes in SDP include a=candidate(.*) a=ice(.*)
Remove Contact and Record-Route Headers in 180 responses	Enable this feature so that the ESBC removes network routing information (Contact headers and Record-Route headers) from SIP messages before forwarding to the SIP PBX.
Add expires header in the 200 response of registration	Expires header: value in seconds e.g., Expires=3600 Inside a REGISTER request, an Expires header designates the lifespan of the registration.
Use the SIP Terminal's IP address as the domain	Enable this feature so that the ESBC composes the host-part of the SIP URI for request messages using the SIP-PBX IP and forwards to the SIP-PBX.
Use "lr=true" for loose routing	Depending on the SIP PBX configuration, the ESBC adds "lr=true" to enable the loose routing feature. In loose routing, as specified in RFC3261, the Request-URI always contains the URI of the destination user agent. As opposed to "strict routing," where the request-URI always contains the URI of

	the next hop.
Use entire SIP address as the authentication name	Use AOR, e.g., “user@domain” as the authentication ID.
Use RFC2543 Hold	RFC2543 is obsoleted by RFC3261. For backward compatibility, the ESBC can allow the use of “c” destination addresses set to all zeroes (0.0.0.0) for call hold operations.
Prefer Route by Identities	If this item is checked, the ESBC validates Caller ID in the order of precedence “P-Preferred-Identity” > “P-Asserted-Identity” > “Remote-Party-Identity” > “From” of the INVITE messages from the SIP PBX. Otherwise the ESBC validates “From” header only. (Please refer to the application notes: ESBC Application Notes-ANI TN Screen.
Remove other media types when sending T.38 offer	When SIP/SDP messages include multiple m= lines for SIP offers, the ESBC removes those m= lines which are not related to T.38 media types.
Ignore domain in Refer-To Header	This option is designed for processing call transfer features using the SIP REFER method with the SIP-PBX. Please refer to the application notes: ESBC Application Notes-Configurations for SIP PBX Call Transfer-REFER

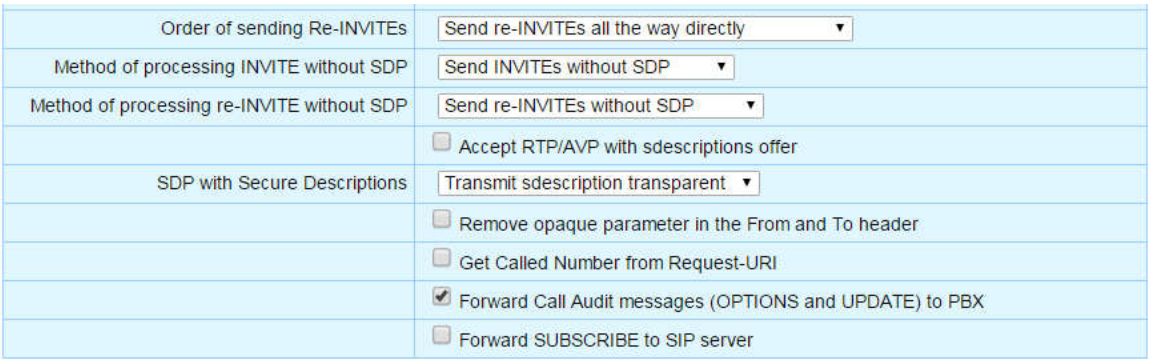


Figure 96. SIP-PBX Interoperability III

SIP Interoperability	**Description**
Order of sending Re-INVITEs	Some particular SIP UAs do not proceed with new sessions with reINVITEs for the current dialog unless the current session is concluded with a response code, such as 200 OK. Leave the default setting unless necessary. Scenario: A – ESBC – B. After call setup, the ESBC receives reINVITE from A.

	• Send re-INVITEs all the way directly. The ESBC sends reINVITE to B, and after receives 200 OK from B, then the ESBC sends 200 OK to A. • Send response before forwarding re-INVITEs. The ESBC replies 200 OK to A with old SDP, and then sends reINVITE to B. After the ESBC receiving 200 OK from B, it will not send 200 OK to A.
Method of processing INVITE without SDP	• Scenario: A – ESBC – B. A calls B, the INVITE message has no SDP. • Send INVITEs without SDP. The ESBC sends INVITE to B without SDP. • Send INVITEs with a fake SDP. The ESBC sends INVITE to B with a fake SDP (g.711). After call setup, the ESBC re-negotiates SDP by sending a reINVITE.
Method of processing re-INVITE without SDP	• Scenario: A – ESBC – B. A calls B, the INVITE message has no SDP. • Send reINVITEs without SDP. The ESBC sends INVITE to B without SDP. • Sends reINVITEs with the old SDP. The ESBC sends reINVITE to B with old SDP. After call setup, the ESBC re-negotiates SDP by sending a reINVITE.
Accept RTP/AVP with sdescriptions offer	Sdescription is short for security descriptions for media streams. Enabling this option allows the ESBC to accept SDP media lines with the value: m=RTP/AVP. Leave the default selection unchanged unless necessary. The RTP/AVP profile is defined for the use of RTP v2 and the associated control protocol (RTCP) within audio and video conferences.
SDP with Secure Descriptions	The RTP/SAVP profile is defined for security services for RTP media and is signaled by use of RTP transport, i.e., SDP media line with the value: m=RTP/SAVP, together with the “crypto” SDP attribute. Select the appropriate item according to the SIP-PBX’s capability to process secured RTP streams. Leave the default selection unchanged unless necessary. • Transmit sdescription transparent • Transmit all sdescription in SAVP • Transmit all sdescription in AVP
Remove opaque parameter in the From and To header	By default, the ESBC transmits any opaque parameter to the SIP PBX. If the SIP-PBX is having issues when receiving this extra parameter in a SIP message, then remove it. Parameter “opaque” is a URI parameter, e.g., sip:me@example.com; opaque=xxxxxx. The opaque parameter is used to achieve the purpose of adding

	extra pieces of information onto an AOR for routing or other purposes, while keeping the AOR intact in the URI.
Get Called Numbers from the Request-URI	By default, the ESBC gets called numbers from the “To” header. Enable this item to allow the ESBC to get (and route) the called numbers from the Request-URI.
Forward Call Audit messages (OPTIONS and UPDATE) to PBX	By default, the ESBC forwards call audit messages (OPTIONS and UPDATE) to the SIP PBX, and the ESBC does not autonomously respond 200 OK to the SIP server. If this item is unchecked, the ESBC responds autonomously to related call audit messages on behalf of the SIP-PBX and does not pass on these messages to the PBX. This setting for the SIP PBX profile should be consistent with the similar parameter in section 4.2.3.2 for the target Trunk SIP profile.
Forward SUBSCRIBE to SIP Server	Enable this option to forward SUBSCRIBE messages from the SIP PBX to the SIP Server and also forward response/NOTIFY message in the opposite direction.

4.4.1.3 ESBC - PBX Security Configuration

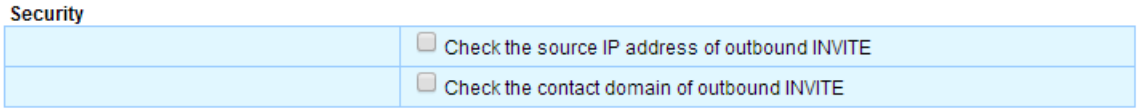


Figure 97. Security configuration for the SIP-PBX

Security	Description
Check the source IP address of outbound INVITE	When this item is enabled, the ESBC replies with 404 not found and rejects outbound INVITE requests from a LAN source IP address which is not configured/registered on the ESBC.
Check the contact domain of outbound INVITE	When this item is enabled, the ESBC replies with 404 not found and rejects outbound INVITE requests from the LAN if the contact domain is not in the registered client list.

4.4.1.4 ESBC - PBX Call Feature Configuration

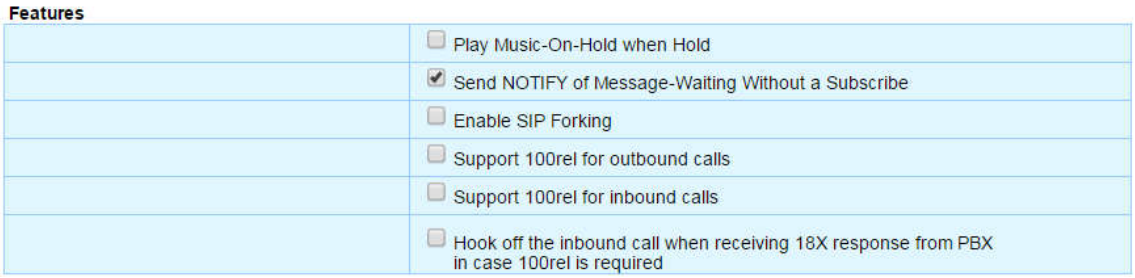


Figure 98. Service or call features designed for the SIP-PBX

Features	Description
Play Music-On-Hold when Hold	By default, the ESBC streams the MOH of the SIP server to the peer party when the PBX user is put on-hold. Enable this feature to have the ESBC play its built-in MOH to the PBX user on behalf of the SIP server.
Send NOTIFY of Message-Waiting without a subscribe	By default, the ESBC always sends NOTIFY messages for MWI to the SIP PBX. Uncheck this item so the ESBC will only send NOTIFY when the SIP PBX subscribes (SUBSCRIBE) to the MWI package.
Enable SIP Forking	Enable this feature to allow the ESBC to “fork” a single SIP call to multiple SIP end points. A single call can ring many end points at the same time.
Support 100rel for outbound calls	When 100rel is enabled, the ESBC will include the 100rel tag in the Supported header and the PRACK method in the Allow header in outgoing INVITE messages. When the called party sends reliable provisional responses, the ESBC will send a PRACK request to acknowledge the response.
Support 100rel for inbound calls	The ESBC supports 100rel/PRACK as the UAS. This setting is used for inbound calls (ESBC in UAS mode). If enabled and the server sends INVITE with no “Supported/Require:100rel” to ESBC, ESBC will respond without 100rel; and if server sends INVITE with “Supported/Require:100rel”, ESBC will respond with 100rel.
Hook off the inbound calls when receiving 18x response from PBX in case 100rel is required	When this parameter is set to disabled, the ESBC does not go into talking state when it receives 183, but instead goes into talking state when it gets 200 OK. By default, this parameter is set to disabled.

4.4.2 SIP-PBX and SIP-Client Authentication

Navigate to **Telephony** > **SIP-PBX** > **Authentication**.

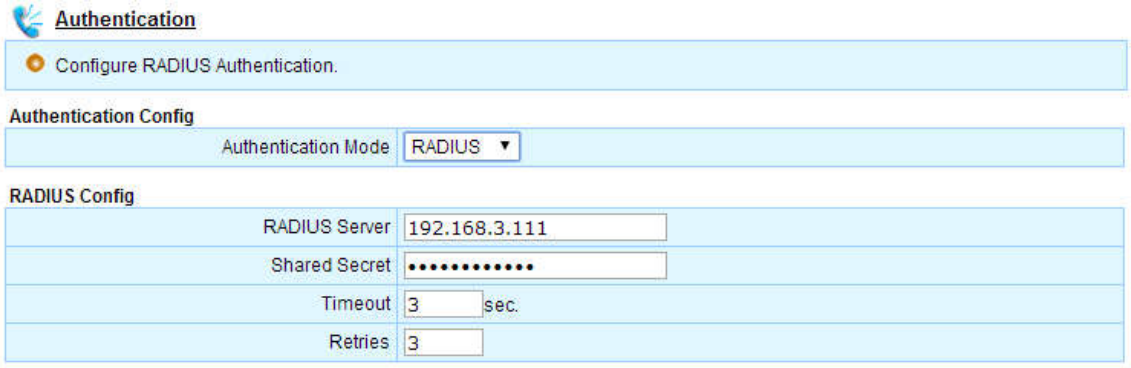


Figure 99. Authenticating the ESBC SIP clients

Three authentication modes are provided for SIP PBX authentication requirements.

- The authentication mode selected is applied to all SIP clients which connect to the ESBC Voice-NAT interfaces (for SIP Trunk Service).

Authentication Mode	Description
None	The ESBC trusts the SIP request attempts from SIP clients. No authentication required.
Local	The ESBC authenticates SIP request attempts from SIP clients by SIP authentication defined in RFC3261, which is a stateless, challenge-based mechanism.
RADIUS	The ESBC supports RADIUS, access server authentication for sip accounts. Enter the target RADIUS server IP or FQDN, shared secret for the ESBC to connect, and threshold values. Note that all sip accounts have to be configured on the RADIUS server when this option is chosen.

Refer to section 10.4 ESBC SIP Authentication Flow.

4.5 ESBC System Global SIP Settings

4.5.1 SIP Parameters

To configure the ESBC's global SIP settings (in addition to those on the "Trunk SIP Profile" and "PBX SIP Profile"), navigate to **Telephony** > **SIP-PBX** > **SIP Parameters**.

4.5.1.1 SIP Parameters

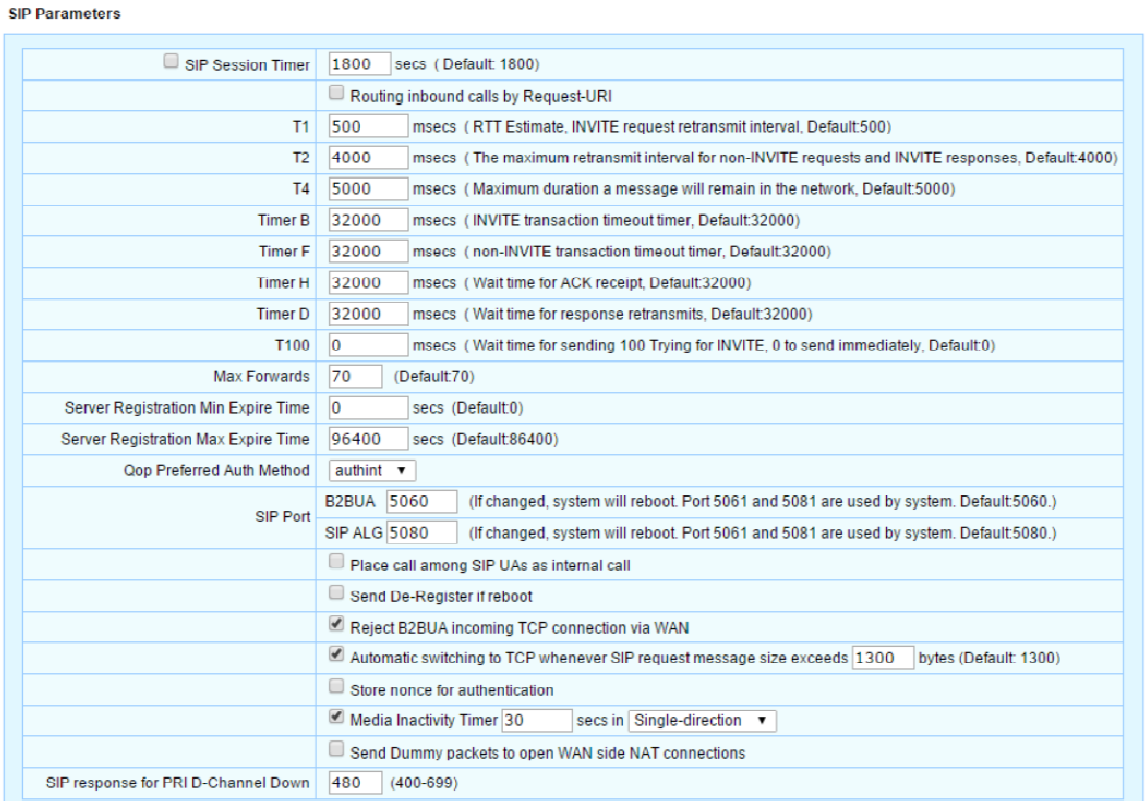


Figure 100 SIP parameter global settings

SIP Parameters	Description
SIP Session Timer	The SIP session timer specifies a keep-alive mechanism for SIP sessions, which limits the time period over which a stateful proxy must maintain state information without a refresh re-INVITE.
Routing incoming calls by Request-URI	The ESBC, by default, routes incoming calls according to the context specified in the "TO" header. If this box is checked, the ESBC routes incoming calls according to "Request-URI" header.

T1, T2, T4, Timer B, Timer F, Timer H, Timer D	Standard SIP timers, defined in RFC 3261
T100	Waiting time for sending 100 Trying for an INVITE message.
Max Forwards	Defined in RFC3261, the Max-Forwards header is used to limit the number of proxies or gateways that can forward the request to avoid looping errors. Default number is 70.
SIP Port	Port used for SIP signaling communicating with north bound SIP servers. Default 5060 for B2BUA (SIP trunk service) and 5080 for SIP-ALG (hosted voice service).
Place call among SIP UAs as internal calls	When this feature is enabled, the ESBC routes calls locally when the called numbers are configured on the ESBC database and does not route to the service provider network. By default, this feature is disabled.
Send De-Register if reboot	When the ESBC reboots, it sends a De-Register message to the service provider network, and initiates the Register and service subscription processes.
Reject B2BUA incoming TCP connection via WAN	By default, the ESBC rejects TCP connection attempts to reduce the possibility of large TCP packets being dropped by routers along the communication path which may result in call attempt failures.
Automatic switching to TCP whenever SIP request message size exceeds xxxx bytes.	Allow automatic switching from UDP to TCP transport when the size of a SIP request message is larger than a threshold value. Default is set to enabled, and the configurable threshold default value is 1300 bytes (not including UDP and IP headers). This configuration applies to both LAN and WAN interfaces, but only to SIP request messages, not responses.
Store nonce for authentication	Depending on the SIP (or IMS) server configuration, the ESBC may store nonce for authentication.
Media Inactivity Timer	Single-direction or Bi-direction time-out timer in second. If checked, the ESBC disconnects calls when no media traffic is detected according to the selection “single-direction” or “both-direction” after the xxx seconds entered.
Send Dummy packets to open WAN side NAT connections	Keep alive message to keep rtp port open if there is NAT equipment deployed in the service provider network to which the ESBC transmits media packets.
SIP response for PRI D-Channel Down	Defines the SIP response code sent to the SIP Server when the PRI D-Channel goes down.

4.5.1.2 System Music on Hold (MOH)

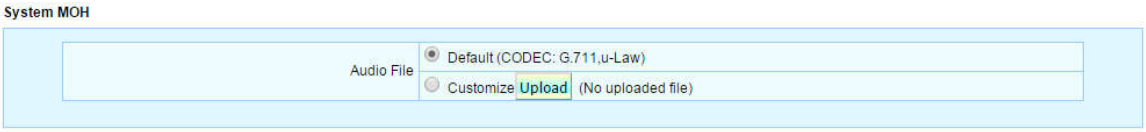


Figure 101. Configuring the ESBC system MOH

MOH	Description
Audio File	The ESBC plays Music On Hold for calls on the FXS ports and SIP UAs during special conditions to prevent the remote side experiencing dead-silence if they are on hold. The ESBC supports a customized MOH audio file of G.711-uLaw of 60 second in length.

4.5.1.3 Filter SIP Method

The ESBC filters SIP messages specified in the list during the call setup process. The configurations affect both inbound and outbound directions to the SIP PBX and the SIP server in the Service Provider network.

Filter SIP Method

No.	Method	Allow	Auth
1	ACK	☑	☐
2	BYE	☑	☐
3	CANCEL	☑	☐
4	INVITE	☑	☑
5	NOTIFY	☑	☐
6	OPTIONS	☑	☐
7	REFER	☑	☐
8	REGISTER	☑	☑
9	SUBSCRIBE	☑	☑
10	MESSAGE	☐	☐
11	INFO	☑	☐
12	PRACK	☑	☐
13	UPDATE	☑	☑
14	PUBLISH	☐	☐
15	SERVICE	☐	☐

Figure 102 Filter SIP Method

4.5.2 Customized SIP response code settings

Navigate to **Telephony** > **SIP-PBX** > **SIP Response** to configure the SIP Response code translation between the enterprise SIP PBX (thru the "NAT and Voice" interfaces) and the SIP server in the service provider network (thru the WAN interface).

The ESBC default mapping tables should meet most SIP requirements. There is no need to input new records to these two tables unless the PBX or SIP Server specify response codes for different processes. If your network is live, make sure that you understand the potential impact of any configuration changes.

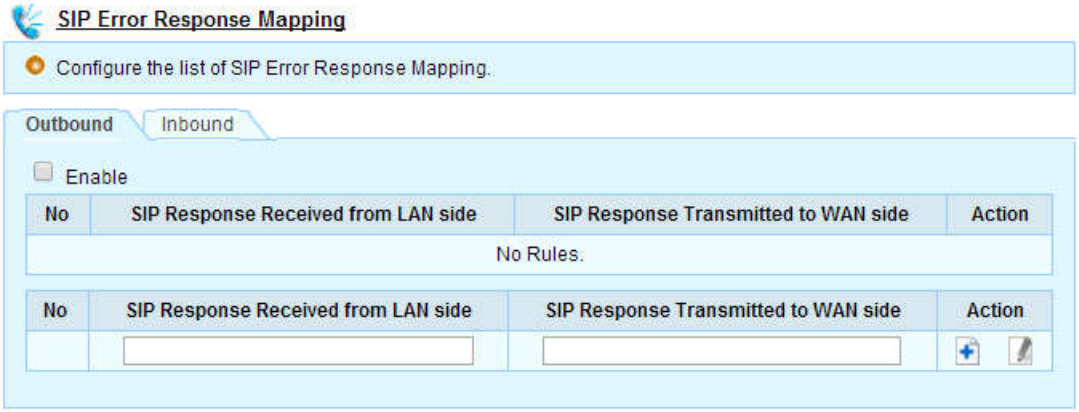


Figure 103. SIP Response Code mapping

The mapping rules are used to match incoming (incoming to the ESBC) SIP error response codes in the order of 'closeness' to the Received Response codes in the list. In this context, closeness is judged by exact matches first, followed by least wild-carded entries. Entries of the same degree of wildcarding are chosen by the rule that matches the most digits before the wild-carded characters.

4.6 Numbering Plan

When the ESBC SIP User accounts (usually DID numbers) and the PBX numbers do not have the same patterns, the “Digit Translation” plan and configuration can be used.

See the ESBC Application Notes--Configurations for digit translations.

4.6.1 Configuring numbers and formulating digit translation rules

The Digit Translation Feature enables the ESBC to prepend or strip certain digits from calling and called numbers in both the outbound and inbound directions by formulating match-pattern and digit map rules. Navigate to **Telephony** > **ADVANCED** > **Digit Translation.**

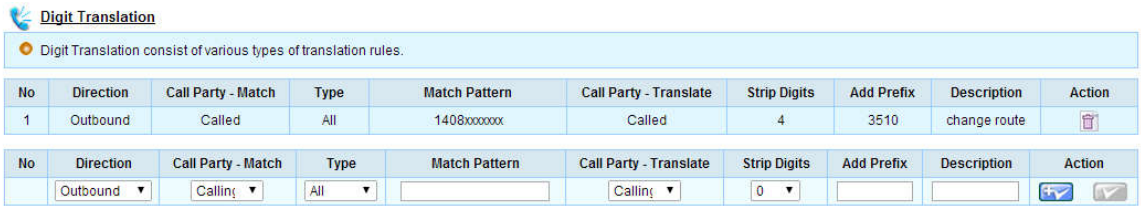


Figure 104. Digit Translation for Called or Calling Parties

Digit Translation	Description
Direction	Direction of the call: Inbound or Outbound
Call Party - Match	Call party to be used to match the digit string rule: Calling Party or Called Party. Note: For Inbound calls, only the called party can be modified.
Type	Types of calls: All, SIP, or PRI
Match Pattern	The Digit Map string that must match before the number can be translated.
Call Party - Translate	Choose whether to translate the number of the calling or called Party.
Strip Digits	The number of digits to strip, starting from the left side.
Add prefix	The digit string to prepend to the number
Description	Add a description of this translation rule.

The following example describes the rule displayed in Figure 104.

1. Strip the first four digits from any telephone numbers which match the pattern (1408xxxxxxx).
2. Add Prefix “3510” is to prepend digit 3510 to the string after step 1 is processed.

The result of this translation will be 3510xxxxxxx (the called numbers sent from the ESBC to the SIP server).

NOTE: To translate CallerID from the PBX for outbound calls, it may be necessary to configure "Set From header for outgoing calls" on the "Trunk SIP Profile." Select the item "Use the original caller" (PBX) and the SIP Server must accept the calling number after translation (see section 4.2.3.2).

4.7 Emergency Call configuration

The ESBC emergency call features permit the service to provide:

- Proper priority for emergency calls
- Line Preemption to always allow emergency calls regardless of session limits--All 911 Emergency Calls always are allowed to be established regardless of any limit in the number of sessions, i.e., the system capacity.
- Overriding of the caller ID and caller name information--Emergency CID and Display Name overrides all other caller id and display name settings when dialing out on an Outbound Route flagged as Emergency.

4.7.1.1 Adding or deleting emergency call numbers

To configure emergency call numbers: navigate to **Telephony** > **ADVANCED** > **Emergency Call**.

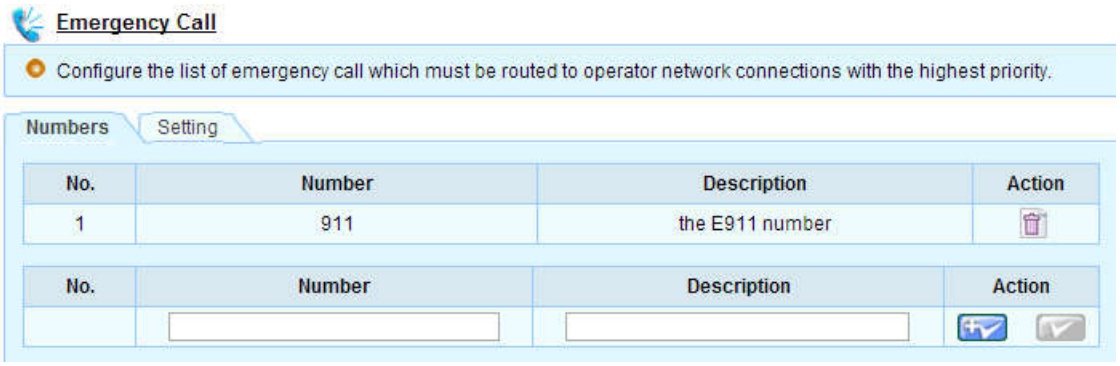


Figure 105. Adding emergency call numbers

Enter an emergency call number in the Number field and its description in the Description Field. Carefully check your local emergency call list. Click the <Add and Apply> button to add this new entry to the ESBC database.

4.7.1.2 Connection settings for emergency call numbers

Click the <Setting> tab.

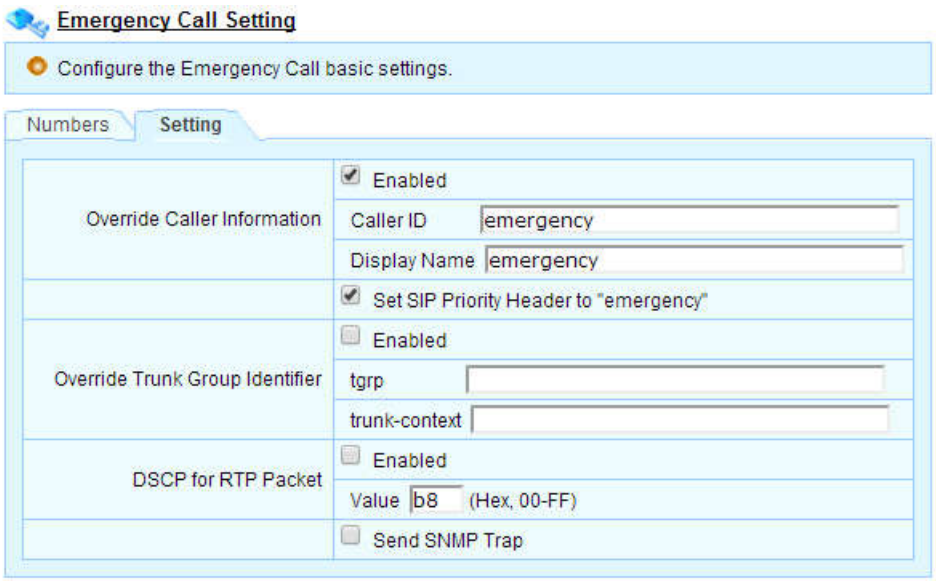


Figure 106. Connection settings for emergency call numbers

Connection Settings	Description
Override Caller Information	If enabled, Emergency CID and Display Name will override all other caller id and display name settings when dialing out.
Caller ID	Emergency caller ID.
Display Name	Emergency caller Display Name.
Set SIP Priority Header to "emergency"	If enabled, the Outbound Route is flagged as Emergency by setting priority: emergency in sip messages.
Set SIP Priority Header to "emergency"	If enabled, the Outbound Route is flagged as Emergency.
Override Trunk Group Identifier	If enabled, the emergency tgrp will override the regular Trunk Group Identifier.
tgrp	Enter an unique Trunk Group Identifier for Emergency Calls
Trunk-context	Enter the Trunk-context for emergency calls
DSCP for RTP Packets	Upstream Internet Protocol (IP) packets are marked with a configurable DSCP to indicate that the IP packet content contains Emergency Media
Value	DSCP value for emergency call media packets.
Send SNMP Trap	Check this box to send an SNMP trap to the EMS or SNMP server when there is an emergency call.

