VoIP and FreeBSD

The daemon meets the phone

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Agenda

- Introduction
- Terms
- Introduction to Asterisk key concepts
- Let's connect to a provider
- What's a dialplan ?
- How cool is an IVR...
- (if time permits) AGI overview

Who am I ?

- First of all, good morning
- I'm Max, nice to meet you all
- I worked on VoIP and FreeBSD for the last 3,5 years implementing technologies for businesses large and small
- I'm now working at BrianTel Srl, delivering voice services over geographical Wi-Fi Networks (more on this @ other talk)

Why am I doing this ?

- VoIP is business, but you have to know how to deal with it (and have right equipment)
- It's not rocket science, but it's a totally different environment for computer professionals
- I have some deal of experience
- I don't feel the need to keep others from doing what I do 1000's miles away
- It's fun !

Let’s start

- A few questions to let me understand the level of knowledge of the class
- If you have any question further on, raise your hand at any time any time

TERMS

Terms 1/5

- Direct Inward Dial
- It's a real PSTN number which lets the call into your VoIP system.
- Normally works on a PRI
- It's normally intended as a phone number
- Can be bought from many different providers and forwarded to your asterisk box via any provider

DID

Terms 2/5

- Voice Circuit (may carry data as well)
- Can carry either 24 (T1) or 30 (E1) b-channels (audio) and 1 d-channel (for data communication across peers). It can also be sliced, and carry less channels, if less are needed
- Voice is carried using a-law or μ-law, depending on the part of the World you are in
- Needs specific hardware to interconnect to our OS and PBX (either Digium or Sangoma cards)

PRI

Terms 3/5

- Namely, when you have a mean for PSTN users to reach you via a telephone number, DID

Origination

- When your calls leave your VoIP system and get into the PSTN network to reach the dialed party

Termination

- Different types of routes have different names (white, gold, so on...)

Terms 4/5

- Interactive Voice Response
- It's a way to have defined procedures to send callers into
- They can have some backed-logic behind, offering real services (directories, payment gateways, e-commerce) using simple languages (PERL, PHP, Python, many more) via AGI.

IVR

Terms 5/5

- Asterisk Gateway Interface
- It’s a means of “interconnetting” the asterisk toolbox with any programming language out there
- Very useful in many situations
- May introduce a lot of overhead, too

AGI

Is Asterisk a PBX ?

- Yes and no...
- It may be better considered a toolkit
- It can be used as a simple PBX, but it can also deliver any type of telephone service out there (IVR, toll free, complete carrier solutions, call centers, calling cards applications)

Is VoIP ready for production use ?

- Lots of people use skype every day, why wouldn't they use a better-suited solution to their needs ?
- Skype is a software, I'm speaking of a <u>real</u> telephone system
- Customization is the key
- Definitely, YES !

What's the deal with faxes ?

- Faxes are a real problem, businesses (still) rely heavily on them, so they have to be perfect
- One of the biggest needs
- T.38 may be the way, but we're far from reality (we're testing it, though)
- Hylafax is ready and rolling, and IAXModem complements it
- **Faxes do not work 100% on IAX/SIP**

The Asterisk toolkit and libraries

- Asterisk is composed of:
  - Libpri
    - Library for PRI devices
  - Zaptel
    - Drivers for Digium hardware, both PRI and BRI
  - SpanDSP
    - Libraries for faxing (to be deprecated)
  - Asterisk itself
    - Core
    - Apps
    - Modules

The ports system

- What's available in ports ?

  Asterisk 1.4.18.1

  Libpri 1.4.1

  Zaptel 1.4.6

  A little mess (partially inherited from linux...)

Pay particular attention to zaptel, as I've had frequent crashes with the version in ports. Use the subversion trunk instead !

What hardware is available ?

- Pretty much anything from one PSTN line to a DS3 card (beware)
- Main makers are Digium, Sangoma and Junghanns
- Sangoma is the most freebsd-friendly (a developer is here at the conference as well)

Installing Asterisk

- We will install Asterisk without any interface, so it should be straightforward

Just type:

Cd /usr/ports/net/asterisk

Make install WITHOUT_ODBC=yes WITHOUT_H323=yes WITHOUT_FAX=yes

Echo 'asterisk_enable=”yes”' >> /etc/rc.conf && /usr/local/etc/rc.d/asterisk.sh start

Done

Asterisk -vvvvvvvr to connect to it

SIP and IAX

- SIP is a wider standard, adopted by many vendors, devices and softwares
- IAX is project-specific, but more powerful and efficient
- SIP uses RTP as transport protocol, IAX is all-in-one, on a single port for both data and voice (4569 by default) (sip uses 5060 + RTP)
- IAX can do “trunking” (SIP does it as well, beta)
- IAX has a jutterbuffer mechanism (SIP in beta)

sip.conf

- Defines both users and “gateways”
- Easy syntax, hard concepts to understand
- A telephone must only have a definition
- A “gateway” must have both a definition and a “register” line
    - Register => username:pass@provider.com/exten

extensions.conf 1/6

- Easy, assembly-like syntax
  - Context
  - Extension number
  - Priority
  - Application
  - (Options)

exten => 1001,1,dial(SIP/1001,20,jo)

exten => 6000,1,Answer()

extensions.conf 2/6

- Old (preferred) behavior is to scroll through priority numbers in order, jump to priority(+101) if application errors
- You have to use options “jo” in applications in order to restore this
- This leads to great freedom in defining powerful rulesets within minutes

```
exten => 1001,1,SetCallerID($;MYNUMBER)
exten => 1001,2,Dial(SIP/$NUMBER)
exten => 1001,3,Dial(IAX/$NUMBER)
exten => 1001,103,goto(3)
exten => 1001,104,busy(16)
exten => 1001,105,hangup()
```

extensions.conf 3/6

- Extensions can (must) be grouped into contexts
- Contexts let you split up your extensions into smaller chunks
  - Easier to manage
  - Chance to have more people with same numbering plans (Hosted PBX's, for example)
- Contexts are heavily used to create IVR's

```
[Default]
exten => 1001,1,SetCallerID($;MYNUMBER)
exten => 1001,2,Dial(SIP/$NUMBER)
exten => 1001,3,Dial(IAX/$NUMBER)
exten => 1001,103,goto(3)
exten => 1001,104,busy(16)
exten => 1001,105,hangup()

[MyNum]
exten => 1001,1,SetCallerID($;MYNUMBER)
exten => 1001,2,Dial(SIP/$NUMBER)
exten => 1001,3,Dial(IAX/$NUMBER)
exten => 1001,103,goto(3)
exten => 1001,104,busy(16)
exten => 1001,105,hangup()
```

extensions.conf 4/6

- Pattern matching can happen in the dialplan
- Every pattern should start with “_”
- X matches any digit from 0 to 9
- Z matches any digit from 1 to 9
- N matches any digit from 2 to 9
- [15-7] Matches any digit or range specified
- “.” is a wildcard

```
exten => _1800.,1,Dial(SIP/${EXTEN})

exten => 1800.,1,Dial(SIP/${EXTEN})
```

extensions.conf 5/6

- Macros can be defined
- Simply create a context whose name starts with macro-$name
- Macros can be given arguments to work with
- They are directly called from the dialplan when needed

```
[macro-greet]
exten => s,1,NoOp("Hello, " ${ARG1}"!")

[Default]
exten => 3000,1,macro(greet, 1)
```

extensions.conf 6/6

- Variables can be defined and used inside your dialplan

USER=john

Defines the var $USER, assigns "john" to it

EXTEN=${NUM:1}

Gives us EXTEN as NUM with first digit truncated

Apps in the dialplan

- Dial()
- Goto()
- GotoIf()
- GotoIfTime()
- Playback()
- Background()
- Answer()
- Hangup()
- Meetme()
- Voicemail()
- MusicOnHold()
- Queue()
- Busy()
- Congestion()

Our IVR

- Here's how to implement a simple IVR to direct calls over to the right person every time
- Use vox to convert your audio files into the right format (i.e. Vox $infile -b -u -r 8000 -v $outfile.wav) and you're done. Otherwise use the voicemail() application to record the messages
- Always remember to use the "m" flag to dial when passing calls over, otherwise people won't hear anything waiting for the other party to answer

AGI Overview

- AGI is an application interface to have interaction between custom-made applications and asterisk
- There are libraries written for almost any language (PHP, Perl, Python, Ruby for example)
- Easy to use and powerful

Easy Faxing

- SpanDSP
- Everything is done in the dialplan, with a shell script to do fax2mail
- No error-control onboard, it's like sending TCP packets via UDP...
- Lines have to be <u>REALLY</u> clean
- It hardly gives good results

More complex faxing

- IAXmodem
- IAXmodem connects as a normal IAX user, passes the call to Hylafax
- Hylafax (a real fax server) processes the fax and does all the magic
- Hylafax can also be on a remote machine
- Very good results (even on SIP trunks)
- Neat quality

More Topics

- Conferencing (app_meetme2, web_meetme, app_conference)
- CDRs
- chan_capi
- SER, sipxpbx
- You name it...

Any more questions ?

- Ask me, if there's still time
- Go to http://www.voip-info.org for a nice guide about VoIP
- Go to http://www.briantel.com/learnvoip to find more materials (online soon)
- mailto:stucchi@briantel.com
- The Asterisk book from O'Reilly is a nice guide

Thanks for coming

enjoy your meal