Interworking SipXcom with TWC SIP Trunks

Introduction

The North American cable companies (often called Multiple Service Operators or MSOs) through their Cablelabs consortium began working on a specification called Packetcable 2.0 starting in the mid-2000s that introduces SIP and IMS concepts into the cable HFC-based access network. In November of 2010, Cablelabs issued the Packetcable PKT-SP-ESG-I01-101103 Enterprise SIP Gateway (ESG) specification, which allows cable companies to offer SIP trunks with quality of service for voice over the HFC access network for small and medium businesses. The ESG high-level architecture from this Packetcable specification looks as follows:
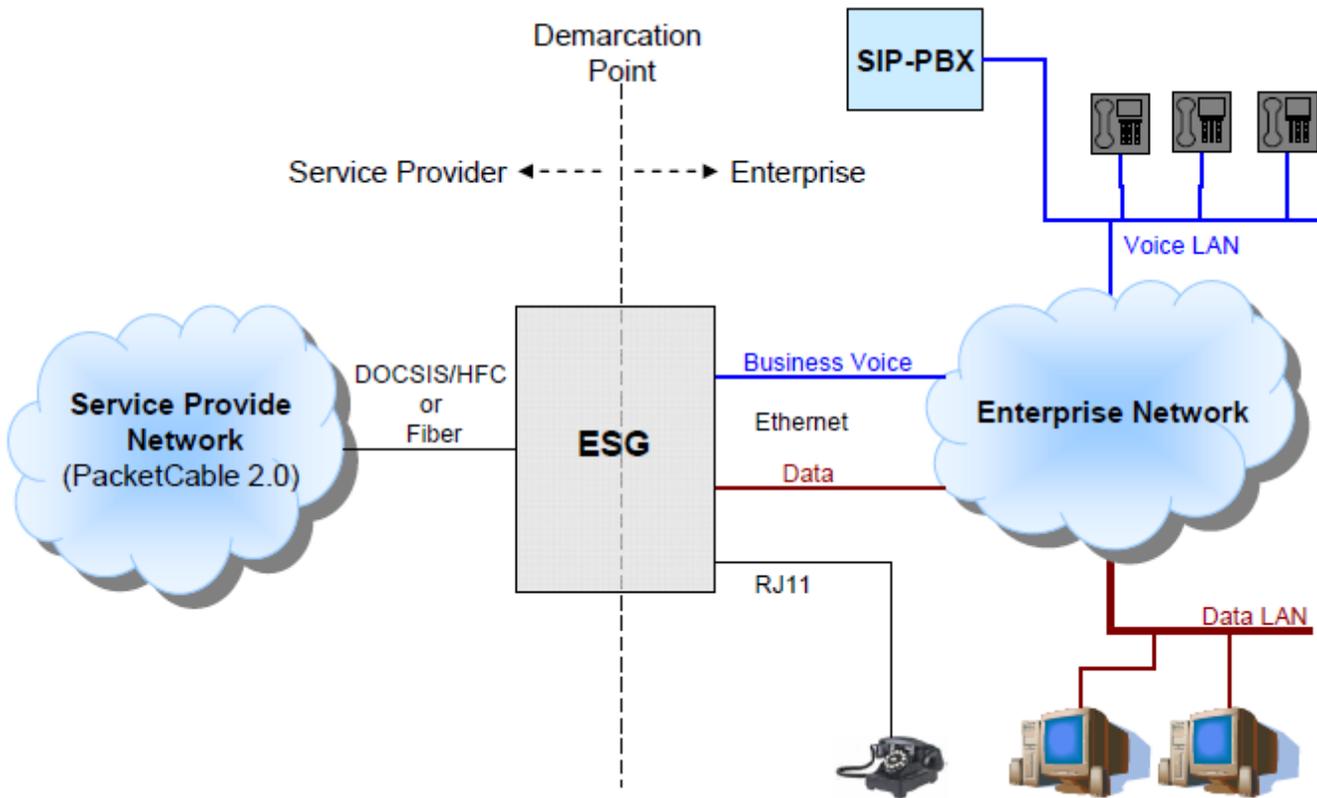
Enterprise SIP Gateways (ESG) are Back-To-Back Users Agents (B2BUA) that deliver registered and unregistered SIP trunks to IP-based PBXs and other devices sitting on small and medium business premises. From the service provider perspective, the ESG connects to the MSO softswitch or IMS platform where calls are then routed to the public phone network. There are two types of ESGs deployed by cable companies:

- Enterprise SIP Gateway with embedded DOCSIS cable modem (DOCSIS stands for Data over Cable Service Interface Specification) - or for the purposes of this wiki page, E-ESG.
- Standalone Enterprise SIP Gateway with **no** embedded cable modem - or for the purposes of this wiki page, S-ESG.

When a MSO installs an E-ESG in a small business, the co-axial cable on the premise connects to the E-ESG. When a S-ESG is deployed, the MSO brings a second cable modem to the customer. The MSO engineers their Hybrid Fiber-Coaxial (HFC) network so that the voice traffic travels down separate channels to the business premise from the data traffic - this approach delivers quality of service for voice applications.The customer environment side of the following diagram illustrates how the ESG/cable modem for voice applications and separate cable modem for data traffic co-exist in a business premise.

Test Environment — **Customer Environment**
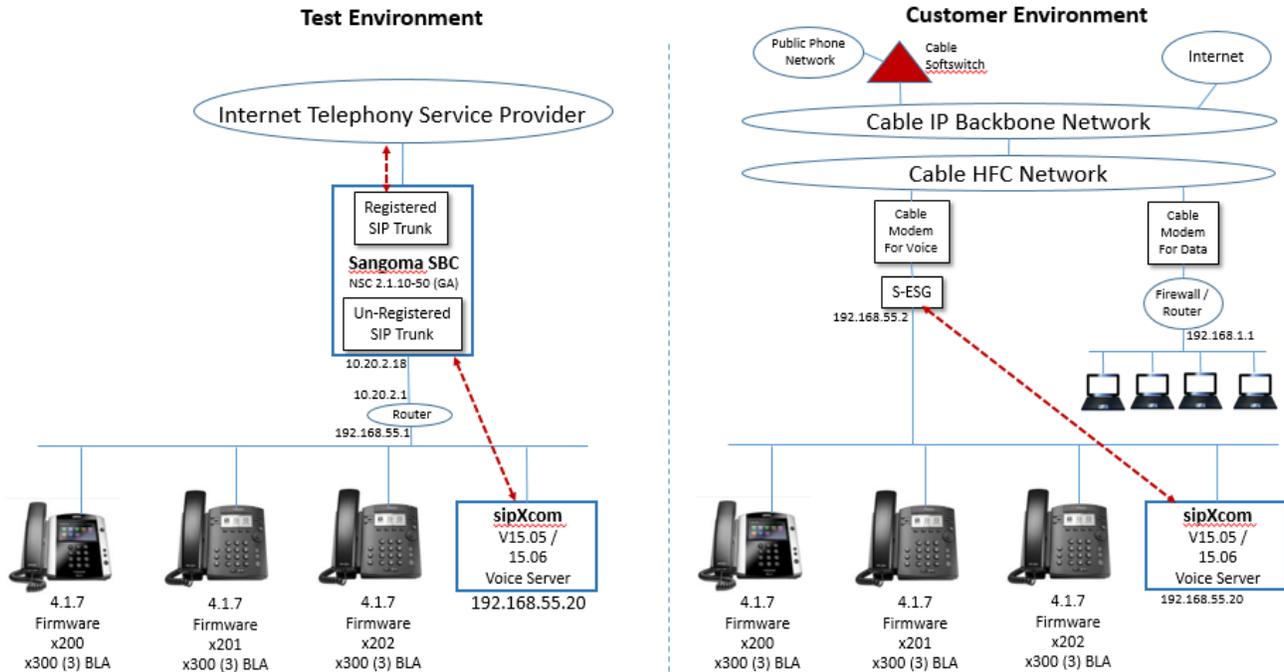
In preparing this SipXcom interoperability document, a Sangoma SBC was used to simulate a MSO ESG (left-hand side of the above diagram). The SipXcom configuration was then validated with a customer using SIP-based voice services connected to a major MSO in the United States. Both the lab and customer configurations used the Bridged Line Appearance (BLA) capability of SipXcom - the setup for BLA is described here http://wiki.sipxcom.org /display/sipXcom/Setting+Up+Bridged+Line+Appearance+Lines+for+Small+and+Medium+Businesses.

The ESG architecture has been standardized by Cablelabs, and deployed by major MSOs across North America for business voice applications. Although there are multiple manufacturers of ESGs used by MSOs, there are variations in implementation between what is described in this document specifically for Time-Warner Cable (TWC) SIP trunks and what is used in other MSO ESG deployments. The differences in implementation appear to be in two main areas (so far):

- When using un-managed gateway from Sipxcom to the ESG, the RTP traffic for the phones behind Sipxcom go directly to the ESG. Some MSO gateways (e.g. Optimum SIP Trunk Adapter using Edgemarc ESG) are configured so that all phone media must pass through the IP-PBX; therefore a registered SIP trunk must be defined between Sipxcom and the ESG so that phone bearer traffic is anchored via Sipxbridge.
- Some ESGs (e.g. TWC) do not enable SIP REFER support by default - features such as DISA will not work without REFER.

The remainder of this interoperability describes how Sipxcom 15.08 was tested with TWC's SIP trunk using an ESG provided by Innomedia.

Step 1 - Define an Un-managed Gateway to the ESG

Go to **Devices->Gateways** and create an un-managed gateway such as **CableESG.** The address field is the IP address assigned to the ESG and is provisioned by the TWC systems engineer depending on the network topology - the address can be on the same subnet as SipXcom or routable to another subnet. The port number is assigned by the MSO engineer when provisioning the ESG - include this port number in the un-managed gateway definition. When using BLA in SipXcom, the Transport protocol for **both** the **phones** and the **unmanaged gateway** to the ESG **must be** UDP - there are issues with BLA and ESG interoperability when using TCP transport.

Finally, populate the **Default Caller ID** field of the un-managed gateway with one of the DIDs assigned to the SIP trunk by TWC; otherwise the ESG may reject calls where a DID was used that was not assigned to the SIP trunk.
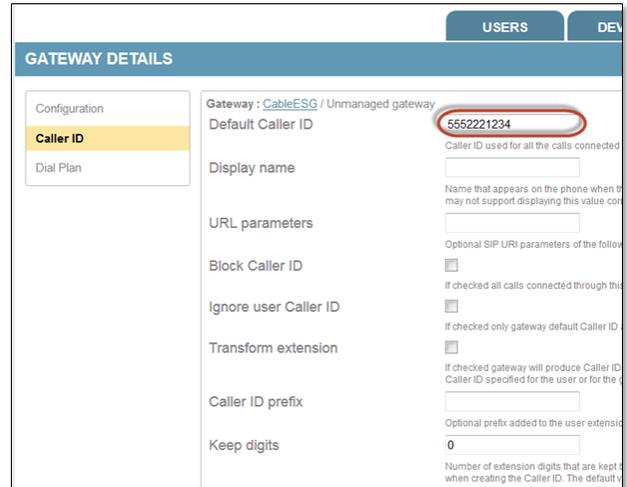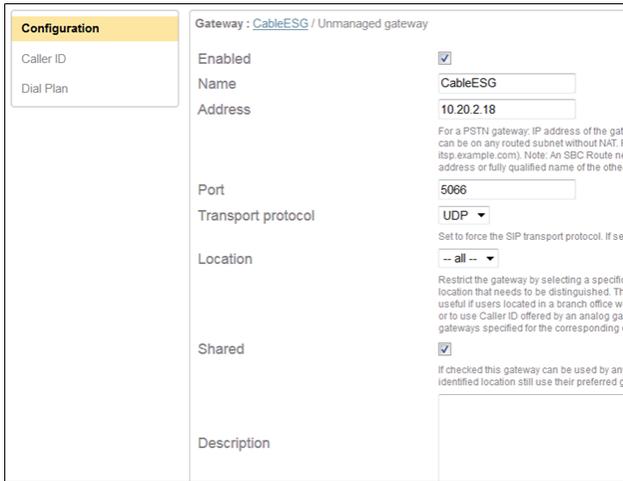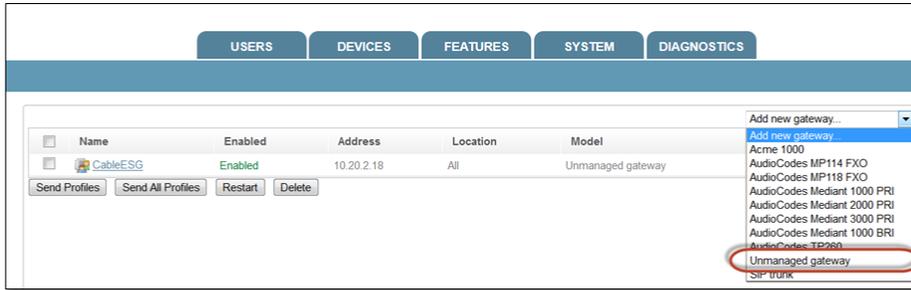
Step 2 - Add Un-Managed Gateways to Dial Plans

Go to **System->Dial Plans** and add the un-managed gateway for the ESG to the appropriate dial plans.

Enabled ☑

Name   Long Distance

Description   Long distance dialing plan

Show Advanced

Permission   Long Distance Dialing ▾
Caller permission needed for this rule to succeed.

PSTN prefix   9
Prefix dialed by the caller and dropped before the call is sent to the gateway.

Long distance prefix   1
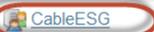Prefix dialed by the caller and sent to the gateway.

Area codes
Optional comma separated list of prefixes that specify dialed numbers to which this rule applies.

External number length   10 digits ▾
Number of digits in the resulting number sent to the gateway. The PSTN prefix and the long distance prefix are not counted.

Schedule   Always ▾

Gateways

More actions...

	Name	Enabled	Address	Location	Model	Description
☐	CableESG	Enabled	10.20.2.18	All	Unmanaged gateway	

Move Up   Move Down   Remove

Step 3 - Disable SIP REFER Method for Transferring Calls

The TWC ESG does not have REFER support enabled by default. Starting in SipXcom Release 15, there is an IVR parameter that allows transfer of calls by the autoattendant via a SIP Re-invite instead of the REFER method. Go the **System->Voicemail** and set the **Transfer by Bridging the Call** field to **True** . When this field is set to **False,** autoattendants that transfer calls to an extension will fail as the ESG responds with a 'not supported' response to the REFER issued from SipXcom.

⚠ Caution - The **Feature->Authorization Code** capability, sometimes known as DISA, also uses SIP REFER capability when placing an external call to an ITSP. Since the TWC ESG does not have SIP REFER support enabled by default on their ESGs, the DISA capability may not work. The workaround is to provision a second low-cost SIP trunk from an ITSP that supports SIP REFERs, define this ITSP trunk via Sipxbridge, and modify the dialplan accordingly to have DISA use this secondary trunk.

Show Advanced Settings

IVR Parameters

Logging Level	NOTICE ▾	*(Default: NOTICE)*
Voicemail encoding format	wav ▾	*(Default: wav)*
	The encoding format to be used for storing voicemails	
Record rate	8000 ▾	*(Default: 8000)*
	Rate (in Hz) to be used for recording voicemails	
Time limit	300	*(Default: 300)*
	Maximum duration (in seconds) of the voicemail recording.	
Fax Format	pdf ▾	*(Default: pdf)*
	The file format for receiving faxes	
HTTP port	8086	*(Default: 8086)*
Transfer By Bridging the call	true ▾	*(Default: false)*
	true	
	false	transfer through IVR / AutoAttendant (default transfer method use SIP REFER method) Enable this option to fix integration with ITSPs / SBCs that does SIP REFER and if you want to have ringback played while transferring call.

Apply

Un-Managed Gateways and NAT Traversal in SipXcom

SipXcom has a number of provisionable parameters in the **System->Internet Calling** and **System->NAT Traversal** menus that are associated with network address translation. The parameters in the **Internet Calling** and **NAT Traversal** menus are used only by **SipXbridge**. **SipXbridge** is the 'internal SBC' within SipXcom and is enabled by creating a **SIP Trunk** gateway within SipXcom.

Un-managed gateways do not use the parameters in the **System->Internet Calling** and **System->NAT Traversal** menus in SipXcom - i.e. no network address translation is performed on any of the SIP headers routed through an unmanaged gateway.

Enterprise SIP Gateways and SIP Trunk Registration

TWC's ESG supports the capability to have SipXcom register a SIP trunk to the ESG using SipXbridge - there is one MSO where this capability is currently in use with bridged line appearance (BLA) enabled for Polycom phones Interoperability with Optimum SIP Trunks. Using the Sangoma SBC lab setup, SIP Trunk registration from SipXcom was tested successfully with Bridged line appearance lines. However BLA did not work properly when SIP trunk registration from SipXcom was enabled with TWC's ESG - the BLA issues were resolved at this customer when an un-managed gateway was defined to the ESG.

Related articles

http://www.cablelabs.com/wp-content/uploads/specdocs/PKT-SP-ESG-I01-101103.pdf

https://en.wikipedia.org/wiki/PacketCable

https://en.wikipedia.org/wiki/CableLabs

http://www.cedmagazine.com/articles/2012/09/billions-up-for-grabs-in-sip-trunking

https://en.wikipedia.org/wiki/DOCSIS

http://wiki.sipxcom.org/display/sipXcom/Setting+Up+Bridged+Line+Appearance+Lines+for+Small+and+Medium+Businesses

Interoperability with Optimum SIP Trunks