4.8 Media Transcoding

4.8.1 Introduction

The media transcoding feature offered in InnoMedia ESBC 9378 and ESBC-10K series provides a solution to the problem where the Service Provider supports different media capabilities to those of the end device located at the enterprise. The ESBC 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).

By configuring the ESBC, 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.

Please refer to the **ESBC Application Notes- Media Transcoding Features** for call flows and signal descriptions**.**

4.8.1.1 Enabling Transcoding Profiles

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 enable the Transcoding Profile, navigate to the **Telephony** > **ADVANCED** > **Transcoding Profile** page. Simply check the "Enable" and hit the <Apply> button.

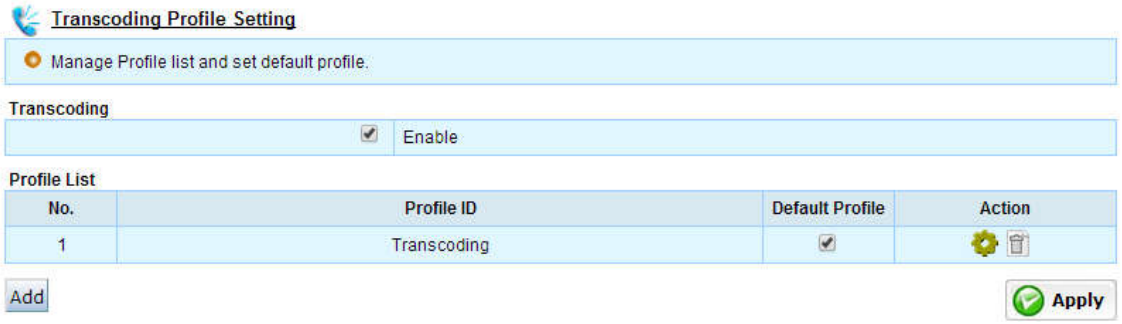


Figure 107 Enabling the transcoding profile

4.8.1.2 Editing or Adding a Transcoding Profile

To add a Transcoding Profile, click the <Add> button and then click the <Setting> button to edit 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:

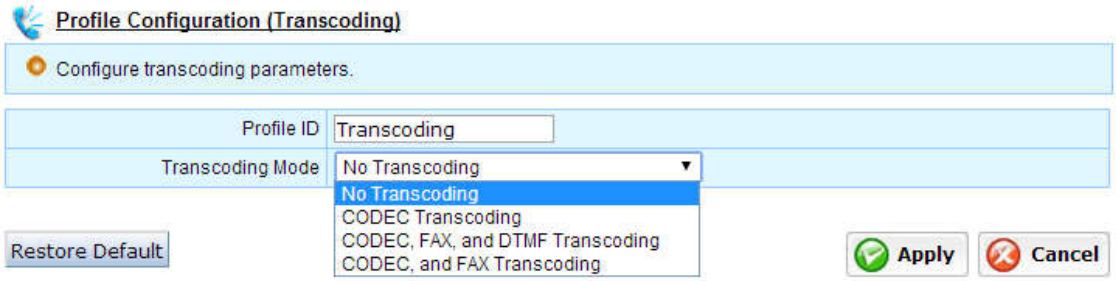


Figure 108. Transcoding mode selections

Transcoding mode	Description
No Transcoding	Transcoding is disabled on UAs assigned to this transcoding profile
CODEC Transcoding	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.
CODEC, FAX, and DTMF Transcoding	Fax, DTMF and CODEC transcoding for all calls.
CODEC, and FAX Transcoding	Fax, CODEC transcoding for all calls, but no DTMF transcoding.

Transcoding options	Description
Allow calls when no supported CODEC in SDP offer	The ESBC will allow the SDP offer to pass through even if the codec is not in the Extended CODEC list (see section 4.8.4). In this case, no transcoding will take place, but this feature allows unsupported transcoding codecs (e.g., 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.

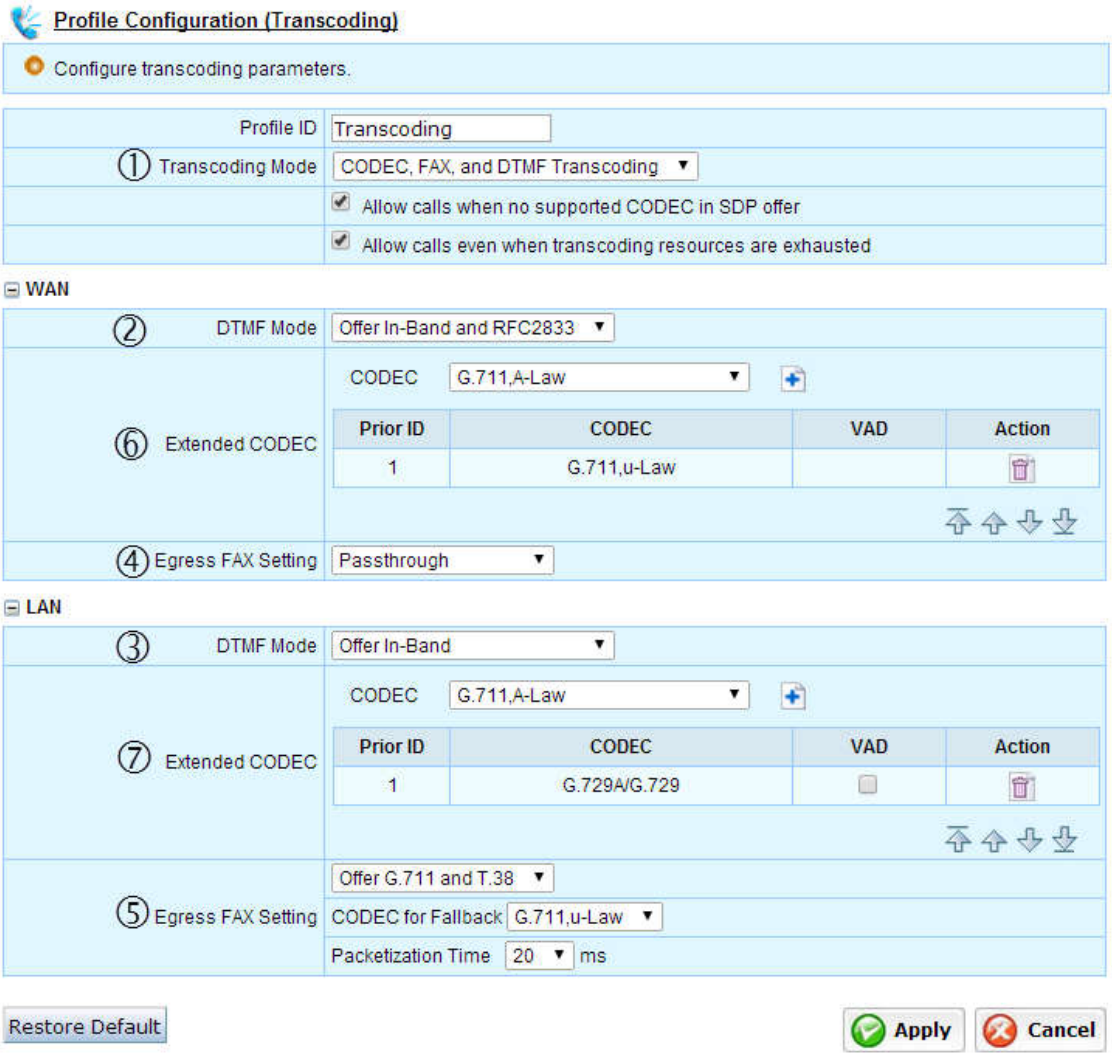


Figure 109. The ESBC Transcoding Profile Configuration

4.8.2 DTMF Transcoding

The ESBC is able to perform DTMF transcoding between the SIP UAs on both the WAN and LAN sides. The DTMF mode can be either 'In-band' or 'RFC2833'.

DTMF Transcoding	Description
Offer In-Band and RFC2833	The ESBC offers both "RFC2833" and "In-band" in SDP messages of SIP Offers.
Offer In-Band	The ESBC offers only "In-band" in SDP messages of SIP Offers.

Note 1 The ESBC automatically handles negotiation when it answers SIP offers from the peer.

In the DTMF configurations displayed in Figure 109,

- ② the ESBC WAN interface offers both RFC2833 and in-band DTMF
- ③ the ESBC LAN interface offers in-band DTMF

The above configurations result in the call flows shown in Figure 110

- For outbound calls, the ESBC transcodes from In-band DTMF to out-of-band RFC2833 DTMF towards the service provider network.
- For inbound calls, the ESBC transcodes out-of-band RFC2833 DTMF to in-band DTMF towards the PBX.

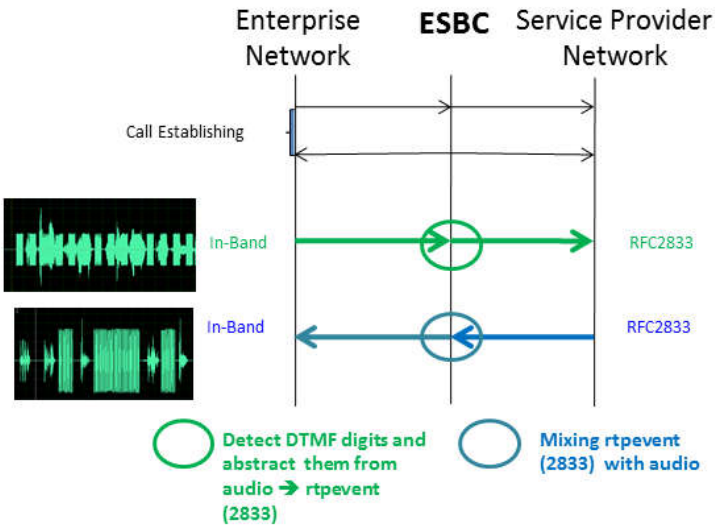


Figure 110 DTMF transcoding: G.711 in-band and RFC2833 processes

4.8.3 FAX Transcoding

FAX over IP communications require a high-quality IP network for proper operation. Please refer to section 3.1.1 for network connectivity assessment.

The ESBC performs FAX transcoding based on an SDP Offer-Answer negotiation. As shown in Figure 109, ④ and ⑤ denote the configuration of the Egress FAX Setting. The ESBC can change its egress SDP signaling using the following three options:

Egress FAX Setting	Description
Passthrough	ESBC allows both LAN UA and WAN UA endpoints to negotiate SDP themselves
Offer G.711 Only	ESBC removes the T.38 codec from the egress SDP offer.
Offer G.711 and T.38	ESBC sends two "m=" lines in its egress SDP including both T.38 and G.711
Packetization Time	Ptime. This parameter is applicable to T.38 packets. Three options are available: 10, 20, and 30 ms.

Note 2. When offered two "m=" lines in SDP, some FAX adaptors will choose the first "m=" line, regardless of the content of the two lines.

To allow FAX Transcoding, the Egress FAX Setting must be selected for both WAN and LAN sides as shown in Figure 109. This will allow the following steps to take place:

1. The ESBC starts processing FAX transcoding flows only when FAX transmission is detected.
2. When FAX transmission is detected by the FAX adaptor on **ONE SIDE** of the fax transmission (usually the receiving side), this side initiates T.38 FAX requests by sending a reINVITE with T.38 information to the ESBC. The ESBC will attempt to honor the T.38 FAX offer from this FAX adaptor.
3. When the condition of item 2 occurs, the ESBC forwards the FAX transmission request to the fax adaptor on the **OTHER SIDE**, according to the ESBC ***Egress FAX Setting Rules*** given below:

    - If Offer G.711 and T.38 is selected, then the ESBC offers two transmission modes (T.38 and G.711) and allows the FAX adaptor to choose which FAX mode to use.
    - If Offer G.711 Only is selected, then the ESBC offers only G.711 transmission mode to the FAX adaptor.
    - If Passthrough is selected on either side, then the ESBC will let the FAX Adaptors determine the final fax transmission mode by allowing both endpoints to negotiate SDP between them.

Note 3. T.38 does not indicate which party (calling party or called party) is responsible for detecting that a FAX transmission is being attempted and initiating the switch from audio mode to T.38 mode by sending reINVITE requests carrying T.38 information. That is, either calling party or called party may initiate T.38 FAX requests.

4.8.4 Voice Codec Transcoding

The ESBC can be configured to add certain codec capabilities (extended codec list, items ⑥ and ⑦ of Figure 109) to transcoding profiles, and perform transcoding in cases where the selected codec in the answer SDP is not available in the original offer.

4.8.4.1 Typical example of voice codec transcoding in deployment

- When the Service Provider network only supports G.711 u-law, then only that CODEC is configured for the ESBC WAN extended codec list.
- When the PBX on the ESBC LAN network only supports G.729, then only that CODEC is configured for the ESBC LAN extended codec list.

Referring to Figure 109, items ⑥ and ⑦ are used to configure the following codec transcoding example. When an outgoing call from the PBX advertises only G.729 in the SDP offer, then the ESBC will add G.711 u-law in the SDP offer when sending out SIP messages to the Service Provider. When an Incoming call from the Service Provider advertises only G.711 u-law in the SDP offer, then the ESBC will add G.729 in the SDP offer when sending out SIP messages to the PBX in the LAN direction.

Once the call is established, the ESBC will transcode the media in both directions with the supported CODECs.

4.9 Routing Calls: ESBC with a PRI-PBX

4.9.1 PRI Spans and Channels

The ESBC T1/E1 module supports ISDN PRI digital lines, and provides up to two T1/E1 ports for connection to the enterprise's TDM PBXs. The T1/E1 TDM voice traffic is converted to VoIP and processed by the ESBC to connect to service provider's SIP trunks and vice versa.

To configure digital lines (PRI T1/E1), navigate to **Telephony** > **T1/E1**> **Digital Line**.

The T1/E1 Status page displays the system configuration and status of all PRI span(s), including D and B channels.

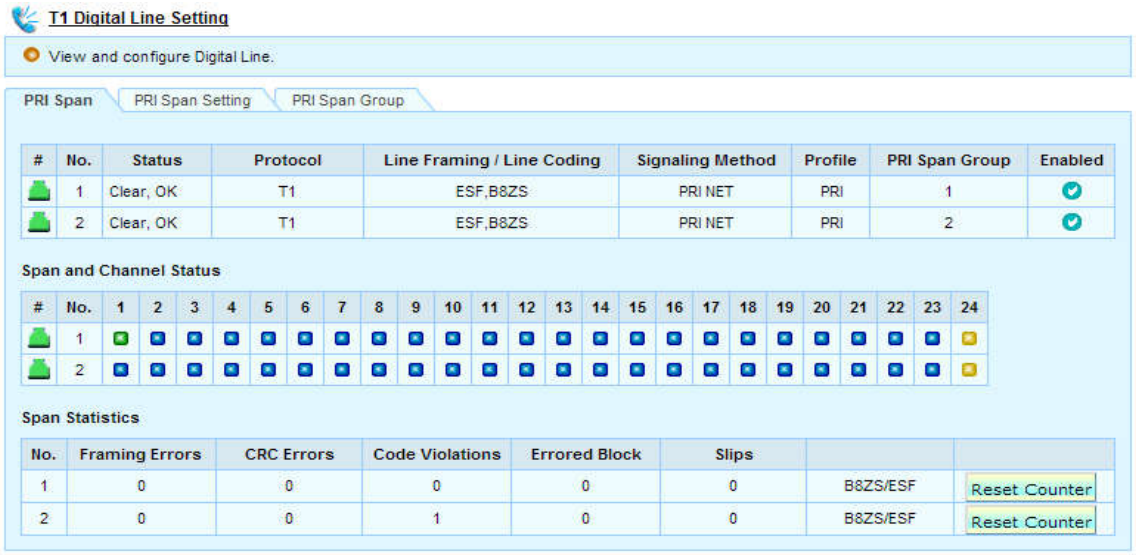


Figure 111. Status display of ESBC digital lines: 2 Span T1 model

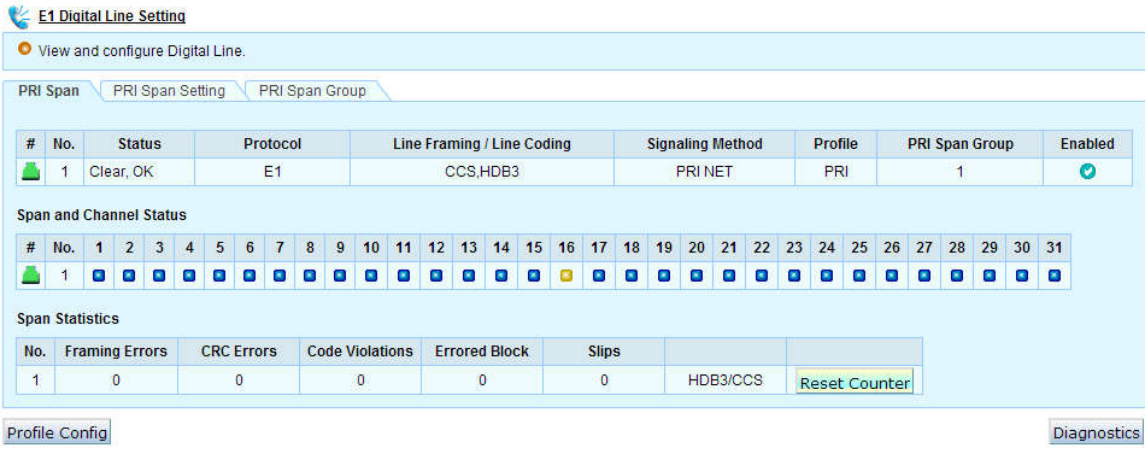


Figure 112. Status display of ESBC digital line: 1 Span E1 model

PRI Span Status Refer to section 4.9.3	Description
Span Status	Disabled: Span is disabled Clear, OK: Span is ready to use D-channel Down: PRI span signaling error or wire unplugged Alarm: Alarm signals are received locally or by the remote PBX
No.	PRI span 1, and span 2
Protocol	T1, J1 or E1
Line Coding	T1 / J1: AMI or B8ZS E1: HDB3 or HDB3 with CRC4
Line Framing	T1 / J1: D4 or ESF E1: CCS
Profile	The media transmission profile defined in the “Profile Config” section for PRI spans. It defines codecs and media related parameters used for PRI spans communicating with the SIP Trunk servers. See detailed description in section 4.9.5.
Span Group	If there are more than one PRI spans (ports), each span can be assigned to a different PRI Span Group to which certain UAs belong. Each span group defines its own PRI span settings and hence a span group may indicate a connection to a different TDM PBX. See detailed description in section 4.9.4.
Enabled	Span enabled or disabled
Channel Status	Idle: channel is ready to use Active: an active call is ongoing on the specified channel Unavailable: channel is disabled ISDN signaling channel (D-Channel) B-Channel Restarting Remote out of service

4.9.2 PRI Span Statistics

The status view of span statistics for digital lines shows Framing Errors, CRC Errors, Code Violations, Errored Blocks, and Slips.

Statistics of Span Health	Description
No.	PRI span (port) number
Framing Errors	Framing errors are counted during synchronous state only. The number increments when incorrect FT and FS-bits in F4, F12 and F72 format or incorrect FAS-bits in ESF format are received.
CRC Errors	No function if CRC6 procedure or ESF format are disabled. In ESF mode, this counter is incremented when a multi-frame has been received with a CRC error. CRC errors are not counted during asynchronous state.
Code Violations	No function if NRZ or CMI code has been enabled. If the B8ZS code is selected, this counter is incremented by detecting violations which are not due to zero substitution. If simple AMI coding is enabled, all bipolar violations are counted.
Errored Block	In ESF format this counter is incremented if a multi-frame has been received with a CRC error or an errored frame alignment has been detected. CRC and framing errors are not counted during asynchronous state. In F4/12/72 format an errored block contain 4/12 or 72 frames. Incrementing is done once per multi-frame if framing errors have been detected.
Slips	The number of frame slips due to clock synchronization between the ESBC and the TDM- PBX
Framing and Coding Status	The Line Framing and Line Coding configuration and the status of the D-channel.
Reset Counter	Resets the counters of all Span Statistics fields

4.9.3 PRI Span Connection Settings

4.9.3.1 Basic Settings

Always ensure that PRI Span parameters are consistent or matched with those settings on the peer TDM-PBX. **Always start from T1/E1 port 1.** Ensure the cable between the interface port and the PBX is connected correctly. ESBC PRI port 1 synchronizes with the clock source, and port 2 follows the timing of port 1. PRI cable(s) must be connected to the port 1 before the port 2 can be used.

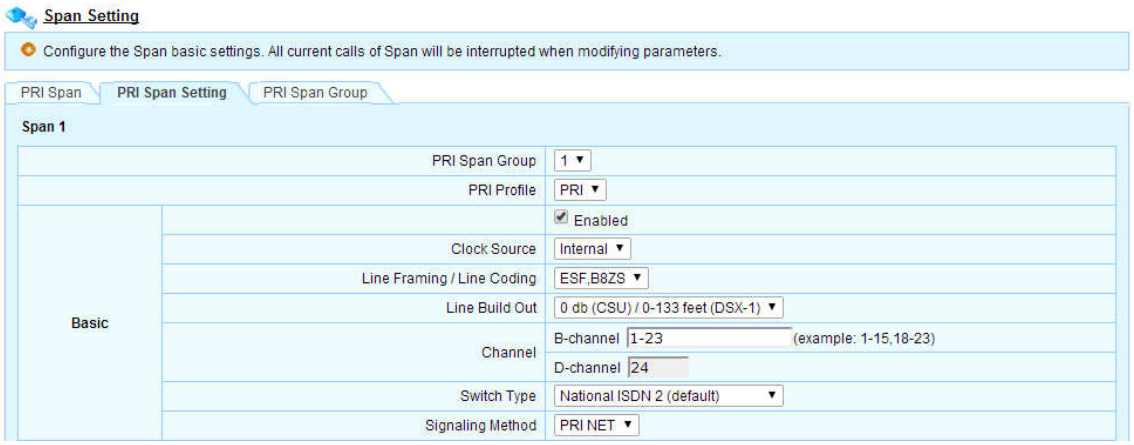


Figure 113. PRI Span Basic Settings

PRI Span: Basic	Description
PRI Span Group	Configure Span 1 (and/or 2) to the target PRI span group which assigns the ESBC user accounts to PRI spans. (PRI Span Groups are described in section 4.9.4)
PRI Profile	The media transmission profile defined in the “Profile Config” for PRI spans. It defines codecs and media related parameters used for PRI spans communicating with the SIP Trunk servers.
Enabled	Check this box to enable the particular span. Span 1 should always be enabled.
Clock Source	Internal. The ESBC default clock source is “Internal”, where voice transmission timings follow the ESBC internal clocking scheme. The connected TDM-PBX clock should be configured to follow the ESBC clock. Line. The ESBC follows the TDM-PBX clocking scheme. If there are two spans, span2 always follows the clock of span1. Span2 does not have its own clocking scheme. If there is only one PRI line, always connect to SPAN1. Do not

	change the ESBC clock mode default settings unless necessary.
Line Framing/Coding	T1 / J1: ESF/B8ZS or D4/AMI E1: CCS/HDB3 or CCS/HDB3 with CRC4 The configuration of the ESBC should be consistent with that of the TDM-PBX.
Line Build Out	Default mode: 0 db (CSU) / 0-133 feet (DSX-1). The configuration of the ESBC should be based on T1 line length and match that of the TDM-PBX.
Channel	Total Channel Numbers: T1-24 channels; E1-31 channels B-Channel: configure the enabled B-channel numbers. D-Channel: T1 fixed at CH 24; E1 fixed at CH 16
Switch Type	T1: National ISDN 2 (default), Nortel DMS100, AT&T 4ESS, Lucent 5ESS, Old National ISDN 1, Q.SIG E1: EuroISDN (common in Europe) The configuration of the ESBC should match that of the TDM-PBX.
Signaling Method	PRI NET/PRI CPE. Always configure the ESBC with PRI NET mode to perform switch side functionality (the TDM-PBX should be in PRI-CPE mode). Do not change this default setting unless necessary.

4.9.3.2 ISDN Interoperability Configuration

The parameters listed here are available for PRI interoperability purposes. They are settings for provisioning Network specific facilities, and ISDN Timers for Q,921/Q.931. Normally, the default settings meet most usage requirements. You will only need to adjust these parameters if the default settings need to be changed to deal with special conditions.

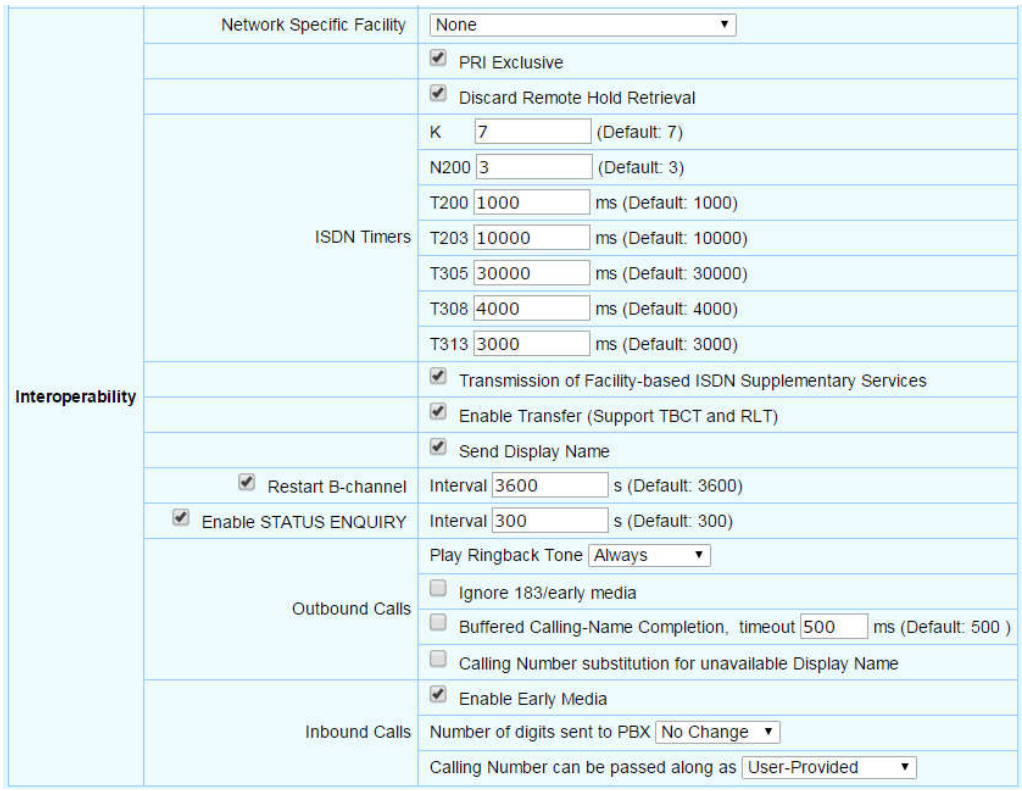


Figure 114. PRI Span Interoperability Settings

PRI Span: Basic	Description
Network Specific Facility	The ISDN protocol allows telephone service providers to add their own custom protocol extensions. These custom protocol extensions provide various localized services that are not defined in the general ISDN specifications. The ESBC supports the following types of Network Specific Facility and acts as a PRI NET role (Central Office switch side). Configure 'none' if the type is unknown or not configured on the TDM-PBX. Available options are: none| sdn | megacom | tollfreemegacom | accunet.
PRI Exclusive	Default mode: checked. Unconditionally picking B channels exclusively. This parameter should be enabled when the ESBC is configured as "PRI NET." Do not change this default setting unless necessary.
Discard Remote Hold Retrieval	To ignore remote side (PRI side) indications and use MOH that is supplied over the B channel. Default mode: checked. Do not change this default setting unless necessary.

ISDN Timers	Do not change these default settings unless necessary. Q921 Timers: • K: the maximum value of outstanding I frames • N200: Maximum number of retransmission attempts • T200: Transmission Timer • T203: Maximum time allowed without Frame Exchange Q931 Timers: • T305: Timer sets how long to wait to get a response such as RELEASE to a DISCONNECT message. • T308: sets how long to wait to get a response such as RELEASE COMPLETE to a RELEASE message. • T313: sets how long to wait to get a response such as CONNECT ACK to a CONNECT message.
Transmission of Facility Based ISDN Supplementary Services	Default mode: checked. Do not change this default setting unless necessary.
Enable Transfer	Default mode: unchecked. Enable this feature when the client subscribes to this supplementary service. Supports business ISDN supplementary services: TBCT/RLT/ECT (National ISDN II/Nortel DMS/Euro ISDN). When enabled, the ESBC uses the SIP REFER method to the SIP Server side and releases B channels when call transfer/forwarding operations are successful.
Send Display Name	Default mode: checked. Send the display name of the caller to the TDM PBX called party for an **inbound call**. Some TDM-PBX's do not support "Display Name" Facility message settings coming with the SETUP message. Uncheck this setting when necessary.

Processing PRI Outbound Calls

Play Ring Back Tone (RBT). The ESBC provides the flexibility to support various forms of RBT generation, either in-band RBT media or out-of-band RBT signals. The default configuration should meet most deployment requirements.

Please take care in fine tuning ring back tone parameters only when necessary. If your network is live, make sure that you understand the potential impact of any configuration changes. In the interworking of ISDN and SIP/SDP signaling between the TDM-PBX and the VoIP network, a common problem scenario is:

A PBX user places a call through the ESBC to an external number and does not hear a RBT before the call is answered, even though the receiving phone rings and the call is answered.

The ESBC processes in-band RBT media, or out-of-band RBT signals are controlled by the following settings:

1. "Play Ringback Tone for outbound call". Three options are available: ALWAYS, NEVER, AS-NEEDED.

   ALWAYS: The ESBC always sends inband RBT media to the PBX, either relaying media from the network or the ESBC generating locally. No out-of-band RBT for “ALWAYS” mode.

   NEVER: The ESBC never generates inband RBT locally. RBT sent to the PBX is either inband RBT media from the SIP Trunk side or out-of-band RBT signals (default setting).

   AS-NEEDED. The ESBC decides the RBT action to the PBX according to the messages it receives (SIP response codes 180/183 from the network; and Q.931 Progress Indicator from the PBX).

2. Check Box: **"**Ignore **183/early media for outbound calls"**. When unchecked, the ESBC honors 183 messages received from the network side; otherwise the ESBC will not process inband RBT media relayed from the network. (default setting: unchecked)

Please refer to Section 10.3.

PRI Span: Processing PRI Outbound Calls	**Description**
Play Ring Back Tone (RBT)	See description above.
Ignore 183/early media	When it is unchecked, the ESBC honors 183 messages received from the network side; otherwise the ESBC will not process inband RBT media relayed from the network. (Default setting: unchecked). See description above.
Buffered Calling-Name Completion, timeout xxx ms. (xxx = 500 by default)	When making outbound calls from the PRI-PBX, the PBX may send a SETUP message without the Calling Party IE, and later the PBX sends a FACILITY message with the Calling Party IE. 1. Disabled: The ESBC will send out the SIP INVITE message without Calling-Name after receiving the SETUP message, and ignores the FACILITY message afterwards. 2. Enabled: the ESBC will not send out the INVITE message immediately. Instead, it will wait xxx ms for the FACILITY message to arrive with the calling party information before sending out the INVITE to the SIP server. The threshold timer is as configured. (Option “Transmission of Facility-based ISDN Supplementary Services” must be enabled for this setting).
Calling Number Substitution for Unavailable Display Name	Enabled: When the Display Name is not available in the SETUP message from the PBX, the ESBC uses the configured calling number to substitute the User part of the From header of the SIP message sent to the SIP Server. Note that the parameter “Set

	From header for Outgoing calls" needs to be set to "Use the Original Caller". See Section 4.2.3.2 SIP Profile Configuration: Interoperability.

Processing PRI Inbound Calls

PRI Span: Processing PRI Inbound Calls	Description
Enable Early Media.	Inband ring back tone (RBT) media may be provided by the PBX when "Enable early media" is checked, which enables the ESBC to send out a 183 message and forward inband media to the service provider's network when the ESBC receives inband signals from the PBX. The default setting is enabled. Configure settings from this section only when a WAN-side user places a call to a PBX user through the ESBC and does not hear a RBT before the call is answered, even though the receiving phone rings and the call is answered. Please refer to *"ESBC Application Notes for PRI RBT Processing*" for a detailed feature description and the ESBC usage of the PRI Progress Indicator.
Number of digits sent to PBX	This setting allows the ESBC to strip off digits and send the number of digits configured to the PBX. Examples as follows: • If the setting is set to "x", then the ESBC keeps the x least significant digits (from the right to the left). When x=4 with an incoming call to the number 17775554567, the ESBC sends "4567" to the PBX. • If the setting is set to "No Change", then the ESBC does not strip off any digits. This specific setting only applies to a TDM PBX connected to the ESBC with PRI span(s). There is another digit translation feature (see section 4.6 Numbering Plan) whose configuration applies to both SIP devices and a TDM PBX.
Calling Number can be passed along as User-Provided or Network-Provided	This parameter is used to set the screening indicator of an inbound call. The options are: • User-Provided • Network-Provided The ESBC default value setting is 'User-Provided'.

4.9.3.3 B-Channel Maintenance

B-channel “RESTART” and “Status Enquiry” mechanisms are used to synchronize the B-channels of the ESBC and PBX so as to ensure that telephony services are operational.

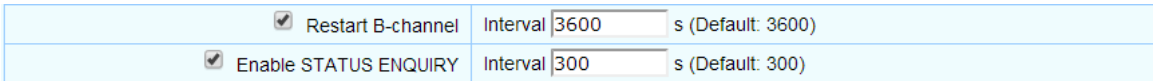


Figure 115. Span Setting Screen— B-Channel maintenance

B-Channel RESTART

The RESTART message requests a restart (set to idle) for a specified B-channel to the peer device. Response to a successful request is the RESTART ACKNOWLEDGE message.

By default, the ESBC triggers B-channel restarts every 60 minutes. The restart requests are performed on idle B channels only. In addition, when an incoming call is placed and the ISDN cause code returned from the PBX is "channel unavailable (44)" or "circuit congestion (34)", the ESBC immediately directs the incoming call to the next available channel, and triggers a Restart Message to the PBX in order to re-initialize this B-channel to an idle state.

If there are no acknowledgement messages (RESTART ACKNOWLEDGE) received from the PBX, then the ESBC marks these channels with “not available for service”.

STATUS Enquiry

The ESBC attempts to continuously monitor the status of active calls by sending STATUS ENQUIRY messages to the PBX periodically, and indicates whether calls are still active from the “STATUS” messages returned from the PBX. The returned STATUS message contains the call state Information element (IE). On the occurrence of certain procedural errors, both sides of the connection will attempt to re-align call states.

4.9.4 User Account Assignment to PRI Span Groups

Click the <PRI Span Group> tab to assign user accounts to each Span Group. If there are more than one PRI spans (ports), each span can be assigned to a different PRI Span Group to which certain UAs belong. Each span group defines its own PRI span settings and hence a span group can allow a connection to a different TDM PBX.

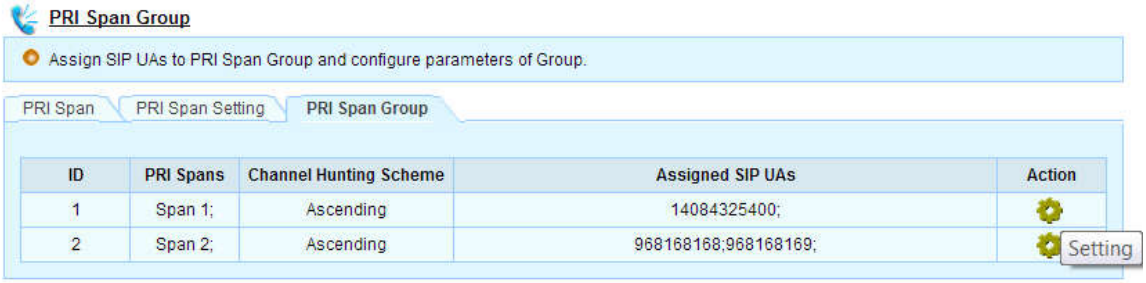


Figure 116. The PRI Span Groups

The above example shows span1 is assigned to span group1, and span2 is assigned to span group2. Span group1 contains the default route user account (14084325400), and hence all numbers that are not configured in the ESBC SIP UA database will be forwarded to span1. (It is also possible that both span1 and span2 are assigned to the same PRI Span group. Please refer to section 4.9.3.1)

PRI Span Group	Description
ID	ID of PRI Span Group. One Span group may be comprised of multiple (two) PRI spans.
PRI Spans	A PRI span (port) can only belong to one PRI Span Group.
Channel Hunting Scheme	Ascending or Descending
Assigned SIP UAs	Displays UAs of each PRI Span Group

4.9.4.1 Assigning UAs to a PRI Span Group

Click the <Setting> icon of the PRI Span Group under the Action column of Figure 116 to complete the UA assignment task for the specified PRI Span Group. Click Arrow keys (⇨⇦) to assign or remove user accounts to/from the particular span group. Refer to Figure 117.

4.9.4.2 Selecting an appropriate B-Channel hunting scheme

Choose the appropriate channel hunting scheme (ascending or descending). The use of a different hunting scheme from that of the PBX is suggested. If the PBX uses an “ascending” channel hunting scheme, then configure the ESBC with “descending” so as to distribute loading evenly on entire PRI spans. Refer to Figure 117.

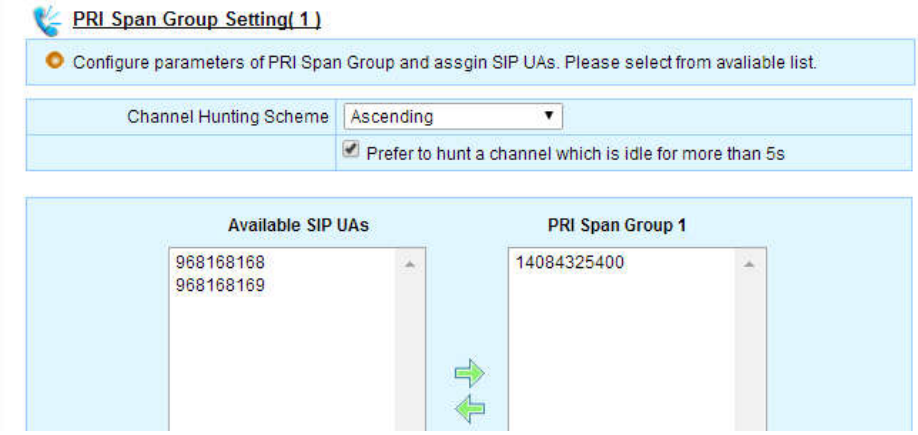


Figure 117. PRI Span Group Settings

4.9.5 PRI Media Profile Settings

To configure media transmission settings for digital lines, click <Profile Config> button on the **PRI Span** main page, Figure 112.

FAX over IP communications require a high-quality IP network for proper operation. Please refer to section 3.1.1 for network connectivity assessment.

Figure 118 Media profile configuration for digital lines

PRI Media Profile Setting	Description
Profile ID	Name of this profile
DTMF mode	“RFC2833” and “In-band” are supported. Choose the correct mode according to what the sip trunk (service provider) indicates.
G.726 packing order	There are two types of byte order for G.726, namely RFC3551 and AAL2. With this setting you can choose the byte order in order to use the same order as the remote entity.
Gain	Control telephone speaker and listen volume. Tx: transmission gain to digital lines (toward the PBX) Rx: receiving gain from the digital lines and sending toward the SIP Trunk side.
CODEC	To change the priority level of the CODECs, select the CODEC and click the up and down arrows at the bottom-right hand corner. To remove a CODEC, click the Delete icon in the Action column.
Fax	The ESBC supports both “T.38 Relay” and Pass_Through modes for fax transmission over an IP network. Parameters for Pass_Through: Fax signals are transmitted in the same way as voice media. Codec used: G.711 (U-Law or A-Law). Parameters for T.38 Relay: • Jitter Buffer. Default value is 120 ms. Do not change the default value unless necessary. • T2. Timeout timer for receiving packets. Default value is 400 ms. Do not change the default value unless necessary. • Low Speed Redundancy. Number of redundant T.38 fax packets to be sent for the low speed V.21-based T.30 fax machine protocol. Default value is 4. Do not change the default value unless necessary. • High Speed Redundancy. Number of redundant T.38 fax packets to be sent for high-speed fax machine image data. Default value is 2. Do not change the default value unless necessary. • Bit Rate. Choose a fax transmission speed to be attempted: 2400, 4800, 9600, or 14400. By choosing 14400, the ESBC can automatically adjust/lower the speed during the transmission training process. The ESBC supports G3 Fax. • Max Buffer Size. This option indicates the maximum number of octets that can be stored on the remote device before an

	overflow condition occurs. Default value is 200. Do not change the default settings unless necessary. • Max Datagram Size. Maximum datagram size. This option indicates the maximum size of a UDPTL packet that can be accepted by the remote device. Default value is 300. Do not change the default settings unless necessary. • ECM. Enable Error Correction Mode (ECM) for the gateway. • Packetization Time (p-time): For fax, the period (in ms) after which a UDPTL packet is sent. The default value is 20ms.

4.9.6 PRI diagnostics

To diagnose the status of the ESBC PRI spans, navigate to **Telephony > T1/E1 > Digital Line**, and click the <Diagnostics> button. The ESBC supports both "Bit Error Rate Test" (BERT) and "Loop Back Test" for PRI trunk lines. Note that the PRI trunk lines will be put out of service when entering diagnostic mode.

4.9.6.1 Bit error rate testing (BERT)

The BERT module tests PRI cables and diagnoses signal problems in the field. BERT generates a specific pattern on the egress data stream of a T1/E1 controller and analyzes the ingress data stream for the same pattern. The bits that do not match the expected pattern are counted as bit errors. Error statistics are displayed in real-time during the testing process.

Environmental factors may affect BERT Results. In a communication system, the bit error rate of the receiver side may be affected by **transmission channel noise, interference, distortion, bit synchronization problems, attenuation due to cable length**, etc. The BERT checks communications between the local and the remote ports.

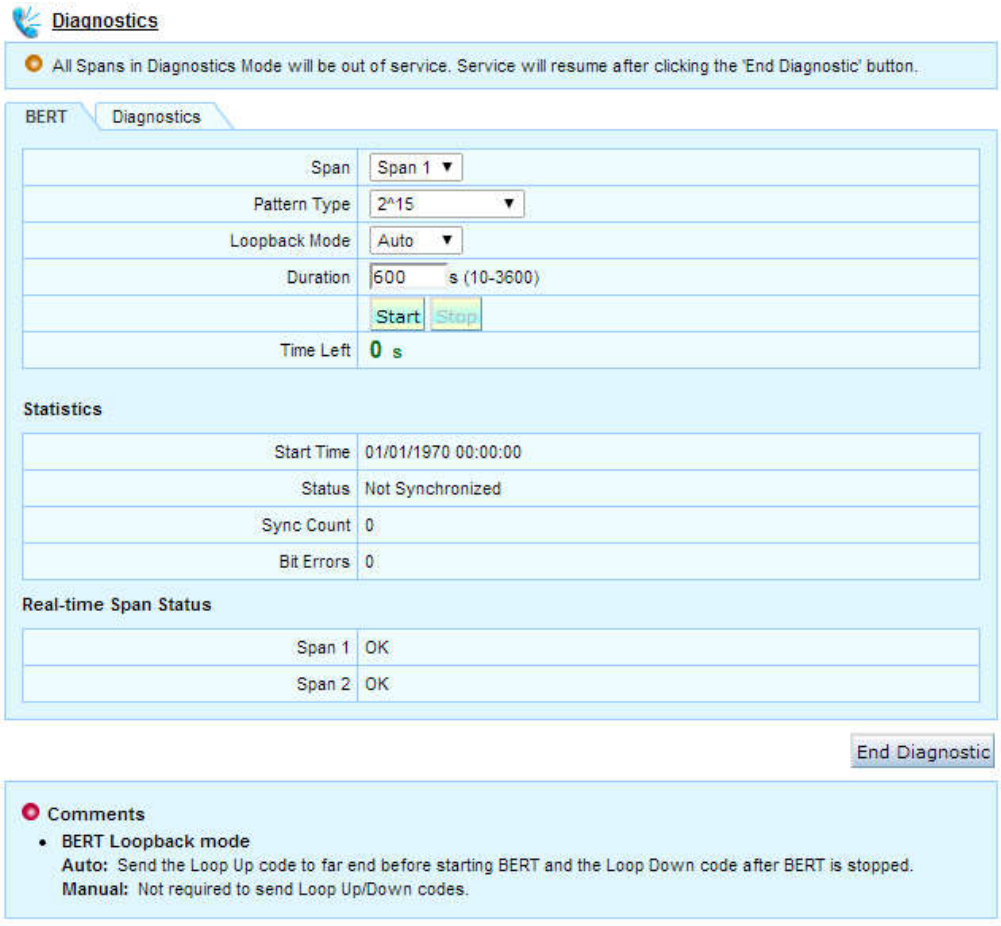


Figure 119. BERT diagnostics page

Choose which span to test, target pattern of bit stream, loop back mode, and test duration, then simply press the <Start> button to perform the BERT diagnostics.

Pattern Type

Depending on the particular sequence of bits (i.e., the data pattern) transmitted through a system, different numbers of bit errors may occur. Patterns that contain long strings of consecutive identical digits (CIDs). When the BER is tested using dissimilar data patterns, it is possible to get different results. A detailed analysis of pattern-dependent effects is beyond the scope of this article, but it is sufficient to note the importance of associating a specific data pattern with BER specifications and test results.

Pattern Type	Description
2^15 2^20	Pseudo-random repeating test pattern that consists of 32,767 (2^15) or 1,048,575 (2^20) bits.
Unframed 2^15 Unframed 2^20	Pseudo-random repeating pattern that is 32,767 (2^15) or 1,048,575 (2^20) bits long. The DS-3/E3 framing bits in the DS-3/E3 frame are overwritten when the pattern is inserted into the frame.

Two modes are supported for BERT: Auto and Manual.

BERT Loop Back Mode	Description
Auto	The ESBC sends the loopup code to the remote port before starting the BERT and sends the loopdown code after the BERT finishes.
Manual	The ESBC does not send loopup or loopdown mode codes to the far end port. When this mode is selected, you must manually enable loopback at the remote port before you start the BERT.

The ESBC displays the total number of error bits and statistics in real-time during the testing process. The number of “Bit Errors” and “Sync Count” may be incremented over time until the duration timer is reached.

BERT Test Results	Description
Status	Synchronized Not Synchronized: when no signal is received.
Sync Count	0: not sync-up at all. 1 or more: the number of sync-up times.
Bit Errors	This is the Count of total number of bit errors detected after Status is "Synchronized."

4.9.6.2 PRI Span LoopBack Diagnostics

This section describes a troubleshooting method known as loopback testing. The ESBC supports three types of LoopBack testing methods to diagnosis **clocking** and/or **line health** states. Note that loopback tests are intrusive and impact services.

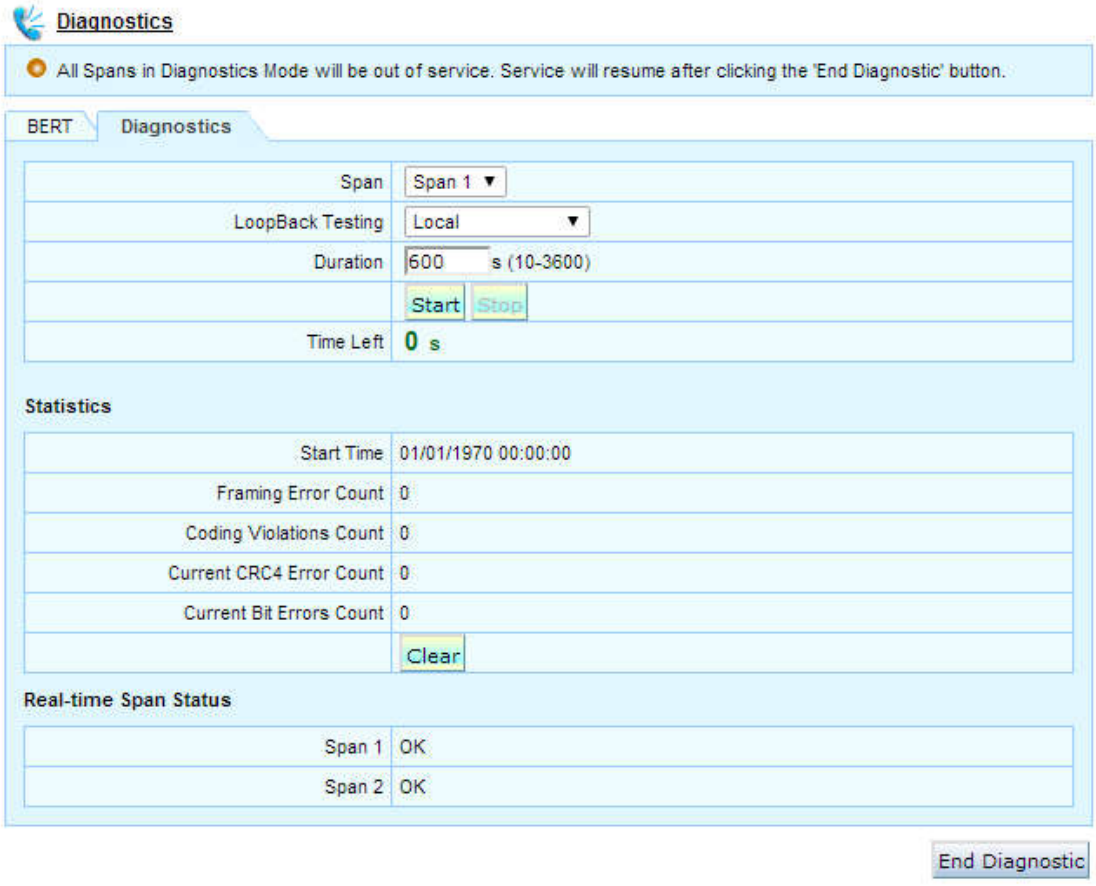


Figure 120. LoopBack diagnostics page

A common issue in VoIP networks with a digital interface connection to a TDM-PBX is that the ISDN circuit does not come up or stay up. Such issues can be complex because:

- Faulty components might reside in several places - for example, within the ESBC or in the TDM-PBX domain.
- Multiple components impact the status of the ISDN PRI. The problem could be mismatched configuration across the PRI lines (which leads to clock slips, line/path violations), a damaged cable, a bad card, or other issues.

Choose either span to test, loop back testing mode, and test duration, then simply press the <Start> button to perform loop back diagnostics.

LoopBack Testing Mode	Description
Local	Tests the inward loopback such that the interface on the ESBC can synchronize on the signal it is sending.
Network Line	Loops the data back towards the network before the framer chip entering the ESBC.
Network payload	Loops the data back towards the network from the T1/E1 framer chip in the ESBC.

Note that the wire used for LoopBack testing has to be made with special cross-over pin wiring. It is illustrated in the following picture.

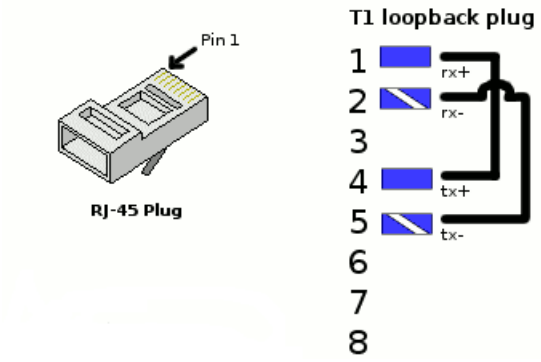


Figure 121. T1/E1 loopback wiring map

Please refer to the "*ESBC Application Notes-T1E1 PRI Troubleshooting Guide*" for further detailed information.

4.9.7 SIP response code – PRI cause code mapping

Navigate to **Telephony** > **T1/E1** > **SIP Response Mapping** to configure the mapping of SIP Response codes to PRI cause codes, and vice versa. Cause codes identify possible reasons for call failures.

The ESBC default mapping tables should already meet most deployment requirements. There is no need to input new records to these two tables unless the PBX or SIP Server specifies proprietary codes. If your network is live, make sure that you understand the potential impact of any configuration changes.

4.9.7.1 Mapping of a received SIP 4xx-6xx response to an outbound INVITE request

The ESBC follows the guidelines below to disconnect calls.

- On receipt of a SIP failure response (4xx-6xx) to an outbound SIP NVITE request, unless the ESBC is able to retry the INVITE request to avoid the problem (e.g., by supplying authentication in the case of a 401 or 407 response), the ESBC transmits a Q.931 DISCONNECT message with the Cause Code value in accordance with the SIP 4xx-6xx response.

- On receipt of a SIP BYE request from the IP domain, the ESBC sends a Q.931 DISCONNECT message with cause value 16 (normal call clearing).
- On receipt of a SIP CANCEL request to clear a call for which ESBC has not sent a SIP final response to the received SIP INVITE request, the ESBC sends a Q.931 DISCONNECT message with cause value 16 (normal call clearing).

Note that when a call is disconnected from the ISDN (PRI PBX) side, the sip Reason header is set to the received Q.850 cause in the appropriate messages (e.g., BYE|CANCEL|FINAL… failure responses) and sent to the SIP server. If the call is disconnected because of a SIP reason, the Reason header is set to the appropriate sip response code.

Refer to "*ESBC Application Notes-- Interworking of SIP Response Codes and ISDN Q.931 Cause Codes*"**.**

Figure 122. Configuring special SIP response to PRI cause code mapping records

SIP Response to PRI Cause Code	Description
No	Record number
Received SIP Response Code	Must specify digits within the range 400-699 which denote SIP trunk side errors.
Transmitted PRI Cause Code	Must specify digits within the range 1-127 which denote PRI Q.931 errors.

4.9.7.2 Mapping of a Received PRI Cause Code to SIP Response

The ESBC follows the guidelines below to disconnect calls.

- If ESBC has received a SIP INVITE request but not sent a SIP final response, ESBC sends a SIP response according to the cause code value in the received Q.931 DISCONNECT message from the PBX. Refer to "ESBC Application Notes: Interworking of SIP Response Codes and ISDN Q.931 Cause Codes."
- If a Q.931 cause value is neither listed in the default mapping nor in the configurable mapping, the default response '500 Server internal error' is used.

Figure 123. The PRI cause code mapping to SIP Response codes

PRI Cause Code to SIP Response	Description
No	Record number
Received PRI Cause Code	Must specify digits within the range 1-127 which denote PRI Q.931 errors.
Transmitted SIP Response Code	Must specify digits within the range 400-699 which denote SIP trunk side errors.

4.10 SIP Trunk Voice Service During Redundant WAN Switchover

For SIP Trunking Voice Services, the ESBC processes all SIP messages on behalf of both SIP servers and its LAN SIP UAs. When the WAN interface switches, the ESBC will

- Terminate on-going calls.
- Perform a registration process on behalf of all SIP UAs to the SIP Server immediately.
- New outgoing calls will be processed through the new active interface.
- If the ESBC receives new inbound calls (SIP INVITEs) from the original active interface, the ESBC will process the calls and send 200 OK replies from the new active interface.

5 ESBC Hosted Voice Service

ESBC SIP ALG module is applicable when the WAN Interface mode is configured as Single Interface.

5.1 ESBC SIP-ALG Functionality Features and Benefits

The ESBC SIP ALG functionality supports hosted voice services to allow enterprises to obtain full-featured IP PBX solutions without the cost of purchasing a PBX or a key system. While provisioning and delivering scalable voice features to enterprise SIP phones, the ESBC SIP ALG offers the ability to allow voice traffic to flow both from the enterprise to service provider networks and vice versa, which enables the service provider to deploy hosted voice services to enterprises seamlessly.

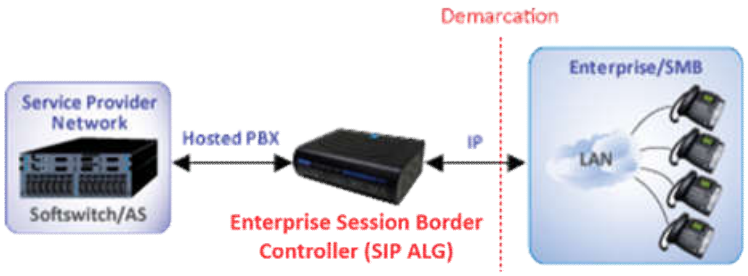


Figure 124. Hosted Voice Service delivered by the ESBC SIP-ALG module

Serving as a proxy, the ESBC's SIP ALG operations include (but are not limited to):

- Solving the VoIP routing issues caused by the introduction of a NAT in the enterprise network. If the SIP message uses an IP address local to the enterprise network when replying to a SIP message originating from the service provider network, it cannot be routed properly without a SIP ALG. This is corrected by the ESBC by inspecting traffic and rewriting information within SIP messages (SIP headers and SDP body) to ensure that the signaling and media traffic communicate correctly and can hold an address:host binding until the session terminates.
- Allowing the SIP phones and SIP Server (the host PBX) to use dynamic UDP ports to communicate with the known ports used by the service provider and SIP Phones. Without the ESBC, the ports would either get blocked by the enterprise firewall, or the network administrator needs to open a large number of pinholes in the firewall, resulting in the network being vulnerable to attacks.
- The ESBC provides security to the enterprise voice network. The ESBC protects against toll fraud and provides needed security and privacy for the connection, using IP layer protection, ACLs and an internal SIP firewall.
- Constantly monitoring voice quality and providing statistics to help diagnose network problems either between the operator's core network and the enterprise, or within the enterprise itself.
- Monitoring dynamic SIP phone registration status for accounting and usage status management

5.2 Configuring SIP Phones for Hosted Services via the ESBC

Follow the steps below to allow SIP devices (phones or gateway) on the enterprise network to register and obtain voice service from the service provider via the ESBC.

5.2.1 Configuring the SIP phones on the ESBC LAN

Arrange the SIP devices to be located in the same network as the ESBC NAT and Voice (VoIP) ports. If the SIP devices are configured as DHCP clients and the ESBC LAN port is configured to offer DHCP server functionality, the SIP devices may obtain an IP address from the ESBC. If the SIP devices are configured with fixed IP, it is necessary to have the default gateway of the SIP devices point to the ESBC LAN IP address.

Apart from the IP addresses configured on the sip devices being on the same network as the ESBC LAN, the registering sip server and other service configurations of the sip devices should be pointed to the service provider network.

5.2.2 Configuring the ESBC SIP ALG Module

To enable ESBC SIP ALG service is straightforward. Navigate to the **Telephony** > **SIP ALG** > **Setting** page.

Figure 125. Configuring the ESBC SIP-ALG module

Field Name	Description
Enable	Check this item to enable SIP ALG service.
RTP Timeout	This item refers to a media inactivity timer. When there are no RTP packets associated with a particular connection for a longer period than this timer, the ESBC drops this connection.
SIP Expired Time	If the registration status of a particular SIP UA becomes stale and exceeds this configured timer, the ESBC removes it from the registration list.
Modify the User part of the Contact header in outgoing	The User part of contact headers usually refers to the SIP accounts. Do not change this default setting unless necessary.

messages	This setting is used for handling inbound calls with SIP Forking features from the SIP server to the IP Phones. When multiple phones are configured with one SIP account, the ESBC composes the contact header for each phone by appending a device-dependent string to ring these phones configured with the same SIP account.
Modify the host and port part of the Contact header to the ESBC's IP and port in outgoing messages	If this option is enabled, the ESBC SIP ALG module replaces the IP:port information of SIP headers and the SDP body with that of the ESBC WAN interface in outgoing messages to the SIP server for NAT traversal purposes.

5.2.3 Hosted Voice Service Survivability

When the ESBC switches its WAN connection from the primary to the secondary interface (or vice versa), during the transitional period ongoing calls may still continue. However, new inbound calls can only be successfully connected once the IP phones complete a SIP registration process with the host SIP server. That is, the IP phones need to send REGISTER messages in order to receive calls. In order to minimize the time duration between the network switchover and REGISTER messages being sent by the IP phones, the ESBC uses two SIP parameters to smooth the voice service transition.

Even when both the primary and secondary WAN interfaces (networks) are out of service, the ESBC can still assure continuity of local voice services. The connected IP phones can still make calls locally.

Navigate to **Telephony** > **SIP ALG** > **Setting**

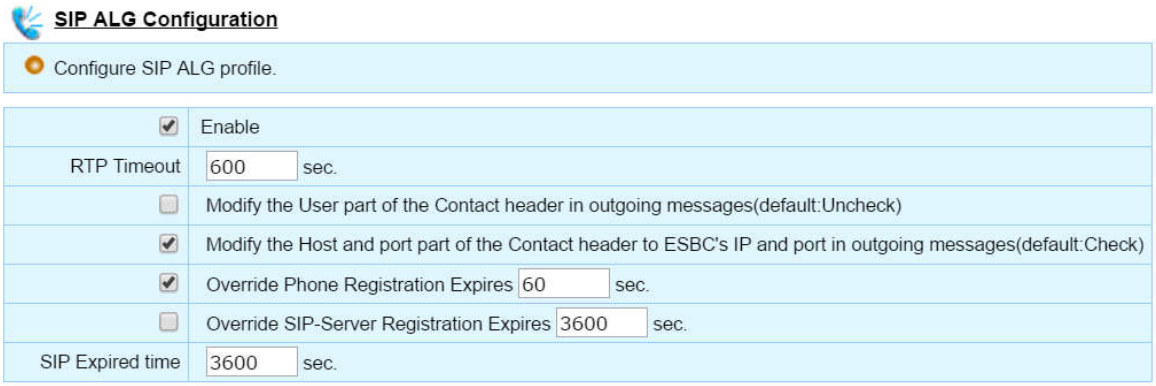


Figure 126. SIP ALG settings for accelerating voice service transition-1

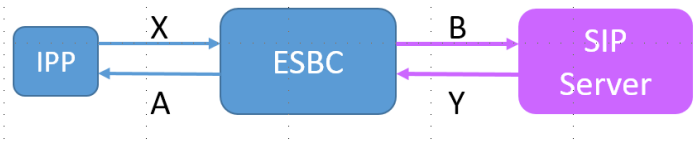


Figure 127. SIP ALG settings for accelerating voice service transition-2

Parameters	Description
Override Phone Registration Expires	The REGISTER expiration value **A** (in SIP 200 OK response to the IP phone) The ESBC uses this configured value, **A**, to replace the Expires value in the 200 OK message sent by the SIP server in reply to the REGISTER request from the IP Phone. The value **X** is the registration expiration time in the REGISTER request from the IP Phone. The value **Y** is the Expires value in the 200 OK message from the SIP server. A ≤ X (A has to be smaller or equal to X) A ≤ Y (A has to be smaller or equal to Y) It is recommended that this parameter is enabled. The intention is to have the IP Phones trigger REGISTER requests as frequently as needed to reduce the time from the switchover of network interfaces to when the IP phones are registered on the new network. Hence a small value is recommended, such as 60 seconds.
Override SIP-Server Registration Expires	The registration expiration value **B** (in the REGISTER request) The ESBC uses this configured value B, to replace the Expires value in the REGISTER request sent to the SIP server. If it is disabled, the ESBC simply forwards the value X (as described above) in the REGISTER requests to the SIP server. B ≤ Y

5.3 FQDN to IP Static Mapping

When there is a need to configure unresolved domain names for sip devices, or occasionally if the FQDNs configured on the sip devices are not resolved by the DNS servers configured on the ESBC, the ESBC may be configured to statically map sip domain names to IP addresses and route calls to the designated service provider networks.

The FQDNs may be included in the Request-URI, Via header, Contact header, Route header etc. The ESBC follows the sequence of resolving and routing sip messages to the service provider network according to the precedence SIP URI: outbound proxy > route header > request URI.

When the ESBC resolves a name, it first checks the static record, then the system DNS cache, and finally, if it is still unresolved, the ESBC will perform a DNS query. The ESBC caches DNS results for 10 minutes.

Navigate to the **Telephony** > **SIP ALG** > **Setting** page.

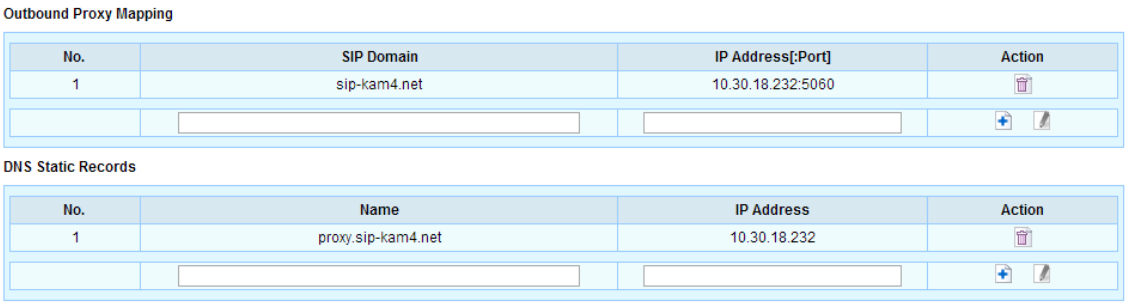


Figure 128. The static FQDN – IP mapping table

Outbound proxy mapping	Description
SIP Domain	The sip domain to which the ESBC shall query for sending SIP request messages.
IP address[:Port]	The IP address (and port number) associated with the SIP Domain. If the Port number is not specified, the ESBC use 5080 as the default SIP ALG communication port.

DNS Static Records	Description
Name	The FQDNs included in the SIP headers.
IP Address	The IP address associated with the FQDN of the same record.

5.4 List of Active Devices for Hosted Service

The ESBC SIP ALG module records all active sip devices which register to the service provider network. When the registration of a particular device becomes stale (see Figure 125), the ESBC removes it from the list. Navigate to the **Telephony** > **SIP ALG** > **Status** page.

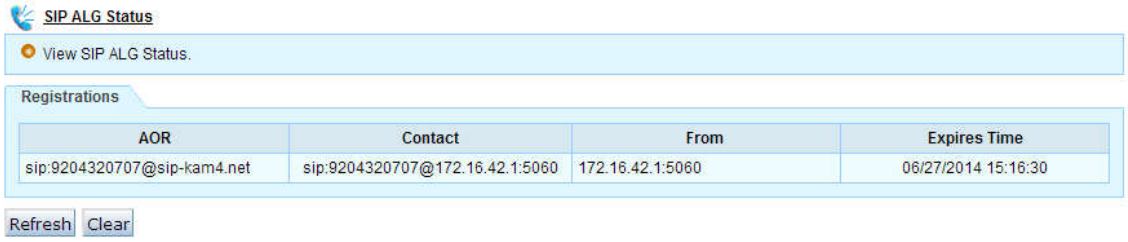


Figure 129. The registration status table of ESBC LAN SIP devices

SIP ALG Client Status	Description
AOR	The Address of Record is usually thought of as the “public address” of the user. It is composed of a user-part (e.g., 9204320707) and a host-part (e.g., sip-kama.net).
Contact	The contact header of the sip device. The ESBC uses the ip:port to reach this sip device.
From	The ip:port of this sip device.
Expires Time	The registration expiration date and time.

5.4.1 Enable Provisioning Service to IP Phones for Hosted Voice Services

For security purposes, the ESBC by default blocks data traffic to or from the hosts on the ESBC Voice-NAT network. To enable provisioning service for hosted IP Phones, refer to section 3.4.6.2 Access Control to conditionally enable the provisioning data traffic.

6 OAMP, Security and Fraud Protection

6.1 User Account Configurations

6.1.1 Local Account Settings

To add or modify user privileges to access the ESBC console.

Navigate to **System** > **Administrator**.

To modify attributes of existing users, click <Setting> icon under column Action. To add a user, click <Add> button. Note that the User ID “admin” is the default administrator ID, and cannot be deleted from the system.

Figure 130. User account administrative page

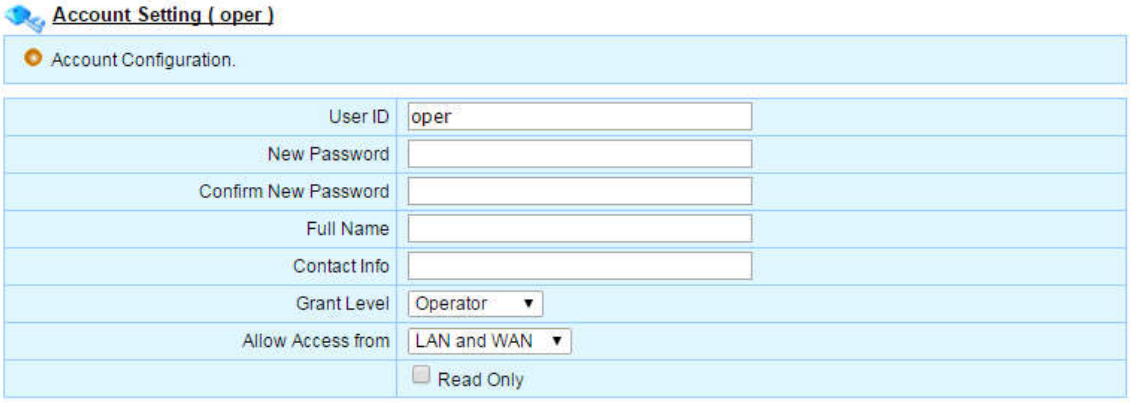


Figure 131. Adding or modifying a user account attributes

Account Setting	Description
Grant Level	Three levels of user accounts. • Admin: access full configurations of the system • Technician: access configurations for installing the ESBC to the enterprise. • Operator: access configurations for connecting the ESBC to the PBX. See “Chapter 8 Installers and Operators” for detailed descriptions.
Allow Access from	Access management console, including WEB and CLI, via the following three interfaces: WAN&LAN | LAN |WAN
Read Only	View configurations only. Applicable to Operator and Technician account types.

6.1.2 TACACS+ Account Settings

TACACS+ allows centralized management of user-device authentication policies, instead of managing those policies separately on each device. A remote TACACS+ server stores user login credentials for TACACS+ enabled devices. During the user login process, the ESBC interacts with the TACACS+ server by providing the user authentication details and gets the results of either “PASS” or “FAIL” for login attempts from the server, and finally responds to the user appropriately.

TACACS+ authentication is provided for the ESBC WEB management console, including both HTTP and HTTPs, through the following interfaces:

- The active WAN interface
- LAN management interface (if enabled)
- LAN VoIP/NAT interface (if management port is not enabled)

If TACACS+ support is enabled, the following scenarios apply:

1. When the TACACS+ server is reachable, the ESBC will only allow users to login with TACACS+ credentials (i.e., user login with locally stored credentials will NOT be allowed). If the user fails to login after 3 attempts, an SNMP trap will be sent to the trap manager.
2. When the TACACS+ server is not reachable or the WAN connection is down, the ESBC will then allow a user to login with locally stored credentials. These users may remain logged in without further TACACS+ authentication until the ESBC inactivity timer times out, or the user actively logs out.
3. While logged in with TACACS+ authentication, the ESBC monitors the TACACS+ connection state for each WEB user request.

4. If the TACACS+ server reachability state changes from up to down during a user's login period, the user will be logged out when they next attempt a WEB user request, and will need to re-login with local credentials.

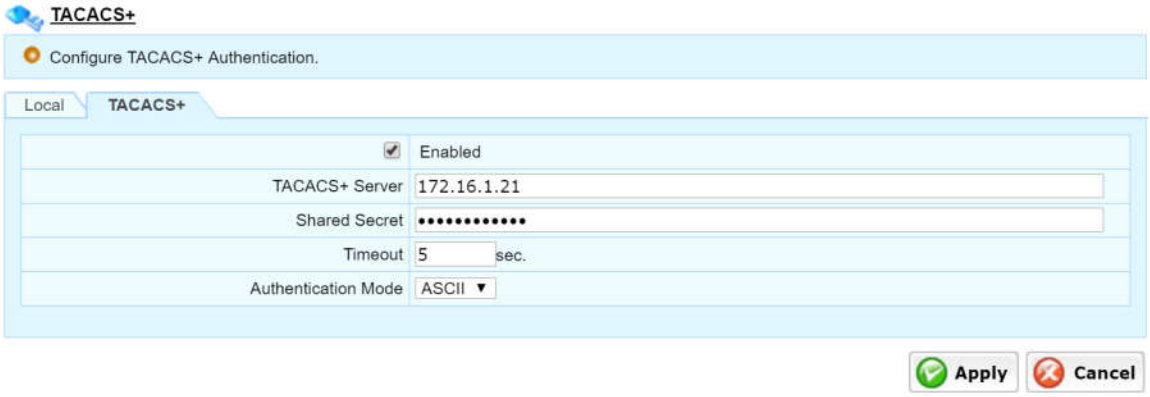


Figure 132. Configure TACACS+ Server

TACACS+ Setting	Description
Enabled	Enable or disable the TACACS+ authentication mode. Default is disabled.
TACACS+ Server	The IP address or FQDN of the TACACS+ server
Server Secret	The shared secret between the TACACS+ server and the ESBC for data obfuscation.
Timeout	The length of time in seconds that the ESBC waits for a TACACS+ response before failing the authentication.
Authentication Mode	Select a TACACS+ authentication mode from the drop-down list: • ASCII • PAP (Password Authentication Protocol) • CHAP (Challenge Handshake Authentication Protocol)

6.2 System Time

To deploy voice services in the field, it is often necessary to have all related devices synchronize with a precise timing mechanism. The ESBC can be configured to synchronize current time with specified Network Time Servers or obtain time information from the connected administrative console, i.e., the computer accessing the admin WEB console. If there is no synchronization source, the ESBC uses the Linux native time which is Jan 01, 2000.

The system time synchronization status shows “Synchronized” once the ESBC is synchronized with the specified SNTP server. Note that the synchronization status will be kept until the next reboot even if the ESBC subsequently loses synchronization with the SNTP server. Once the system reboots, the ESBC will update the status.

Navigate to **System** > **System Time**.

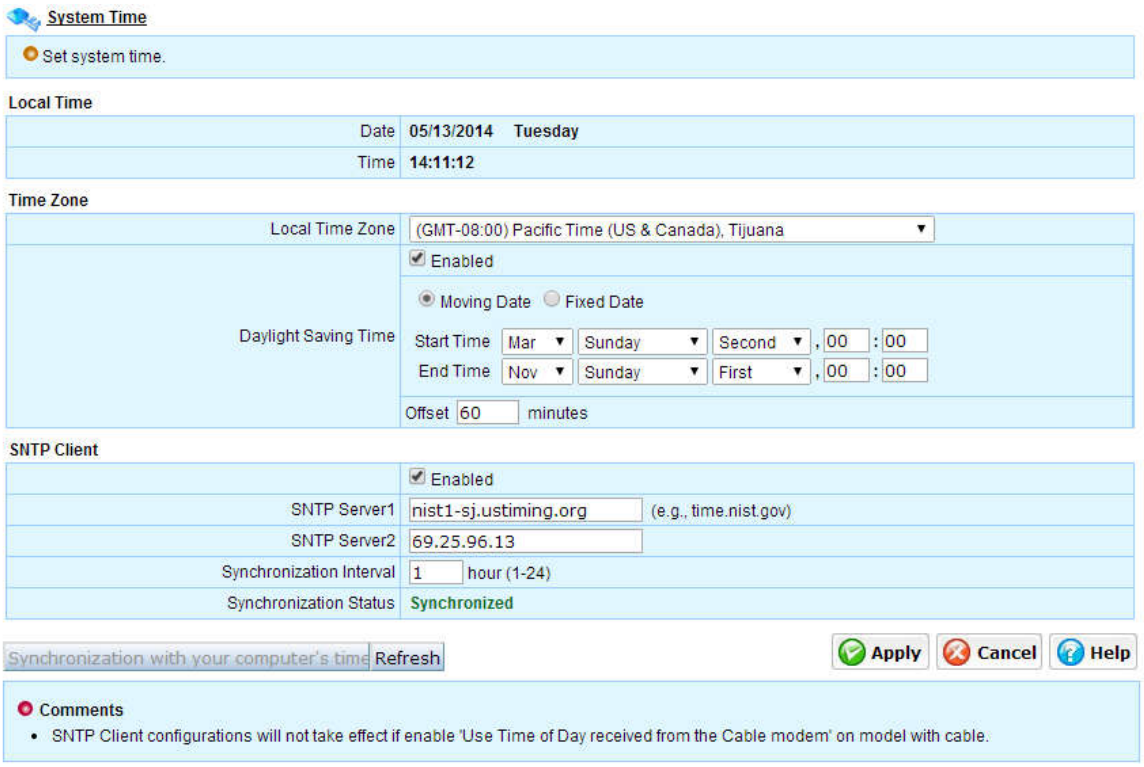


Figure 133. Configuring the ESBC system time

Item	Description
Local Time	Local time information is based on the “Time Zone” specified on this page. The ESBC sends standard “UTC” time information to OAM&P servers, e.g., SIP server or SNMP server.

Time Zone	Select the Time Zone where the ESBC is physically deployed. • Enable or disable the “Daylight Saving Time” (DST) option. • If DST is enabled, choosing “Moving Date” or “Fixed Date” for the starting and ending date of DST. In North America, the start day usually is the second Sunday in March, and end day is the first Sunday in November, and hence “Moving Date” should be selected. Offset: The offset refers to the offset between DST and “normal time.”
SNTP Client	• Configure the ESBC to synchronize time with network time servers (SNTP server). • Enable the SNTP Client to synchronize time with the SNTP server. Input the FQDN or IP addresses of the target SNTP servers (primary and secondary). • Synchronization Interval. (default is 2 hours) • The ESBC displays the synchronization status.
Synchronization with your computer’s time	When “SNTP Client” is unchecked (disabled), the ESBC may synchronize time information with the management computer (the device running the web console). Click this button to trigger time sync immediately.

6.3 Management Control

The management control function allows the service provider to shut down voice services temporarily for maintenance purposes, and then restart these services automatically by configuring a prescheduled time or under certain conditions.

Navigate to **System** > **Management Control** page.

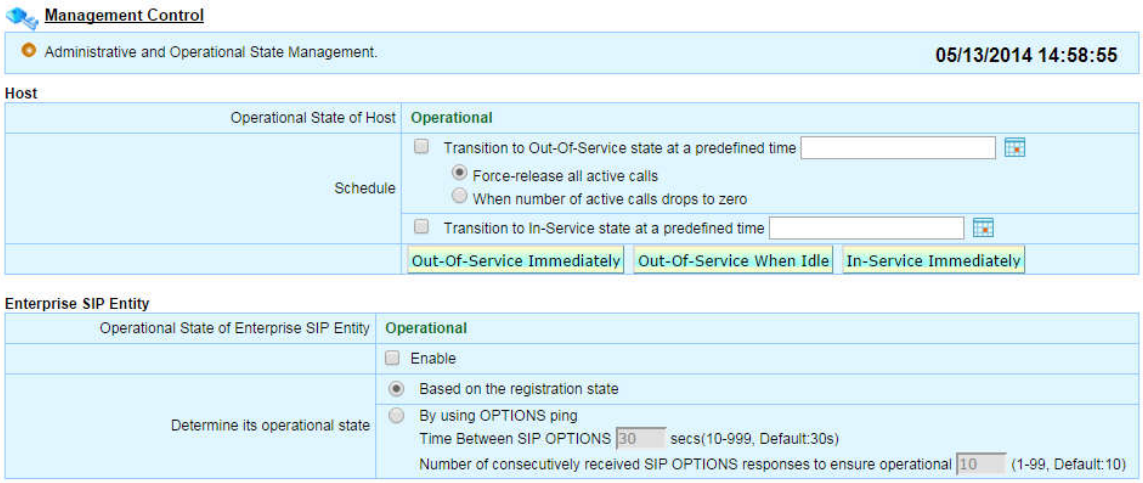


Figure 134. Administrative and Operational State Management

Host	Description
Current Time Display	The current system time is displayed at the upper right corner.
Operational State of Host	Displays the current state: Operational, or Not Operational. If the ESBC is in “Not Operational” state, some condition exists which prevents the ESBC from providing Business Voice service. For example, it could be administratively out-of-service, or missing some critical configuration data, or other physical issues which block its ability to provide service.
Schedule	• Transition to out-of-service state at a predefined time. There are options available only to shut down the service under configured conditions. • Transition to In-service at a predefined time.
Actions	• Out-of-Service immediately: Have the ESBC enter maintenance state NOW with no conditions. • Out-of-Service when idle: Have the ESBC enter maintenance state automatically when there are no active calls. • In-service immediately: Have the ESBC enter service mode NOW.

Enterprise SIP Entity	Description
Enable	When this feature is enabled, if the Enterprise SIP entity (i.e., SIP PBX) is removed from service, the ESBC will release all active calls towards the service provider network, then de-register the Enterprise SIP Entity from the service provider network.
Operational State of Enterprise SIP Entity	Displays the current state of the connected SIP PBX "Operational", or "Not Operational". If the IP PBX is in "Not Operational" state, some condition exists which is preventing the IP PBX from providing Business Voice service. For example, it could be administratively out-of-service, or the IP PBX may be missing some critical configuration data, or other physical issues may be present which are blocking its ability to provide service.
Determine its operational state	Select the desired method for the ESBC to determine whether the SIP Entity is operating properly. • Based on registration state. This is applicable to the SIP Entity which uses SIP REGISTER to connect to the ESBC (as opposed to the Static operational mode). • By using SIP OPTIONS ping. This SIP method is used to send keep-alive messages. This is applicable to the SIP Entity which supports SIP OPTIONS pings.

6.4 Maintenance

The ESBC maintenance features are used to change the system status.

Navigate to **System** > **Maintenance** page.

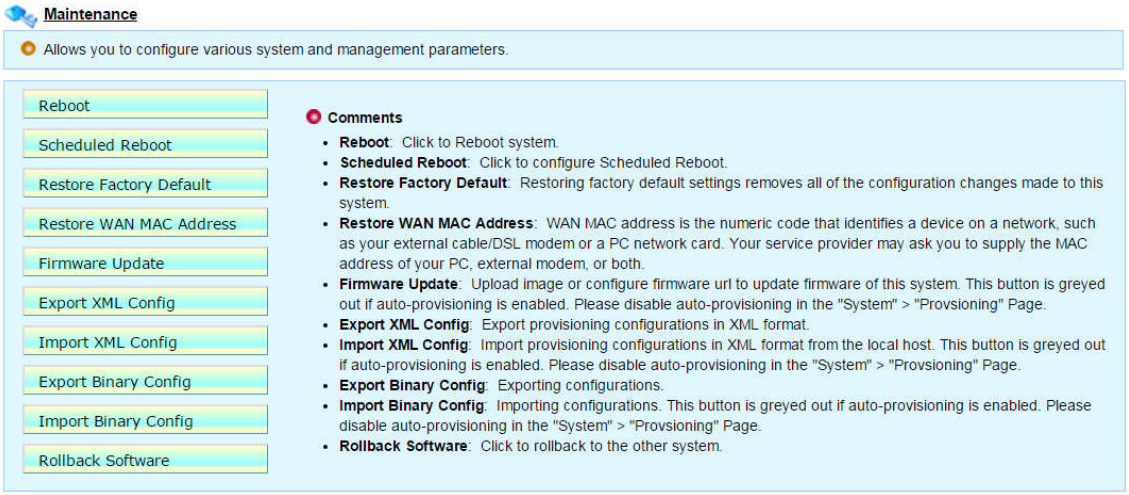


Figure 135. System Maintenance

6.4.1 Reboot |Scheduled Reboot | Restore Factory Default | Restore WAN MAC Address

The comments for each function displayed on this page are self-explanatory.

Item	Description
Reboot	Performs a soft-reboot process.
Scheduled Reboot	Enables the ESBC to perform an automatic administrative reboot at a scheduled frequency and time. Scheduled reboot will be triggered under the following conditions: • Uptime is more than 20 hours, and • When the system time is synchronized with a SNTP server. Configuration Control • Disable or enable scheduled reboot settings. • Frequency: every day | every week, during the configured one hour window. If there is an active call, the reboot will be delayed until 5 seconds after the active call ends in the time window.
Restore Factory Default	Clears all updates and restores the unit to default values. It is

	recommended that the config is backed up (Export XML or Binary) before performing this task.
Restore WAN MAC Address	Restore the ESBC WAN MAC address back to its factory value. Use this command only when the WAN MAC has been cloned.

6.4.2 Firmware Update | Rollback Software

Item	Description
Firmware Update	Either upgrade or downgrade the ESBC's currently running firmware to the target version. The ESBC supports updating the image file stored at • local drive (Upload). Browse the image file from the local drive, or • enter the URL complete path, such as protocol://FQDN<http/tftp>_server/image_file_path_and_name Note: • Supported protocol types: **HTTP** and **TFTP** • The ESBC always updates the firmware on the backup partition and migrates the backup database to the current database. Once the ESBC updates successfully, the system reboots and the partitions swap. • When auto-provisioning is enabled, the firmware update button is greyed out on the WEB GUI.
Rollback software	The rollback function allows the backup partition image and database to become active. With the rollback function, the database will not be migrated. The ESBC will simply swap partitions.

6.4.3 Import XML or Binary Config | Export XML or Binary Config

Item	Description
Export XML Config Export Binary Config	Manually backs up the ESBC database to an external file. • XML format. Files can be edited after export. Use the ESBC provisioning tags to assign appropriate values. XML files are firmware version independent. If the currently running firmware version does not recognize any provisioning tags, the ESBC just ignores them. • Binary format. Files include the complete database of the ESBC for the current partition. Binary files are firmware dependent. They can be imported to a system running the same firmware version as was used during the export. The file is read only and ensures data

	integrity with the system. Note that manual backup can be used together with the “auto backup” function, see section 6.5 for a detailed description.
Import XML Config Import Binary Config	Manually restore an ESBC configuration file to the current system. Note: • When auto-provisioning is enabled, the “Import” buttons are grey out. • Importing a Binary Config needs a “matching” firmware version. • During import of an XML Config, since it is version dependent, some parameters could be ignored if the target firmware version does not support these parameters.

6.5 Auto backup system configuration periodically

The ESBC system configurations can be scheduled and backup to an external FTP server automatically and periodically. If Auto Backup fails, the ESBC logs the event to Audit Log.

Navigate to **System** > **Auto Backup**.

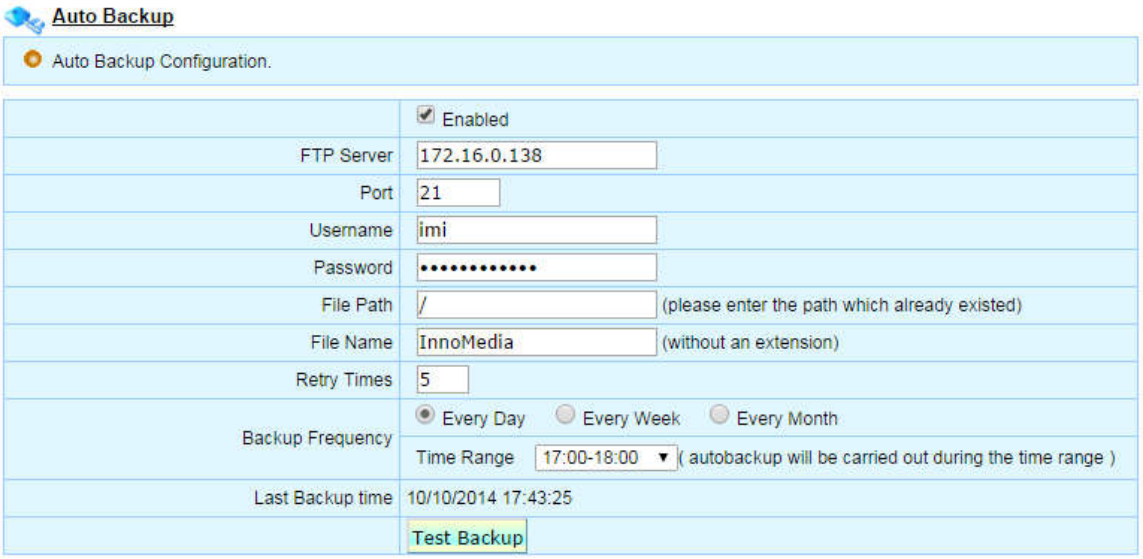


Figure 136. Auto backup the ESBC Configuration to an FTP Server

Item	Description
Enabled	Check this box to enable the Ftp
FTP Server	The FTP server IP address or FQDN
Port	Communication port for FTP protocol. The default port number is 21.
Username	Enter the FTP Username provided by the FTP server administrator.
Password	Enter the FTP password provided by the FTP server administrator.
File Path	Enter the FTP server path for the ESBC to upload config file.
File Name	Enter a file name designed for the ESBC config file. If the auto-backup file name is left blank, the filename will be completed using the following format: [Pilot Number][WAN IP Address][Model][SW version] [DATE/TIME]
Retry Times	FTP server connection retry times
Backup Frequency	Frequency: Every Day | Every Week | Every Month Time Rang. The auto backup procedure is activated anytime within the specified clock hour.
Test Backup	Click to “Backup” NOW.

Note that the ESBC does not perform auto backup procedure if one of the following conditions happens.

- “Every Day” is selected. The current hour is behind the scheduled hour.
- “Every Month” is selected. The current date is behind the scheduled date.
- The previous backup event happens less than one hour (3,600 seconds) of the current time.
- The ESBC boot-up time is within the scheduled auto backup hour.

6.6 Battery Status

The ESBC unit comes with a built-in smart battery for continuing voice service in case of power outage event happening.

Navigate to **System** > **UPS**.

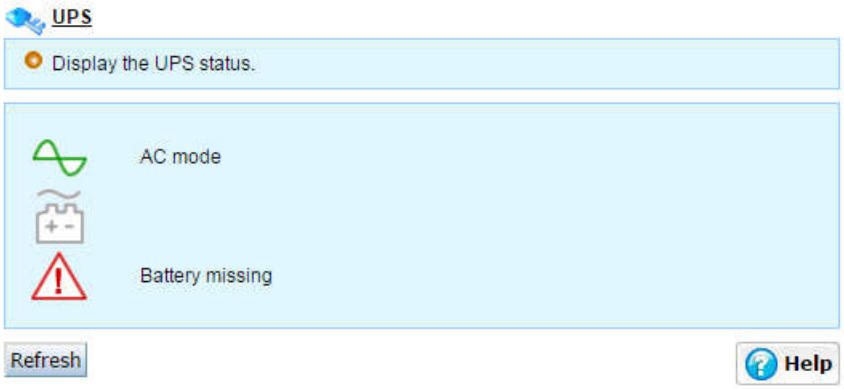


Figure 137. The UPS-Battery status page

6.7 Call History and Logs

6.7.1 Call History Settings

In order for the ESBC to record each call detail record, navigate to **Telephony** > **TOOLS** > **Call History** > **Setting**. Check all desired call types to enable call history for these calls.

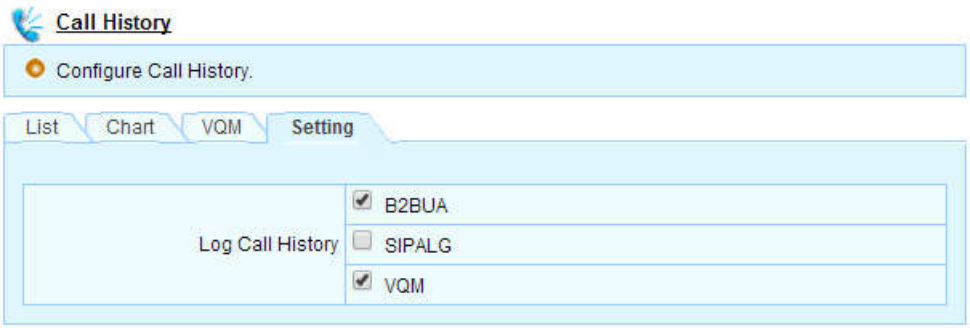


Figure 138. Call History Setting Options

Call History Setting	Description
Log Call History	B2BUA: check this item to enable Call History Records for SIP Trunk Service Calls SIPALG: check this item to enable Call History Records for Hosted Service Calls
VQM	VQM: check this item to enable voice quality measurement (R-Factor, MOS calculation) for selected call types (B2BUA and/or SIP ALG)

Note: To enable "Log Call History" is needed for the Voice Quality calculation and display. See section 6.8 for details.

6.7.2 Call History Record

The ESBC records all calls through the system, if configured to do so. The Call History page displays calls with various filtering criteria: Call Type, Caller ID, Callee ID, and Dates. The CDRs can be exported to an external csv file for accounting use. Navigate to **Telephony** > **TOOLS** > **Call History.**

Note: It is necessary to enable "Log Call History" in order for the ESBC to calculate and display Voice Quality Measurement information (see section 6.7.1)

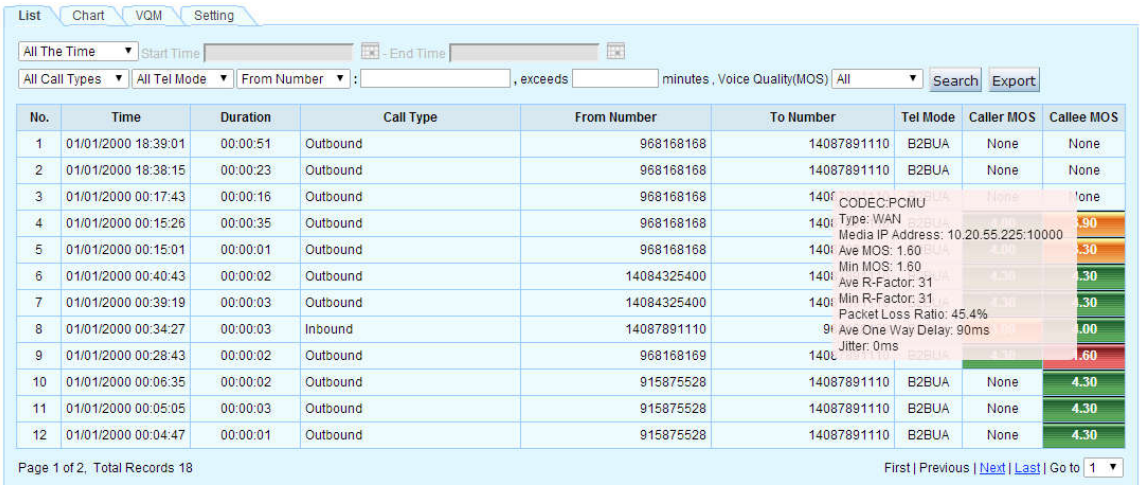


Figure 139. Call history records (list view)

As the mouse points to any MOS score area, the associated voice metrics statistics are displayed.

Call History	Description
Caller MOS Callee MOS	Based on IP network factors, the ESBC calculates R-factor and MOS scores for each call. Hence, only IP connections can generate VQM results and MOS scores are displayed for both parties for IP connections. No MOS scores are available for PRI connections.
Search	Filter and display records according to the various inquiry criteria
Export	Export call history records to a text based CSV file

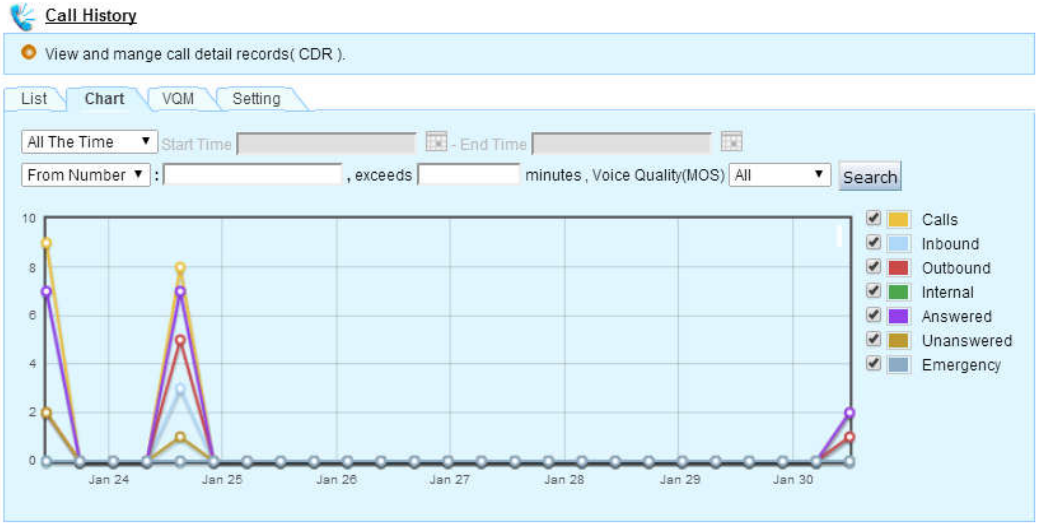


Figure 140. Call History (Chart View)

6.7.3 VQM (Voice Quality Measurement)

Call statistics are gathered at the end of each call based on packets received and sent by the ESBC during the call.

Figure 141. Call History (Voice Quality details)

VQM Factors	Description
Time	The time and date of end of call reporting.
PRI, SETA, SIP-ALG, B2BUA; trace-id	Category of call and Internal trace number (SETA: SIP End Point Test Agent)
Start-time; end-time	Internal time stamps of a session: start – end.
Call-side	The calling party, either from the WAN or LAN side.
Direction	OUTBOUND or INBOUND call
qos-type	BE: best effort; UGS: cable modem guaranteed service flow
codec-type	Audio codec types
NLR	Network Packet Loss Rate
RTD	Round trip delay
IAJ	Inter-arrival jitter
amos	Average mos
mmos	Minimum MOS
Afactor	Average R-factor
mfactor	Minimum R-factor

CALLID	Call ID in SIP message
LocalID	The ESBC User Account in SIP URI format
RemoteID	The remote User Account in SIP URI format
OrigID	The caller User Account in SIP URI format
LocalAddr:PORT	The ESBC WAN interface IP address: RTP Port
RemoteAddr:PORT	The remote SIP entity IP address: RTP port
SSRC	Synchronization source identifier uniquely identifies the source of a stream. The synchronization sources within the same RTP session will be unique.
Remote Group/Local Group	The User Accounts of remote party and of the ESBC

6.8 Voice Quality Measurement and SLA Assurance

Rating Factor (R-Factor) and **Mean Opinion Score** (**MOS**) are two commonly used measurements of overall VoIP call quality. The ESBC employs R-factor to evaluate and rate the quality of telephony voice traffic and translate it to MOS values. The voice quality performance of each call over time is calculated from the RTP traffic to/from the service provider side on the ESBC WAN interface, and on the ESBC LAN interface with connected SIP devices for both SIP Trunk (B2BUA) and Hosted (SIP-ALG) services.

R-Factor: The R-Factor provides a powerful and repeatable way to assess whether a data network is capable of carrying VoIP calls with high quality. A value is derived from network factors such as latency, jitter, and packet loss per *ITU-T Recommendations.* Typical scores range from 50 (poor) to over 90 (best quality).

MOS: Subjective MOS scores are gathered through exhaustive tests with large groups of human listeners who listen to audio and give their opinion of the call quality. The *ITU-T Recommendations P.800 series* describes how these tests are conducted and the types of scores that are produced.

There are strong correlations between R-Factor and MOS and the mapping between R-factor and MOS-CQE scores is described in ITU-T G.107 standard. The ESBC calculates the voice quality metric R-factor and uses this mapping to translate it to a MOS value.

To configure the Voice Quality parameters and view statistics, navigate to **Telephony** > **TOOLS** > **Voice Quality**

6.8.1 Voice Quality Parameter Basic Configuration

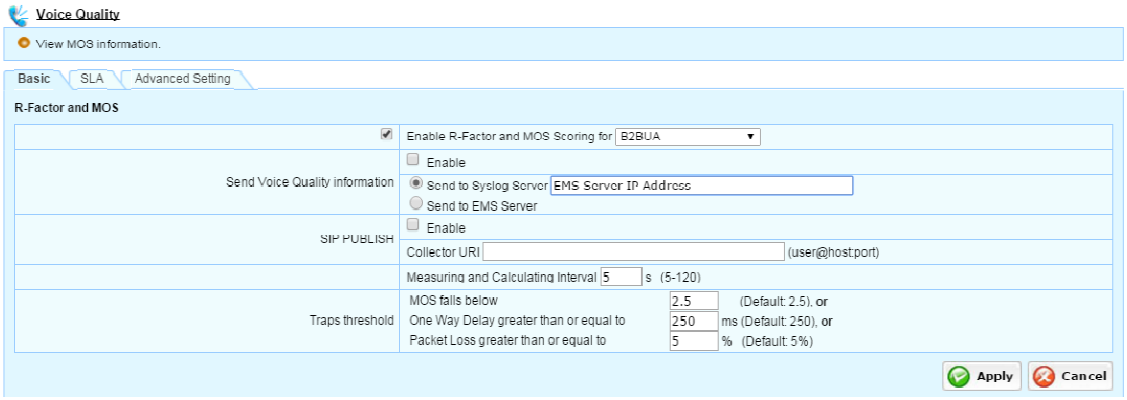


Figure 142. Voice Quality Parameter Basic Configuration

Note: In order to display Voice Quality Chart and SLA information, it is necessary to enable the Call History feature which is described in section 6.7.1.

R-Factor and MOS	Description
Enable R-Factor and MOS scoring	Check the option box if you would like to enable R-factor and MOS scoring calculations for calls. Calls for SIP Trunk services and SIP Hosted (ALG) modes are both supported.
Send Voice Quality Information to Syslog Server	Enter the Syslog Server IP addresses or FQDNs to which the ESBC sends VQM (voice quality measurement) statistics for each call. The ESBC supports the ability to send VQM messages to up to 3 (three) syslog servers simultaneously.
Send to EMS Server	If the InnoMedia EMS is deployed alongside the ESBC, enter the IP address or FQDN of the EMS server.
SIP PUBLISH	Enable this feature and enter the Telemetry Collector URI to allow statistics to be carried in SIP PUBLISH messages.
Measuring and calculating interval	R-factor calculation interval in seconds. The range is from 5 to 120 seconds.
Traps threshold	Enable the ESBC to send SNMP traps to an SNMP server if the voice quality is considered poor. Please note that the “Send SNMP trap alarm” and “Alarm” features must be enabled (see section 2.7, and section 6.9.12.7)

6.8.2 Voice quality statistics line chart

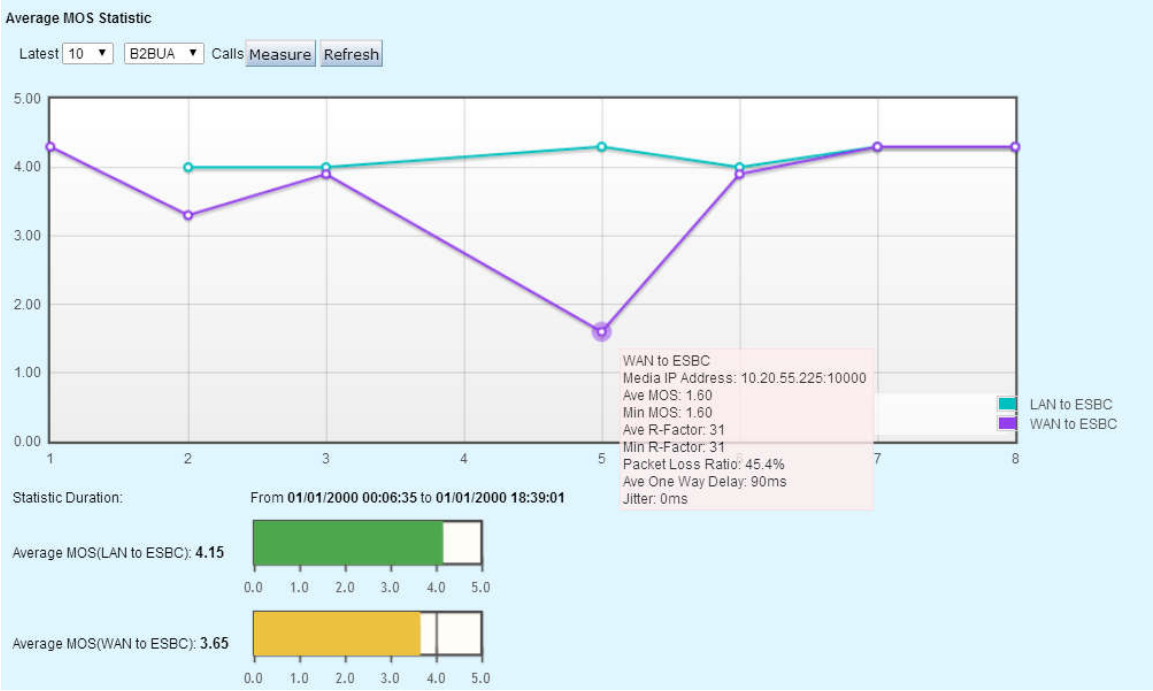


Figure 143. Voice Quality Statistics Line Chart

Point the mouse to any spot on the chart to display the related VQM parameters. Figure 143 shows that the WAN side “Packet Loss Ratio: 45.5%” is most likely the factor which results in a low MOS value for this call leg.

R-Factor and MOS	Description
WAN to ESBC	The RTP (media) packets coming from the WAN side to the ESBC.
LAN to ESBC	The RTP (media) packets coming from the LAN side to the ESBC.

6.8.3 SLA (Service Level Agreement) Parameters

The SLA page provides a high level view of overall performance for quality of experience.

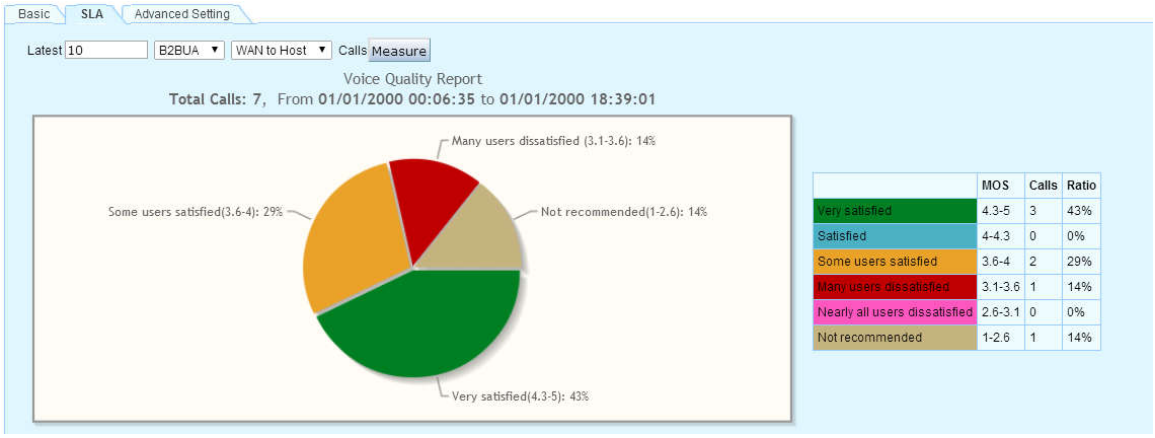


Figure 144. Overall quality of experience display

6.8.4 Advanced Settings

The values in this table are used as some of the input parameters to the R-factor calculation. Do not change these values unless instructed to do so.

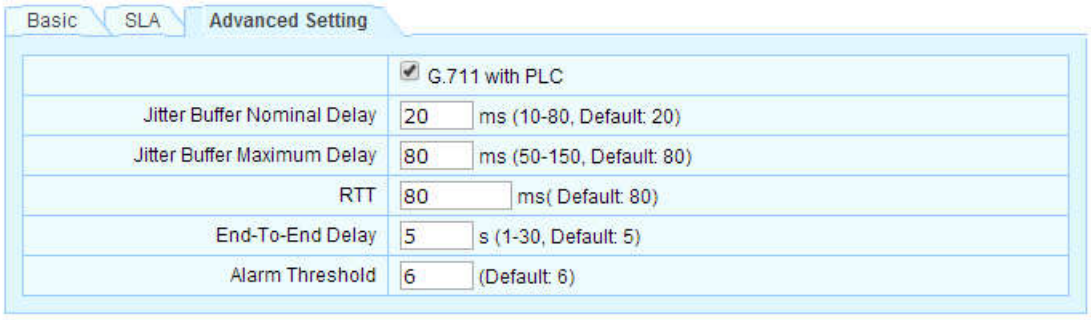


Figure 145. R-Factor Parameter Setting

R-Factor and MOS	Description
G.711 with PLC	If this option is checked, ESBC will calculate the R-factor as if the remote endpoint supports PLC with G.711.
Jitter Buffer Nominal Delay/ Jitter Buffer Maximum Delay	A jitter buffer holds datagrams at the receiving side. If x ms nominal delay setting is used, the first voice sample received is held for x ms before it is played out. The maximum delay is used as an input parameter to the R-factor calculation.
RTT	Round Trip Time If RTT cannot be collected by the ESBC, this configuration value can be used as a default RTT to calculate R-factor
End-To-End Delay	If endpoint to endpoint delay is greater than the configured value, it will consider the delay as excessive and send out a SNMP Trap alarm.
Alarm Threshold	Specify how many occurrences of poor MOS values will trigger the ESBC to send out an SNMP Trap alarm.

6.9 Alert Notification:

The ESBC will send SNMP traps and/or email alerts to the specified destination(s) described in **sections 2.7 and 2.8** when any of the following alerting events occur.

6.9.1 SNMP Trap Alarms

Navigate to **System** > **Alert Notification**.

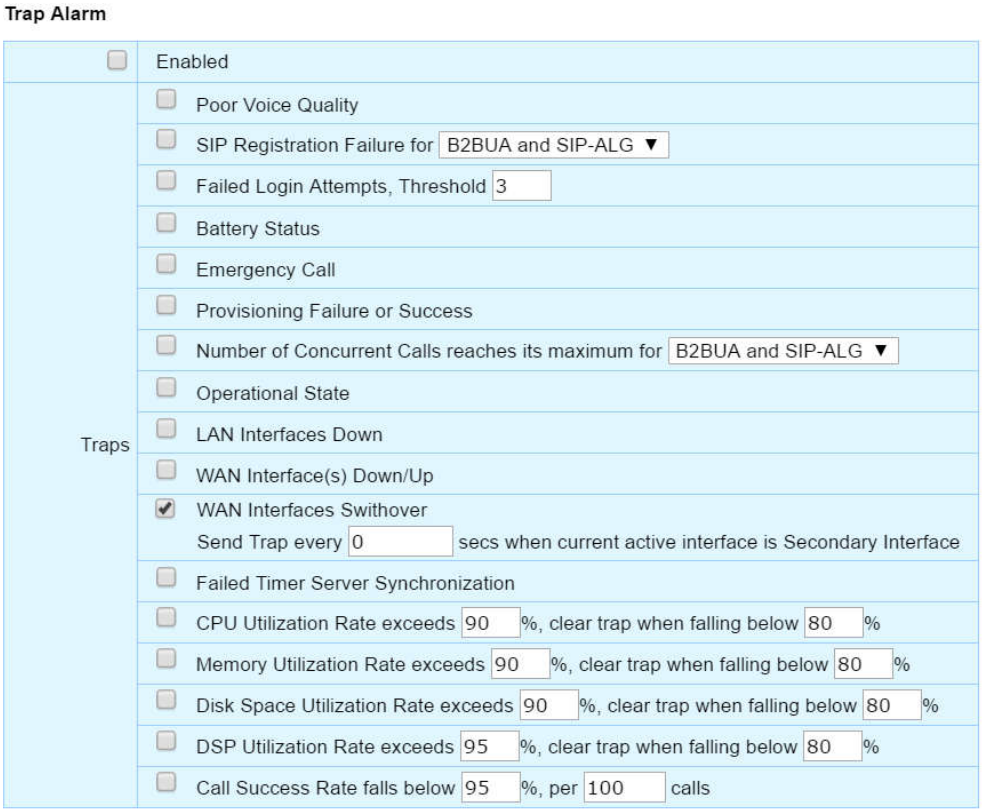


Figure 146. SNMP Trap Alarm Configuration

SNMP Traps	Description
Enabled	Check this box to process any of the selected traps.
Poor Voice Quality	When one of the voice quality levels: MOS, one way delay, or packet loss exceeds the specified threshold (see Section 6.8).
SIP Registration Failure	When one or more SIP User Accounts fail to register to the proxy server (see Section 4.2).
Failed Login Attempts	Number of failed login attempts to the ESBC WEB console or SSH connections. (The ESBC Audit Log logs all login attempt information, including date-time, user name and source IP (see

	Section 6.10.6).
Battery Status	When the battery is low, missing, or status changes.
PRI Alarm	When a PRI alarm happens (e.g., D channel down, or any red/yellow alarms). Please refer to **"***ESBC 9x80 PRI SNMP and MIBs***"** document for managing Alarm events through SNMP traps.
Emergency Call	When there is an emergency call from a PBX subscriber (see Section 4.7).
Provisioning Failure or Success	When there are provisioning events (see Section 2.10).
Number of concurrent calls reach its maximum	When the number of calls reaches the integrated licensed number, the ESBC sends out a trap (See Section 2.2).
Operational State	When the operational state of the ESBC changes (See Section 6.3).
LAN Interface Down	When the LAN interface is not accessible for management access, e.g., data link layer down, lost connection to the connected switch or cable unplugged (See Section 2.5.2.1).
WAN Interface Down/Up	Send out alert trap for WAN interface link-down and link-up events. These traps are only sent out when the WAN interface recovers.
WAN Interfaces Switchover	Send out a trap when the WAN connection is transitioning to the secondary interface. A periodic trap with a configurable timer can also be sent when the secondary interface is in use.
Failed Timer Server Synchronization	When the ESBC loses connection with the SNTP server and fails to synchronize system time (See Section 6.1).
System Utilization Rate exceeds x and clear trap when falling below y	CPU | Memory | Disk Space: send out alert trap when the number exceeds x %; and clear trap when falls below y%. By default, x=90, and y=80. Call Success Rate: send out alert trap when the number falls below x% per y calls. By default: x= 95, and y=100

6.9.2 Email Alarms

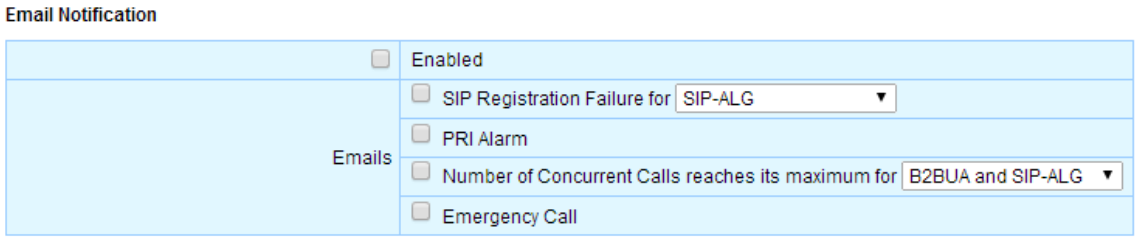


Figure 147 Configuring Email Notification Items

Please refer to Section 6.9.1 for descriptions of associated alerts.

6.10 Security

6.10.1 System access control: Basic

Navigate to **System** > **Access Contr**ol > **Basic**.

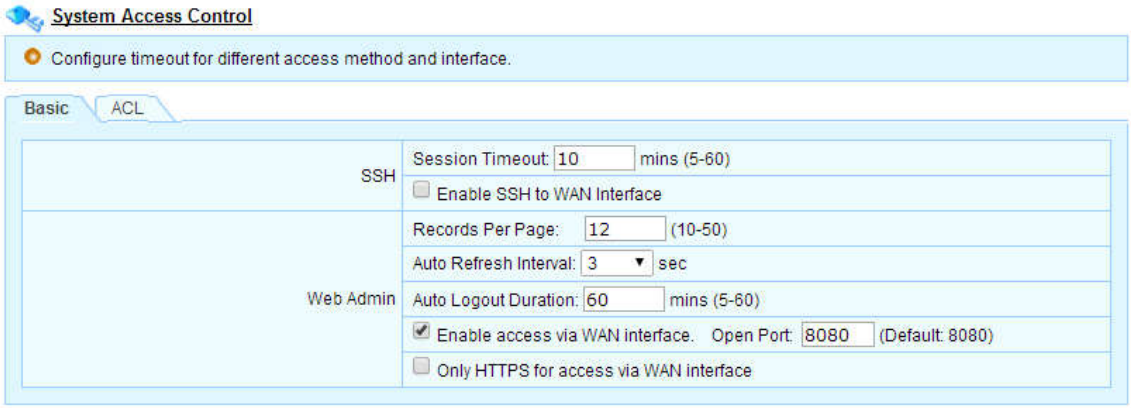


Figure 148. System Access Control -- Basic

System Access Control- Basic	Description
SSH	
Session Timeout.	Default: 10 minutes. If there is no action on the SSH console, the ESBC automatically closes this connection.
Enable SSH to WAN Interface	The ESBC, by default, does not allow SSH access via the WAN interface for security purposes.
Web Admin	
Records per Page	The number of records displayed per page on the Audit log, SIP Firewall Log, or other logging tables. Default: 12 records per page.
Auto Refresh Interval	The interval for the ESBC WEB GUI to refresh the system current status. Default: 3 seconds.
Auto Logout Duration	If there is no action on the WEB GUI, the ESBC automatically logs out the user from the WEB console.
Enable access via WAN interface	Access WEB GUI via WAN. The ESBC, by default, disables access via the WAN interface for security purposes. Change the access port when necessary. Default port: 8080. (http://WAN_IP:8080)
Only HTTPS for access via WAN Interface	When this item is enabled, the ESBC always switches the access protocol to HTTPS. (e.g., http://WAN_IP:8080 and/or https://WAN_IP)

6.10.2 IP Layer Protection: Access Control List

The use of an ACL (Access Control List) is recommended to protect the ESBC on the specified interfaces from undesired access attempts, scanning etc. The ACL rules for the WAN and LAN interfaces are processed independently. That is, if the rule is configured for the WAN, it applies to traffic on the WAN interface only. Traffic that comes into the ESBC is compared to the ACL rules based on the order in the list. The ESBC continues to match the packet against the rules until it finds a match. If no matches are found, the traffic is dropped. In other words, if the ACL feature is enabled, there is an implicit 'drop' rule that will block packets that do not match any rules for that interface.

For detailed configuration guideline, please refer to the *ESBC Application Notes – ACL Configurations*. Navigate to **System** > **Access Control** > **ACL**.

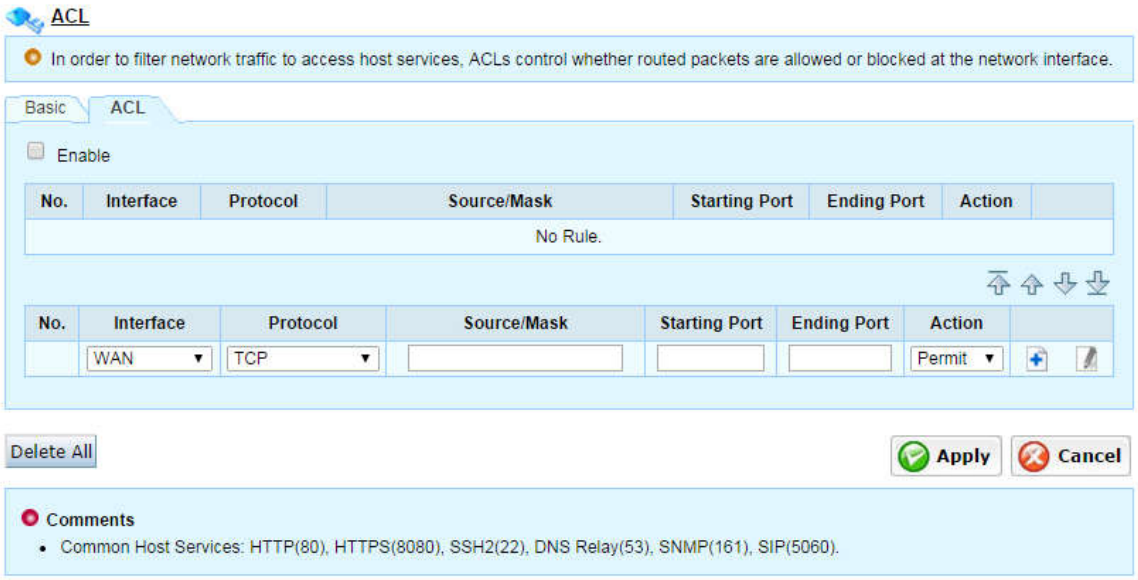


Figure 149. The ESBC Access Control List (IP Layer Protection)

ACL	Description
Enable	Special Note: When the ACL feature is enabled, there is an implicit drop rule that will block packets that do not match any rules for that interface.
No.	Sequential number of rule
Interface	Apply ACLs to WAN or LAN
Protocol	TCP, UDP, and TCP+UDP
Source/Mask	Source IP or Network /Mask: (e.g., 0.0.0.0/0.0.0.0, 192.168.1.1/255.255.255.255, 172.16.1.1/255.255.0.0, or

	192.168.1.0/24)
Starting Port | Ending Port	Service Port, indicating the TCP or UDP port numbers. A service port range can be supported
Action	"Permit", "Deny" and "Drop." "Deny" means reject a request, and "Drop" means no response for a request

6.10.3 SIP Layer Protection

6.10.3.1 SIP Firewall Rules

The ESBC SIP firewall rules (SFW) enable the operator to design and select predefined rule-sets that define all messages to be examined by the ESBC. SFW is script-based, and follows the same structure and syntax as SIP Header Manipulation Rules (SHMR).

SFW first filters all traffic according to the firewall rules (if a firewall rule script is applied) before handling the resulting traffic that needs to be processed. Firewall rules can be applied independently on the following interfaces:

- ESBC WAN interface
- ESBC LAN interface (only for NAT-Voice ports)

SIP Firewall rules are presented in a structured manner within a script file which can be imported into the ESBC for the applicable LAN or WAN interface.

For SIP Trunk Telephony Services: Navigate to **Telephony** > **ADVANCED** > **Firewall**.

For Hosted Telephony Services: Navigate to **Telephony** > **SIP ALG** > **Firewall**

Please refer to "ESBC SIP Firewall Rules" for instructions on composing firewall rules.

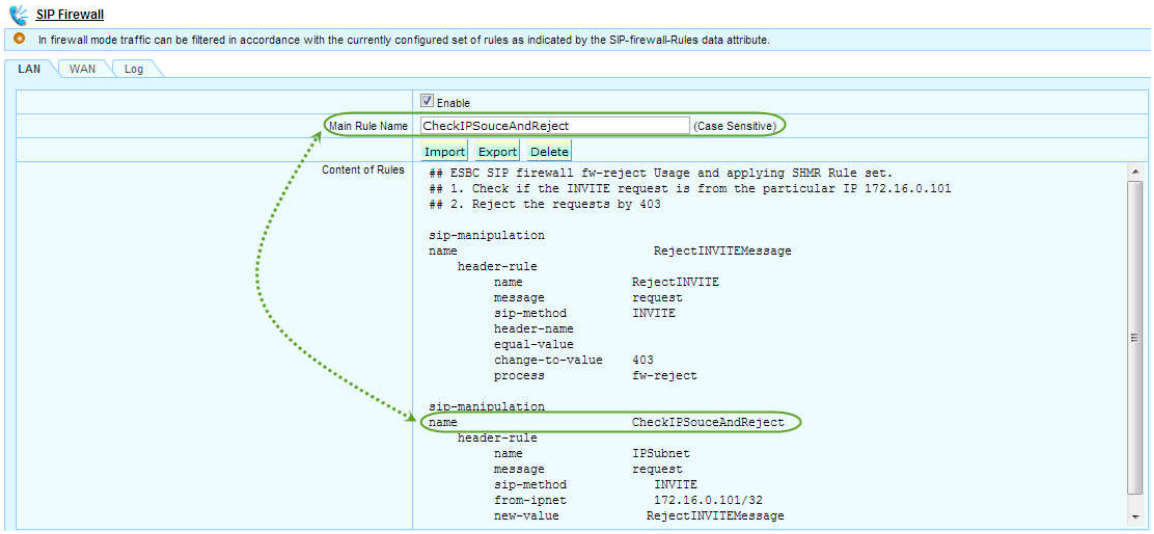


Figure 150. ESBC SIP firewall rules: importing scripts

SIP Firewall Rules allow administrators to define the following categories.

- IP, IP-Subnet, port
- From and To directions
- SIP method
- Black or white lists

For any access attempts which match the SIP firewall rules, the ESBC processes those access attempts according to the defined actions (e.g., disposition event)

- fw-accept: allow the messages to pass through the ESBC
- fw-drop: discard the messages
- fw-reject: reject incoming sip messages and reply with sip error response code.
- sip-manip: firewall rule set manipulation.

All the actions taken by the ESBC SIP firewall rules are recorded in the Firewall log (see section 6.10.4).

6.10.4 SIP Firewall logs

All access attempts which match the ESBC SIP firewall rules are logged.

For SIP Trunk Telephony Services: Navigate to **Telephony** > **ADVANCED** > **Firewall > Log**.

For Hosted Telephony Services: Navigate to **Telephony** > **SIP ALG** > **Firewall > Log.**

The administrator can search for sources of attack according to the following recorded items for each access attempt.

Date & Time | Protocol | SIP Identity | Source IP | Destination IP | Source Port | Destination Port | Message Type | Disposition Event | Reason

6.10.5 SIP Message | domain | IP examination to prevent attack or fraud

The ESBC can be configured to block incoming SIP messages from both LAN/WAN interfaces (or either one) by examining their originating IP or domains.

6.10.5.1 Fraud from the LAN interface

An INVITE from an unregistered LAN side rogue CPE can be examined and/or blocked from initiating an outbound call.

Navigate to **Telephony** > **SIP-PBX** > **PBX SIP Profile**. Choose the relevant target PBX profile for the LAN SIP PBX or SIP User Agents.

See section 4.4.1.3 for a detailed description.

6.10.5.2 Fraud or attack from the WAN interface

The IP or domain of incoming SIP messages will be examined and/or be blocked to prevent spoofed source IP, SIP attacks or fraud from the WAN. The ESBC blocks all REGISTER attempts from the WAN interface.

Navigate to **Telephony** > **SIP TRUNKS** > **Trunk SIP Profile**. Choose the relevant target trunk sip profile for the SIP server in the service provider network.

See section 4.2.3.3 for detailed descriptions.

6.10.6 Audit logs

The ESBC logs major operations which are issued by the system administrator or events triggered by the system.

Operations such as:

Login/Logout | Importing/Exporting Configurations | WAN/LAN interface settings | Provisioning settings | Firmware updates | DMS/EMS settings | Maintenance commands | T1/E1 D-Channel up/down | etc.

Navigate to **System** > **Audit Log** to view or export audit log records.

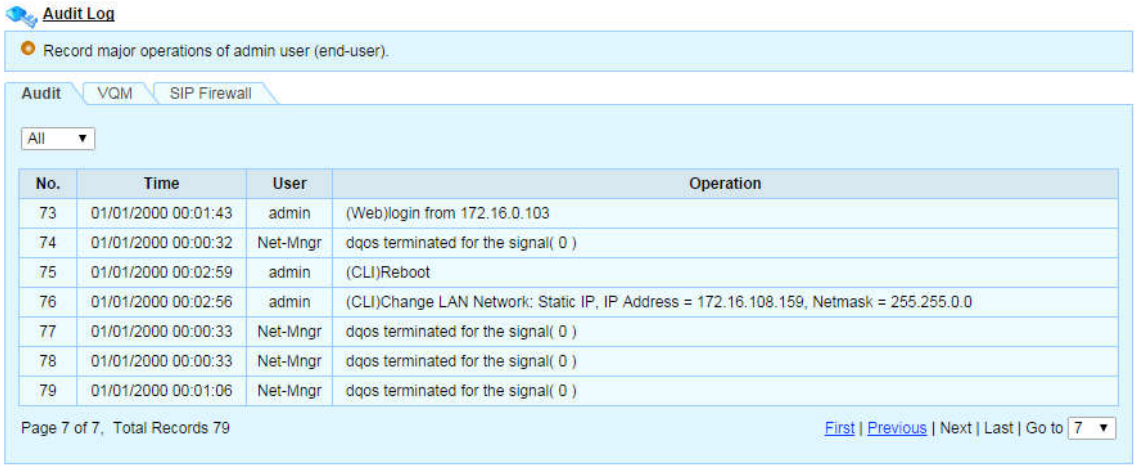


Figure 151. The ESBC Audit Log page

6.11 System Monitor

The ESBC system monitor page provides the system administrator with real time system performance statistics. They include:

Uptime | CPU Utilization | System Memory Utilization | Disk Space |

Statistics on WAN Interface (link layer statistics) | Statistics on LAN Interface (link layer statistics)

Navigate to **System** > **Monitor**.

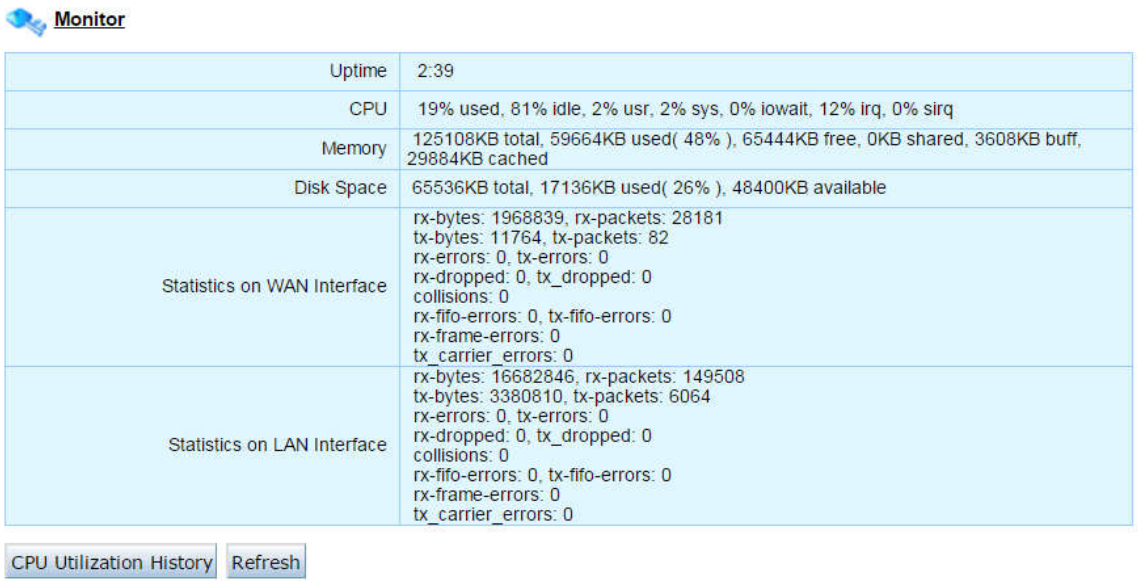


Figure 152. Real time system performance statistics

Note: Please refer to Standard IF-MIB descriptions for Statistics on WAN/LAN Interfaces.

6.11.1 CPU Utilization History

Click the CPU Utilization History button shown in Figure 152 to display ESBC CPU utilization historical statistics.

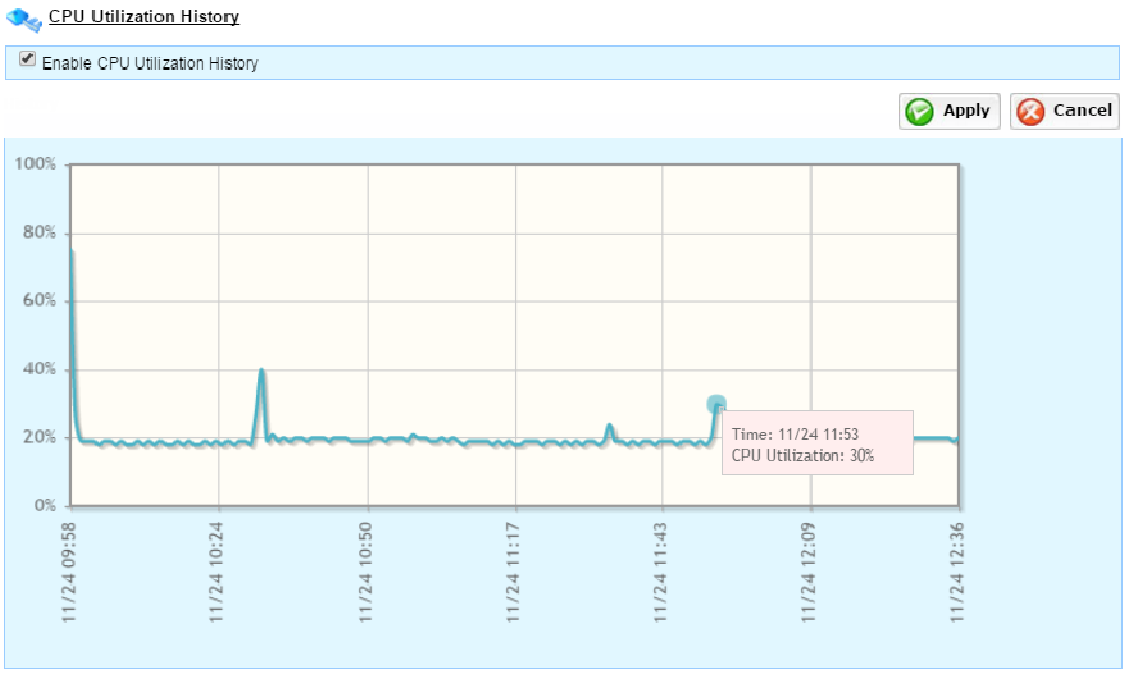
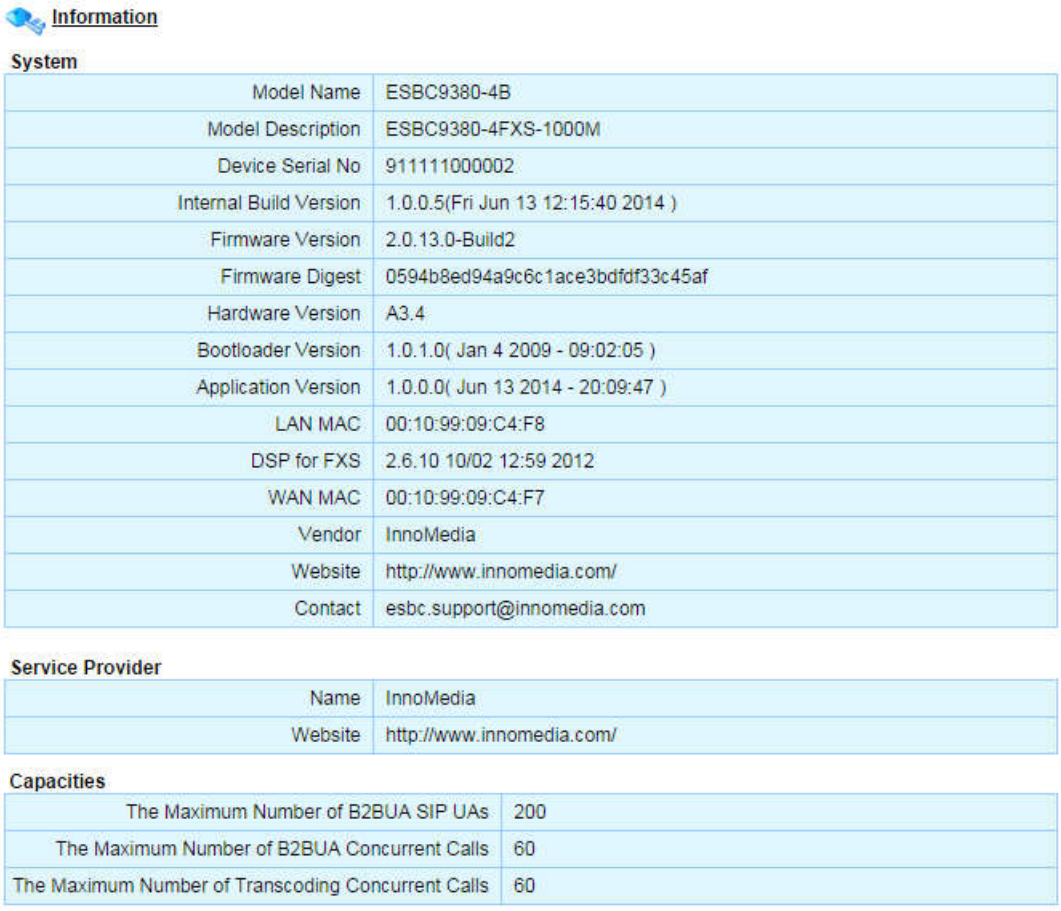


Figure 153. CPU utilization rate historical chart

6.12 System information

Navigate to the **System** > **Information** page to view system information for the current ESBC unit.



Information

System

Model Name	ESBC9380-4B
Model Description	ESBC9380-4FXS-1000M
Device Serial No	911111000002
Internal Build Version	1.0.0.5(Fri Jun 13 12:15:40 2014)
Firmware Version	2.0.13.0-Build2
Firmware Digest	0594b8ed94a9c6c1ace3bdfdf33c45af
Hardware Version	A3.4
Bootloader Version	1.0.1.0(Jan 4 2009 - 09:02:05)
Application Version	1.0.0.0(Jun 13 2014 - 20:09:47)
LAN MAC	00:10:99:09:C4:F8
DSP for FXS	2.6.10 10/02 12:59 2012
WAN MAC	00:10:99:09:C4:F7
Vendor	InnoMedia
Website	http://www.innomedia.com/
Contact	esbc.support@innomedia.com

Service Provider

Name	InnoMedia
Website	http://www.innomedia.com/

Capacities

The Maximum Number of B2BUA SIP UAs	200
The Maximum Number of B2BUA Concurrent Calls	60
The Maximum Number of Transcoding Concurrent Calls	60

Figure 154. System information page

7 Diagnosis

7.1 Test Calls

Navigate to **Telephony** > **TOOLS** > **Test Agent**.

Test calls are used to verify successful registration from the SIP Trunk Interface to the service provider network. Enter the telephony number to be called (for example, the technician's cell phone number) to complete the test. The called number will be sent a .WAV file and a series of tones for approximately 60 seconds. See Test Agent (section 4.3) for detailed configuration.

7.2 Syslog

7.2.1 Debugging syslog

The debugging syslog is used for debugging system issues or interoperating with the network or other equipment, e.g., SIP server, PBX or any other devices. The debugging syslog is disabled by default. “Debugging syslog” is designed for debugging purposes only. Uncheck all options by choosing “None” during normal operation.

Navigate to the **System** > **Syslog** > **Debugging** page.

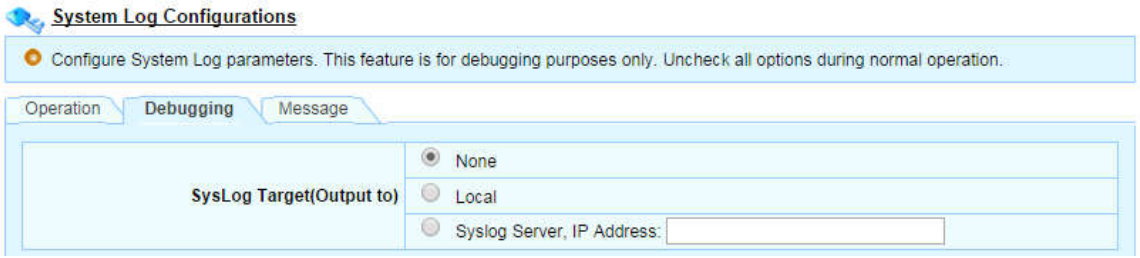


Figure 155. Debugging Syslog Configuration

The debugging syslog messages can be output to

- Local. The local flash memory. Click the “Message” tab to display debugging syslog messages. If the number of records exceeds 5000, the new record will overwrite the earliest record. Click the “Message” tab to view, and/or export syslog messages saved at the local storage location.
- An external syslog server. Enter its IP address.

Debugging syslog messages are categorized as follows:

Kernel | System and Network | B2BUA | SIP ALG | PRI

Check or uncheck related features for sending out debugging syslog messages.

7.2.2 Operational syslog

Operators can determine system operational states and/or special incidents happening on deployed ESBC units by making use of the operational syslog service.

Navigate to **System** > **Syslog** > **Operation** page.

When valid operational syslog server IP addresses or FQDNs are entered, the ESBC sends operational syslog messages to the server once any of the following events occur. The ESBC supports the ability to send syslog records to up to 3 (three) syslog servers simultaneously.

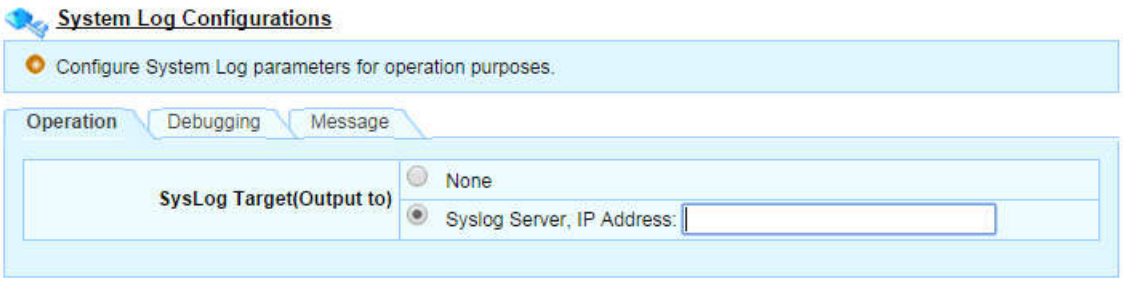


Figure 156. Enabling operational syslog server

When a valid operational syslog server IP address is entered, the ESBC sends syslog messages to the server once any of the following events occurring.

Operational Syslog Events	Description
Network Interface Link Status	Internet and LAN: Network up or Network down.
DNS query	Success or Failure
send REGISTER send DEREGISTER	
Receiving network initiated de-registration	
REGISTER	Success/Failure (Received 480 Temporarily Unavailable, Timer F expired)
UA send SUBSCRIBE	Success/Failure (including initial SUBSCRIBE and refresh)
Update UA configure	
LAN side PBX REGISTER	
LAN side PBX DEREGISTER	
Bootup	Sending version information
NTP server sync	Success/failure
ESBC NTP synced	Success/failure
Resolve NTP server DNS name	Failure
Proxy discovery	Failure
INVITE to the Server	Failure (with Timer B expiring, or other causes)
Check AOR of Notify	Warning. All AORs in the NOTIFY message of reg-event do not match the UA.
Receive INVITE from other Proxy or address	
SIP firewall notifications	When a message is dropped or rejected by the ESBC's internal SIP Firewall, the ESBC sends a warning message with "WARNING: SIP Firewall" as the prefix.

DQoS failures	Send alert messages when B2BUA reserve DQoS fails. • B2BUA: if reserve DQoS fails, send an error log; if the DQoS is BE, send a warning log. • SIP ALG: if reserve DQoS fails, send a warning log.

7.3 Call Trace

The built-in ESBC call trace capability can be used to capture and store SIP+PRI signaling traces. The <Tracing> utility displays call signal traces in a ladder diagram format, and the <Capture> utility captures packets of all calls during the recording period, including SIP, RTP, ISDN Q921 and ISDN Q931 packets.

7.3.1 Tracing - Ladder diagram

Navigate to **Telephony** > **TOOLS** > **Call Trace**.

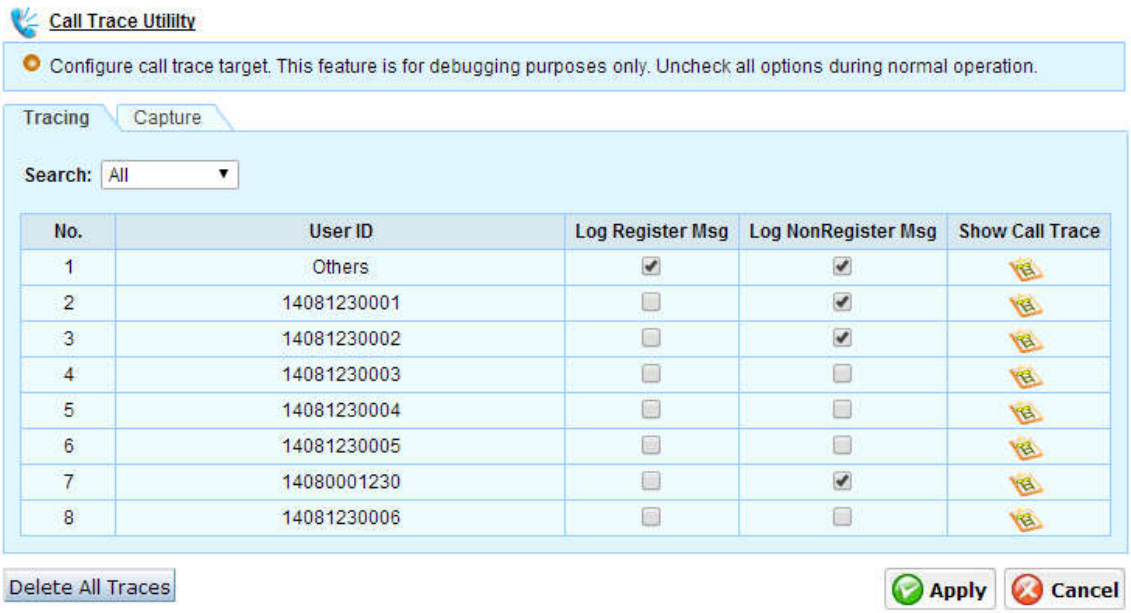


Figure 157. SIP Account List

Select target User IDs and track SIP and PRI (Q.931) signals for all connections.

- Log Register Msg: display SIP flows for SIP UA REGISTER exchanges with the SIP Server.
- Log NonRegister Msg: display SIP and PRI flows for all call attempts.

Click the <Show Call Trace> button to display information for the selected User ID account.

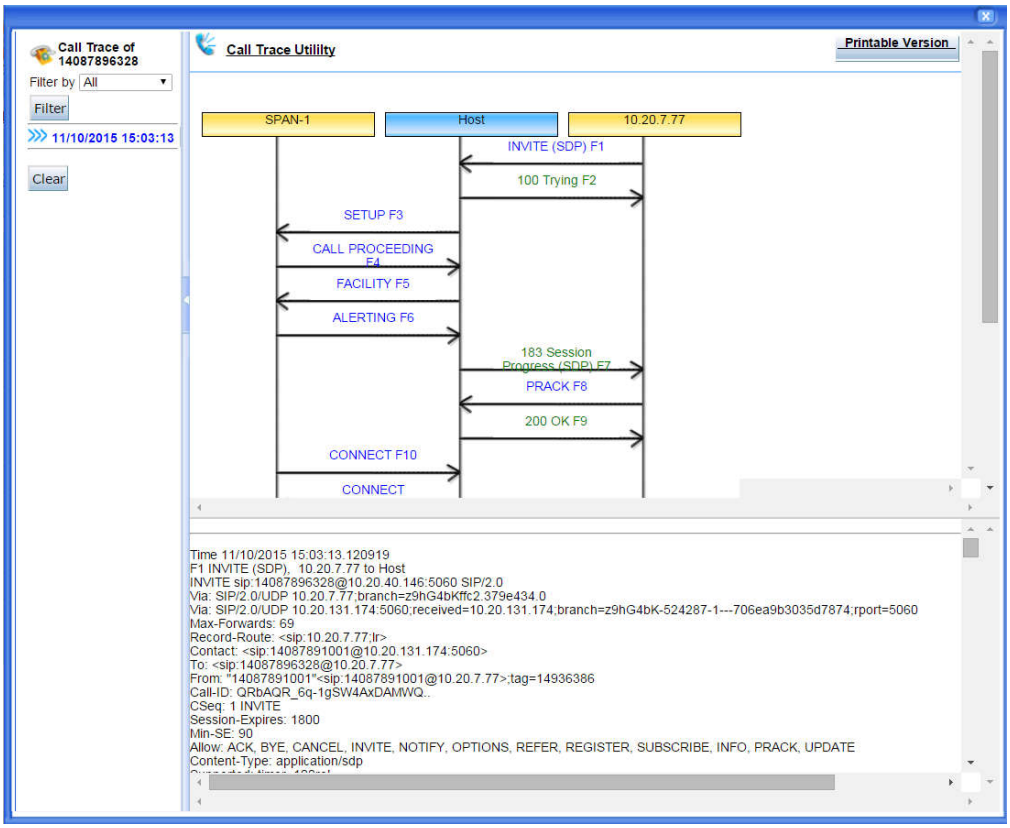


Figure 158. SIP and PRI signal trace of the selected account and call

7.3.2 Packet capture

The ESBC can capture packets, including signaling and voice packets for live calls . Recorded files can be opened by wireshark or other utilities.

The ESBC built-in capture tool is capable of capturing packets for both LAN and WAN, ingress and egress directions, and outputs them to a single file. This feature greatly aids in the investigation of telephony related interoperability issues. Packet types include the following.

- Signaling: SIP signaling (for both WAN and LAN)
- Signaling: ISDN (Q931, Q921)
- Media: RTP packets

Captured files can be uploaded directly to a remote ftp server, local storage or external USB storage.

Navigate to **Telephony** > **TOOLS** > **Call Trace** > **Capture**.

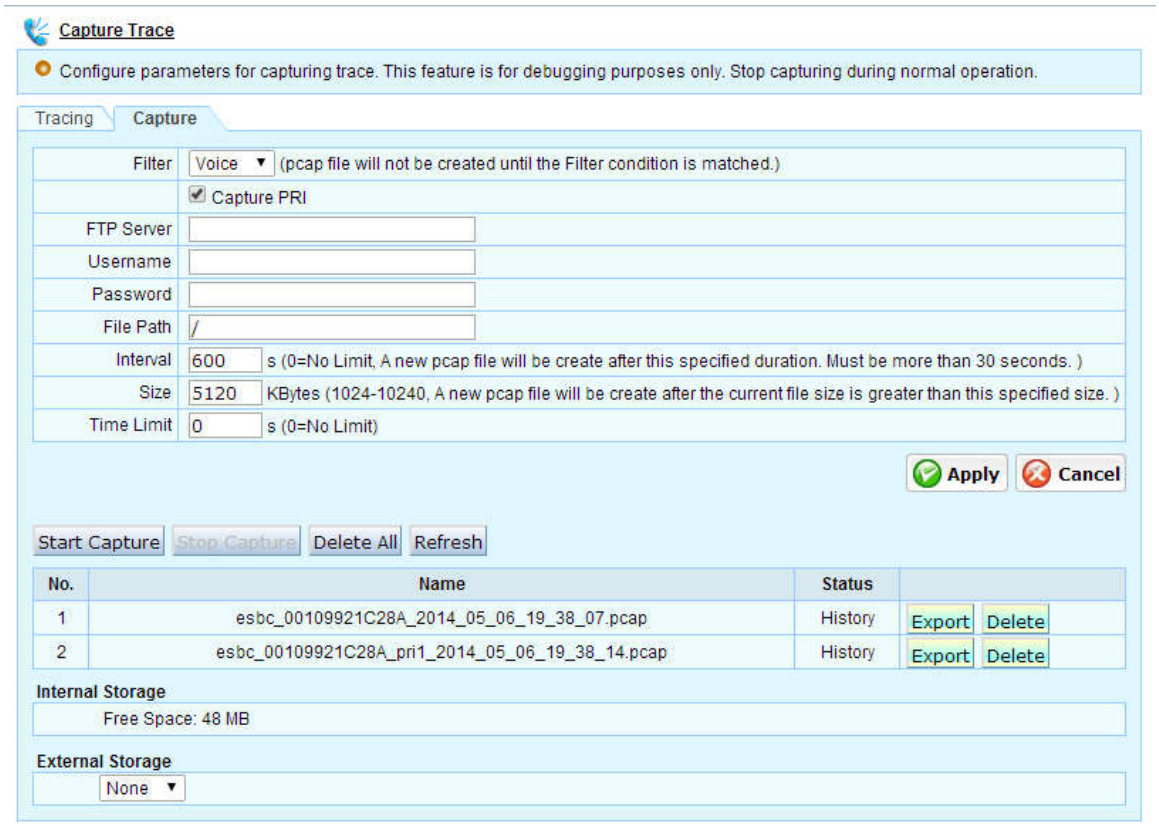


Figure 159 Packet Capture

Capture Trace	Description
Filter	Voice, both SIP and RTP packets Signaling, SIP packets Media, RTP
Capture PRI	Enable this item when it is necessary to capture ISDN Q931 and Q921 signals in call flows (note that you must use q931 or q921 in the filtering box within wireshark).
FTP Server	Enter the IP address or FQDN of the target FTP server
Username/Password	The Username and password to access the FTP server
File Path	The path for the ESBC to upload captured files
Interval	In seconds. 0=No Limit, A new pcap file will be created after this specified duration. Must be more than 30 seconds.
Size	Kbytes. 1024-10240. A new pcap file will be created after the current file size is greater than this specified size.
Time Limit	The capturing duration in seconds (0=No Limit).

Storage	Description
Internal	Storing captured files to the ESBC internal flash memory.
External	If an USB flash disk is inserted, it is possible to specify an external storage space for the captured files.

7.4 Network diagnostic utilities

The network administrator can use the test tools on the ESBC WEB console to verify the connectivity of the ESBC system and trace the path of data through the network.

Navigate to **Network** > **Advanced** > **Diagnostics**.

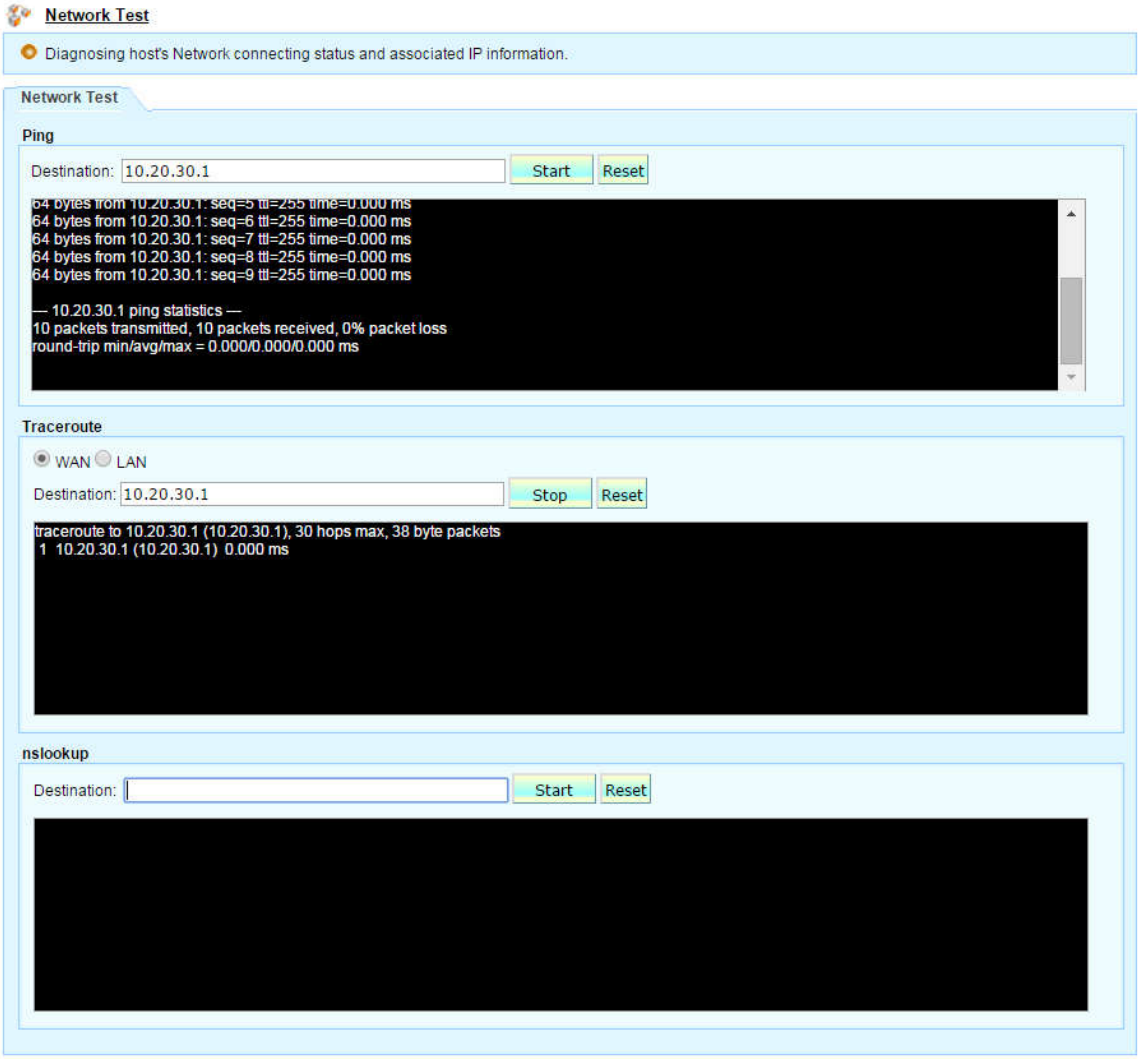


Figure 160. The network diagnosis utilities

7.4.1 Ping Test

Ping is the most common test used to verify basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address. Successful ping tests do not guarantee that all data messages are allowed between the system and the test IP address.

7.4.2 Traceroute

Traceroute is used to track the progress of a packet through the network. This test can be used to verify that data destined for a WAN device reaches the remote IP address via the desired path. Similarly, network paths internal to a company can be traced over the LAN to verify the local network topology. Selecting LAN or WAN determines the path (direction) of the trace route test.

7.4.3 Nslookup

Nslookup is a network administrator tool for querying the Domain Name System (DNS) to obtain the corresponding IP address of a domain (example, "abc.com"). It will also do a reverse name lookup and find the host name for an IP address you specify.

8 Installers and Operators

In addition to the administrator web console, the ESBC supports simplified web pages for technicians or operators to configure the ESBC's features. Technicians are technical staff who install the ESBC at customer sites. Operators are enterprise administrative staff who facilitate daily routine jobs for the company's telephony or data network.

Note that when logged in with the administrative account, the Technician and Customer features are displayed under the "Installation" category of the web console.

8.1 Installation via Technican WEB console

1. Configure the technician's PC with an appropriate IP address within the same network as the ESBC LAN interface, either for the management port or NAT and Voice port.
2. Assuming the PC connects to the NAT and Voice port, start your web browser and enter e.g., http://172.16.1.1, in the address field to connect to the ESBC. The login page will appear. The default user name is "tech" and the password is "123". Click the login button to enter the ESBC main page.

8.1.1 The ESBC9x78, 9x28, ESBC10K series models

Once logged in, the main page displays as follows. The telephony configurations either have been provisioned or are pre-configured on the target ESBC system. The technician may have to perform the following tasks in order to install the ESBC to the enterprise's network.

- Network configurations and/or diagnosis
- Have IP-PBX connect/register to the ESBC

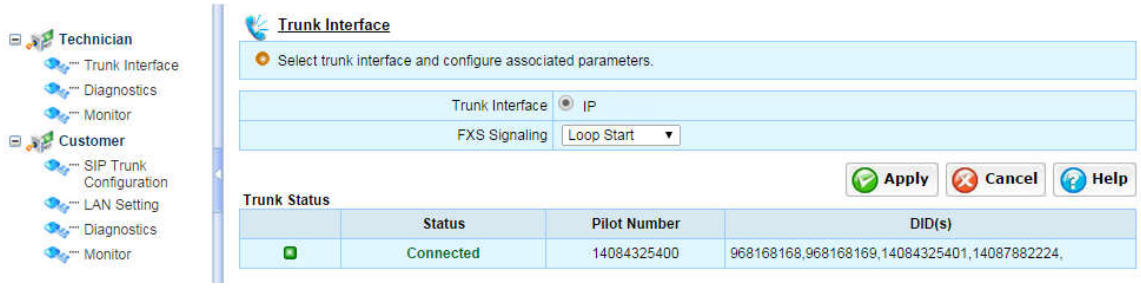


Figure 161. the Technician main page

8.1.1.1 Technician-Trunk Interface

Trunk Interface	Description
Trunk Interface	IP. The LAN side of the ESBC should have the SIP PBX or SIP Phone connected or registered.
FXS Signaling	Choose between the "Loop Start" or "Ground Start" options.

Trunk Status	Status: The ESBC registration/connection status with the north bound SIP server. Pilot Number: The registration agent (RA) configured via the administrative web GUI or provisioned. Note that if there is no RA configured on the ESBC, the DID(s) column is blank. DID(s): The DIDs (SIP user agents) configured with the RA. DIDs with no RA configured will not be displayed.

8.1.1.2 Telephony and Network Diagnostics

Navigate to **Technician** > **Diagnostics**, or **Customer** > **Diagnostics**.

Call Test. Use the ESBC built-in SIP device, the Test Agent (TA), to verify telephony connectivity and voice quality with the service provider network. See section 4.3 for detailed descriptions.

On the WEB GUI, the technician needs to enter the destination phone number which could be a designated test call TN, or the technician's mobile phone TN, and press "Dial".

Network Test. Use Network Test utilities to verify the network connectivity status from the ESBC to both LAN and WAN interfaces. See section 7.4 for detailed descriptions.

8.1.1.3 Connect/Register SIP PBX to the ESBC

To have the SIP PBX connect/register to the ESBC LAN Voice-NAT interface, navigate to **Customer** > **SIP Trunk Configuration**.

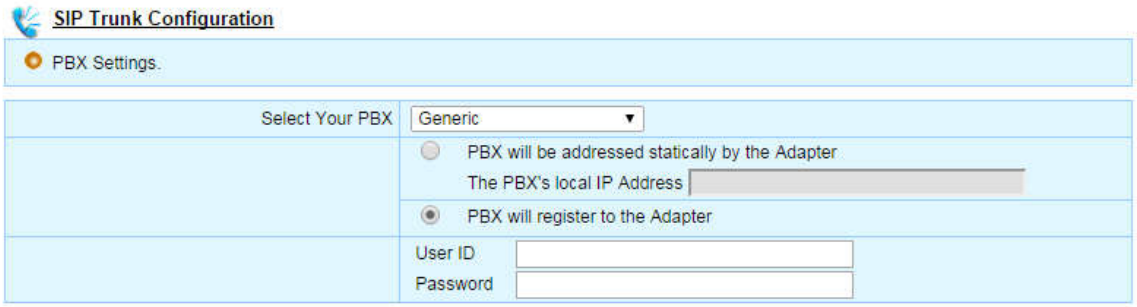


Figure 162. The SIP PBX Connecting/Registering Page

PBX Settings	Description
Select Your PBX	Choose the SIP PBX type that connects to the ESBC. The selection items are the PBX profiles configured in the administrator web console. See section 4.4.1 for detailed descriptions.
Choose among "Static" or	Static mode: PBX will be addressed statically by the Adapter. The

REGISTER operation mode	IP address of the target SIP PBX (the interface toward the ESBC) is needed for this mode. REGISTER mode: PBX will register to the Adapter. The User ID and Password (of the main pilot number) for the SIP PBX is needed to register to the ESBC for this mode.

8.1.1.4 LAN Setting

Configure the ESBC LAN Voice-NAT interface to interoperate with the enterprise telephony network. Enable the DHCP Server option when the ESBC is to offer IP addresses to SIP UAs or hosts in this network. Note that when the SIP PBX connects to the ESBC by the static operation mode, it is recommended that this SIP PBX is configured with a static IP which should be out of the range of the DHCP server IP address range.

Navigate to **Customer** > **LAN Setting**.

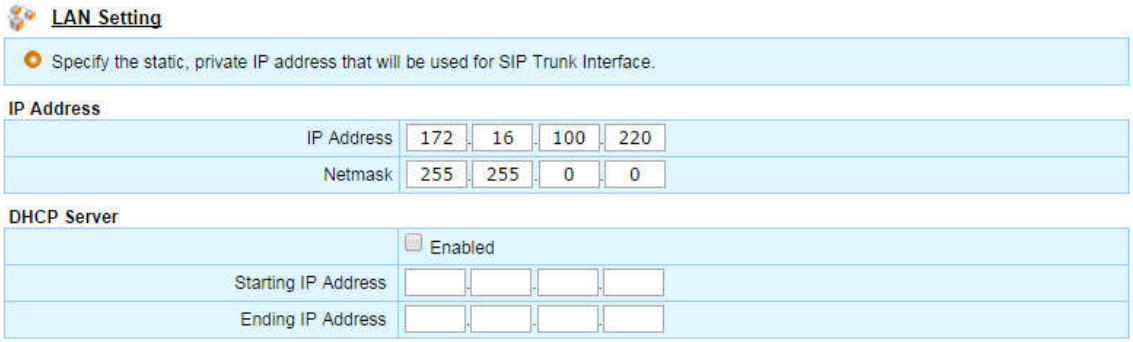


Figure 163. The Installer page: ESBC LAN settings

8.1.1.5 Monitor

The monitor page displays the SIP UAs registration/connection status for both the North bound and South bound interfaces of the ESBC.

Navigate to **Technician** > **Monitor**, or **Customer** > **Monitor**.

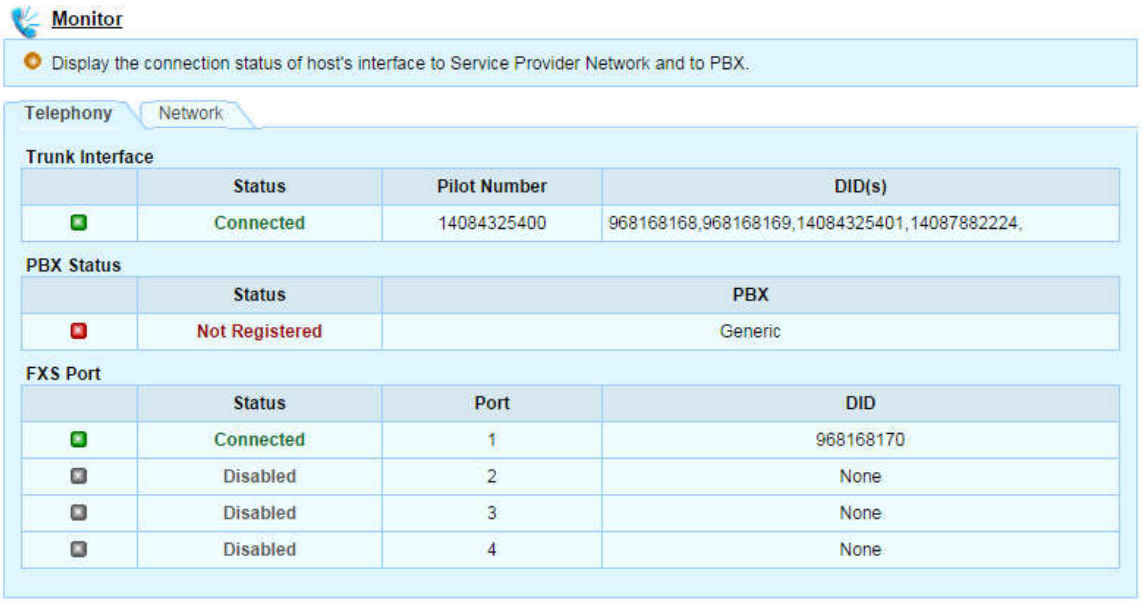


Figure 164. SIP UA Connection/Register status

8.1.2 ESBC-9x80 series models (switch between T1/E1 and transcoding)

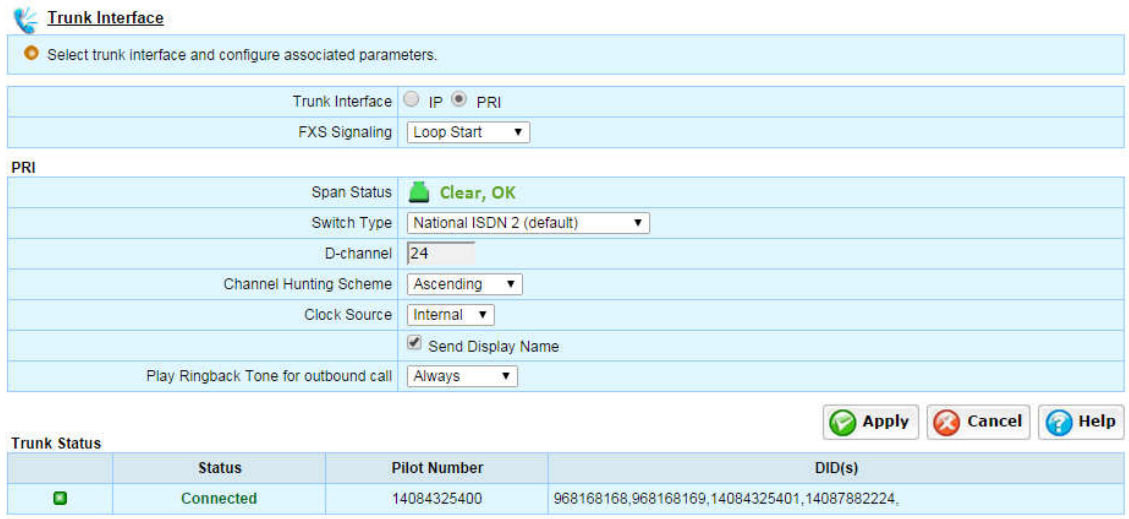


Figure 165. T1/E1 configuration page for installers

The ESBC9x80 (E1/T1) models offer service providers with the flexibility of selecting either a PRI interface (connecting to a TDM PBX) or an IP interface (connecting to a SIP PBX) at the customer site. The IP or PRI options can be chosen through the technician account login.

Trunk Interface:

- When choosing “IP” mode, all configurations are the same as those described in section 8.1.1.

- When choosing “PRI” mode, the technician will need to configure the PRI interface to connect to the TDM-PBX. See section 4.9 for a detailed description of PRI configurations.

For all PRI configurations, please refer to section 8.1.1 for a detailed description.

PRI	Description
Span Status	To display the D Channel connection status, which reflects the q921 signaling for Up or Down.
Switch Type	To choose the switch type, which needs to be exactly the same as that of the TDM-PBX to which the ESBC connects.
D-Channel	Display the fixed channel number for the D-Channel. CH 24 for T1; CH 16 for E1.
Channel Hunting Scheme	Choose the appropriate channel hunting scheme (ascending or descending). It is suggested that a different hunting scheme is used from that of the PBX. E.g., if the PBX uses an “ascending” channel hunting scheme, then configure the ESBC with a “descending” scheme so as to distribute loading evenly on entire PRI spans.
Clock Source	The ESBC default clock mode is “Internal”, using which voice transmissions follow the ESBC’s internal clocking scheme. The connected TDM-PBX clock should be configured to follow the ESBC clock. If there are two spans, span2 always follows the clock of span1. span2 does not have its own clocking scheme. Do not change the ESBC clock mode default settings unless necessary.
Send Display Name	Default mode: checked. Send display names to called parties for inbound calls. Some TDM-PBX’s do not support “Display Name” IE settings. Uncheck this setting when necessary.
Play Ringback Tone for outbound call	"Play Ringback Tone for outbound call". Three options are available: ALWAYS, NEVER, AS-NEEDED. **ALWAYS:** The ESBC always sends inband RBT media to the PBX, either relaying media from the network or the ESBC generating locally. No out-of-band RBT for “ALWAYS” mode. **NEVER**: The ESBC never generates inband RBT locally. RBT sent to the PBX is either inband RBT media from the SIP Trunk side or out-of-band RBT signals (default setting). **AS-NEEDED:** The ESBC decides the RBT action to the PBX according to the messages it receives (SIP response codes 180/183 from the network; and Q.931 Progress Indicator from the PBX).

8.2 Operator Management via the Operator WEB Console

1. Configure the operator's PC with an appropriate IP address within the same network as the ESBC LAN interface, either for the management port or NAT_and_Voice port.
2. Assuming the PC connects to the NAT_and_Voice port, start your web browser and enter e.g., http://172.16.1.1, in the address field to connect to the ESBC. The login page will appear. The default user name is "oper" and the password is "123". Click the login button to enter the ESBC main page.

The operator console is designed for the end customer to configure the PBX (TDM or SIP) information on the ESBC when there are network setting updates.

Please refer to section 8.1 for descriptions of related features.

Figure 166. The operator (end customer) login page

9 SIP Firewall and Header Manipulation Rules (SHMR)

To provide finer control of SIP messages traversing through the SIP PBX and the SIP server, the ESBC allows the service providers to create SIP Header Manipulation Rules (SHMR) to achieve this purpose.

The SHMR function consists of a sophisticated scripting language that can be used to create scripts that modify SIP message contents both at the LAN/WAN ingress and egress in the following directions:

- ESBC WAN interface, inbound
- ESBC WAN interface, outbound
- ESBC LAN interface (NAT-Voice port), inbound
- ESBC LAN interface (NAT-Voice port), outbound

SHMR can be used to modify SIP headers, parameters as well as SDP contents. Regular expressions also allow complex matching rules to be constructed. Another feature of the SHMR function is multi-level programmability which enables rules to reference each other and pass parameters between them.

To import, verify, and activate Firewall rules and SHMR rules,

- SIP trunk voice services—SIP B2BUA mode (including SIP PBX, TDM PBX), navigate to **Telephony > ADVANCED > Firewall, and Telephony > ADVANCED > SHMR**.
- SIP hosted voice services – SIP ALG mode, navigate to **Telephony > SIP ALG > Firewall**, **and Telephony > SIP ALG > SHMR**.

9.1 SIP Header Manipulation and Firewall Scripts

The ESBC SHMR and Firewall are scripting rules, refer to the following documents for detailed descriptions of this script language. "*ESBC Application Notes- SHMR Usage Guide" and "SIP Firewall Rules"*.

The SHMR rules are composed of the following constructs:

- Objects: headers and header elements. (Headers are SIP headers, and header elements include all subparts of a header, such as header values, header parameters, and URI parameters)
- Rules: header rules and element rules
- Processes
- Regular expressions for matching and giving a new value to an object.

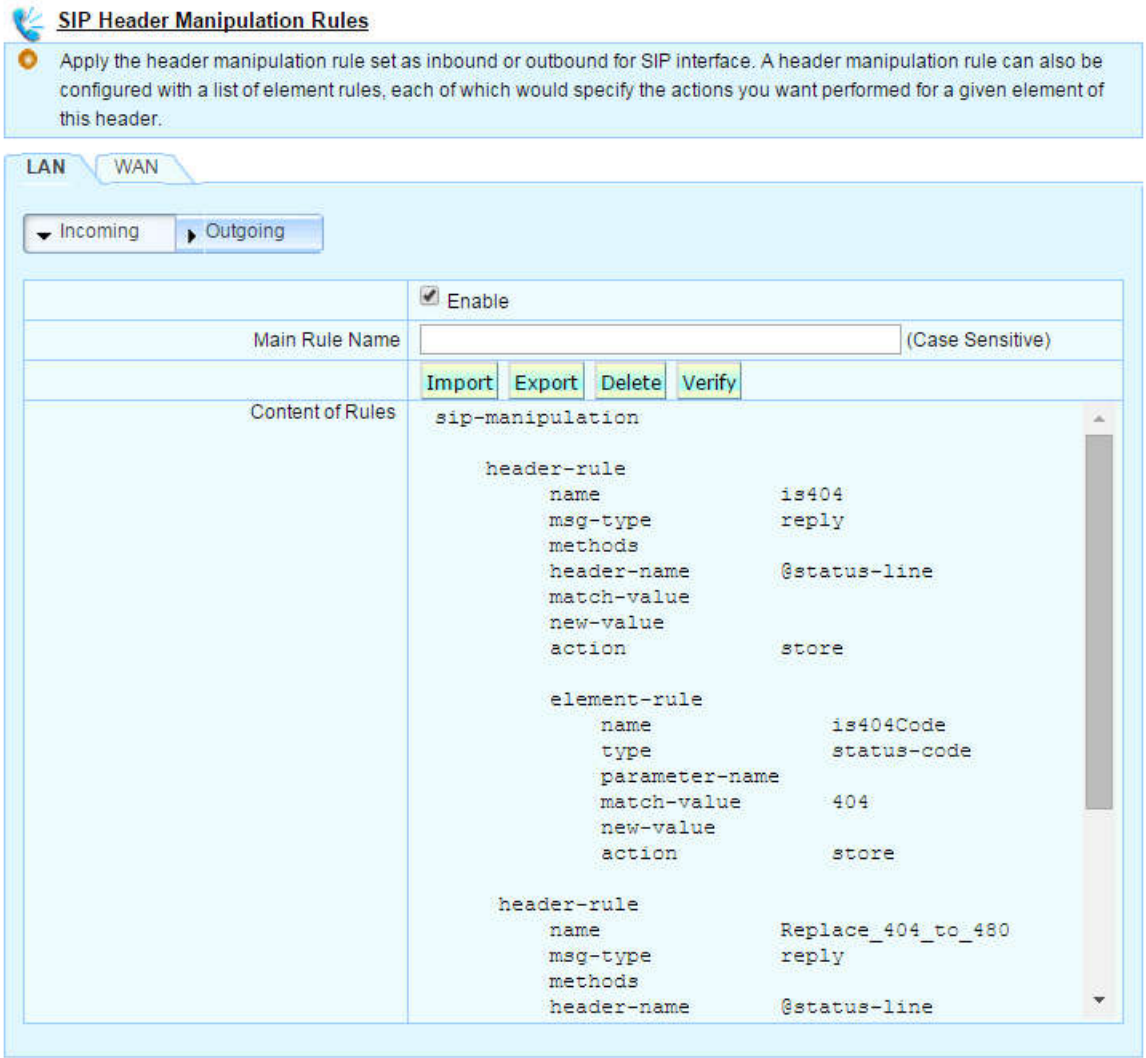


Figure 167. The ESBC SHMR script configuration screen

ESBC SHMR	Description
Enable	Check this box to activate a SHM rule (or rule set) of the target interface-direction.
Import	Importing SHMR script from a file in text file format.
Export	Exporting the SHMR script from the ESBC system to a text file.
Delete	Deleting the current SHMR rule file from the ESBC system.
Verify	Verify the SHMR with a sample SIP message.

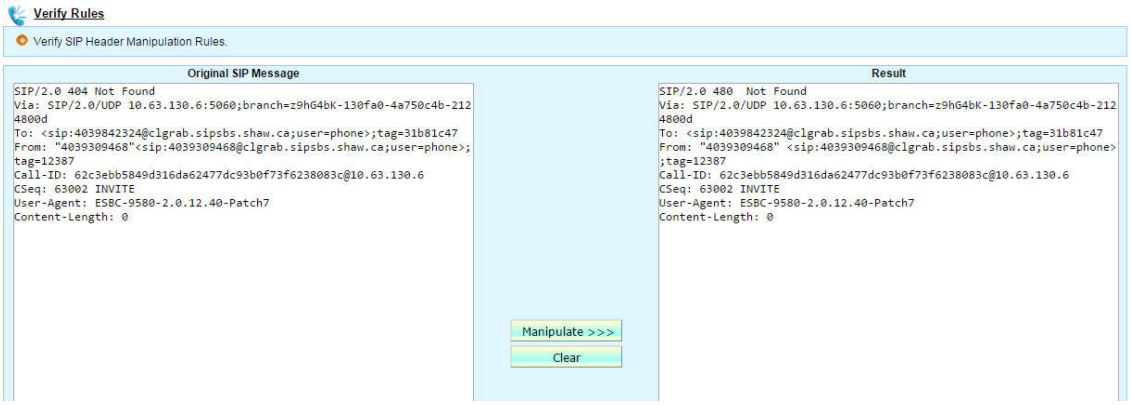


Figure 168. The SHMR SIP message verification screen (changing 404 to 480)

9.2 SIP Firewall

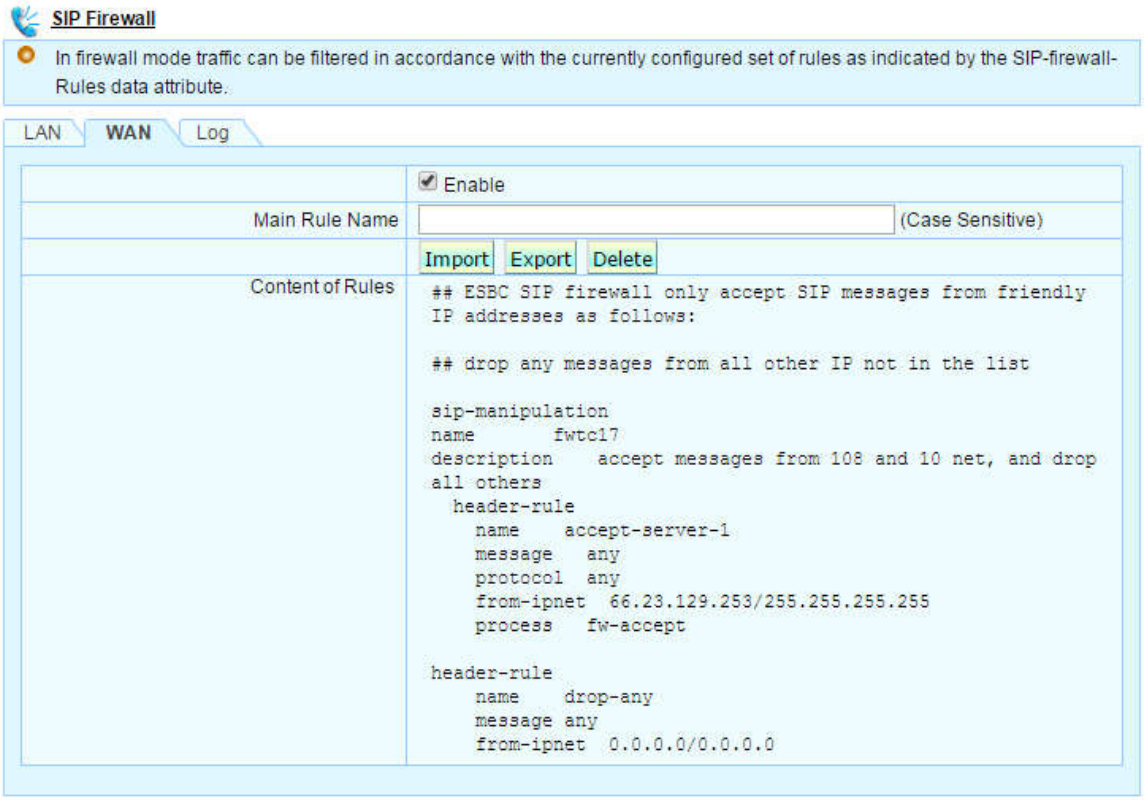


Figure 169. The ESBC SIP firewall configuration screen

ESBC SIP Firewall	Description
Enable	Check this box to activate SIP firewall rules of the target interface.
Import	Importing firewall rules from a file in text file format.
Export	Exporting the firewall script from the ESBC system to a text file.
Delete	Deleting the current firewall rule file set from the ESBC system.
Log	Viewing the SIP firewall log

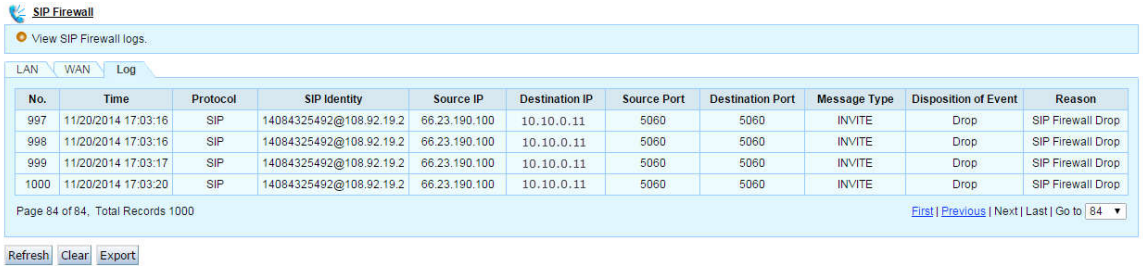


Figure 170. The ESBC SIP firewall log

10 Appendix

10.1 SIP Reason Header

The SIP Reason header (see IETF RFC3326) can be added to either “BYE and CANCEL” request messages or SIP responses to provide information on the possible causes of call failures. Some of the reasons covered including the following (although this list is not exhaustive):

1. Abnormal conditions cause the ESBC to drop calls. (see Table 2. Abnormal conditions)
2. Calls are disconnected from the T1/E1 LAN interface, and their related Q.850 cause codes will be carried in the Reason header with the appropriate SIP messages and sent to the SIP Server. In this case, the PRI Q.850 cause codes include both normal call clearance and abnormal failures.

Table 2. Abnormal conditions

Item	Description	Reason Header	Sample scenarios
1	Reload ESBC configuration during active calls	Reason: SIP;text="Reload configurations"	Change configuration on SIP Parameters page during the call
2	Trunk lost registration	Reason: SIP;text="Trunk lost registration"	The ESBC de-registers this UA from the server during the ongoing call
3	SDP negotiation failure	Reason: SIP;text="SDP negotiation failed"	A call where caller and callee do not have common CODEC
4	PRI-D channel down	Reason: SIP;text="PRI D-Channel down"	Unplug T1 cable during the call to enter OOS condition
5	PRI channel is restarted (by the PBX)	Reason: SIP;text="PRI channel is restarted"	Receive RESTART from PBX on the channel of the current active call
6	PRI STATUS enquiry failed	Reason: SIP;text="PRI STATUS enquiry failed"	PRI STATUS timeout
7	PRI Call state error	Reason: SIP;text="PRI Call state error"	PRI call state machine error
8	DQoS Reservation failed	Reason: SIP;text="Reserve DQoS failed"	When the number of B2BUA concurrent calls reaches the ‘Active Dynamic Service Flows’ limit, and the option “allowing calls of best effort” is disabled. Upon making a new call, this

			call will be rejected by the ESBC
9	Commit DQoS failed	Reason: SIP;text="Commit DQoS failed"	ESBC reserves QoS (DSA messaging) successfully for the SDP offer, but fails to update QoS (DSC messaging) when it receives the SDP answer
10	PRI Enter diagnostics mode in call (ringing/talking) state.	Reason: SIP;text="Reload configurations"	When call is in ringing or talking state, click the 'Diagnostics' button on 'Digital Line' section of the GUI
11	Security check failed (Security section on Trunk/PBX SIP profile pages)	Reason: SIP;text="Security check failed"	Enable security options on Trunk and/or PBX SIP profile page SIP request messages are sent from host(s) other than the server to which the ESBC registers; or the SIP PBX from which the ESBC receives REGISTER messages
12	Loop detected (Max-Forwards header, History-Info headers)	Reason: SIP;text="Loop detected "	When the "Loop Detected option" in Trunk SIP profile is enabled, and the ESBC detects MAX-Forwards header value equal to 0
13	Media Inactivity	Reason: Q.850; cause=41;text="Temporary failure"	When the "Media Inactivity Timer" option is enabled, and the associated direction(s) has no RTP traffic passing through it until the timeout activates
14	Caller ID has reserved char (', the apostrophe)	Reason: SIP;text="Caller ID has reserved char"	INVITE messages to the ESBC contain reserved chars in the caller ID string – Note: if it is an outbound call, the "default route" has to be enabled

15	Called number has reserved char (‘, the apostrophe)	Reason: SIP;text="Called number has reserved char"	INVITE messages to the ESBC contain reserved chars in called party ID string. Note: if it is an inbound call, the “default route” has to be enabled
16	No supported codec for transcoding (Transcoding profile setting, "Allow calls when no supported CODEC in SDP offer" is disabled.)	Reason: SIP;text="No supported codec for transcoding"	Disable option "Allow calls when no supported CODEC in SDP offer" on transcoding setting page Configure caller to use SDP which is not supported in transcoding Make outbound call
17	Ongoing calls terminated by new emergency calls	Reason: SIP;text="Killed by emergency call"	When the number of “Concurrent calls” reaches the B2BUA user license number, new emergency calls preempt system resources and terminate existing calls
18	**(Certain customers only)** When the ESBC receives an INVITE from the PBX while the PBX registration is either expired or lost, the ESBC rejects this call attempt	Reason: SIP;text="PBX is not registered"	Deregister PBX from the ESBC, then make call from PBX to ESBC.

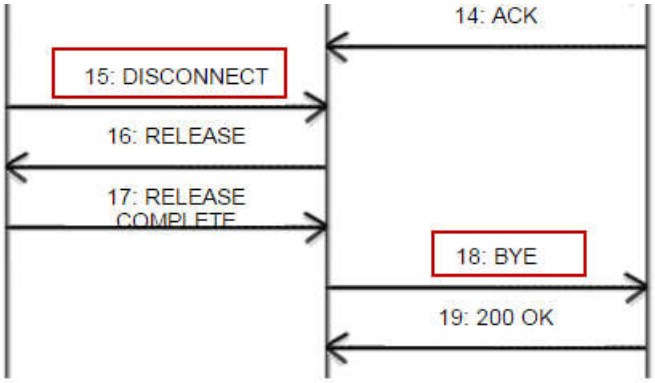


Figure 171. A partial SIP to PRI call flow

Table 3. Example of a SIP Reason Header with Q.850 cause code sent to the SIP Server

ISDN Q.931 Message	F15 DISCONNECT, SPAN-1 to Host Protocol Discriminator: Q.931 (8) len=9 TEI=0 Call Ref: len= 2 (reference 44/0x2C) (Sent to originator) Message Type: DISCONNECT (69) [08 02 81 90] Cause (len= 4) [Ext: 1 Coding: CCITT (ITU) standard (0) Spare: 0 Location: Private network serving the local user (1) Ext: 1 Cause: **Normal Clearing (16)**, class = Normal Event (1)]
SIP Reason Header with Q.850 cause	F18 BYE, Host to 10.20.7.77 BYE sip:15104320001@10.20.46.250:5063 SIP/2.0 Via: SIP/2.0/UDP 10.20.40.146:5060;branch=z9hG4bK-524287-1---2674dc287af43c73;rport Max-Forwards: 70 Route: <sip:10.20.7.77;lr> Contact: <sip:14087896328@10.20.40.146:5060> To: "My Name"<sip:15104320001@10.20.7.77>;tag=1450022260 From: <sip:14087896328@10.20.7.77;user=phone>;tag=66098017 Call-ID: 1498610344@10.20.46.250 CSeq: 2 BYE User-Agent: ESBC9380-4B-2.0.14.0-ER63 **Reason: Q.850;cause=16;text="Normal Clearing"** Content-Length: 0

10.2 SIP Remote-Party-ID header parameter mapping with the PRI SETUP message

When the PRI SETUP message of an outbound call from the PBX has “Presentation Restricted”, the ESBC treats it as an anonymous call.

PRI SETUP Message ➔	**SIP Headers**

Presentation Allowed ➔
`Calling Party Number (len=15) [ Ext: 0 TON: National Number (2) NPI: ISDN/Telephony Numbering Plan (E.164/E.163) (1)`
 	
`Presentation: Presentation allowed, User-provided, not screened (0) 14087896328 ]`
	From:
	`"MTA6328"<sip:14087896328@10.20.7.77;user=phone>;tag=5730ad65`
	Remote-Party-ID:
	`"MTA6328"<sip:14087896328@10.20.7.77;user=phone>;`**`privacy=off;screen=no`**

Presentation Restricted ➔
`Calling Party Number (len=15) [ Ext: 0 TON: National Number (2) NPI: ISDN/Telephony Numbering Plan (E.164/E.163) (1)`
 	
`Presentation: Presentation restricted, User-provided, verified and passed (33) 14087896328 ]`
	From:
	`"Anonymous"<sip:anonymous@anonymous.invalid;user=phone>;tag=640e059c`
	
	Remote-Party-ID:
	`<sip:14087896328@10.20.7.77;user=phone>;`**`privacy=full;screen=yes`**

Table 4. SIP Remote-Party-Header mapping with PRI SETUP message

10.3 ESBC-TDM PBX ringback tone behavior summary

Outbound calls

Calling Direction: Outbound Calling originating from PBX	ESBC Setting: Play Ringback tone for outbound calls	ESBC Setting: Ignore 183 Early Media for outbound calls	ESBC Behavior:	Party responsible for playing ringback:
	As Needed	Uncheck	When the SIP trunk side responds by sending inband RBT media (183 response code) to the ESBC, the ESBC relays inband media to the PBX.	SIP Trunk side (the service provider network) provides the inband RBT media

			When the SIP Trunk side responds with out-of-band RBT signals (180 response code) to the ESBC but the SETUP message of caller expects inband RBT media, the ESBC generates inband RBT media locally and sends to PBX.	ESBC provides the inband RBT media
			When the SIP Trunk side responds with out-of-band RBT signals (180 response code) to the ESBC, and the SETUP message from the caller indicates out-of-band RBT, the ESBC will let the PBX play RBT.	PBX provides the out-of-band RBT
		Checked	When the SETUP message from the caller expects inband RBT media, the ESBC generates inband RBT media locally and sends to the PBX	ESBC provides the inband RBT media
			When the SETUP message from the caller indicates out-of-band RBT, the ESBC will let the PBX play RBT.	PBX provides the out-of-band RBT
	Always	Uncheck	When the SIP trunk side responds with sending inband RBT media (183 response code) to the ESBC, the	SIP Trunk side (the service provider network) provides the inband RBT media

			ESBC relays inband media to the PBX.	
			When the SIP Trunk side responds with out-of-band RBT signals (180 response code) to the ESBC but the SETUP message of caller expects inband RBT media, the ESBC generates inband RBT media locally and sends to PBX.	ESBC provides the inband RBT media
		Check	Regardless of what the SIP trunk side responds with regarding RBT, the ESBC always generates inband RBT media locally and sends to PBX.	ESBC provides the inband RBT media
	Never	Uncheck	When the SIP trunk side responds with sending inband RBT media (183 response code) to the ESBC, the ESBC relays inband media to the PBX.	SIP Trunk side (the service provider network) provides inband RBT media
			When the SIP Trunk side responds with out-of-band RBT signals (180 response code) to the ESBC, the ESBC will let the PBX play RBT.	PBX or device behind PBX provides out-of-band RBT media
		Check	Regardless of what	PBX or device behind PBX

			the SIP trunk side responds with regarding RBT, the ESBC will let the PBX play RBT.	provides out-of-band RBT media

Table 5. The ESBC – PRI : Outbound call ringback tone behavior summary

10.4 ESBC SIP Authentication Flow

10.4.1 Authenticate SIP Request Messages from SIP Server

When the ESBC receives incoming SIP requests from the SIP server, the ESBC checks the setting "Challenge inbound SIP requests for authentication".

Navigate to **Telephony** > **SIP Trunks** > **Trunk SIP Profile**.

- If enabled, the ESBC sends 401 to a request without the correct credentials, and challenges the SIP request.
- If disabled, the ESBC processes the SIP request.

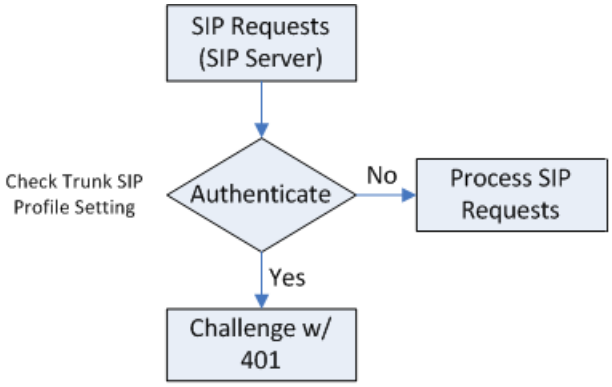


Figure 172. Authentication Flow: SIP Request Messages from the SIP Proxy Server

10.4.2 Authenticate SIP Request Messages from SIP-PBX

When the ESBC receives incoming SIP requests from SIP-PBX, the ESBC processes them according to the following logic:

1. Check "Authenticate" settings from the "Filter SIP Method" table. Navigate to **Telephony** > **SIP-PBX** > **SIP Parameters**.
   - If it is enabled for this SIP method, go to step 2.
   - If it is not enabled for this SIP method, go to step 4.
2. Check "Authenticate Mode" setting. Navigate to **Telephony** > **SIP-PBX** > **Authentication**
   - If it is "None", go to step 4.
   - If it is "Local" or "Radius", go to step 3
3. The ESBC challenges the SIP message with 401 if it is sent without the correct credentials.
4. The ESBC does not challenge the SIP message.

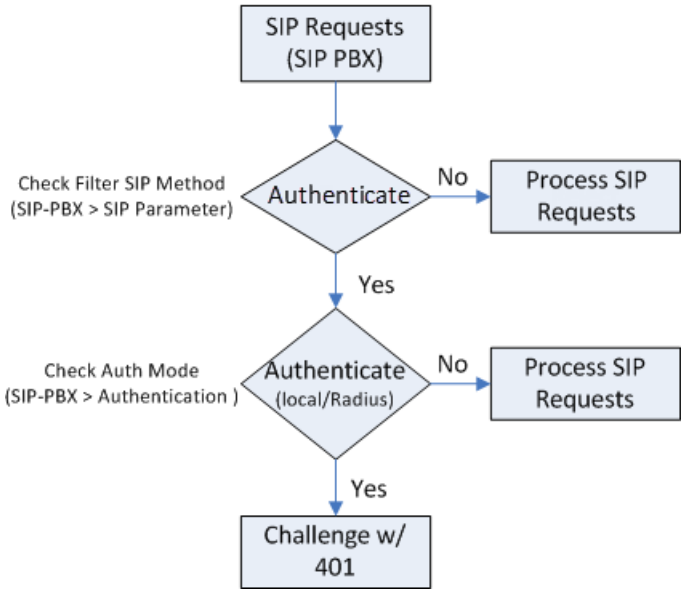


Figure 173. Authentication Flow: SIP Request Messages from the SIP-PBX

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