Introduction to Asterisk

PacNOG4 VoIP Workshop
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Introduction

- Quick Overview of Asterisk
- A look at trixbox, an Asterisk based ‘pretty’ PABX
- Basic configuration
- ‘Advanced’ Configuration
- Examples

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
  - 1.2 - becoming obsolete rapidly, but it’s good and stable
  - 1.4 - the current release of choice for most, stable
  - 1.6 - all the new features in here, still many bugs
  - 1.8 - bleeding edge development - don’t use it!

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 :)

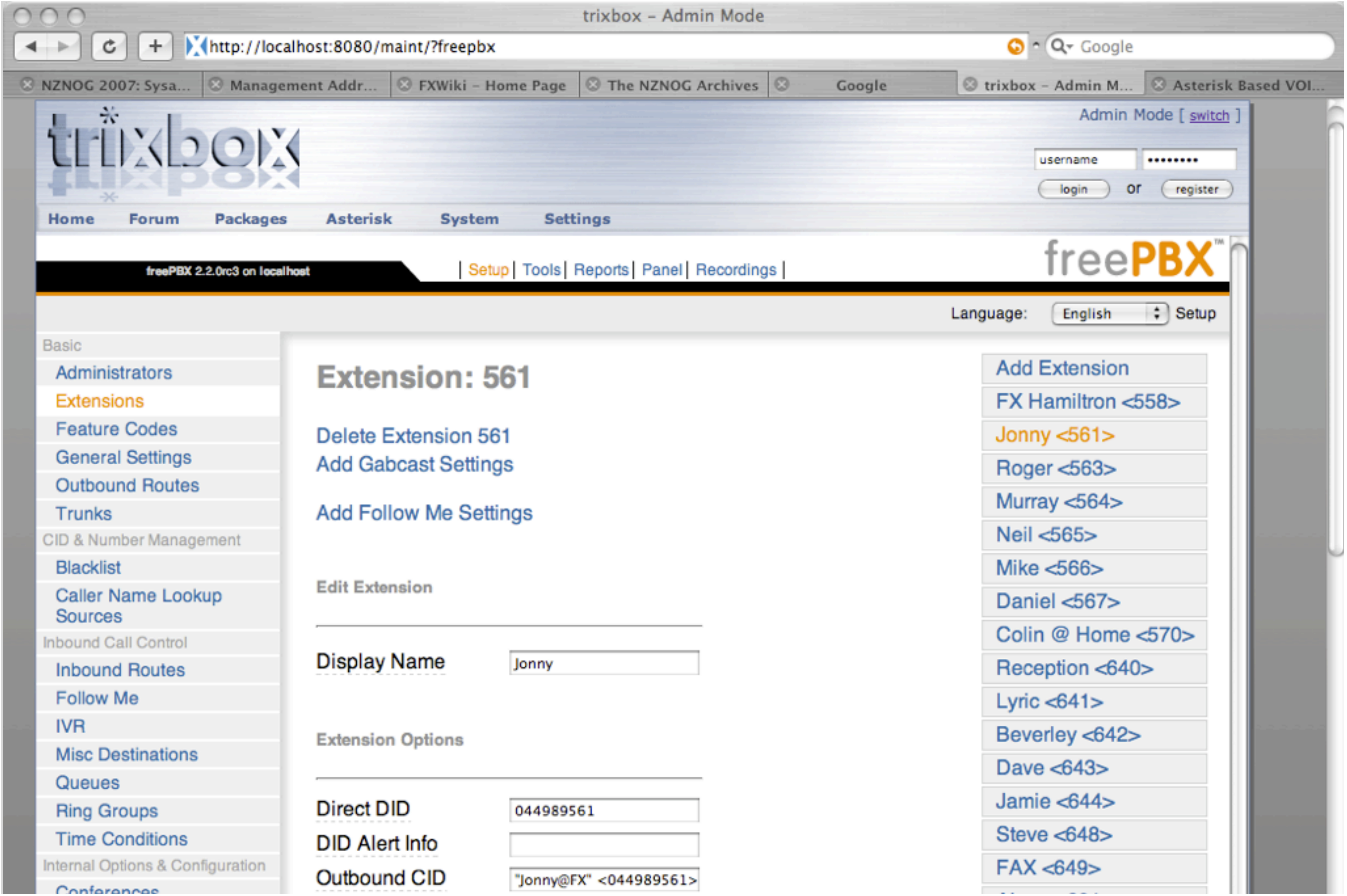
- Asterisk is a hybrid TDM and packet voice PBX
- Interfaces any piece of telephony hardware or software to any application
- Prime components: channels and extensions.conf - the Asterisk dial plan
- Channels can be many different technologies - SIP, IAX, H323, skinny, Zaptel, and others as they are 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 = 2.6, based on Asterisk 1.4
- 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
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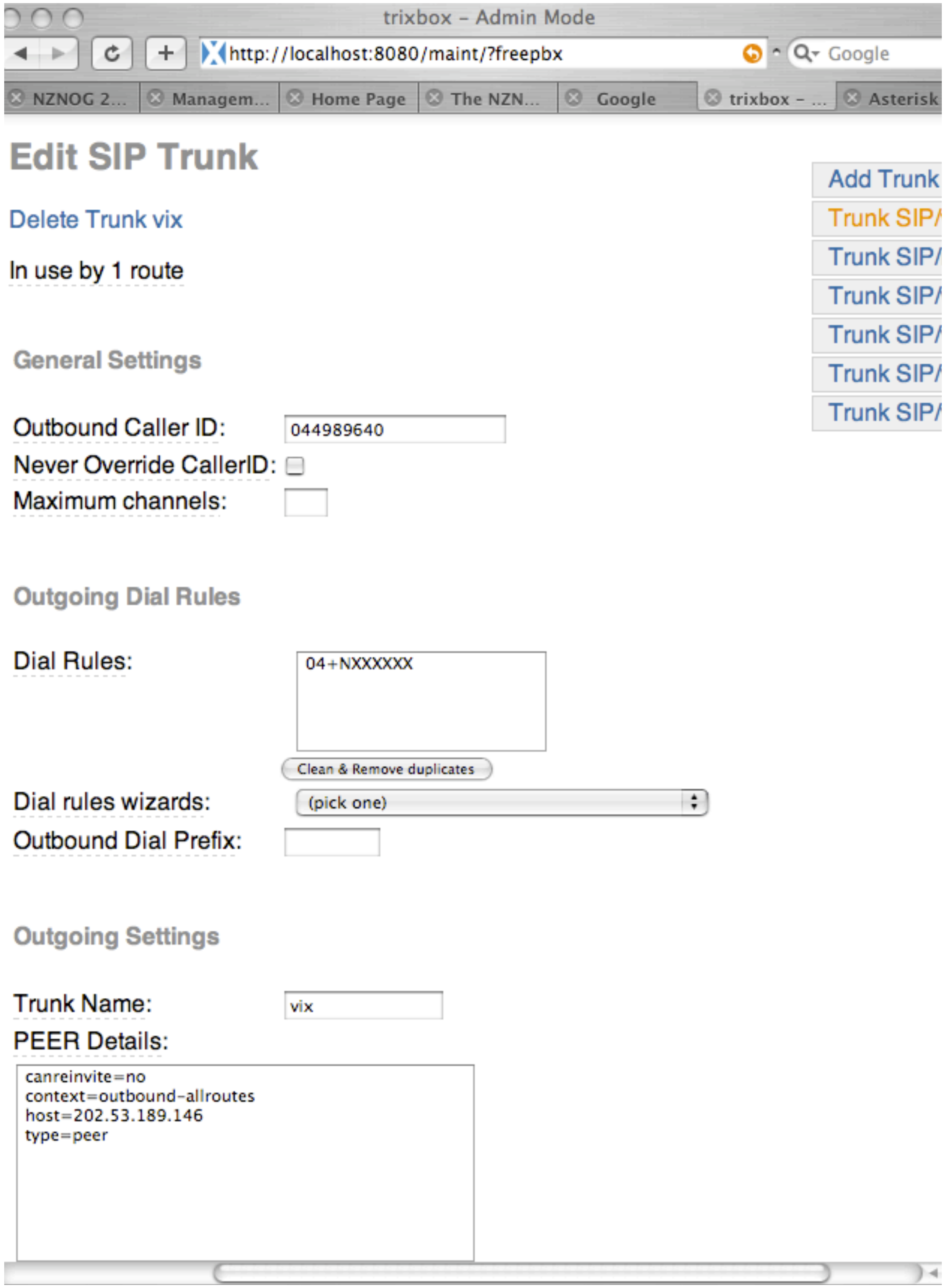
Asterisk Based VOI...
trixbox
Admin Mode [switch]
username
login
or
register
Home
Forum
Packages
Asterisk
System
Settings
freePBX 2.2.0rc3 on localhost
| Setup | Tools | Reports | Panel | Recordings |
freePBX
Language: English Setup
Basic
Administrators
Extensions
Feature Codes
General Settings
Outbound 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
NZNOG 2...
Managem...
Home Page
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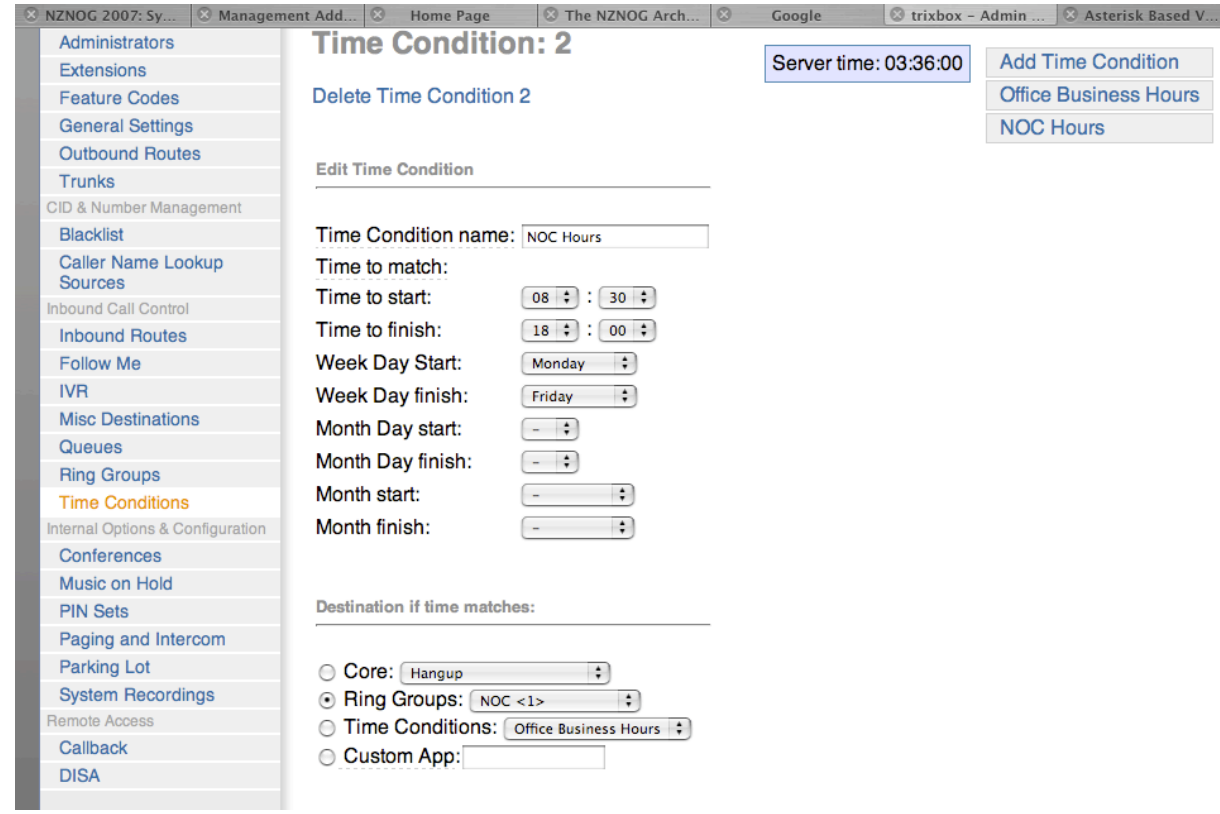
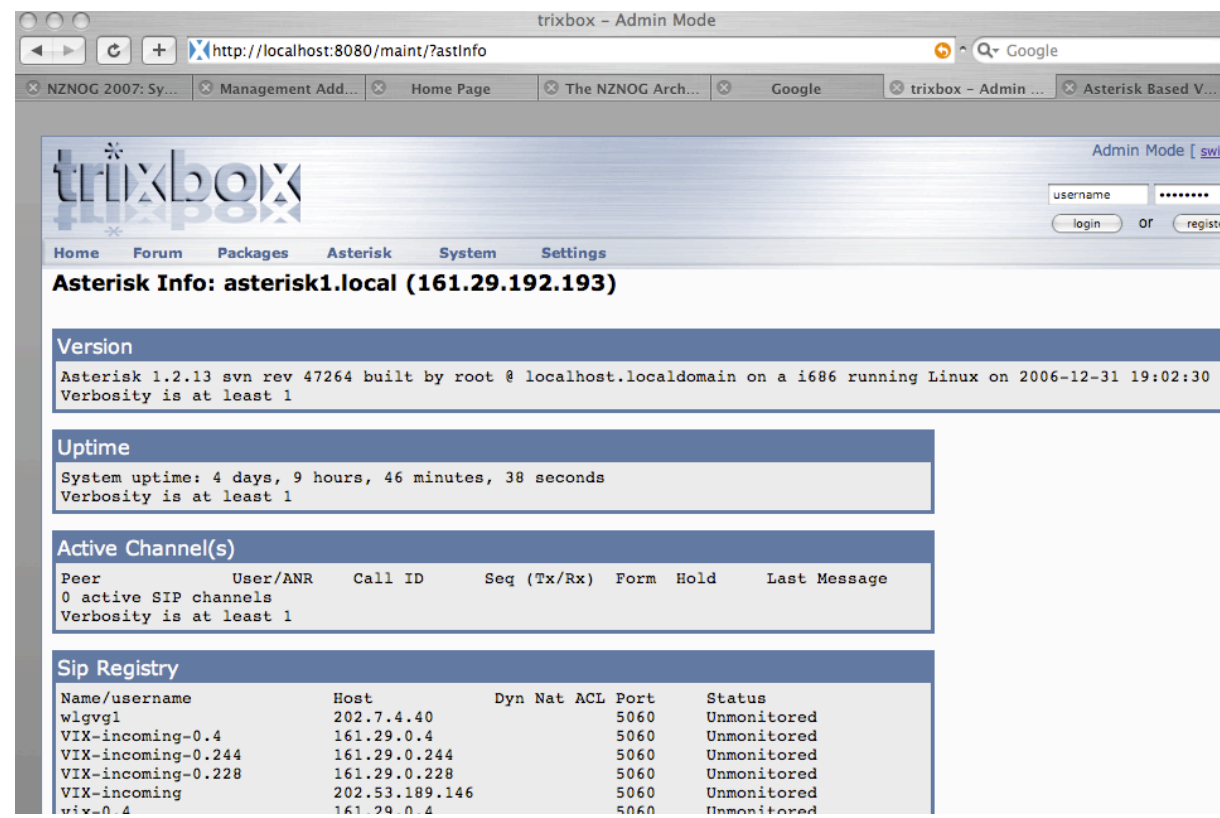
trixbox – ...
Asterisk
Edit SIP Trunk
Delete Trunk vix
In use by 1 route
Add Trunk
Trunk SIP/
Trunk SIP/
Trunk SIP/
Trunk SIP/
Trunk SIP/
Trunk SIP/
Trunk SIP/
General Settings
Outbound Caller ID:
044989640
Never Override CallerID:
Maximum channels:
Outgoing Dial Rules
Dial Rules:
04+NXXXXXX
Clean & 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


NZNOG 2007: Sy...
Management Add...
Home Page
The NZNOG Arch...
Google
trixbox – Admin ...
Asterisk Based V...
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
Conferences
Music on Hold
PIN Sets
Paging and Intercom
Parking Lot
System Recordings
Remote Access
Callback
DISA
Time Condition: 2
Delete Time Condition 2
Server time: 03:36:00
Add Time Condition
Office Business Hours
NOC Hours
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: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Google
trixbox – Admin ...
Asterisk Based V...
trixbox
Admin Mode
username
login
or
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...

trixbox

- Caution, the trixbox CD will erase everything on the hard drive!!!
- Several versions to download
  - CE (Consumer Edition) -- free
  - PRO (Standard Edition) -- free
  - PRO (Enterprise Edition) -- $$$
  - PRO (Call Centre Edition) -- $$$

trixbox LAB

- Many good how-to guides
- http://www.sureteq.com/asterisk/trixboxv2.4.htm
  - Written for version 2.4 but still good
- Set all passwords to ‘pacnog2008’
-

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

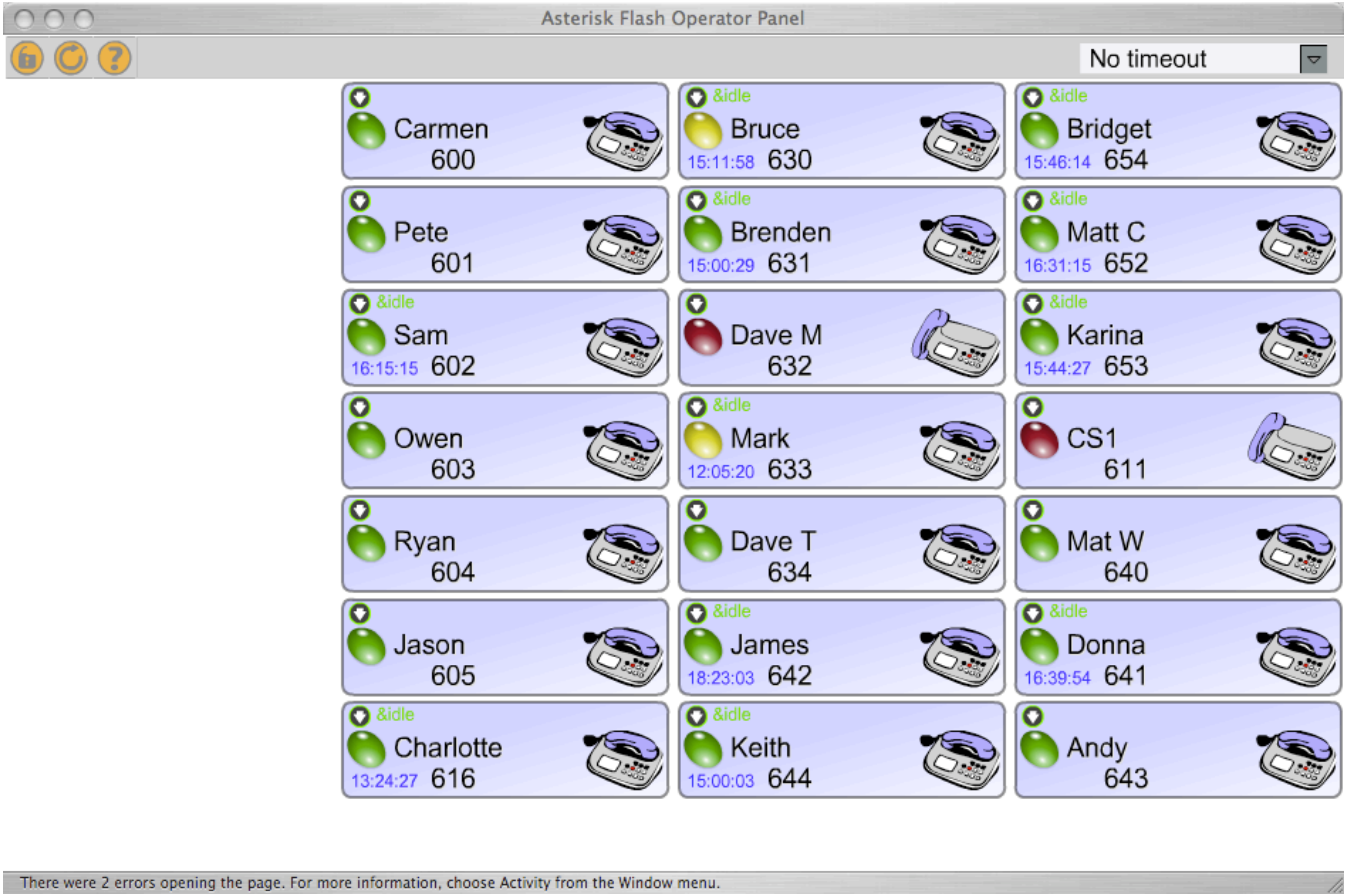
```
exten => 1,1,Festival('Record your message now')
exten => 1,n,Record(filename:alaw)
exten => 1,n,Festival('You recorded')
exten => 1,n,Playback(filename)
exten => 1,n,Festival('message saved.')
exten => 1,n,Goto(s,1)
```

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
```