EarthLink Business
SIP Trunking

Cisco CME IP PBX
Customer Configuration Guide

Publication History

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CHANGE HISTORY

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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 Cisco CME 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. The SIP Trunking product can be offered as an overlay to several of EarthLink's existing products such as Internet and MPLS based products. 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.

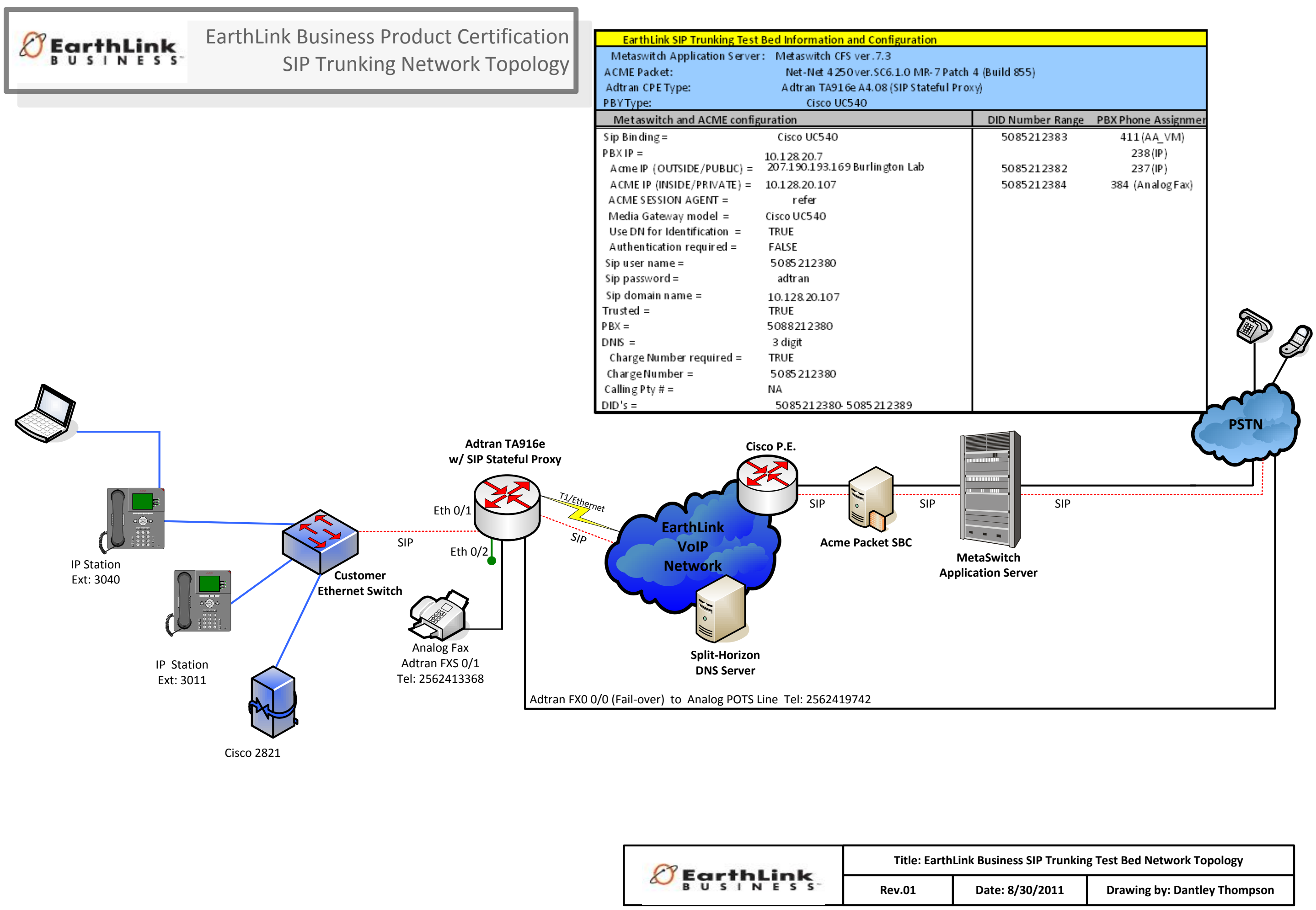

Figure 1-EarthLink SIP Trunking-Network Topology

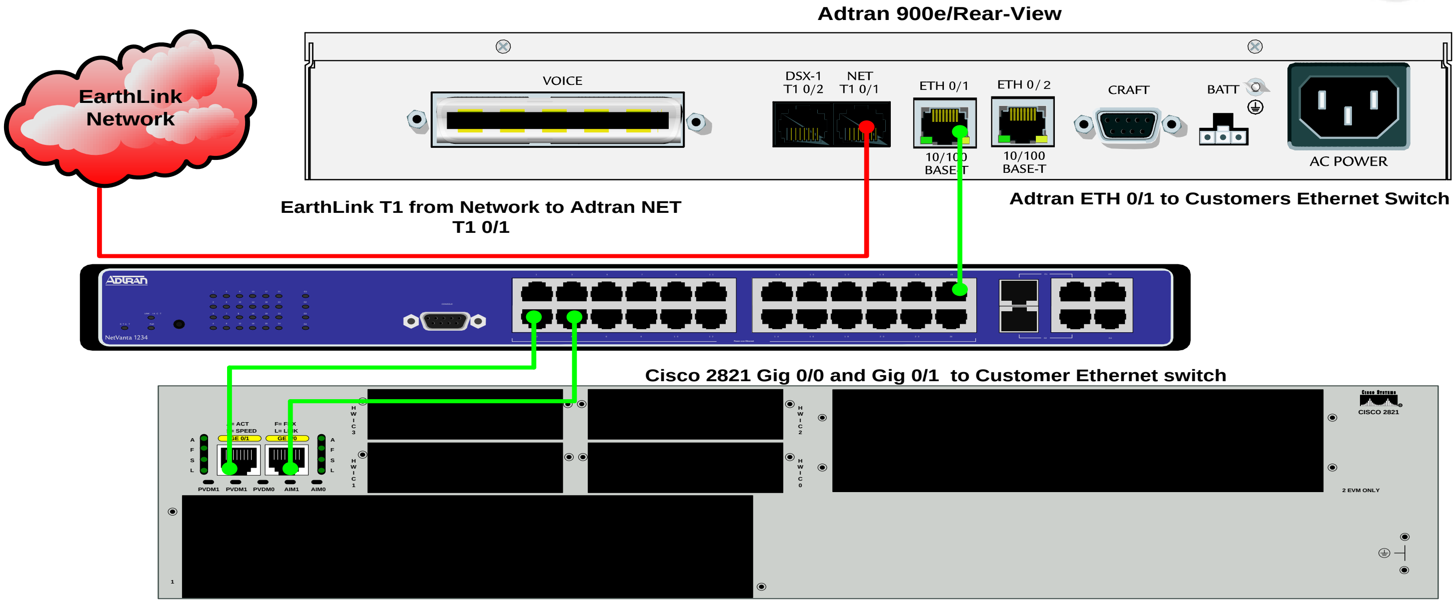

Figure 2-EarthLink SIP Trunking-Connections from Adtran CPE to IP PBX

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 A4.08

IP PBX Software Version Tested

- Cisco IOS version c2800nm-adventerprisek9-mz.124-24.T.bin
- CME version 8.1

EarthLink Open Issues & Non-Supported Features

- Registration is currently not supported for the EarthLink SIP Trunking Product.
- When the originating calling number in 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.

Cisco CME Open Issues & Non-Supported Features

- Twinning to mobile will continue to ring IP phone after call has been answered on the mobile side. Open issue with Cisco.

IP PBX Configuration for EarthLink SIP Trunking with Adtran SIP Proxy

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

Cisco CME IP PBX Configuration

The following is the configuration of the CME for SIP trunking:

DHCP configuration for phones

```
ip dhcp pool phone
  network 10.1.1.0 255.255.255.0
  default-router 10.1.1.1
  option 150 ip 10.1.1.1
```

VOIP configuration

```
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 fax protocol none
 no fax-relay sg3-to-g3
 sip
  registrar server expires max 3600 min 3600
  asserted-id ppi
  no update-callerid
  sip-profiles 1000
```

Codec Preference

```
voice class codec 1
 codec preference 1 g711ulaw
```

Voice Translation Rules

```
voice translation-rule 410
 rule 1 /^9\(.*\)/ /\1/
 rule 15 /^...$/ /5085212380/
!
voice translation-rule 411
```

```
 rule 1 /^9\(.*\)/ /ABCD9\1/
!
voice translation-rule 412
 rule 1 /^ABCD\(.*\)/ /\1/
!
voice translation-rule 1000
 rule 1 /.*/ //
!
voice translation-rule 1111
 rule 15 /^...$/ /5085212380/
!
voice translation-rule 1112
 rule 1 /^9/ //
!
voice translation-rule 2222
 rule 1 /^91900......./ //
 rule 2 /^91976......./ //
```

Interface connected to switch to Adtran 9XX

```
interface GigabitEthernet0/0.2
 encapsulation dot1Q 52
 ip address 192.168.1.2 255.255.255.0
```

Interface as default gateway for phones

```
interface GigabitEthernet0/1
 ip address 10.1.1.1 255.255.255.0
 duplex auto
 speed auto
```

IP route to send everything to Adtran

```
ip route 0.0.0.0 0.0.0.0 192.168.1.1
```

Telephony Service – phone features

```
elephony-service
 max-ephones 5
 max-dn 5
 ip source-address 10.1.1.1 port 2000
 caller-id block code *67
 load 7960-7940 P00308000500
 max-conferences 8 gain -6
```

```
call-forward pattern .T
transfer-system full-consult
transfer-pattern 9.T
transfer-pattern .T
create cnf-files version-stamp Jan 01 2002 00:00:00
```

Voip Dial Peers

```
dial-peer voice 2000 voip
 description ** cue voicemail pilot number **
 destination-pattern 401
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 voice-class sip outbound-proxy ipv4:10.1.10.1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad

dial-peer voice 1000 voip
 permission term
 description ** Incoming call from SIP trunk (One Communications) **
 session protocol sipv2
 session target sip-server
 incoming called-number .%
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1001 voip
 corlist outgoing call-local
 description ** star code to SIP trunk (One Communications) **
 destination-pattern *..
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad
```

```
dial-peer voice 1003 voip
 description ** Passthrough Inbound Calls for PSTN from CUE **
 translation-profile incoming SIP_Passthrough
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 incoming called-number ABCDT
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad

dial-peer voice 1005 voip
 description ** Passthrough Inbound Calls for MWI from CUE **
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 incoming called-number A80T
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad

dial-peer voice 1009 voip
 description ** Passthrough Inbound Calls for Internal Extensions from CUE **
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 incoming called-number ^...$
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad

dial-peer voice 1020 voip
 corlist outgoing call-national
 description **CCA*North American-10-Digit*Long Distance**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91[2-9]..[2-9]......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
```

```
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1021 voip
 corlist outgoing call-international
 description **CCA*North American-10-Digit*International**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 9011T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1022 voip
 corlist outgoing call-local
 description **CCA*North American-10-Digit*Service Numbers**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 9[2-9]11
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1023 voip
 corlist outgoing call-emergency
 description **CCA*North American-10-Digit*Emergency**
 translation-profile outgoing CALLER_ID_TRANSLATION_PROFILE
 preference 1
 destination-pattern 911
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad
```

```
dial-peer voice 1024 voip
 corlist outgoing call-emergency
 description **CCA*North American-10-Digit*Emergency**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 9911
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1025 voip
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91855.......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1026 voip
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91866.......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1027 voip
```

```
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91877.......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1028 voip
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91888.......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1029 voip
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 91800.......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 1030 voip
 corlist outgoing call-local
 description **CCA*North American-10-Digit*10-Digit Local**
```

```
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 9[2-9]..[2-9]......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad

dial-peer voice 2001 voip
 description ** cue auto attendant number **
 translation-profile outgoing PSTN_CallForwarding
 destination-pattern 444
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 voice-class sip outbound-proxy ipv4:10.1.10.1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad

dial-peer voice 1031 voip
 corlist outgoing call-local
 description star code dial peer
 translation-profile outgoing PSTN_Outgoing
 preference 1
 destination-pattern 9*..[2-9]..[2-9]......
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 dtmf-relay rtp-nte
 no vad
```

Ephone examples

```
ephone-dn  2  octo-line
 number 238
 name Mike Machnik
 call-forward busy 401
```

```
 call-forward noan 401 timeout 20
 translation-profile incoming CallBlocking

!
ephone  2
 device-security-mode none
 mac-address 0016.C79C.B1F0
 button  1:2
!
```

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**

Cisco Support (TAC): http://www.cisco.com/cisco/web/support/index.html

- **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: