SIP Trunking Workshop

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VP Product Management

Agenda

- What is SIP Trunking
- What is Open Source Asterisk
- Why are they important together
- One perspective on future of SIP Trunking
- How to take the next step

What is SIP Trunking?

- The new dialtone
- SIP is technical name for a phone line
  - **S**ession **I**nitiation **P**rotocol
  - Standards-based: RFCs
  - “IP” version of TDM Trunks
- Allows basic interoperability between heterogeneous equipment and services
  - SIPConnect is the SIP Forum’s interop test suite/specs
  - Customers still require “piece of mind”
- Some companies just have introduced SIP, others have had for years
- Today, “SIP Trunking” Certification is critical success factor to market acceptance

What are the Major Benefits of SIP?

- Reduces costs
- Facilitates IP convergence simplifying networks and providing end-to-end unified communications
- Facilitates IP applications which are more powerful, more economical, and more flexible than TDM
- Eliminates gateways reducing complexity
- Enables wide variety of services from your SIP Service Provider

What does a user look for from their PBX for SIP?

- SIP Standards
- SIP Forum Member
- SIP Connect Certified
- SIPit members/participants
- Certified with several SIP Service providers

What are the biggest SIP Trunking Challenges?

- Different implementation of SIP RFC standards
- Use of reserved fields in SIP standards for proprietary extensions
- Constant changes by SIP Providers in access protocol
- Availability for multi-city/multi-country connections
- Understanding of SIP
- Trusting reliability after years of proven TDM

Asterisk: the Open Source Project

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InfoWorld **2008 Technology of the Year Award: BEST IP PBX**

The Asterisk Community

- **Over 1,540,000 downloads in 2008**
- **Over 1.03M downloads 1st half 2009**
- **Over 63,000 active participants on forums**
- **Over 28,500 Topics**
- **Over 92,000 Posts**
- **Over 17,700 on active Asterisk mailing lists**
- **Over 7,248 on our Bug Tracker**
- **Over 820 active contributors**
- **Over 2,000 new code commits in 2009**
- **Over 200 service providers worldwide using Asterisk**
- **Dedicated Industry Events:**
  -AstriCon for Asterisk Developers and Users
  -Digium Asterisk World for Business Users

The Asterisk Community was voted

#1 Most Influential Individual in VoIP, VoIP Info 11/2006

End-User OSS Adoption

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Buy a complete OSS-based product

- IP PBX (based on OSS)
- Media Gateways (based on OSS)
- Soft Switch (based on OSS)

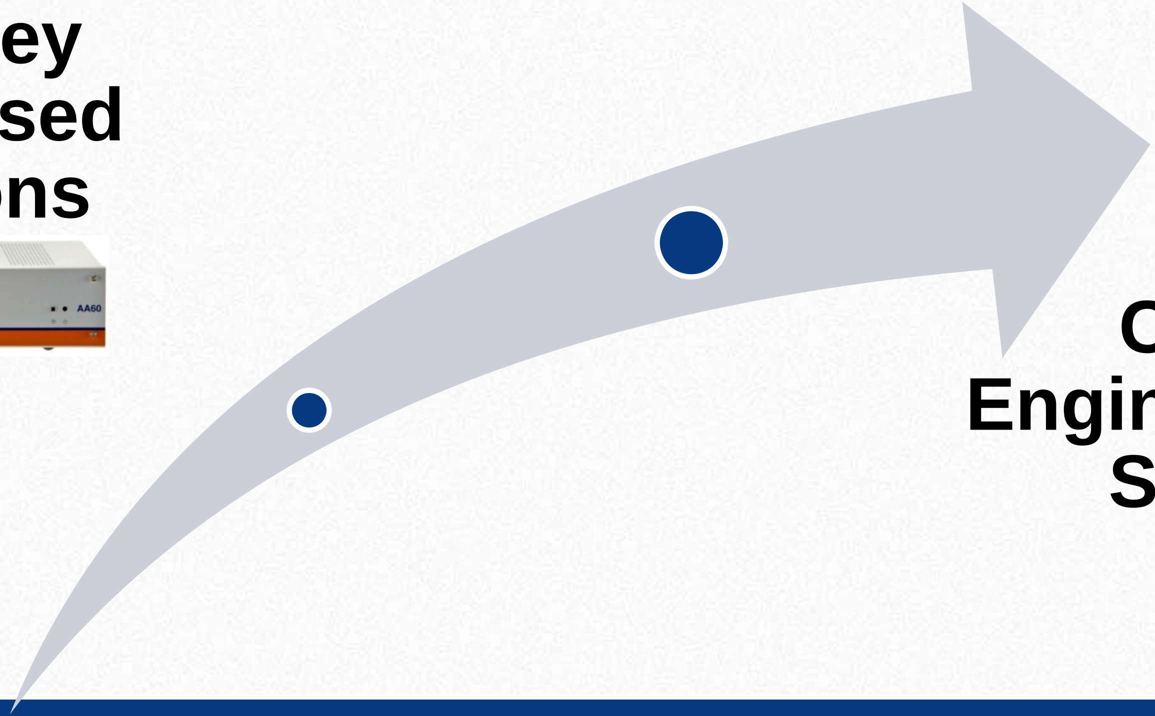
Build you own – or have an integrator build a solution for you

- OSS Software
- Commercial Software
- Hardware Platforms
- Training
- Services

Tool Kit Components

OSS Adoption as a Function of End User Technical Skill

Turn-Key OSS-Based Solutions



Custom Engineered OSS Solution

Lo …. End-User Technical Competency ……. Hi

Open Source to SIP Trunks

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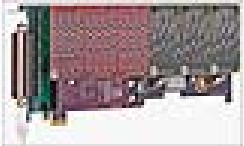
SIP Trunking
to Service Provider



Turn-Key Business Phone Systems

Custom Telephony Solutions

Lo ....End-User Technical Competency .... Hi

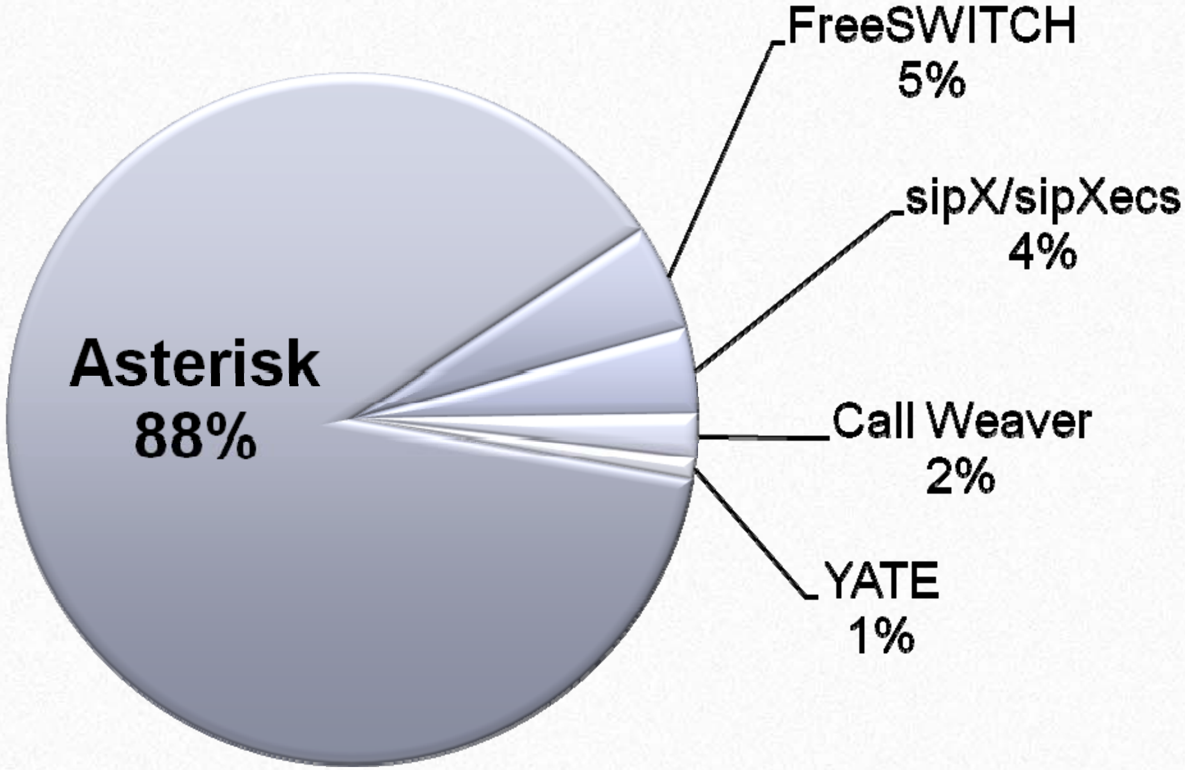
What is needed to connect Asterisk to a SIP Trunk?

- SIP (SP) Provider Name
- Account ID: is the username SP provided.
- Password: user/administrator assigned
- Hostname/IP Address
- Callback Extension: The default extension to ring when receiving a call over this
- DTMF Mode: The DTMF mode to use when sending and receiving DTMF tones to/from SP
- Advanced Options: caller ID, Port #, various other params, codecs
- Set calling rules inbound/outbound, DIDs

(Abridged)

Why is SIP trunking and Asterisk Important?

Based on most recent implementation



Source: Eastern Management Group

Digium Perspective on the SIP Trunking Industry

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- Small # Users
- Small companies
- Trials
- Analysts “Reports”
- Interoperability “Early”
- SIPConnect Important

2007-2008

- SMB Ramp-up
- Small Companies
- Enterprise Trials
- Analysts Market Share
- Interoperability “Good”
- SIPConnect Important

2009-2010

- SMB Stronghold
- Enterprise Acceptance
- Analysts Market Share
- Interoperability “non-issue”
- SIPConnect assumed

2011-2012

- What’s Next?
- RFC Updates?
- Who Knows?
- Embedded elements for social media? New media?

2013-2014

Taking the Next Step with Asterisk?

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1. See Steve Sokol in next session after this panel who will share some infrastructure applications
2. Stop into the Digium booth (#206) during exhibits to see open source in action
3. Attend Thursday sessions on open source: Tristan Degenhardt (8:30 AM) and Steve Sokol (2 PM)
4. Download AsteriskNOW (www.asterisknow.org) or Switchvox Free (www.digium.com/switchvox) and connect to a SIP Trunk and you have a business phone system

The End
Thank You

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