EarthLink Business
SIP Trunking

Allworx 6x IP PBX – SIP Proxy
Customer Configuration Guide
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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 Allworx 6x in Proxy mode with the Adtran CPE. 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) and will not be configured as a SIP Proxy. The Allworx 6x will handle the natting and the 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 will 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
- Allworx 6x 7.7.5.5
- Allworx Phones 9204 and 9212 version 2.3.3.3

EarthLink Open Issues & Non-Supported Features
- Registration is currently not supported for the EarthLink SIP Trunking Product.

Allworx 6x Open Issues & Non-Supported Features
- SIP REFER is not sent to the network. SIP REFER is handled on the LAN by the Allworx between phones.
IP PBX Configuration for EarthLink SIP Trunking with Adtran CPE

The steps below provide a step by step guide for configuration of the Allworx 6x IP PBX in SIP Proxy mode for the EarthLink SIP Trunking Product. Basic configuration of the Allworx 6x should be complete and the Allworx 6x MUST be connected to the LAN prior to configuring the system for SIP Trunking.

Allworx 6x IP PBX Configuration

SIP trunk configuration for Allworx 6x using SIP Proxy.

From the home page go into Configuration:
Modify the Network and configure the WAN IP and Gateway. The gateway will be the CPE interface IP.
Configure static default route to Earthlink CPE.
Configure the VOIP Server under Servers on the dashboard. RTP and SIP DSCP will be marked EF and DTMF payload is 101.
Configure the SIP Proxy from the dashboard under Phone System, Outside Line. SIP Server is the domain name for Earthlink’s SBC
Modify Dial Plan to route external calls out the SIP Proxy. Phone System – Dial Plans. Create a Service Group to use the Proxy and apply the group to services to be used by the Proxy.

DHCP and DNS configurations if needed and not provided by customer LAN.
From the dashboard under Maintenance restart the Allworx. Follow prompts to save and reboot.
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/support/support.xea
  - 24x7 Support Availability

Allworx: http://www.allworx.com/support
  - 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: