EarthLink Business
SIP Trunking

Adtran 7100 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 Thompon
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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 Adtran 7100 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.

EarthLink Business Product Certification
SIP Trunking Network Topology

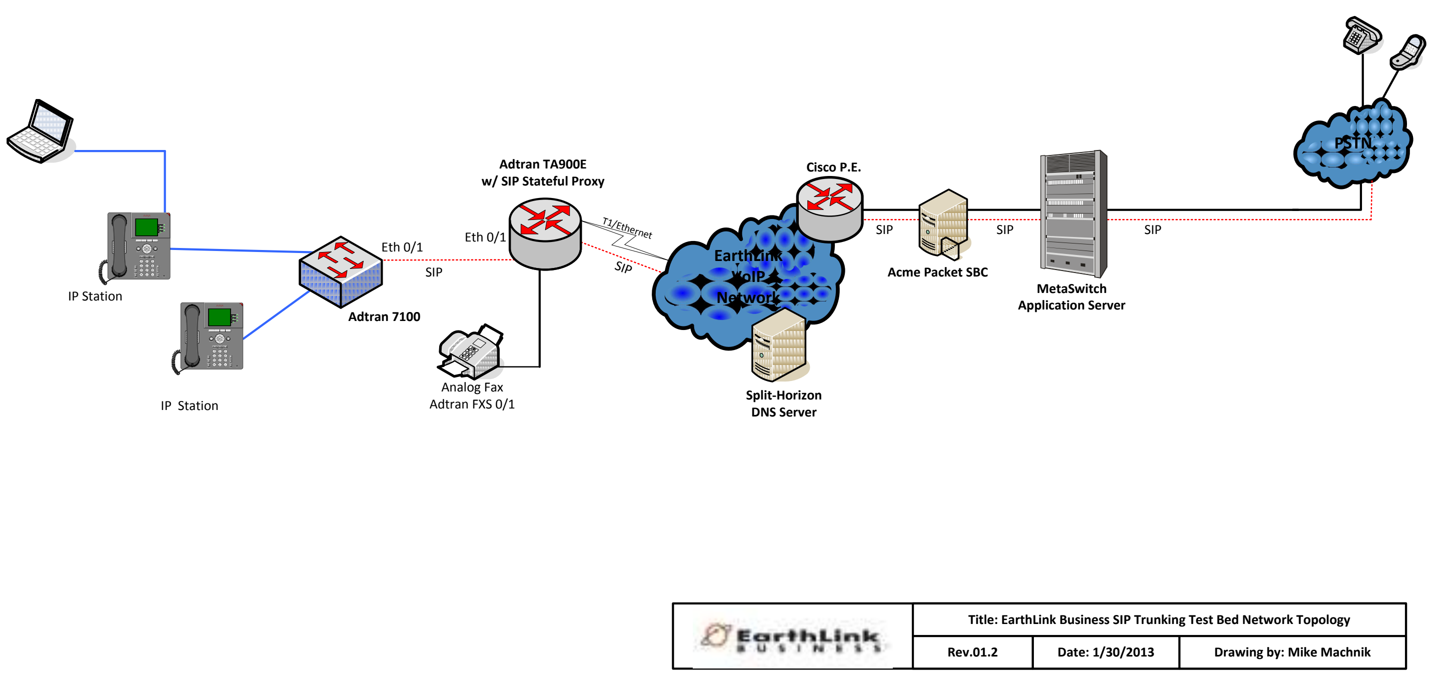

Figure 1-EarthLink SIP Trunking-Network Topology

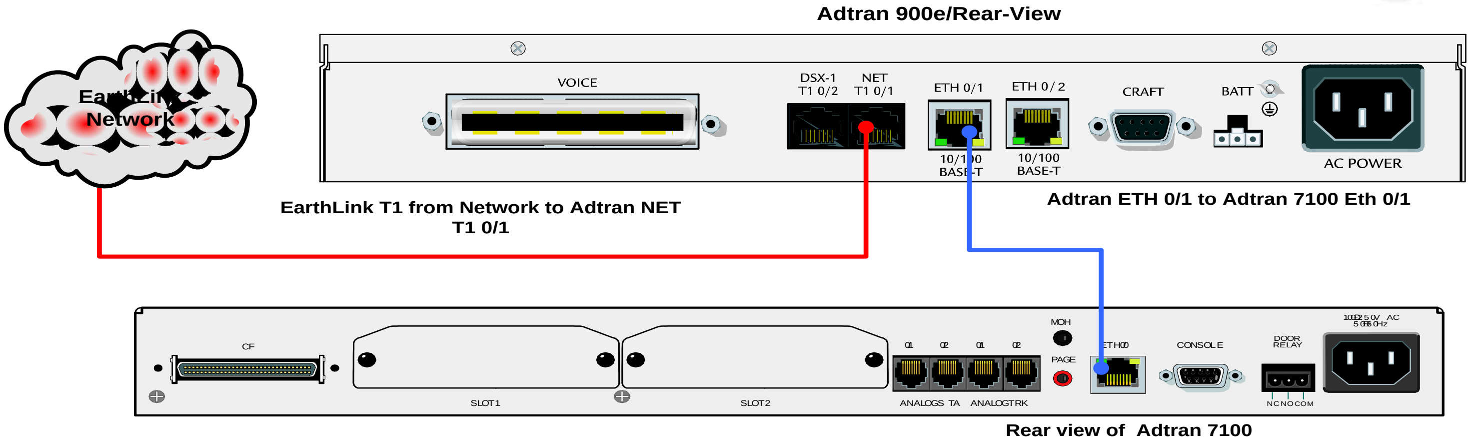

Figure 2-EarthLink SIP Trunking-Connections from Adtran CPE to Adtran 7100 IP PBX

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

- Adtran 7100 R10.3.2

IP PBX Configuration for EarthLink SIP Trunking with Adtran SIP Proxy

The Running Configuration provided below can be used as a guide for the configuration of the Adtran 7100 IP PBX for the EarthLink SIP Trunking Product. Basic configuration of the Adtran 7100 should be complete and the Adtran 7100 MUST be connected to the LAN prior to configuring the system for SIP Trunking.

Adtran 7100 IP PBX Configuration

SIP trunk configuration for Adtran 7100

Building configuration...

!

! ADTRAN, Inc. OS version R10.3.2.E

! Boot ROM version 15.01.00

! Platform: NetVanta 7100, part number 1200796E1

! Serial number LBADTN0842AC238

!

hostname "NV7100"

enable password password

!

clock timezone -6-Central-Time

!

ip subnet-zero

ip classless

ip routing

ipv6 unicast-routing

domain-proxy

name-server X.X.X.X X.X.X.X *←Primary and secondary Name Servers*

!

no auto-config

!

event-history on

no logging forwarding

no logging email

!

no service password-encryption

!

username "admin" password "password"

username "polycomftp" password "password"

!

ip firewall

ip firewall stealth

no ip firewall alg msn

```
no ip firewall alg mszone
no ip firewall alg h323
!
no dot11ap access-point-control
!
ip dhcp database local
!
ip dhcp pool "LAN_pool"
  network 10.10.10.0 255.255.255.0
  dns-server X.X.X.X  X.X.X.X  ←Primary and secondary Name Servers  netbios-node-type h-node
  default-router 10.10.10.1
  tftp-server tftp://10.10.10.1
  ntp-server 10.10.10.1
  timezone-offset -6:00
  option 157 ascii
TftpServers=0.0.0.0,FtpServers=10.10.20.1:/ADTRAN,FtpLogin=polycomftp,FtpPassword=password,Layer
2Tagging=True,VlanID=2
!
ip dhcp pool "VoIP_pool"
  network 10.10.20.0 255.255.255.0
  dns-server 10.10.20.1
  netbios-node-type h-node
  default-router 10.10.20.1
  tftp-server tftp://10.10.20.1
  ntp-server 10.10.20.1
  timezone-offset -6:00
  option 157 ascii
TftpServers=0.0.0.0,FtpServers=10.10.20.1:/ADTRAN,FtpLogin=polycomftp,FtpPassword=password,Layer
2Tagging=True,VlanID=2
!
vlan 1
  name "Default"
!
vlan 2
  name "VoIP"
!
vlan 11
  name "VLAN0011"
!
vlan 100
  name "VLAN0100"
```

```
!
interface loop 33
 ip address  172.24.17.49  255.255.255.255
 no shutdown
!
interface eth 0/0
 ip address  192.168.100.2  255.255.255.0
 ip access-policy Public
 media-gateway ip primary
 no shutdown
!
interface eth 0/1
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/2
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/3
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/4
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/5
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/6
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
```

```
interface eth 0/7
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/8
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/9
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/10
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/11
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/12
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/13
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/14
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/15
```

```
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/16
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/17
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/18
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/19
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/20
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/21
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/22
 spanning-tree edgeport
 no shutdown
 switchport mode trunk
!
interface eth 0/23
 spanning-tree edgeport
```

```
 no shutdown
 switchport mode trunk
!
interface eth 0/24
 no shutdown
 switchport mode trunk
!
interface gigabit-eth 0/1
 no shutdown
 switchport mode trunk
 switchport trunk allowed vlan 11
!
interface gigabit-eth 0/2
 no shutdown
 switchport mode trunk
!
interface vlan 1
 ip address  10.10.10.1  255.255.255.0
 media-gateway ip primary
 no shutdown
!
interface vlan 2
 ip address  10.10.20.1  255.255.255.0
 ip access-policy Private
 media-gateway ip primary
 no shutdown
!
interface vlan 100
 no ip address
 shutdown
!
interface fxs 0/1
 no shutdown
!
interface fxs 0/2
 no shutdown
!
interface fxo 0/1
 no shutdown
!
interface fxo 0/2
```

```
  no shutdown
!
isdn-number-template 0 prefix "" subscriber 911
isdn-number-template 1 prefix "" subscriber NXX-XXXX
isdn-number-template 2 prefix "" national NXX-NXX-XXXX
isdn-number-template 3 prefix 011 international X$
isdn-number-template 4 prefix "" unknown NXX
isdn-number-template 5 prefix "" unknown NXXX
isdn-number-template 6 prefix 1 national NXX-NXX-XXXX
!
ip access-list standard NAT
  remark Internet Connection Sharing
  permit any
!
ip access-list extended Admin
  remark Admin Access
  permit tcp any  any eq https   log
  permit tcp any  any eq ssh   log
  permit tcp any  any eq telnet   log
  permit tcp any  any eq www   log
  permit tcp any  any eq ftp   log
  permit tcp any  any eq ftp-data   log
!
ip access-list extended InterVLAN
  remark Voice / Data VLAN Traffic
  permit ip 10.10.10.0 0.0.0.255  10.10.20.0 0.0.0.255
  permit ip 10.10.20.0 0.0.0.255  10.10.10.0 0.0.0.255
!
ip access-list extended Private
  ! Implicit permit (only for empty ACLs)
!
ip access-list extended self
  remark Traffic to Netvanta
  permit ip any  any     log
!
ip access-list extended SIP
  remark SIP Service Provider Traffic
  permit udp any  any eq 5060
!
ip policy-class Private
  allow list self self
```

```
 allow list InterVLAN stateless
 nat source list NAT interface eth 0/0 overload
 allow list Admin self
!
ip policy-class Public
 allow list SIP self
 allow list Admin self
!
ip route 0.0.0.0 0.0.0.0 192.168.100.1 ← Default route to TA900 CPE
!
tftp server
tftp server overwrite
http server
http session-timeout 4200
http secure-server
no snmp agent
ip ftp server
ip ftp server default-filesystem cflash
no ip scp server
ip sntp server
ip sntp server send-unsynced
!
ip sip
ip sip udp 5060
ip sip tcp 5060
!
voice dial-plan 0 always-permitted 911
voice dial-plan 1 always-permitted 9-911
voice dial-plan 2 internal-operator 0
voice dial-plan 3 extensions MXXX
voice dial-plan 4 local 9-NXX-NXX-XXXX
voice dial-plan 5 long-distance 9-1-NXX-NXX-XXXX
voice dial-plan 6 toll-free 9-1-800-NXX-XXXX
voice dial-plan 7 toll-free 9-1-888-NXX-XXXX
voice dial-plan 8 toll-free 9-1-877-NXX-XXXX
voice dial-plan 9 toll-free 9-1-866-NXX-XXXX
voice dial-plan 10 operator-assisted 9-0-NXX-NXX-XXXX
voice dial-plan 11 international 9-011-$
voice dial-plan 12 900-number 9-1-900-NXX-XXXX
voice dial-plan 13 900-number 9-1-976-NXX-XXXX
voice dial-plan 14 900-number 9-976-XXXX
```

```
!
voice class-of-service normal_users
 override-passcode 6789
 default-level
 aa-dnd
 block-caller-id
 call-privilege extensions
 call-privilege local
 call-privilege long-distance
 call-privilege operator-assisted
 call-privilege specify-carrier
 call-privilege toll-free
 camp-on
 disable-callwaiting
 dnd
 door-phone
 external-fwd
 forward
 hold
 hotel
 logout-group
 message-waiting
 overhead-paging
 park
 program-user-speed
 redial
 remote-fwd
 retrieve-park
 return-last-call
 station-lock
 system-speed
 transfer
 user-speed
!
voice class-of-service public_phones
 override-passcode 1234
 call-privilege extensions
 call-privilege local
 call-privilege toll-free
 hold
 overhead-paging
```

```
 park
 retrieve-park
 transfer
!
voice class-of-service executive_users
 aa-dnd
 aa-initiate-permit $
 block-caller-id
 call-privilege extensions
 call-privilege international
 call-privilege local
 call-privilege long-distance
 call-privilege operator-assisted
 call-privilege specify-carrier
 call-privilege toll-free
 call-privilege 900-number
 camp-on
 disable-callwaiting
 dnd
 door-phone
 external-fwd
 forward
 hold
 hotel
 logout-group
 message-waiting
 overhead-paging
 park
 program-user-speed
 redial
 remote-fwd
 retrieve-park
 return-last-call
 station-lock
 system-mode
 system-speed
 transfer
 unlock-door
 user-speed
!
voice class-of-service "door phone"
```

```
 call-privilege extensions
!
voice codec-list g711_first
 default
 codec g711ulaw
 codec g729
!
voice codec-list g711
 codec g711ulaw
!
voice codec-list g729
 codec g729
 codec g711ulaw
!
voice codec-list g729_only
 codec g729
!
voice trunk T03 type sip
 description "T03"
 no reject-external
 sip-server primary 192.168.100.1 ← Interface on TA900 CPE - SIP Proxy
 codec-group g729
!
!
voice grouped-trunk "SIP TRUNK"
 trunk T03
 accept NXX-XXXX cost 0
 accept 1-NXX-NXX-XXXX cost 0
 accept 1-800-NXX-XXXX cost 0
 accept 1-888-NXX-XXXX cost 0
 accept 1-877-NXX-XXXX cost 0
 accept 1-866-NXX-XXXX cost 0
 accept 1-855-NXX-XXXX cost 0
 accept 011-$ cost 0
 accept 411 cost 0
 accept 611 cost 0
 accept 911 cost 0
 accept 0-NXX-NXX-XXXX cost 0
 accept 10-10-XXX-$ cost 0
 reject 976-XXXX
 reject 1-900-NXX-XXXX
```

```
  reject 1-976-NXX-XXXX
!
voice autoattendant "DefaultAA" extension 8200
  entry-filename "defaultAA.xml"
!
voice directory "SYSTEM"
  description "The system directory"
!
voice coverage go_to_voicemail
  coverage vm
!
voice mail max-login-attempts 3
!
voice mail class-of-service normal_voicemail
  default-level
  prompt-delete
!
voice mail class-of-service executive_voicemail
  greeting-length-max 120
  greeting-quota 5
  message-length-max 600
  message-quota 30
  prompt-delete
!
voice user 2001
  connect sip
  cos "normal_users"
  first-name "One"
  last-name "Comm"
  password "1234"
  sip-authentication password "1234"
  codec-group g711_first
  no directory-include
  voicemail cos normal_voicemail
  voicemail password "1234"
  voicemail notify schedule Sunday 12:00 am
!
!
!
voice user 2010
  connect sip
```

```
 cos "executive_users"
 first-name "670"
 last-name "2010"
 password "1234"
 group-ring-call-waiting
 did "9783781161"
 coverage global go_to_voicemail
 sip-authentication password "BGzhIHmDiuxmwqSr"
 codec-group g729
 voicemail auth-mode password
 voicemail cos normal_voicemail
 voicemail password "1234"
 voicemail notify schedule Sunday 12:00 am
!
voice user 2011
 connect sip
 cos "executive_users"
 first-name "501"
 last-name "2011"
 password "1234"
 group-ring-call-waiting
 did "9783781162"
 sip-authentication password "TH3lCPgwz5SHoDj9"
 codec-group g711_first
 voicemail auth-mode password
 voicemail password "1234"
 voicemail notify schedule Sunday 12:00 am
!
voice user 2012
 connect sip
 cos "executive_users"
 first-name "706"
 last-name "2012"
 password "1234"
 group-ring-call-waiting
 did "9783781163"
 coverage global go_to_voicemail
 sip-authentication password "AkLYfnPyx4MBhx8Y"
 codec-group g711_first
 voicemail auth-mode password
 voicemail greeting default
```

```
 voicemail cos normal_voicemail
 voicemail password "1234"
 voicemail notify schedule Sunday 12:00 am
!
!
!
voice user 2013
 connect sip
 cos "executive_users"
 first-name "550"
 last-name "2013"
 password "1234"
 group-ring-call-waiting
 did "9783781164"
 sip-authentication password "4Zeg2lRtWUmZqEXV"
 codec-group g729
 voicemail auth-mode password
 voicemail cos normal_voicemail
 voicemail password "1234"
 voicemail notify schedule Sunday 12:00 am
!
!
voice operator-group
 type all
 num-rings 4
 member 2001
 login-member 2001
 member 2002
 coverage aa 8200
 originator-id user
 voicemail password "1234"
!
!
voice ring-group 8004
 type all
 member 2003
 login-member 2003
 member 2001
 login-member 2001
 voicemail password "1234"
!
```

```
voice ring-group 8001
  type linear
  description Test ring auto attendant
  num-rings 0
  coverage vm 5001
  voicemail password "1234"
!
voice music-on-hold mode internal
!
voice music-on-hold player System
  default
  file 5 JazzClub.wav
!
no ip sip authenticate
!
ip sip registrar
!
ip sip database local
!
ip rtp dtmf-relay min-duration 100
!
ip rtp quality-monitoring
ip rtp quality-monitoring udp
ip rtp quality-monitoring sip
!
line con 0
  no login
!
line telnet 0 4
  login local-userlist
  no shutdown
line ssh 0 4
  login local-userlist
  no shutdown
!
sntp server time.nist.gov
!
end
NV7100#
```

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**

Allworx: http://www.allworx.com/support/support_overview.aspx

- **Mon – Fri 8am-8pm EST 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: