Open Source VoIP
...it’s easy, and actually quite useful!

NZNOG 2008
SysAdmin Miniconf

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Introduction

- VoIP *systems* - the legacy telephone bits are important still!
- Open source VoIP options
- Quick Overview of Asterisk
- A look at TrixBox, an Asterisk based ‘pretty’ PABX
- Asterisk CLI Configuration
- Real examples

Analogue Telephony

- Where it all started!
- PSTN allows connection between any two endpoints
- Human speech typically in the range 250 - 3,000Hz
  - Humans can hear in the region of 20 - 20,000Hz
- PSTN analogue channel originally designed to carry 300 - 3,500Hz
- Most analogue lines delivered via copper from the local exchange (or CO, Central Office)
  - Average line in NZ ~3Km. Longest lines >7Km

Analogue Telephony

- Even in the day and age of VoIP, this is still important!
  - Analogue telephone adapters (ATAs)
  - Fax - it just won’t go away :)
  - Echo
  - Voice and sound is most definitely analogue
    - First and last conversions in a VoIP call

The Analogue Telephone

- Analogue telephones connect to a copper pair
  - A two wire circuit
- Analogue telephones are comprised of five major parts:
  - Ringer
  - Dial Pad
  - Hybrid
  - Hook switch
  - Handset

Ringer

- The exchange provides DC (~48vDC) to power the phone
  - Exchange = big centralised UPS
- Exchange provides a burst of AC (~80vAC) to ring the phone's bell
  - Originally a mechanical bell, these days an electronic buzzer
- These days phone have a Ringer Equivalence Number (REN)
  - Exchange can power up to a sum of 5 RENs
  - Phones these days typically < 0.5 REN
  - ATAs have same limitation

Dial Pad

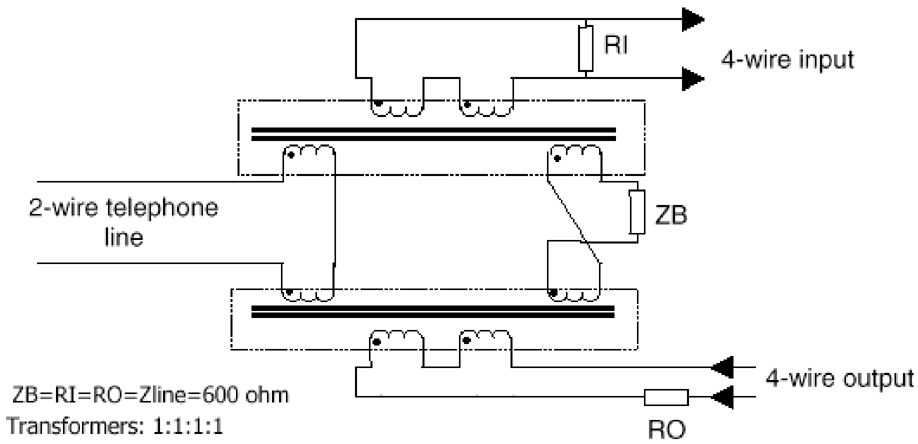
- Telephones need to signal back to the exchange
- Originally done with a rotary dialler making and breaking the copper loop
    - Pulse Dial, still typically supported by exchanges and some VoIP kit
- All done with audio tones now
    - Dual Tone Multi Frequency (DTMF)
    - Telephone handsets a matrix of switches
    - One tone per column, one per row
    - Each switch generates two tones, hence Dual Tone

DTMF Tones

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Hybrid Network

- The heart of an analogue telephone
- The transformer that couples two signals onto one line
  - Send (Tx) and receive (Rx)
- Creates sidetone ('good echo')
  - Allow speaker to hear himself
- Creates echo unless perfectly balanced

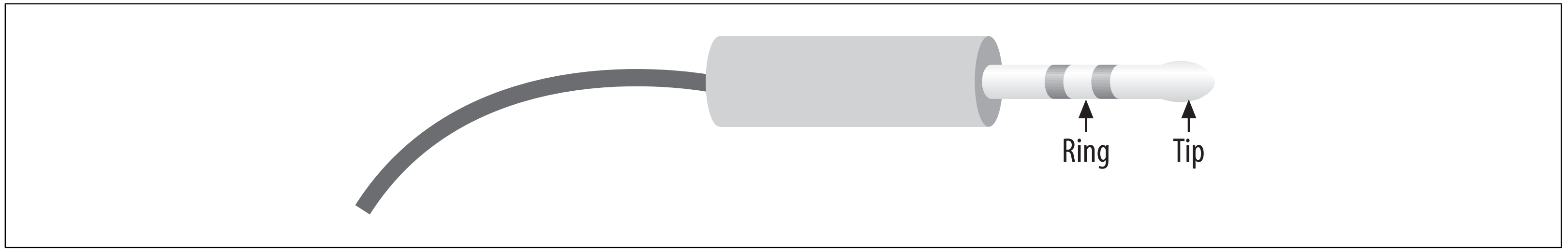


Hook Switch

- Telephone uses it to signal state to the exchange
  - On Hook, closes the copper loop
    - Phone idles, waiting for incoming ring
  - Off Hook, breaks the copper loop
    - Requests dial tone from the exchange, and then allows audio to pass
- Also used to signal ‘advanced’ features, e.g. call waiting
  - Hook Flash - a timed closure of the hook switch, typically ~300ms

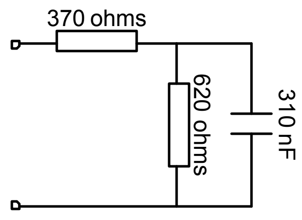
Tip and Ring

- Telephony world often refers to ‘Tip’ and ‘Ring’
- Historical term from the days when exchanges were literally switchboards
- Operator manually patched lines together
- Tip (red) = +ve polarity (0v)
- Ring (green) = -ve polarity
    - -48v on hook, -7v off hook



Telephone and Line Impedance

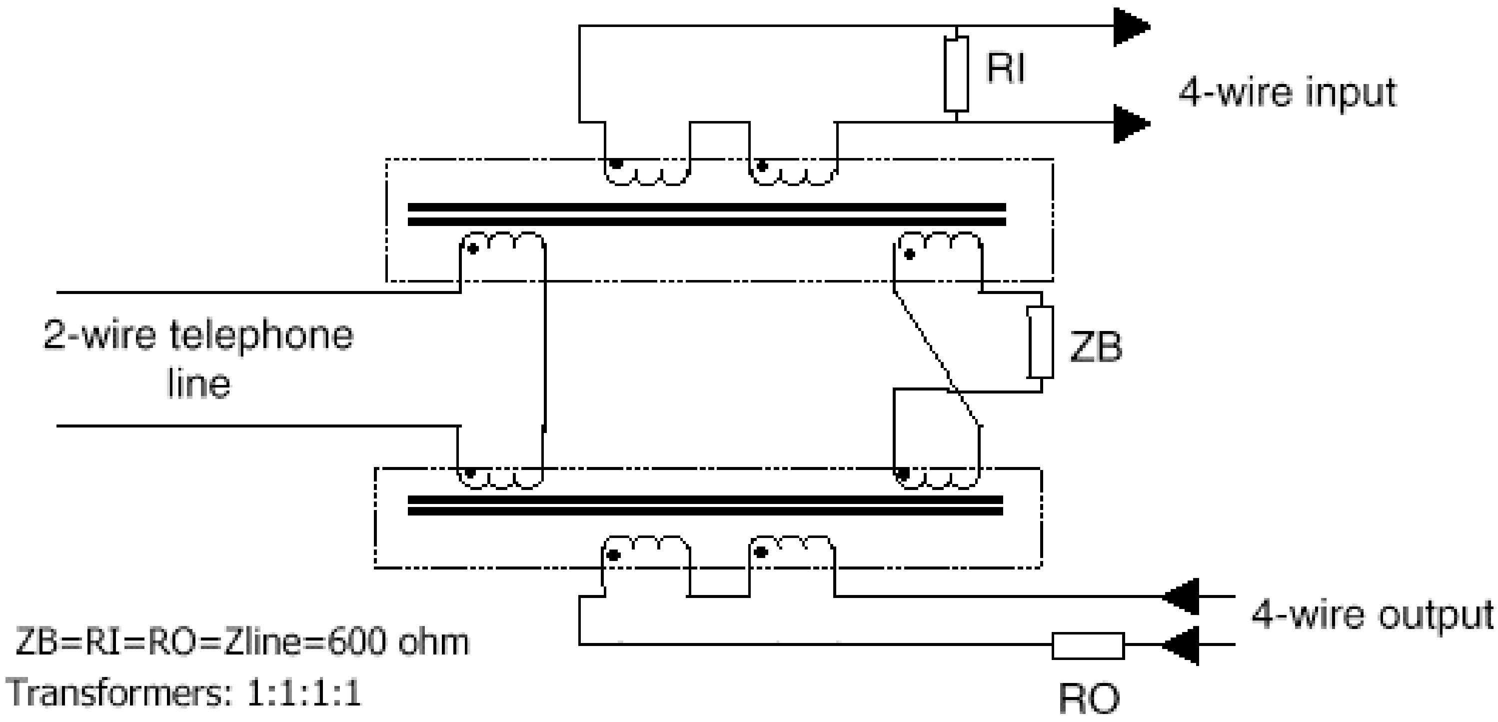
- Impedance = technical way of saying resistance
  - Varies with both frequency and phase
- American telephone impedance is 600 ohms
  - Approximation of the impedance of 0.4mm twisted copper pair at voice frequencies
- British (and NZ) telephone impedance is complex (in the resistive sense of the word), called BT3
  - 370 ohms in series with (620 ohms in parallel 310nF)
  - Attempt to better match line impedance



Echo

- VoIP does not cause Echo!
  - Hybrids cause echo
  - Echo becomes apparent as latency increases
  - VoIP creates higher latency than circuit switched circuits
- Hybrids must be balanced to the line to effect maximum power transfer and minimal signal reflection
  - Reflection back down the line = echo
  - Reflection back towards the handset = sidetone

Echo - Telephone Hybrid



Echo

- Sidetone is used to let the user know that the phone ‘is working’
  - It’s somewhat unnatural to not her oneself
  - Too much sidetone and you can only hear yourself
  - Too little and it appears the line is dead
- Echo is present on most lines
  - When latency is low ($<$ 20ms or so) the far end perceives it as sidetone

Acoustic Echo

- Caused by the output of the handset's speaker entering the microphone
  - Due to the speaker volume being too loud or microphone sensitivity too loud
    - Very bad with softphones when not using a headset
  - Or flimsy handset construction (acoustic coupling through the handset itself)
  - The telephone handset design hasn't changed much over the years as it is a very good one!
- Indistinguishable to the far end from echo caused by the local hybrid

Reducing Echo

- There are only four ways to reduce echo
    - Remove the two wire (analogue) portion of the call
    - Balance the analogue portion of the call better
        - Hard to do even if you do have access to the endpoint(s)
    - Reduce the latency
        - Often impossible, e.g. long distance calls
    - Cancel the echo

Echo Cancellers

- Measure signal on the line, predict the echo, and create a signal to cancel it
- Echo cancellers are configured for a ‘tail’ length - the maximum latency of an echo which it can possibly cancel
- Takes time to converge to an echo cancelled state, dependant on the tail length of the canceller
- Echo cancellers aren’t perfect, so best to prevent echo in the first place
- Popular misconception that software based echo cancellation is bad.
  - Hardware echo cancellers have very good, often patented algorithms
  - No really good open source software implementations (yet...)
  - Software echo cancellation is not bad - if you have a good algorithm!

Digital Telephony

- Telephony moved digital for the same reason everything else did
- Voice turned to a digital signal using Pulse Code Modulation (PCM)
    - Sample signal in time
- Two important factors:
    - Number of samples per second (highest frequency is half of the sample rate - Nyquist's Theorem)
    - Number of bits used to encode signal
- Tradeoff between quality and bandwidth - standard is 8bits at 8kHz sampling

Digital Telephony

- Standard voice channel (timeslot, or DS0) is 64kbit/s
- Most common codec is G711, a companding codec
    - Two types, ulaw (US) and alaw (Europe)
- Majority of telephone conversation is 'quiet'
- More bits are allocated to quiet signals to improve overall quality

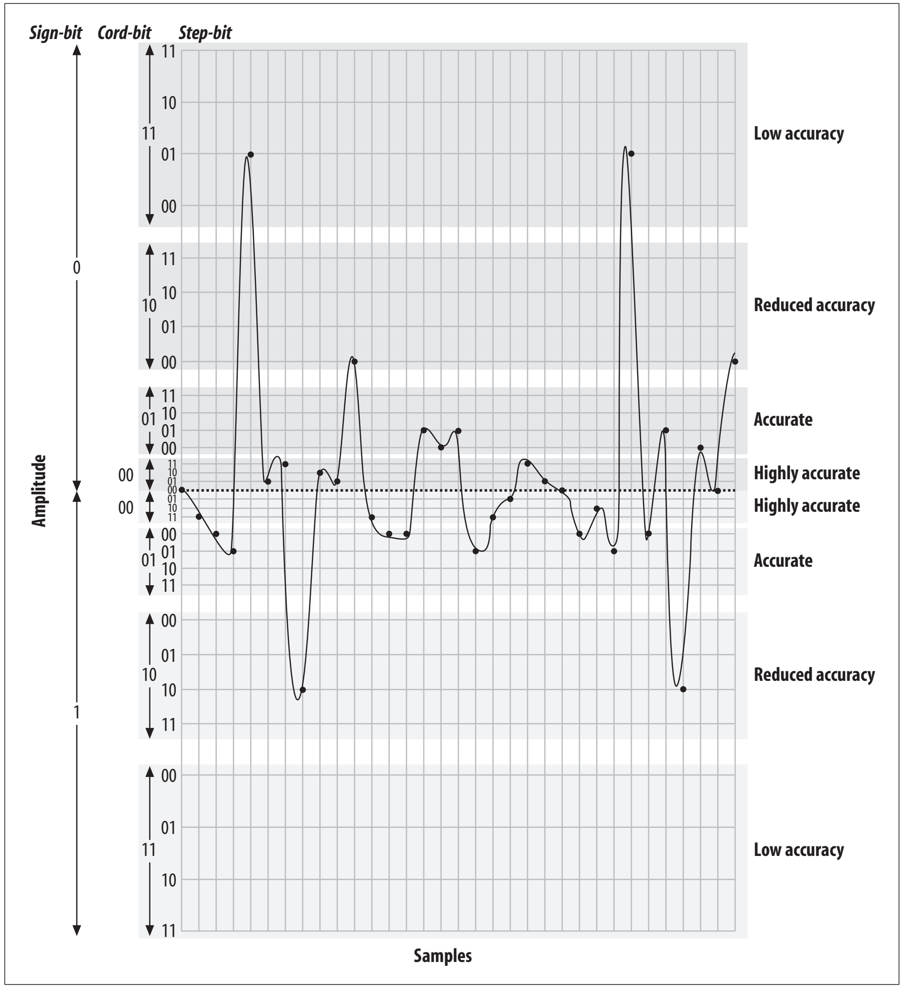


Figure 7-12. Quantized and companded at 5-bit resolution from Asterisk, The Future of Telephony

PSTN Circuits

- Analog line
- ISDN
  - Basic rate, two voice 64kbit/s voice channels + 16kbit/s data channel -> 144kb/s
  - Primary rate
    - US - T1, 24 64kbit/s voice channels -> 1.544Mb/s line rate
    - Europe - E1, 30 64kbit/s voice channels -> 2.048Mb/s line rate
- Proprietary circuits between key phones and PBXs - not covered here

VoIP

- Natural progression from digital telephony
    - Circuit switched --> packet switched
    - Still a need to sample and encode signals
- Many different codecs in the VoIP world
- Many different signalling protocols

Open Source VoIP Options

- Open source call processing components (PBX, SIP Proxy)
- Softphones (SIP User Agents)
- Tools

PBXs / SIP Proxies

- Asterisk - www.asterisk.org
- SIP Express Router (SER) - www.iptel.org/ser/
- sipX - www.sipfoundry.org
- FreeSwitch - www.freeswitch.org

Softphones

- Ekiga (standard in a lot of Linux window environments)
- Kphone (Linux)
- SFLPhone (multiplatform)
- OpenWengo (multiplatform)
- xten xlite softphone (well, not open source, but free version available)

VoIP Tools

- SIPbomber - www.metalinkltd.com/downloads.php
  - SIP proxy testing
- SIPp - sipp.sourceforge.net
  - Performance testing tool
- pjsip-perf - www.pjsip.org
  - SIP call performance testing

What is Asterisk

- Asterisk, *The* Open Source PBX. www.asterisk.org
- A complete PBX in software
- Runs on virtually any OS
- Support for most VoIP protocols
- Most full-featured PBX features already built in
    - MOH, conferencing, queues, voicemail, IVR...
- Supports many different hardware telephony cards

Asterisk Documentation

- There’s lots of info all over the place, some of it contrary though
- www.voip-info.org
  - Lots of really good information, lots of plain wrong information too!
  - Defacto documentation store at this stage
- www.asterisk.org
- www.digium.org - hardware cards
- Asterisk CLI !

Asterisk Versions

- Three versions currently in popular use:
  - 1.0 - obsolete, but it’s good and stable and I like it
  - 1.2 - stable in most releases still
  - 1.4 - most use this
  - 1.6 - latest and greatest

Asterisk File Locations (debian)

- /etc/asterisk/ - Asterisk configuration files
- /var/lib/asterisk/ - contains the astdb, firmware and keys
- /usr/share/asterisk/sounds - in built asterisk sound prompts
- /var/spool/asterisk/ - temporary files and voicemail files
- /var/log/asterisk/ - Asterisk log files
- /var/log/asterisk/cdr-csv/ - Asterisk call detail records

How Asterisk Works, in one slide or less :)

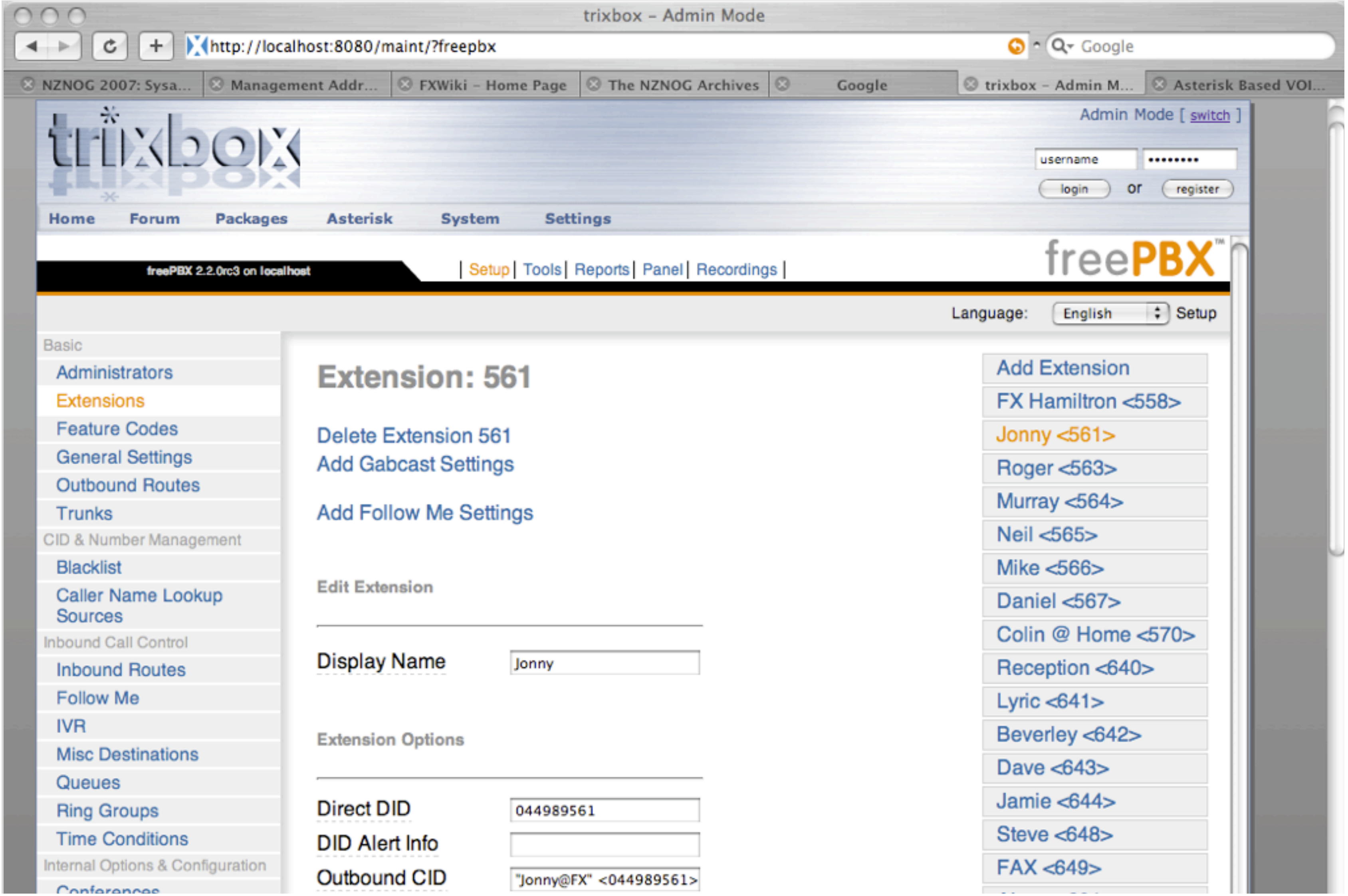
- Asterisk is a hybrid TDM and packet voice PBX
- Interfaces any piece of telephony hardware or software to any application
- Prime components: channels and extensions.conf - the Asterisk dial plan
- Channels can be many different technologies - SIP, IAX, H323, skinny, Zaptel, and others as they are created
- extensions.conf is basically a programming language controlling the flow of calls
- Applications do the work - answer a channel, ring a channel, voicemail, etc.

TrixBox

- www.trixbox.org
- Asterisk PBX up and running in one hour
- The PBX formally known as Asterisk@Home
- Latest version = 2.4, based on Asterisk 1.2
- Full featured PBX system including all the regulars:
    - Voicemail, conferencing, call forwarding, extensions
- Provides web based interface, which in turn drives Asterisk configuration files

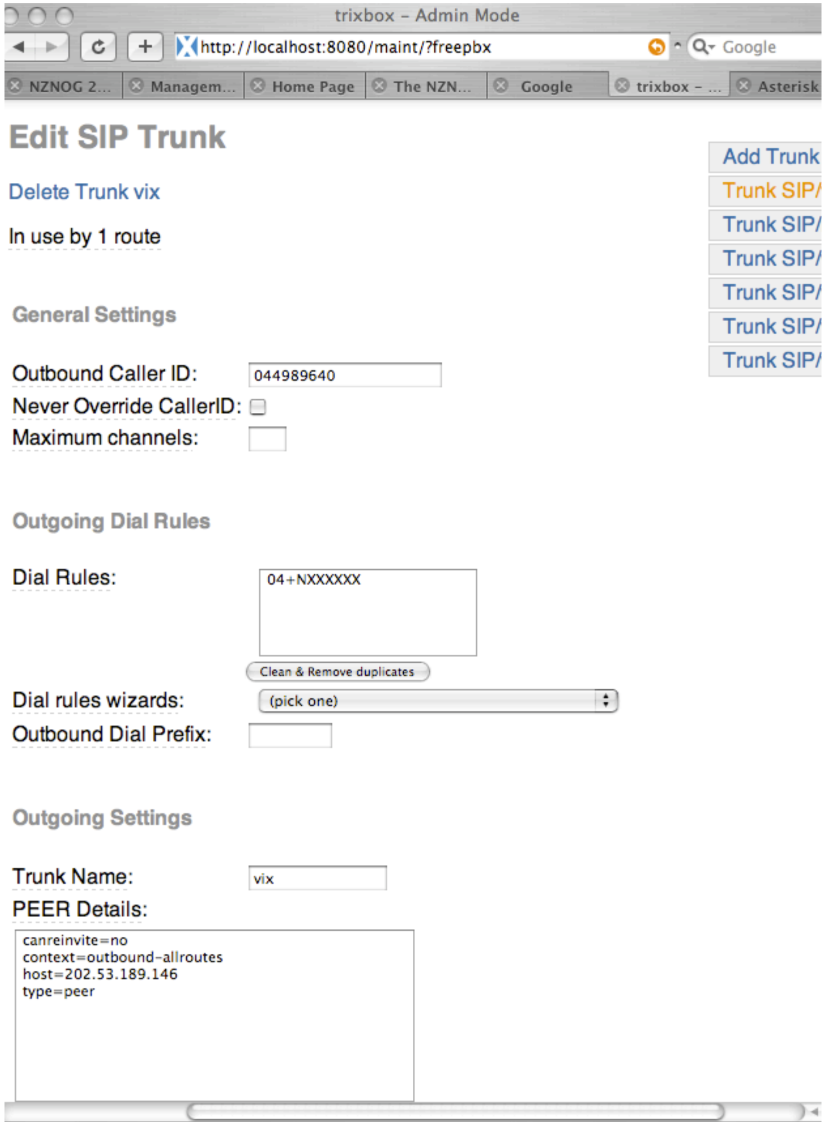
TrixBox

- Ties together several applications:
  - freePBX - the web interface configurator
  - A2Billing - call reporting
  - Flash Operator Panel (FOP) - telephone status panel
  - Munin - host monitoring
  - Several Others

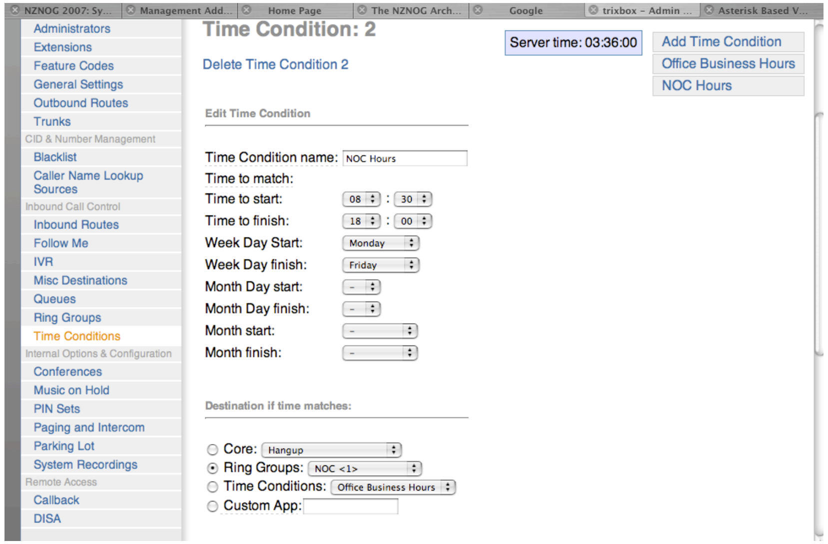


trixbox - Admin Mode
http://localhost:8080/maint/?freepbx
Google
NZNOG 2007: Sysa...
Management Addr...
FXWiki - Home Page
The NZNOG Archives
Google
trixbox - Admin M...
Asterisk Based VOI...
trixbox
Admin Mode [switch]
username
login
or
register
Home
Forum
Packages
Asterisk
System
Settings
freePBX 2.2.0rc3 on localhost
| Setup | Tools | Reports | Panel | Recordings |
freePBX
Language: English Setup
Basic
Administrators
Extensions
Feature Codes
General Settings
Outbound Routes
Trunks
CID & Number Management
Blacklist
Caller Name Lookup Sources
Inbound Call Control
Inbound Routes
Follow Me
IVR
Misc Destinations
Queues
Ring Groups
Time Conditions
Internal Options & Configuration
Extension: 561
Delete Extension 561
Add Gabcast Settings
Add Follow Me Settings
Edit Extension
Display Name
Jonny
Extension Options
Direct DID
044989561
DID Alert Info
Outbound CID
"Jonny@FX" <044989561>
Add Extension
FX Hamiltron <558>
Jonny <561>
Roger <563>
Murray <564>
Neil <565>
Mike <566>
Daniel <567>
Colin @ Home <570>
Reception <640>
Lyric <641>
Beverley <642>
Dave <643>
Jamie <644>
Steve <648>
FAX <649>

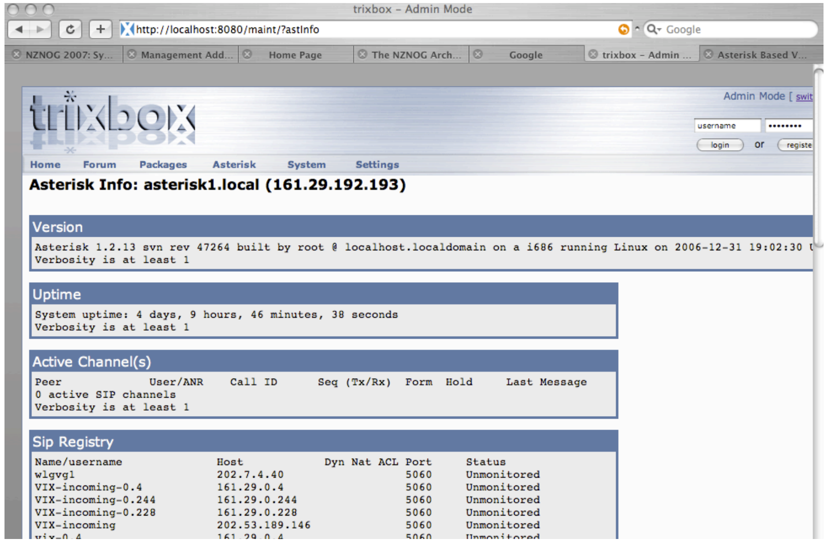
TrixBox



trixbox - Admin Mode
http://localhost:8080/maint/?freepbx
Edit SIP Trunk
Delete Trunk vix
In use by 1 route
Add Trunk
General Settings
Outbound Caller ID: 044989640
Never Override CallerID:
Maximum channels:
Outgoing Dial Rules
Dial Rules: 04+NXXXXXX
Clean & Remove duplicates
Dial rules wizards: (pick one)
Outbound Dial Prefix:
Outgoing Settings
Trunk Name: vix
PEER Details:
canreinvite=no
context=outbound-allroutes
host=202.53.189.146
type=peer



Time Condition: 2
Delete Time Condition 2
Server time: 03:36:00
Add Time Condition
Office Business Hours
NOC Hours
Administrators
Extensions
Feature Codes
General Settings
Outbound Routes
Trunks
Blacklist
Caller Name Lookup Sources
Inbound Routes
Follow Me
IVR
Misc Destinations
Queues
Ring Groups
Time Conditions
Conferences
Music on Hold
PIN Sets
Paging and Intercom
Parking Lot
System Recordings
Callback
DISA
Edit Time Condition
Time Condition name: NOC Hours
Time to match:
Time to start: 08 : 30
Time to finish: 18 : 00
Week Day Start: Monday
Week Day finish: Friday
Month Day start:
Month Day finish:
Month start:
Month finish:
Destination if time matches:
Core: Hangup
Ring Groups: NOC <1>
Time Conditions: Office Business Hours
Custom App:



trixbox - Admin Mode
http://localhost:8080/maint/?astinfo
trixbox
Home Forum Packages Asterisk System Settings
Asterisk Info: asterisk1.local (161.29.192.193)
Version
Asterisk 1.2.13 svn rev 47264 built by root @ localhost.localdomain on a i686 running Linux on 2006-12-31 19:02:30
Verbosity is at least 1
Uptime
System uptime: 4 days, 9 hours, 46 minutes, 38 seconds
Verbosity is at least 1
Active Channel(s)
Peer User/ANR Call ID Seq (Tx/Rx) Form Hold Last Message
0 active SIP channels
Verbosity is at least 1
Sip Registry
Name/username Host Dyn Nat ACL Port Status
wlgvgl 202.7.4.40 5060 Unmonitored
VIX-incoming-0.4 161.29.0.4 5060 Unmonitored
VIX-incoming-0.244 161.29.0.244 5060 Unmonitored
VIX-incoming-0.228 161.29.0.228 5060 Unmonitored
VIX-incoming 202.53.189.146 5060 Unmonitored

TrixBox

TrixBox

- Behaves the way the developers envisage a ‘PBX System’ operating
  - Sometimes different to what you would expect
  - Trade off between roll your own and pre-packaged
- Can easily customise the dial plan - If you know what you are doing!
- Many inputs are still in ‘Asterisk Dial Plan Language’
- Good to know what’s happening under the hood...

Asterisk Configuration Details

- Text based configuration files
  - sip.conf
  - extensions.conf
  - voicemail.conf
  - agents.conf
  - queues.conf

sip.conf

/etc/asterisk/sip.conf

```
[general]
context=default                 ; Default context for incoming calls
port=5060                       ; UDP Port to bind to (SIP standard port is 5060)
bindaddr=0.0.0.0                ; IP address to bind to (0.0.0.0 binds to all)
srvlookup=yes                   ; Enable DNS SRV lookups on outbound calls


[2000]
type=friend                     ; both send and receive calls from this peer
host=dynamic                    ; this peer will register with us
username=2000
secret=j3nny
canreinvite=no                  ; don't send SIP re-invites (ie. terminate rtp stream)
nat=yes                         ; always assume peer is behind a NAT
context=phones                  ; send calls to 'phones' context
dtmfmode=rfc2833                ; set dtmf relay mode
allow=all                       ; allow all codecs

[pstn-gateway]
type=friend
disallow=all
allow=alaw
context=from-pstn-gateway
host=pstn-gateway.jonnynet.net
canreinvite=no
dtmfmode=rfc2833
allow=all
```

extensions.conf

```
/etc/asterisk/extensions.conf

[general]
static=yes               ; default values for changes to this file
writeprotect=no          ; by the Asterisk CLI

[globals]
; variables go here

[default]
; default context

[phones]
; context for our phones
exten => 2000,1,Dial(SIP/2000)
exten => 2000,2,Voicemail(u2000)

exten => 500,1,Answer()
exten => 500,2,Playback(demo-echotest)            ; Let them know what's going on
exten => 500,3,Echo                               ; Do the echo test
exten => 500,4,Playback(demo-echodone)            ; Let them know it's over
exten => 500,5,Hangup

exten => _1.,1,Dial(SIP/${EXTEN:1}@pstn-gateway)  ; match anything and send to wlg-gateway
exten => _1.,2,Hangup

[from-pstn-gateway]
; context for calls coming from wlg-gateway
exten => 4989560,1,GoTo(phones,2000,1)
exten => _.,1,Congestion()                     ; everyone else gets congestion
```

voicemail.conf

/etc/asterisk/voicemail.conf

```
[general]
format=wav49|gsm|wav
serveremail=voicemail@jonnynet.net
mailcmd=/usr/sbin/sendmail -t
attach=yes
maxmsg=100
maxmessage=180
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3

emailbody=Dear ${VM_NAME}:\n\n\tjust wanted to let you know you were just left a
 ${VM_DUR} long message (number ${VM_MSGNUM})\nin mailbox ${VM_MAILBOX} from ${V
M_CALLERID}, on ${VM_DATE}, so you might\nwant to check it when you get a chance
.  Thanks!\n\n\t\t\t\t--Asterisk\n
emaildateformat=%A, %B %d, %Y at %r

[default]
; all our mailboxes here
2000 => 1234,Jonny,jonny@jonnynet.net
```

Dial Plan - Contexts

- extensions.conf split into sections called contexts
- [context-name]
- contexts isolated from one another - can have the same extension in multiple contexts
- Calls from a channel land in the context specified by that channel,
- Calls land in default context if nothing is specified
- Be careful with what is in the default context - it is easy to give access to more than is intended

Dial Plan - Extensions

- One or more extensions in each context
- An extension is followed by an incoming call or digits dialled on a channel
    - exten => name,priority,application()
    - exten => 2000,1,Dial(SIP/2000)
- Priorities are numbered and followed sequentially from ‘1’
    - Asterisk will stop processing an extension if you skip a priority
- Each priority executes one specific application

Dial Plan - ‘n’ priority

- Asterisk 1.2 onwards understands the ‘n’ priority

```
exten => 2000,1,FirstApplication()
exten => 2000,n,NextApplication()
exten => 2000,n(priority_label),AnotherApplication()
```

- Saves renumbering your extensions if you add/remove a priority

- labels can make dial plan more readable, particularly when branching using gotos.

Dial Plan - Variables

- Three types of variables available in the dial plan
- Global
    - Set in the [globals] section of extensions.conf
- Channel
    - Variables set automatically, andusing the set command on a per channel basis
- A number of pre-defined channel variables - e.g. ${EXTEN}

Dial Plan - Variables

- Some of the pre-defined channel variables:

```
${CALLERID}
${CALLERIDNAME}
${CALLERIDNUM}
${CHANNEL}
${CONTEXT}
${EXTEN}
${SIPUSERAGENT}
```

Dial Plan - Extension Matching

- exten => _04NXXXXXX,1,SomeApplication()
- exten => _.,1,SomeApplication()
  - _ denotes a pattern matching extension
  - N matches any number from 2 through 9
  - X matches any single digit
  - . matches one or more of any digit
  - [2-6] matches any of 2,3,4,5,6

agents.conf

- Users can log in as an Agent
- Maps current extension to that users Agent
- Agent can then be logged into queues
- Agents can log in / out at will, follow-me functionality
- Agents functionality still quite buggy - best not to use for anything complex

agents.conf

/etc/asterisk/agents.conf

```
[general]
; Define whether callbacklogins should be stored in astdb for persistence
persistentagents=yes

[agents]
;autologoff=15 ; time (s) before agent auto logoff if no answer
;ackcall=no
wrapuptime=1000
;musiconhold => default
;updatecdr=no
; Enable recording calls addressed to agents. It's turned off by default.
recordagentcalls=yes
;recordformat=gsm
; This section contains the agent definitions, in the form:
; agent => agentid,agentpassword,name

group=1
; Junior NOC staff
agent => 600,1234,Lilly

group=2
; Senior NOC staff
agent => 610,1234,Steve
```

queues.conf

- Reasonable queuing support within Asterisk
- Queues can have static or dynamic members
- Members can be channels, or Agents
- Automatic distribution of calls based on queue strategy

queues.conf

```
/etc/asterisk/queues.conf

[general]
; Store each dynamic agent in each queue in the astdb for persistence
persistentmembers = yes

; Note that a timeout to fail out of a queue may be passed as part of
; an application call from extensions.conf:
; Queue(queuename|[options]|[optionalurl]|[announceoverride]|[timeout])
; example: Queue(dave|t|||45)

[noc]
musiconhold = default
strategy = ringall ; ringall, roundrobin, leastrecent, fewest calls, random, rrmemory

servicelevel = 30 ; SLA setting (s). stats for calls answered in this time
timeout=15  ; How long the phone rings before it's considered a timeout
retry=0  ; How long do we wait before trying all the members again?
; Weight of queue - when compared to other queues, higher weights get preference
weight=2
wrapuptime=5 ; how long before sending agent another call
maxlen = 0 ; of queue, 0 for no maximum

; How often to announce queue position and/or estimated holdtime to caller (0=off)
announce-frequency = 0
;announce-holdtime = yes|no|once
;announce-round-seconds = 10
; How often to make any periodic announcement (see periodic-announce)
;periodic-announce-frequency=60
```

queues.conf

/etc/asterisk/queues.conf ...ctd

```
monitor-format = wav
monitor-join = yes ; join both monitor files (sides of call) together

joinempty = no
leavewhenempty = yes

reportholdtime = no ; report caller hold time to member when answered
memberdelay = 0 ; delay before connecting member too caller

; Static NOC members
; member => technology/dialstring,penalty
member => Agent/600,1
member => Agent/610,2
```

/etc/asterisk/extensions.conf

```
; Log Agent in
; Asks the agent to login to the system with callback.
; AgentCallbackLogin([AgentNo|][Options|][exten]@context)
exten => *0,1,AgentCallbackLogin(${CALLERID(NUM)}@default)
```

Queues Example

```
; Using Agents

; agent login to helpdesk queue
exten => *4,1,Answer()
exten => *4,n,AddQueueMember(noc|Agent/${CALLERID(NUM)})
exten => *4,n,AgentCallbackLogin(${CALLERID(NUM)}||q${CALLERID(NUM)}@sip)
exten => *4,n,Hangup()

; agent logout from noc queue
; note # is sent through by as a %23 in some sip headers
; so may need to repeat with exten => %23
exten => #4,1,Answer()
; send trigger to flash panel
exten => #4,n,System(/usr/sbin/asterisk -rx "agent logoff Agent/${CALLERID(NUM)}")
exten => #4,n,RemoveQueueMember(noc|Agent/${CALLERID(NUM)})
exten => #4,n,Playback(agent-loggedoff)
exten => #4,n,Hangup

; Or
; Using dynamic login of channel instead of agents, doesn't send triggers to flash panel
exten => *4,1,Answer()
exten => *4,n,AddQueueMember(noc|${CALLERID(NUM)})
exten => *4,n,Playback(logged-in)
exten => *4,n,Hangup()

exten => #4,n,RemoveQueueMember(noc|${CALLERID(NUM)})
exten => #4,n,Playback(agent-loggedoff)
exten => #4,n,Hangup
```

‘Advanced’ Configuration

- dial plan macros
- Asterisk DB
- Festival - text to speech engine
- Flash Operator Panel (FOP)
- Asterisk Gateway Interface (AGI) Scripts

Dial Plan Macros

- Avoids repetition in the dial plan
- Akin to building a function in the dial plan
- Useful for building standard phone dialing logic
- Uses extra specific channel variables:

  ${ARGn}: The nth argument passed to the macro
  ${MACRO_CONTEXT}: Context of the extension that triggered this macro
  ${MACRO_EXTEN}: The extension that triggered this macro
  ${MACRO_PRIORITY}: The priority in the extension where this macro was triggered

Dial Plan Macros

```
[macro-stdexten]

;
; Standard extension macro:
;    ${ARG1} - Extension  (we could have used ${MACRO_EXTEN} here as well
;    ${ARG2} - Device(s) to ring
;
; ring the interface for 20sec max
exten => s,1,Dial(${ARG2},20)
; jump based on status (NOANSWER,BUSY,CHANUNAVAIL,CONGESTION,ANSWER)
exten => s,2,Goto(s-${DIALSTATUS},1)

exten => s-NOANSWER,1,Voicemail(u${ARG1})              ; If unavailable, send to voicemail
exten => s-NOANSWER,2,Goto(default,s,1)                ; If they press #, return to start

exten => s-BUSY,1,Voicemail(b${ARG1})                  ; If busy, send to voicemail w/ busy announce
exten => s-BUSY,2,Goto(default,s,1)                    ; If they press #, return to start

exten => _s-.,1,Goto(s-NOANSWER,1)                     ; Treat anything else as no answer

exten => a,1,VoicemailMain(${ARG1})                    ; If they press *, send the user into VoicemailMain
```

Asterisk Database

- astdb - simple database forms part of Asterisk
- Dial plan and CLI can insert and remove data
- Data stored in a file, so is retained across Asterisk reloads and server reboots
- Data stored in groupings of families containing keys
    - exten => s,1,Set(DB(family/key)=${some_variable})
    - exten => s,1,Set(DB(system/nightmode_on)=1)

Asterisk Database

```
; start counting and store count progress in astdb (Asterisk 1.2)

exten => 510,1,Set(COUNT=${DB(test/count)})
exten => 510,2,SayNumber(${COUNT})
exten => 510,3,SetVar(COUNT=$[${COUNT} + 1]
exten => 510,4,Set(DB(test/count)=${COUNT})
exten => 510,5,Goto(1)
exten => 510,102,Set(DB(test/count)=1)
exten => 510,103,Goto(1)
```

Festival Text to Speech

- Installed as part of asterisk-addons
- Text to speech is a bit rough, but useable
- Easy to use once installed
- Useful for putting together quick IVRs

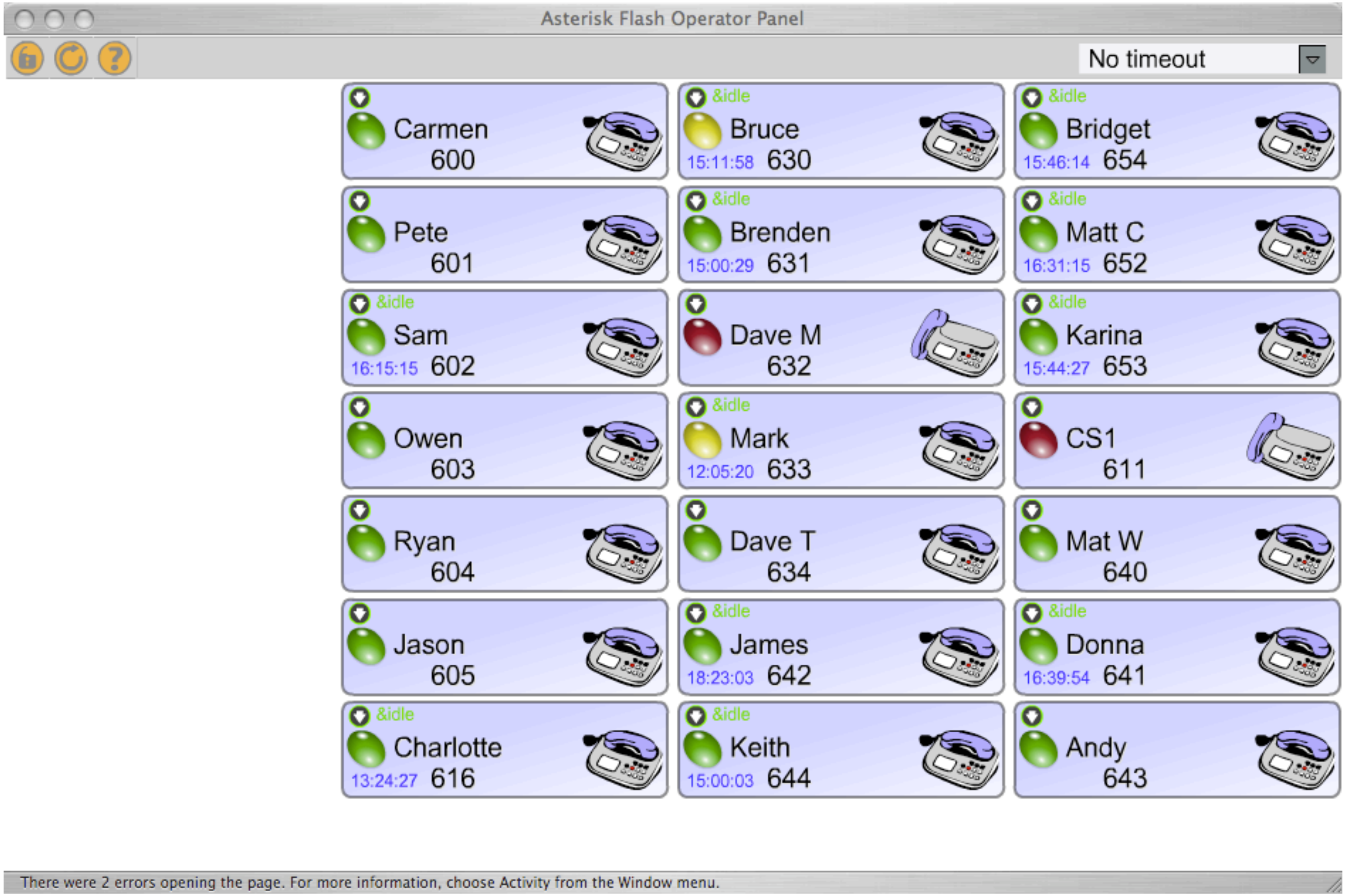
```
exten => 1,1,Festival('Record your message now')
exten => 1,n,Record(filename:alaw)
exten => 1,n,Festival('You recorded')
exten => 1,n,Playback(filename)
exten => 1,n,Festival('message saved.')
exten => 1,n,Goto(s,1)
```

AGI Scripts

- Asterisk Gateway Interface
- Dial plan can call Perl, Python, PHP scripts
- AGI script reads from STDIN to get information from Asterisk
- AGI script writes data to STDOUT to send information to Asterisk
- AGI script can write to STDERR to send debug information to the console
- Scripts stored in /usr/share/asterisk/agi-bin/ on Debian
- exten => 520,1,AGI(agi-script.agi)

Flash Operator Panel

- Gives visual state of extensions and trunks
- PERL script runs on web server, Flash client in browser
- Not quite perfect, but pretty good
- Monitors Asterisk Manager interface for events
- Details at www.asternic.org
- Layout configuration text based - tedious but flexible



Flash Operator Panel

Flash Operator Panel

/usr/local/op_panel/op_buttons.cfg

```
[QUEUE/helpdesk]
Position=1-5
Label="Helpdesk Queue"
Extension=-1    ;transfers disabled at this stage
Privacy=false

[QUEUE/noc]
Position=6-7
Label="NOC Queue"
Extension=-1    ;transfers disabled at this stage
Privacy=false


[SIP/2000]
Position=8
Label="Jonny Martin%0a     2000"
Extension=-1    ;transfers disabled at this stage
Context=sip
Icon=1
Background=bg.jpg
VocieMailExt=2000@default
Privacy=false
```

Flash Operator Panel

/usr/local/op_panel/op_server.cfg

```
[general]
; host or ip address of asterisk
manager_host=127.0.0.1
manager_port=5038
; user and secret for connecting to * manager
manager_user=admin
manager_secret=supersecret
```

/etc/asterisk/manager.conf

```
[general]
enabled = yes
port = 5038
bindaddr = 127.0.0.1
;displayconnects = yes

; flash operator panel access
[admin]
secret = supersecret
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1/255.255.255.255
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
```

Standard Extension Macro

```
[macro-new-stdext]
; variables passed to macro, and turned into channel variables
;   ARG1 how long to initially ring (timer_ring)
;   ARG2 how long to ring on the divert portion (timer_divert)
;
; channel variables populated from db:
;   ext_dialstring
;   divert_dest
;   divert_on   (0 or empty = no, anything else = yes)
;   (eventually will be ring w,h,o (if ring_w/h/o_ext is true) for timer_initial_ring)
;
; dial appropriate devices for timer_ring
; if no answer, check divert_on
; if divert=yes, ring divert_dest for timer_divert, then VM if no answer
; if divert=no, go to VM

; varibalise arguments
exten => s,1,Set(timer_ring=${ARG1})
exten => s,n,Set(timer_divert=${ARG2})

exten => s,n(dbvars),Set(ext_dialstring=${DB(ext/${MACRO_EXTEN}/ext_dialstring)})
exten => s,n,Set(divert_dest=${DB(ext/${MACRO_EXTEN}/divert_dest)})
exten => s,n,Set(divert_on=${DB(ext/${MACRO_EXTEN}/divert_on)})
```

Standard Extension Macro

```
; dial appropriate devices
exten => s,n(dial),Dial(${ext_dialstring},${timer_ring})

 if divert_on=false goto priority divert_no, if true then go to priority divert_yes
;  asterisk throws up a warning here if divert_on=null string.
;  need to put in a null string check on divert_on here.
exten => s,n,GotoIf($[${divert_on}]?divert_yes:divert_no)

; we're not diverting...
exten => s,n(divert_no),Voicemail(su${MACRO_EXTEN})
exten => s,n,Hangup

; we're diverting...
;  set original callerid name and number, and diverting extension in chan vars
;  then send call to divert-callout with a caller id of the diverting ext
exten => s,n(divert_yes),Set(orig_calling_name=${CALLERID(name)})
exten => s,n,Set(orig_calling_num=${CALLERID(num)})
exten => s,n,Set(diverting_ext=${MACRO_EXTEN})
exten => s,n,Set(CALLERID(all)=${CALLERID(num)}diverted<${MACRO_EXTEN}>)
exten => s,n,Goto(divert-callout,${divert_dest},1)
```

Standard Macro Extension

```
[globals]
STD_TIMER_RING=16        ; standard time to ring when an extension is dialled
STD_TIMER_DIVERT=16      ; standard time to ring on diversion portion
STD_GW_STRING=Zap/g0     ; Zap/g0 is the standard one at this stage


[phones]
exten => 2000,1,Macro(new-stdext,${STD_TIMER_RING},${STD_TIMER_DIVERT})
exten => 2001,1,Macro(new-stdext,${STD_TIMER_RING},${STD_TIMER_DIVERT})
exten => 2002,1,Macro(new-stdext,${STD_TIMER_RING},${STD_TIMER_DIVERT})
```

INOC-DBA integration

- INOC-DBA - Inter NOC hotline service provided by PCH
- Need to contact an AS? Dial the ASN
- http://www.pch.net/inoc-dba/

INOC-DBA Integration

/etc/asterisk/sip.conf

```
[general]
register => 9503*561:supersecret:jonny@inoc-dba.pch.net/jonny-inoc

[inoc-dba]
type=peer
host=inoc-dba.pch.net
username=jonny
fromuser=9503*561
secret=supersecret
canreinvite=yes
context=from-inoc-dba
insecure=very
nat=no
```

/etc/asterisk/extensions.conf

```
; This extension will ring SIP extension 100 for 40 seconds then hangup
exten => jonny-inoc,1,Dial(SIP/100,40)
exten => jonny-inoc,2,Hangup

; This extension is for outgoing calls to inoc-dba
; 9 for an outside-inoc-dba-line
exten => _9.,1,SetCIDName(Jonny Martin)
exten => _9.,2,SetCIDNum(9503*561)
exten => _9.,3,Dial(SIP/${EXTEN:1}@inoc-dba)
exten => _9.,4,Congestion
exten => _9.,5,Hangup
```