EarthLink Business
SIP Trunking

Cisco UC540 IP PBX
Customer Configuration Guide

Publication History

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CHANGE HISTORY

Version	Date	Change Details	Changed By
1.0	8/30/2011	Original Document Draft	Dantley Thompson
1.0	8/30/2011	Distributed draft for Peer Review	Dantley Thompon
1.1	9/1/2011	Modified for the Cisco UC540	Mike Machnik

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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 UC540 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.

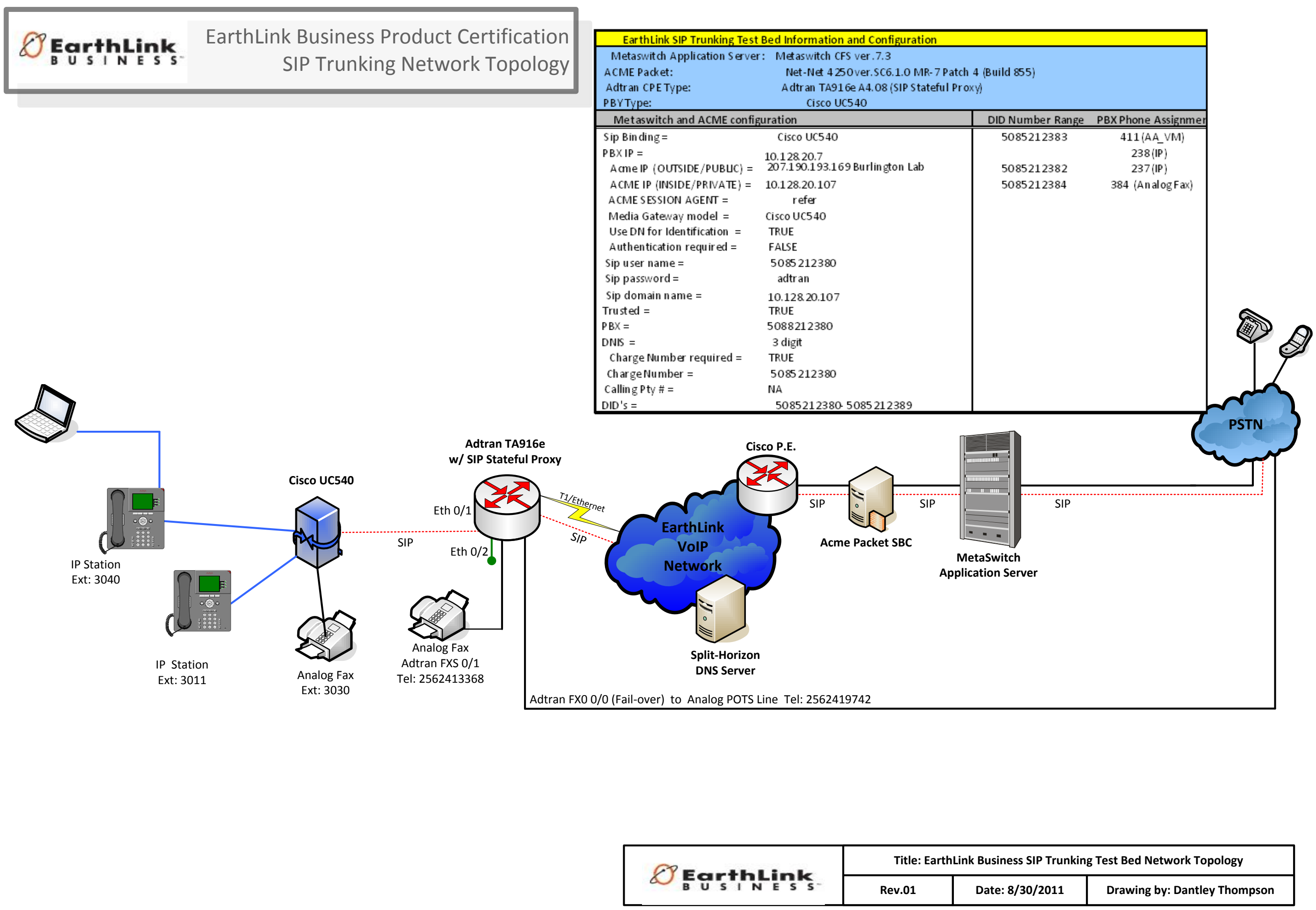

Figure 1-EarthLink SIP Trunking-Network Topology

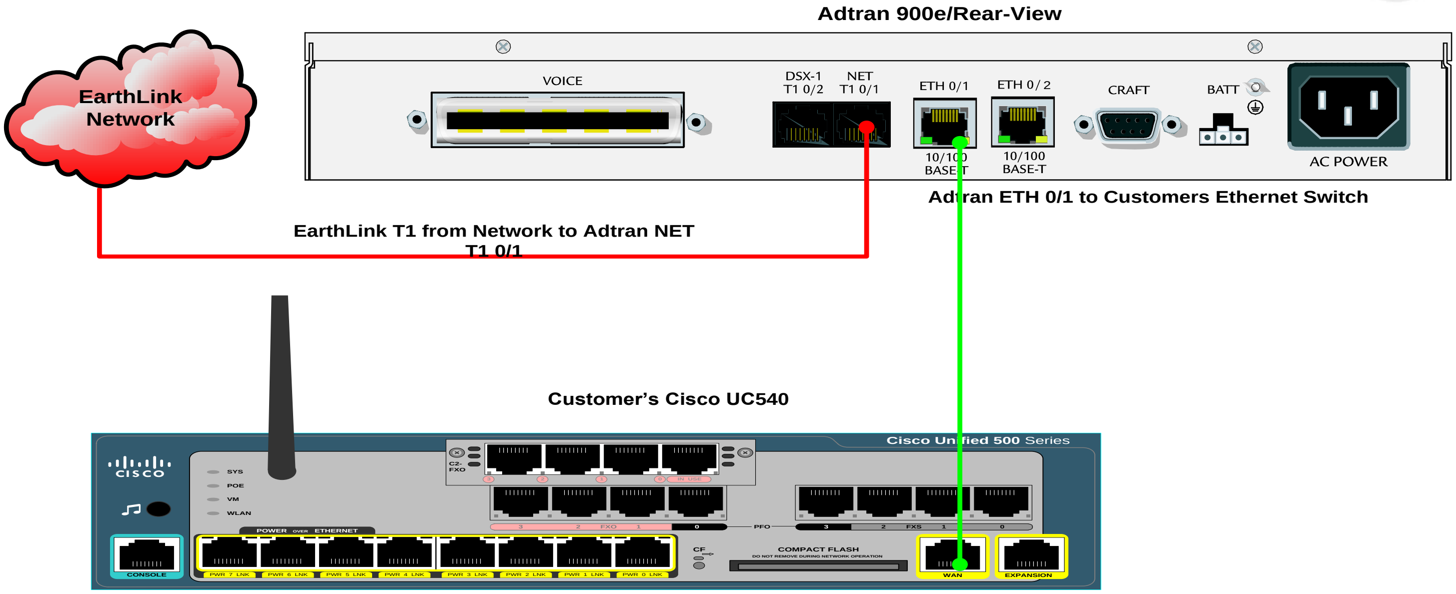

Figure 2-EarthLink SIP Trunking-Connections from Adtran CPE to IP PBX

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 UC540 version uc500-advipservicesk9-mz.151-2.T4
- 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 UC540 Open Issues & Non-Supported Features

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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 UC540 IP PBX for the EarthLink SIP Trunking Product. Basic configuration of the Cisco UC540 should be complete and the Cisco UC540 MUST be connected to the LAN prior to configuring the system for SIP Trunking.

Cisco UC540 IP PBX Configuration

The following is the configuration of the UC540 for SIP trunking:

DHCP configuration for phones

```
ip dhcp relay information trust-all
ip dhcp excluded-address 10.1.1.1 10.1.1.10

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 call send-alert
voice rtp send-recv

voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 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 pai
  no update-callerid
  sip-profiles 1000
```

Codec Preference

```
voice class codec 1
 codec preference 1 g729r8
codec preference2 g711ulaw

voice class sip-profiles 1000
 request ANY sdp-header Connection-Info remove
 response ANY sdp-header Connection-Info remove
voice class cause-code 1
 no-circuit

voice source-group CCA_SIP_SOURCE_GROUP_CUE_CME
 access-list 2
 translation-profile incoming SIP_Incoming

voice source-group CCA_SIP_SOURCE_GROUP_EXTERNAL
 access-list 3
```

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......./ //
```

```
voice translation-profile CALLER_ID_TRANSLATION_PROFILE
 translate calling 1111

voice translation-profile CallBlocking
 translate called 2222

voice translation-profile OUTGOING_TRANSLATION_PROFILE
 translate called 1112

voice translation-profile PSTN_CallForwarding
 translate redirect-target 410
 translate redirect-called 410

voice translation-profile PSTN_Outgoing
 translate calling 1111
 translate called 1112
 translate redirect-target 410
 translate redirect-called 410

voice translation-profile SIP_Incoming
 translate called 411

voice translation-profile SIP_Passthrough
 translate called 412

voice translation-profile nondialable
 translate called 1000


voice-card 0
 dspfarm
 dsp services dspfarm

fax interface-type fax-mail


license udi pid UC540W-FXO-K9 sn FHK1348727J
archive
 log config
```

```
 logging enable
 logging size 600
 hidekeys
username cisco privilege 15 secret 5 $1$DZLe$W1sAt0VEliAEhocqXOH161

bridge irb

interface Loopback0
 description $FW_INSIDE$
 ip address 10.1.10.2 255.255.255.252
 ip access-group 101 in
 ip nat inside
 ip virtual-reassembly in
```

Interface connected to Adtran

```
interface FastEthernet0/0
 description $FW_OUTSIDE$
 ip address 192.168.1.2 255.255.255.0
 ip nat outside
 ip inspect SDM_LOW out
 ip virtual-reassembly in
 load-interval 30
 duplex auto
 speed auto

interface Integrated-Service-Engine0/0
 description cue is initialized with default IMAP group
 ip unnumbered Loopback0
 ip nat inside
 ip virtual-reassembly in
 service-module ip address 10.1.10.1 255.255.255.252
 service-module ip default-gateway 10.1.10.2
```

Switchports with Phone VLAN 100

```
interface FastEthernet0/1/0
 switchport voice vlan 100
```

```
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/1
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/2
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/3
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/4
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/5
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/6
 switchport voice vlan 100
 macro description cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/7
 switchport access vlan 20
 switchport voice vlan 100
 macro description cisco-phone | cisco-phone
 spanning-tree portfast

interface FastEthernet0/1/8
 switchport mode trunk
```

switchport voice vlan 100
macro description cisco-switch

interface Vlan1
no ip address
bridge-group 1
bridge-group 1 spanning-disabled

Voice VLAN for Phones

interface Vlan100
no ip address
bridge-group 100
bridge-group 100 spanning-disabled

ip forward-protocol nd

ip http server
ip http authentication local
ip http secure-server
ip http path flash:/gui
ip dns server
ip nat inside source list 1 interface FastEthernet0/0 overload

IP route to send everything to Adtran

ip route 0.0.0.0 0.0.0.0 192.168.1.1
ip route 10.1.10.1 255.255.255.255 Integrated-Service-Engine0/0

logging esm config
access-list 1 remark SDM_ACL Category=2
access-list 1 permit 10.1.1.0 0.0.0.255
access-list 1 permit 192.168.10.0 0.0.0.255
access-list 1 permit 10.1.10.0 0.0.0.3
access-list 2 remark CCA_SIP_SOURCE_GROUP_ACL_INTERNAL
access-list 2 remark SDM_ACL Category=1
access-list 2 permit 10.1.10.0 0.0.0.3
access-list 2 permit 192.168.10.0 0.0.0.255
access-list 2 permit 10.1.1.0 0.0.0.255
access-list 3 remark CCA_SIP_SOURCE_GROUP_ACL_EXTERNAL
access-list 3 remark SDM_ACL Category=1
access-list 3 permit 207.190.193.167

```
bridge 1 route ip
bridge 100 route ip

voice-port 0/0/0
 station-id name 5085212385
 station-id number 285
 caller-id enable

voice-port 0/0/1
 caller-id enable

voice-port 0/0/2
 caller-id enable

voice-port 0/0/3
 caller-id enable

voice-port 0/1/0
 trunk-group ALL_FXO 64
 connection plar 201
 caller-id enable

voice-port 0/1/1
 trunk-group ALL_FXO 64
 connection plar 202
 caller-id enable

voice-port 0/1/2
 trunk-group ALL_FXO 64
 connection plar 203
 caller-id enable

voice-port 0/1/3
 trunk-group ALL_FXO 64
 connection plar 204
 caller-id enable

voice-port 0/4/0
```

```
 auto-cut-through
 signal immediate
 input gain auto-control -15
 description Music On Hold Port

sccp local Loopback0
sccp ccm 10.1.1.1 identifier 1 version 4.0
sccp

sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 2 register mtp00270d4b7ae0

dspfarm profile 2 transcode
 description CCA transcoding for SIP Trunk One Communications
 codec g711ulaw
 codec g729abr8
 codec g729ar8
 maximum sessions 10
 associate application SCCP

dial-peer cor custom
 name internal
 name local
 name local-plus
 name international
 name national
 name national-plus
 name emergency
 name toll-free
 name PSTN-fax


dial-peer cor list call-internal
 member internal

dial-peer cor list call-local
 member local

dial-peer cor list call-local-plus
 member local-plus
```

```
dial-peer cor list call-national
 member national

dial-peer cor list call-national-plus
 member national-plus

dial-peer cor list call-international
 member international

dial-peer cor list call-emergency
 member emergency

dial-peer cor list call-toll-free
 member toll-free

dial-peer cor list user-internal
 member internal
 member emergency

dial-peer cor list user-local
 member internal
 member local
 member emergency
 member toll-free

dial-peer cor list user-local-plus
 member internal
 member local
 member local-plus
 member emergency
 member toll-free

dial-peer cor list user-national
 member internal
 member local
 member local-plus
 member national
 member emergency
 member toll-free
```

```
dial-peer cor list user-national-plus
 member internal
 member local
 member local-plus
 member national
 member national-plus
 member emergency
 member toll-free

dial-peer cor list user-international
 member internal
 member local
 member local-plus
 member international
 member national
 member national-plus
 member emergency
 member toll-free

dial-peer cor list call-fax
 member PSTN-fax


dial-peer voice 1 pots
 destination-pattern 285
 port 0/0/0
 no sip-register

dial-peer voice 2 pots
 port 0/0/1
 no sip-register

dial-peer voice 3 pots
 port 0/0/2
 no sip-register

dial-peer voice 4 pots
 port 0/0/3
 no sip-register

dial-peer voice 5 pots
```

```
 description ** MOH Port **
 destination-pattern ABC
 port 0/4/0
 no sip-register

dial-peer voice 6 pots
 description catch all dial peer for BRI/PRI^T
 translation-profile incoming nondialable
 incoming called-number .%
 direct-inward-dial

dial-peer voice 50 pots
 description ** incoming dial peer **
 incoming called-number ^AAAA$
 port 0/1/0

dial-peer voice 51 pots
 description ** incoming dial peer **
 incoming called-number ^AAAA$
 port 0/1/1

dial-peer voice 52 pots
 description ** incoming dial peer **
 incoming called-number ^AAAA$
 port 0/1/2

dial-peer voice 53 pots
 description ** incoming dial peer **
 incoming called-number ^AAAA$
 port 0/1/3

dial-peer voice 54 pots
 description ** FXO pots dial-peer **
 destination-pattern A0
 port 0/1/0
 no sip-register

dial-peer voice 55 pots
 description ** FXO pots dial-peer **
 destination-pattern A1
 port 0/1/1
```

```
 no sip-register

dial-peer voice 56 pots
 description ** FXO pots dial-peer **
 destination-pattern A2
 port 0/1/2
 no sip-register

dial-peer voice 57 pots
 description ** FXO pots dial-peer **
 destination-pattern A3
 port 0/1/3
 no sip-register

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 58 pots
 trunkgroup ALL_FXO
 corlist outgoing call-emergency
 description **CCA*North American-10-Digit*Emergency**
```

```
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 9911
 forward-digits all
 no sip-register

dial-peer voice 59 pots
 trunkgroup ALL_FXO
 corlist outgoing call-emergency
 description **CCA*North American-10-Digit*Emergency**
 preference 5
 destination-pattern 911
 forward-digits all
 no sip-register

dial-peer voice 60 pots
 trunkgroup ALL_FXO
 corlist outgoing call-local
 description **CCA*North American-10-Digit*10-Digit Local**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 9[2-9]..[2-9]......
 forward-digits all
 no sip-register

dial-peer voice 61 pots
 trunkgroup ALL_FXO
 corlist outgoing call-local
 description **CCA*North American-10-Digit*Service Numbers**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 9[2-9]11
 forward-digits all
 no sip-register

dial-peer voice 62 pots
 trunkgroup ALL_FXO
 corlist outgoing call-national
 description **CCA*North American-10-Digit*Long Distance**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
```

```
 destination-pattern 91[2-9]..[2-9]......
 forward-digits all
 no sip-register

dial-peer voice 63 pots
 trunkgroup ALL_FXO
 corlist outgoing call-international
 description **CCA*North American-10-Digit*International**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 9011T
 forward-digits all
 no sip-register

dial-peer voice 64 pots
 trunkgroup ALL_FXO
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 91800.......
 forward-digits all
 no sip-register

dial-peer voice 65 pots
 trunkgroup ALL_FXO
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 91888.......
 forward-digits all
 no sip-register

dial-peer voice 66 pots
 trunkgroup ALL_FXO
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 91877.......
```

```
 forward-digits all
 no sip-register

dial-peer voice 67 pots
 trunkgroup ALL_FXO
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 91866.......
 forward-digits all
 no sip-register

dial-peer voice 68 pots
 trunkgroup ALL_FXO
 corlist outgoing call-toll-free
 description **CCA*North American-10-Digit*Toll-Free**
 translation-profile outgoing OUTGOING_TRANSLATION_PROFILE
 preference 5
 destination-pattern 91855.......
 forward-digits all
 no sip-register

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


no dial-peer outbound status-check pots
sip-ua
 no remote-party-id
 retry invite 2
 retry register 10
 timers connect 100
 sip-server ipv4:192.168.1.1:5060
 host-registrar
 g729-annexb override
```

Telephony Service – phone features

```
telephony-service
```

```
sdspfarm units 5
sdspfarm transcode sessions 10
sdspfarm tag 2 mtp00270d4b7ae0
video
fxo hook-flash
max-ephones 40
max-dn 300
ip source-address 10.1.1.1 port 2000
auto assign 1 to 1 type bri
caller-id block code *67
service phone videoCapability 1
service phone ehookenable 1
service dnis overlay
service dnis dir-lookup
service dss
timeouts interdigit 5
system message UC540
url services http://10.1.10.1/voiceview/common/login.do
url authentication http://10.1.10.2/CCMCIP/authenticate.asp
load 7960-7940 P00308010200
load 7941 SCCP41.8-5-4S
load 7941GE SCCP41.8-5-4S
load 7942 SCCP42.8-5-4S
load 7945 SCCP45.8-5-4S
load 7961 SCCP41.8-5-4S
load 7961GE SCCP41.8-5-4S
load 7962 SCCP42.8-5-4S
load 7965 SCCP45.8-5-4S
load 7970 SCCP70.8-5-4S
load 7971 SCCP70.8-5-4S
load 521G-524G cp524g-8-1-17
load 525G spa525g-7-4-8
load 501G spa50x-30x-7-4-8a
load 502G spa50x-30x-7-4-8a
load 504G spa50x-30x-7-4-8a
load 508G spa50x-30x-7-4-8a
load 509G spa50x-30x-7-4-8a
load 525G2 spa525g-7-4-8
load 301 spa50x-30x-7-4-8a
load 303 spa50x-30x-7-4-8a
time-zone 5
```

keepalive 30 auxiliary 4
voicemail 401
max-conferences 8 gain -6
call-forward pattern .T
call-forward system redirecting-expanded
moh flash:/media/music-on-hold.au
multicast moh 239.10.16.16 port 2000
web admin system name cisco secret 5 1cj8/$LURWpxRYbnUkX6ka5fFWB.
dn-webedit
time-webedit
transfer-system full-consult dss
transfer-pattern 9.T
transfer-pattern .T
secondary-dialtone 9
night-service day Sun 17:00 09:00
night-service day Mon 17:00 09:00
night-service day Tue 17:00 09:00
night-service day Wed 17:00 09:00
night-service day Thu 17:00 09:00
night-service day Fri 17:00 09:00
night-service day Sat 17:00 09:00
fac standard
create cnf-files version-stamp 7960 Aug 12 2011 12:43:28

Ephone Templates and Ephone examples

ephone-template 15
url services 1 http://10.1.10.1/voiceview/common/login.do VoiceviewExpress
softkeys remote-in-use Newcall
softkeys idle Redial Newcall Cfwdall Pickup Gpickup Dnd Login
softkeys seized Cfwdall Endcall Redial Pickup Gpickup Callback
softkeys connected Hold Endcall Trnsfer Confrn Acct Park
button-layout 7931 2

ephone-template 16
url services 1 http://10.1.10.1/voiceview/common/login.do VoiceviewExpress
softkeys remote-in-use Newcall
softkeys idle Redial Newcall Cfwdall Pickup Gpickup Dnd Login
softkeys seized Cfwdall Endcall Redial Pickup Gpickup Callback
softkeys connected Hold Endcall Trnsfer Confrn Acct Park

```
ephone-template  17
 url services 1 http://10.1.10.1/voiceview/common/login.do VoiceviewExpress
 softkeys remote-in-use  CBarge Newcall
 softkeys idle  Redial Newcall Cfwdall Pickup Gpickup Dnd Login
 softkeys seized  Cfwdall Endcall Redial Pickup Gpickup Callback
 softkeys connected  Hold Endcall Trnsfer Confrn Acct Park


ephone-template  18
 url services 1 http://10.1.10.1/voiceview/common/login.do VoiceviewExpress
 softkeys remote-in-use  CBarge Newcall
 softkeys idle  Redial Newcall Cfwdall Pickup Gpickup Dnd Login
 softkeys seized  Cfwdall Endcall Redial Pickup Gpickup Callback
 softkeys connected  Hold Endcall Trnsfer Confrn Acct Park
 button-layout 7931 2


ephone-dn  9
 number BCD no-reg primary
 description MoH
 moh out-call ABC


ephone-dn  294
 number 5085212380
 description SIP Main Number registration
 preference 10


ephone-dn  295  octo-line
 number 238 no-reg primary
 name abcdef
 call-forward busy 401
 call-forward noan 401 timeout 20
 translation-profile incoming CallBlocking


ephone-dn  296  octo-line
 number 237 no-reg primary
```

```
 name abcd
 mobility
 call-forward busy 401
 call-forward noan 401 timeout 20
 translation-profile incoming CallBlocking


ephone-dn  297  octo-line
 number 223 no-reg primary
 label 223
 name testy testing
 call-forward busy 401
 call-forward noan 401 timeout 20
 translation-profile incoming CallBlocking


ephone-dn  298  dual-line
 number 222 no-reg primary
 label 222
 name test testing
 call-forward busy 401
 call-forward noan 401 timeout 20
 translation-profile incoming CallBlocking


ephone-dn  299
 number A801... no-reg primary
 mwi off


ephone-dn  300
 number A800... no-reg primary
 mwi on


ephone  1
 device-security-mode none
 mac-address D0D0.FDE9.7526
 ephone-template 16
 max-calls-per-button 2
 username "test1234" password 123456
```

```
 type 525G2
 button  1:298


ephone  2
 device-security-mode none
 mac-address 0015.6247.EE22
 ephone-template 16
 username "test12345" password 123456
 type 7971
 button  1:297

ephone  3
 device-security-mode none
 mac-address 0003.6BDD.35AB
 ephone-template 16
 username "xxxxx" password 1234
 type 7960
 button  1:296

ephone  4
 device-security-mode none
 mac-address 0003.6BDD.32C0
 ephone-template 16
 username "xxxx" password 123456
 type 7960
 button  1:295
```

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: