



Asterisk Update

OpenSIPS Summit 2017

David Duffett

Worldwide Community Director, Asterisk

Twitter: @dduffett

Email: dduffett@digium.com





- **Asterisk 13**
- **Asterisk 14**
- **Hope, Encouragement and Inspiration**
- **The Future**
- **The Challenge**

But first...

- Let's get to know each other better!

**You don't have to be great to start,
but...
you do have to start to be great!**

Joe Sabah

Background

- Mark Spencer creates Asterisk in 1999
 - Started a Linux Support business
 - Needed a phone system
 - Did not like the options
 - Cost
 - Vendor lock in
 - Decided to create his own
 - Made it Open Source
 - Digium is born in 2002

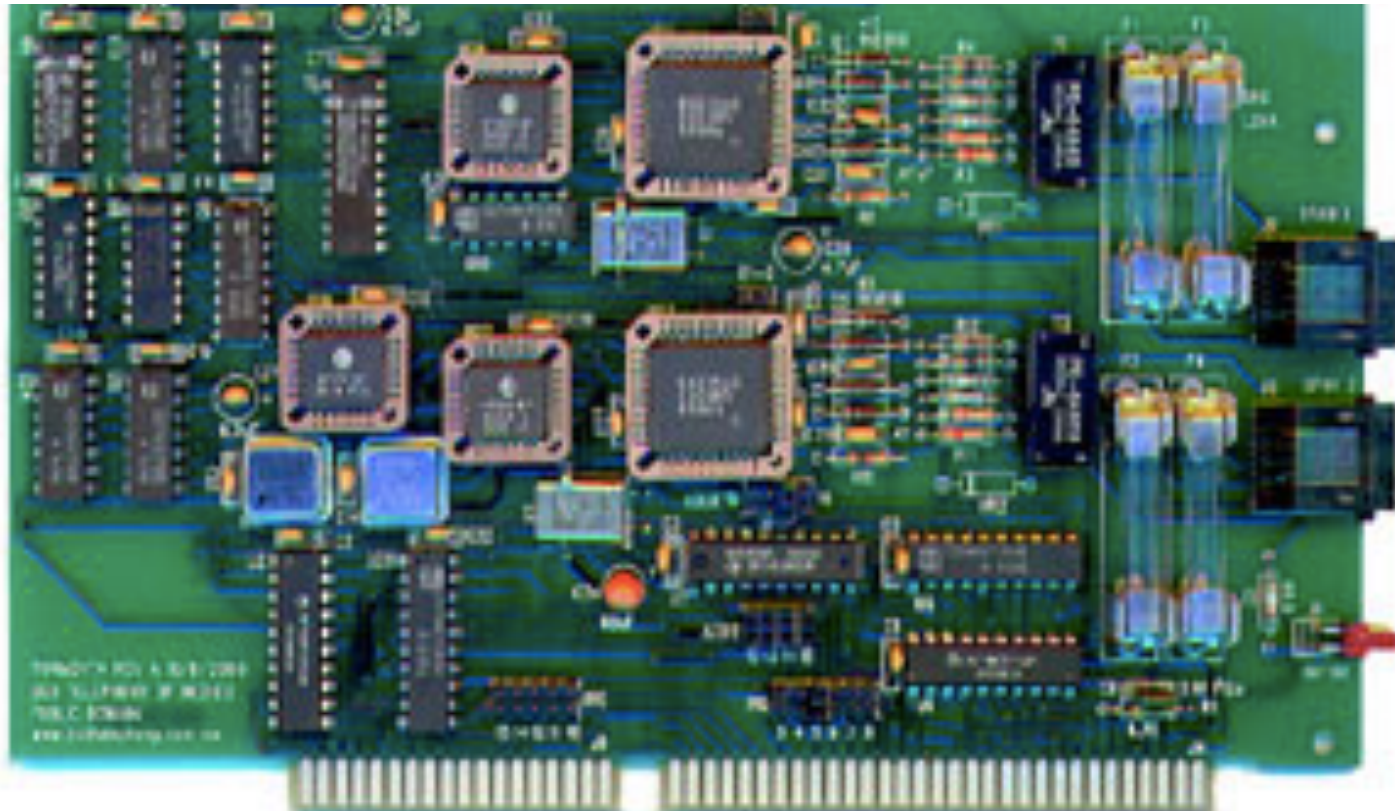


Background

- Zapata Telephony project, Jim Dixon c.2000
 - Saw that the advanced in general purpose CPUs meant that they would be able to do things previously done only by expensive DSPs
 - Created Open Source Telephony Hardware
 - Used with Asterisk



Zapata 'Tormenta' Card



Asterisk 13

- Current LTS, Building on Asterisk 12 – which introduced
 - New SIP stack – PJSIP
 - New API – the ARI, or Asterisk RESTful Interface
 - New BRIDGING model

Asterisk 14

- DNS Overhaul
- Publish Extension States to a SIP subscription server
- Playback of media from a remote HTTP server via a URI
- Enhanced ARI media manipulation
- ARI Channel creation more sophisticated
- wiki.asterisk.org

Asterisk 14

- Now we can publish extension state to an external entity with the SIP PUBLISH method.
- PUBLISH sip:presentity@example.com SIP/2.0
Via: SIP/2.0/UDP
<pua.example.com;branch=z9hG4bK652hsge>
[To: <sip:presentity@example.com>](mailto:sip:presentity@example.com)
[From: <sip:presentity@example.com>;tag=1234wxyz](mailto:sip:presentity@example.com;tag=1234wxyz)
[Call-ID: 81818181@pua.example.com](mailto:81818181@pua.example.com)
[CSeq: 1 PUBLISH](#)
[Max-Forwards: 70](#)
[Expires: 3600](#)
[Event: presence](#)
[Content-Type: application/pidf+xml](#)

Asterisk 14: Why Publish?

- **Remove state from Asterisk**
- **Improved scalability**
- **Allows offloading of individual subscription management**
- **Separating concerns**

Supported Body Types

- **application/dialog-info+xml**
- **application/pidf+xml**
- **application/xpidf+xml**

pjsip.conf

[test-esc]

type=outbound-publish

server_uri=sip:172.16.0.100

from_uri=sip:172.16.0.100

event=dialog

@body=application/dialog-info+xml

@context=^users

@exten=^1000

Autohint support

**[users]
autohints=yes**

equivalent to:

exten => alice, hint, PJSIP/alice

What can I connect with?

- **OpenSIPS**
- **Asterisk**
- **Kamailio**
- **Or anything else that can act as a SIP presence aggregator**

Credits

Many thanks to:

Josh Colp <jcolp@digium.com>

[Kevin Harwell <kharwell@digium.com>](mailto:kharwell@digium.com)

[Asterisk development team](#)

The Asterisk Wiki

- An excellent resource. Full of useful information and tutorials, etc.
- A couple of examples:

wiki.asterisk.org/wiki/display/AST/New+in+13

wiki.asterisk.org/wiki/display/AST/New+in+14

How do we learn?



The power to PLAY

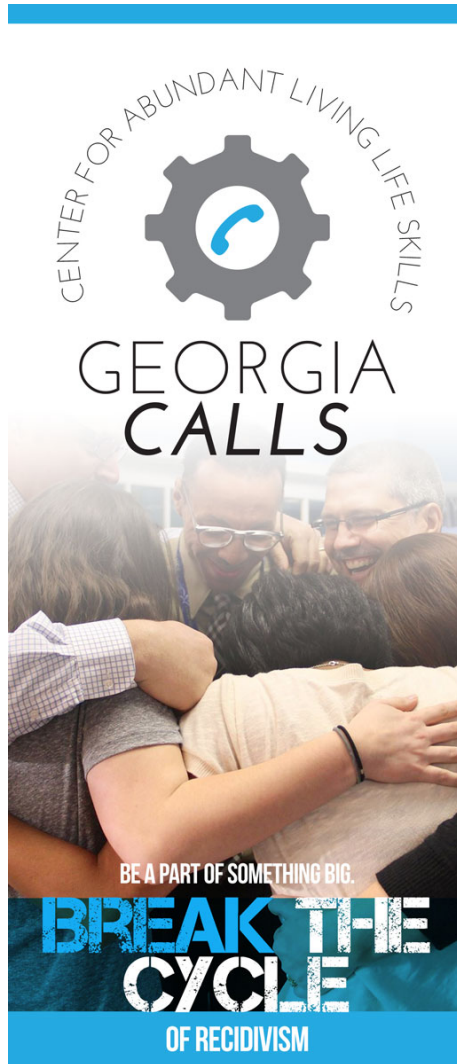


The Gift of HOPE

- Bicycle Powered (and Solar Powered) Asterisk!
- Brings communications to remote communities
- Deployments include Western Uganda

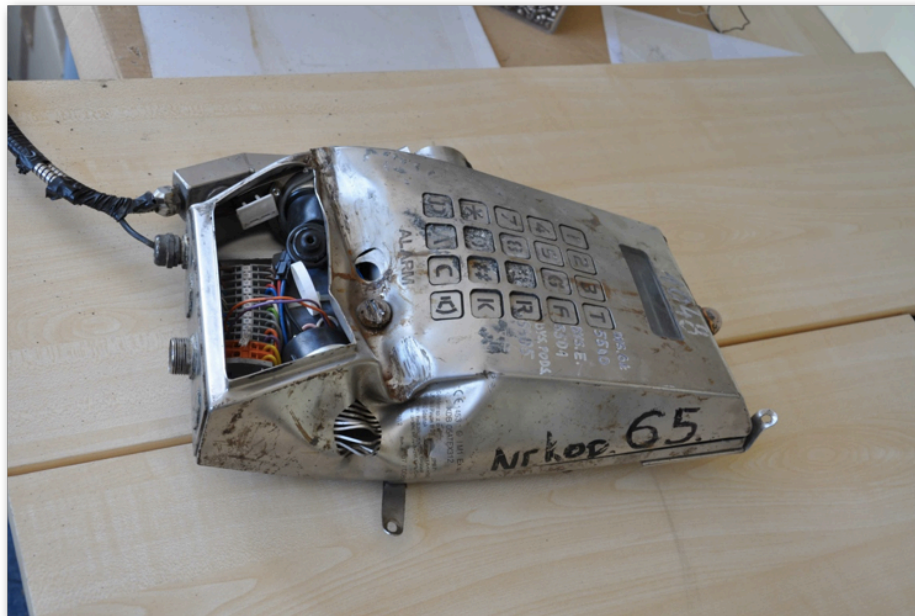


The gift of ENCOURAGEMENT



The gift of INSPIRATION

- The Coal Mine - a heavily regulated environment
- Mission Critical – literally Life or Death!



Jakub Klaus, SS7 Technologies

The Power to PLAY!



Asterisk fosters GROWTH!

- The next generation of RT Communications
- Asterisk as a dynamic media server
- Scalable, resilient systems
- Partnering Projects
 - OpenSIPS
 - Homer
 - Docker

AstriCon 2017

- October 3-5
- Orlando, FL in the USA
- www.astricon.net
 - Speaking
 - Exhibiting
 - Participating

Digium® IP Phones for Asterisk®

Digium IP Phones allow you to take full advantage of the flexibility and customization of Asterisk.



Simple Configuration

—
Plug-and-Play Provisioning

—
Powerful Apps

—
HDVoice Quality

www.digium.com/products/ip-phones

Copyright © 2012 Digium Inc. All rights reserved. Digium, Asterisk, and Asterisk for Linux are trademarks of Digium Inc. All other trademarks are the property of their respective owners.

Digium Phones

- The only phones designed specifically with Asterisk in mind
- Standard SIP, but with extra goodies when used with a properly configured Asterisk
- Buy Digium to help the Asterisk project

We are all on the same side!

- We are all part of something bigger:
Open Source Communications Community
- Asterisk
- FreeSWITCH
- OpenSIPS
- Kamailio
- And others...

You can have everything in life you want, if you will just help enough other people get what they want.

Zig Ziglar

We have received so much.
What can we give?

dduffett@digium.com

Twitter: @dduffett

