EarthLink Business
SIP Trunking

Asterisk 11.2 IP PBX
Customer Configuration Guide

Publication History

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CHANGE HISTORY

Version	Date	Change Details	Changed By
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AUTHOR:
Dantley Thompson
EarthLink Engineering

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Document Purpose

The purpose of this document is to provide a detailed technical description and best practices for successful implementation of the EarthLink SIP Trunking Product for the Asterisk 11.2 with the Adtran SIP Proxy. This document provides information relative to the overall network topology as well as definition and configuration standards for each device associated with the product. Also described within this document are product guidelines and product limitations. This document is to serve as product reference and guide to EarthLink Customers.

Product Summary

The EarthLink Business SIP Trunking product is a complete VoIP (Voice over IP) solution based on the SIP (Session Initiation Protocol) signaling protocol. The SIP Protocol is responsible for set-up and tear-down of voice calls and overall feature and functionality. EarthLink Business' SIP Trunking solution will be served off a MetaSphere Call Feature Server (CFS) fronted by an ACME packet SBC (Session Border Controller). The CFS acts as the centerpiece for call control and feature interaction. The EarthLink Business SIP Trunking Product will primarily use Adtran CPE (Customer Premise Equipment) configured as a SIP Proxy. The MetaSphere CFS Platform is a geo-redundant, high availability solution and serves as the primary element in EarthLink's Hosted Voice and SIP Trunking Product families.

In addition to the basic call control, advanced call routing functionality is available with EarthLink's SIP Trunking product with MetaSphere Enhanced Application Server (EAS) Platform which consists of multiple applications and servers integrated into high availability solution.

The Acme Packet SBC masks private to public IP Address space to provide a safe and secure means of communication between the SIP Server and IP PBX. All SIP traffic destined to, or originating from the MetaSphere CFS, traverses through the ACME Packet SBC. The same policy relates to the CPE device installed at the customer premise. The Acme Packet SBC and Adtran CPE, utilizing SIP Proxy, both resolve NAT (Network Address Translation) related issues exposed when SIP traffic passes through a firewall.

Network Architecture and Design

The EarthLink Business SIP Trunking solution consists of several key network elements that are connected to the existing core routing infrastructure. The MetaSwitch Call Feature Server, IP/TDM Gateways, and Acme Packet SBC's are geographically diverse with reach-ability at both layer two and layer three to provide failover capability and redundancy. Split-Horizon DNS servers are used to resolve the SIP domain to the appropriate regional SBC. Adtran CPE will be connected to the EarthLink network via the traditional means such as Ethernet, PPP (Point to Point Protocol), or MLPPP (Multilink Point-to Point Protocol). T1, or bonded T1 services MUST be provisioned to either the Adtran TA5000 or directly to the Cisco 7609 (Edge Router) to allow for proper QoS (Quality of Service) behavior.

As mentioned earlier in this document, EarthLink's SIP Trunking product can be offered as an overlay to other Earthlink Products and Services. The first diagram below provides a high level look at the primary components that complete the SIP Trunking product. The second diagram provides a detailed layout for the connections between the Adtran CPE and Customers IP PBX.

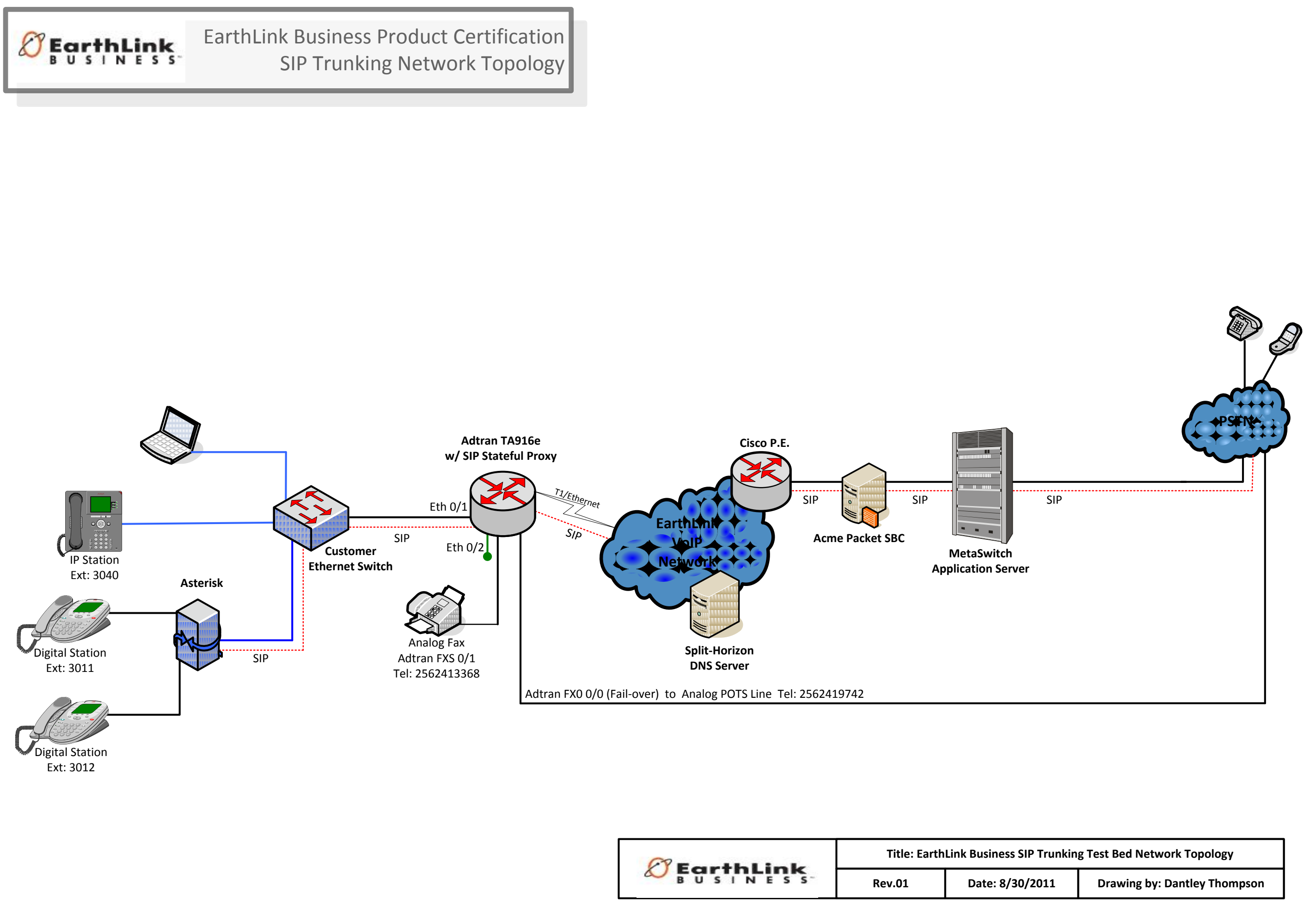

Figure 1-EarthLink SIP Trunking-Network Topology

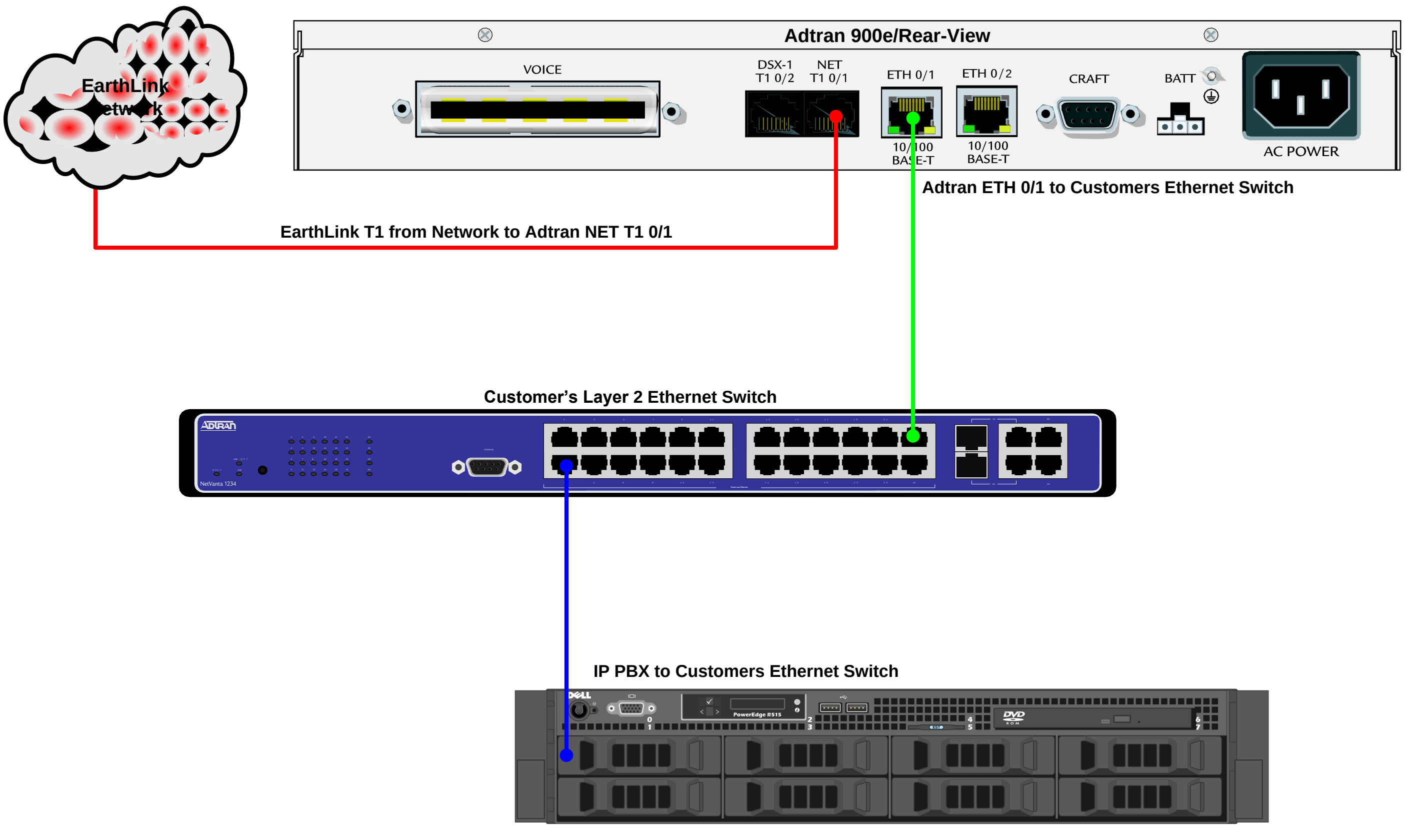

Figure 2-EarthLink SIP Trunking-Connections from Adtran CPE to IP PBX

Media Attributes and Codec Negotiation

Codec Support

A voice codec (coder/decoder) is a hardware/software module/algorithm that takes an analog or digital voice stream and encodes it into an IP packet. For the EarthLink Business SIP Trunking Product, we currently support two (2) of the most common codec's utilized in the continental United States, G.711u and G.729a. The preferred codec offered by EarthLink in the default configuration model is G.711u, then G.729a. Basically this means that the call will negotiate using the G.711u codec first, as long as the terminating end sends G.711u as the first or primary offered codec. The paragraphs below provide more detailed information related to the codec's and other requirements associated with proper negotiation of the media/RTP.

G.711u

G.711u is the most common uncompressed audio codec deployed in the US. Because it is uncompressed, it supports the highest level of quality for the call. Typically the G.711u consumes 90Kbps-100Kbps per call. The standard sampling rate of 8kHz is used for the G.711u codec.

G.729a

G.729a is the most common codec utilized to support compressed audio utilized in the US. Because it is compressed, it is perceived to have a lower voice quality than that of G.711u, however most people would never be able to tell the difference. Typically the G.729 consumes 30Kbps-40Kbps per call. The standard sampling rate of 8kHz is used for the G.729a codec.

Packetization Time

Packetization Time determines how often the audio stream is sampled and how often an IP packet is created. The standard packetization times are 10ms, 20ms, 30ms, and 40ms. EarthLink Media Gateway's have been statically configured to use a 20ms packetization time. IP Phones and/or Voice Applications will need to configure their equipment for a 20ms packetization time before audio traffic can be reliably passed across the EarthLink IP Voice network.

DTMF Support

EarthLink supports the transmission of Dual-Tone Multi-frequency (DTMF) digits through the implementation of RFC2833. This RFC covers the basis of including DTMF digits within the media/RTP path of the call. EarthLink recommends for Customers to configure their IP PBX's and/or Voice Applications to use RFC2833 to allow for DTMF to be passed properly and detected across the EarthLink IP Voice network.

Fax and Modem Support Requirements

Currently, analog devices such as faxes and modems MUST be provisioned using the G.711u codec only. “SIP” to analog lines are supported as SIP Lines off the Adtran FXS Ports or a Cisco 2102 ATA (Analog Terminal Adapter). The customer may also configure the IP PBX to use analog extensions for faxes and modems. This method can be supported utilizing the G.711u codec only. T.38 is currently not supported.

North American Numbering Plan Format

Currently, the EarthLink Business Hosted Voice product only supports the North American Numbering Plan Format. A Global Numbering Plan Format, such as E.164, is currently not supported.

Quality of Service Policy

To ensure the best possible voice quality, EarthLink will mark and match all VoIP traffic related to SIP (Session Initiation Protocol) and RTP (Real-Time Transport Protocol). EarthLink VoIP and/or Real-Time based appliances and applications are configured to use DSCP (Differentiated Services Code Point) “46” for all signaling traffic (SIP) and DSCP “46” for all Real-Time traffic (RTP) for Layer three priority. The Customers IP PBX MUST also be configured to use DSCP “46” to provide prioritization for SIP and RTP. Marking the DSCP field in the IP packet header will allow for packet classification to be matched and provide priority across EarthLink’s network. This also ensures QoS specifications outlined in SLA (Service Level Agreements) can be sufficiently met between EarthLink and the customer.

EarthLink SIP Trunking to IP PBX Interoperability

SIP Trunking interoperability testing was performed between EarthLink and the IP PBX. All phases of the test plan were executed against the actual configuration used in a customer deployment. The information below provides the Adtran and IP PBX software versions tested as well as an issue summary and non-supported elements discovered during compliance testing in the EarthLink Lab.

Adtran Software Version Tested

- Adtran TA908e version A4.09

IP PBX Software Version Tested

- Asterisk 11.2

EarthLink Open Issues & Non-Supported Features

- Registration is currently not supported for the EarthLink SIP Trunking Product.

- When the originating calling number is present in the FROM Header, the main billing telephone number or DID belonging to the trunk group must be provided via the PAI (P-Asserted Identity) Header or via the Diversion Header on Call Transfer and Call Forward calls for the call to pass through the Metaswitch and billed correctly.

- Correct calling name not being sent correctly when on-net calls are made on the same Metaswitch from Business Group to IP PBX.

Asterisk 11.2 Open Issues & Non-Supported Features

- Registration is currently not supported for the EarthLink SIP Trunking Product.

IP PBX Configuration for EarthLink SIP Trunking with Adtran SIP Proxy

The steps below provide a step by step guide for configuration of the Asterisk 11.2 IP PBX for the EarthLink SIP Trunking Product. Basic configuration of the Asterisk 11.2 should be complete and the Asterisk 11.2 MUST be connected to the LAN prior to configuring the system for SIP Trunking.

Asterisk 11.2 IP PBX Configuration

The following configurations are done in sip.conf and extensions.conf of the Asterisk configuration directory, /etc/asterisk/. The configuration below will connect Asterisk with EarthLink's SIP Trunking product via the Adtran SIP Proxy. More detailed information can be found about Asterisk and its configuration at www.asterisk.org.

Asterisk SIP Device Configuration – /etc/asterisk/sip.conf

Some of the configurations below will change depending on the test being performed.

```
[test-trunk]        ; a label used for referencing the trunk in the dialplan
type=peer
secret=1234
host=192.168.1.1          ; place the IP address of the Adtran here
insecure=invite,port
context=test-incoming  ; needs to match the context name in extensions.conf
disallow=all
allow=ulaw
```

Asterisk Dialplan Configuration – /etc/asterisk/extensions.conf

Some of the configurations below will change depending on the test being performed.

```
[test-incoming]                       ; context for calls incoming from the test-trunk
exten => 7742371820,1,NoOp()          ; match incoming DID and ring test phone
same => n,Dial(SIP/test-phone)
same => n,Hangup()

[test-outgoing]                       ; context for calls incoming from the test-phone
exten => _X.,1,NoOp()                 ; match all digits dialed and send them out the trunk
same => n,Dial(SIP/test-trunk/${EXTEN},20,t)
same => n,Hangup()
```

Product Support and Contact Information

The information below provides contact information for assistance in configuration and troubleshooting EarthLink's SIP Trunking service.

EarthLink Support: (800)239-3000 or http://www.earthlinkbusiness.com/

- **24x7 Support Availability**

Digium Support: (256) 428-6000

- **http://www.digium.com/en/products/asterisk/support**
- **24x7 Support Availability**

EarthLink SIP Trunking Turn-up Testing Procedure

To ensure proper call negotiation can be established between EarthLink and the IP PBX, the test steps below MUST be executed during the initial turn-up process.

SIP Trunking Test Steps:

1. Test an outbound call to a Local Number. Check for Ring-back, 2-way Audio, and Call Quality.
2. Test an outbound call to a Long Distance Number. Check for Ring-back, 2-way Audio, and Call Quality.
3. Test an outbound call to an International Number. Check for Ring-back, 2-way Audio, and Call Quality.
4. Test an outbound call to a Toll-Free Number. Check for Ring-back, 2-way Audio, and Call Quality.
5. Test an inbound call that lasts greater than 10 minutes
6. Test an outbound call that lasts greater than 10 minutes
7. Test simultaneous inbound and outbound calls to PSTN
8. Test an outbound Call to Operator “0”
9. Test an outbound Call to Directory Assistance “411”
10. Test a “911” Call (IDENTIFY TO THE 911 OPERATOR THAT THIS IS A TEST). Ask them to provide phone number, address and secondary or alternate number if available.
11. Test an inbound call to an internal DID. Check for Ring-back, 2-way Audio, and Call Quality.
12. Test an inbound call to Auto-Attendant. Check DTMF and Call Quality
13. Test an outbound call to an Auto-Attendant/IVR and verify DTMF
14. Test Call Transfer off-site
15. Test Call Forward off-site

Notes: