ASTERISK & PHP

Hans-Christian Otto
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ABOUT ME

PHP since 2004

Asterisk since 2007

working as a freelancer for various companys

computer science student at TU Dortmund

active member of phpugdo
... AND YOU?
ASTERISK

open source PBX

actually dual-licensed

created by Mark Spencer in 1999

runs on *nix and windows

VoIP

ISDN

bluetooth

and more
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<th>InterAsterisk eXchange</th>
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FEATURES

music on hold
voicemail
phone conferences
IVR
  speech recognition
„follow me“
scriptable

AstDB
queues
call parking
pickup
USE CASES
DIALING
INCOMING CALLS
SOME TERMS

device
dialplan
extension
context
channel
DEVICES

SIP peers

SIP phones
softclients
legacy phones using ATA
sip providers
isdn phones

IAX peers
bluetooth devices
etc.
DIALPLAN

contexts

extension

priority

applications

macros

control structures
SIP.CONF

[alice]
type=friend
context=from-sip
secret=passwordAlice
host=dynamic
disable=all
allow=ulaw
allow=alaw

[bob]
type=friend
context=from-sip
secret=passwordBob
host=dynamic
disable=all
allow=ulaw
allow=alaw
EXTENSIONS.CONF

[from-sip]

exten => 10,1,Dial(SIP/alice&SIP/bob)
exten => 11,1,Dial(SIP/alice)
exten => 11,hint,SIP/alice
exten => 12,1,Dial(SIP/bob)

exten => 81,1,Answer()
exten => 81,2,AGI(weather.agi)
exten => 81,3,Hangup()
IVR

```plaintext
exten => 90, 1, Answer()
exten => 90, n, Playback(marryme)

exten => 1, 1, Playback(thank-you-cooperaation)
exten => 1, n, Hangup()

exten => 2, 1, Playback(sorry)
exten => 2, n, Hangup()
```
CONTROL STRUCTURES

exten => 123,1,Answer

exten => 123,n,Set(i=1)

exten => 123,n,While($[$i] < 5])

exten => 123,n,SayNumber($i})

exten => 123,n,Set(i=$[$i] + 1])

exten => 123,n,EndWhile
context ael-demo {

123 => {

Answer();
for (x=0; ${x} < 5; x=${x} + 1) {
    SayNumber(${x});
}
}
}
function demo_start(context, exten)
    app.wait(1)
    app.answer()
    demo_congrats(context, exten)
end

extensions = {
    demo = {
        s = demo_start;
        ["2"] = function()
            app.background("demo-moreinfo")
            demo_instruct()
        end;
    }
}
FUNFACT

AEL and extensions.conf support goto for a long time ;-)
PHP?
dialplan
AGI
FastAGI
AMI
AJAM
callfiles
STATE OF ASTERISK & PHP

multiple php libraries

freepbx
USE CASES
DIALING SUCKS.

dialing results in …

missdialing

procrastination; not misdialing calls (laziness)

using (more expensive) cellphones

solution: computer based dialing

AMI

callfiles
CALLFILES

text files initiating a call
key-value pairs
placed in a special directory
watched by asterisk
schedule calls by mtime modification

Channel:    SIP/alice
Context:    from-sip
Extension:  12
Priority:   1
WaitTime:   30
RetryTime:  60
MaxRetries: 2
<?php

$filename = tempnam(
    '/var/spool/asterisk/tmp/',
    'callfile'
);

file_put_contents($filename, $callFile);
touch($filename, time() + 60);
rename(
    $filename,
    tempnam(
        '/var/spool/asterisk/outgoing/',
        'callfile'
    )
);
}
INCOMING CALL

display notifications

on client computers

using funny gadgets (emergency lights, anyone?)

turn off espresso maker (so your staff gets back to work)

entertain caller

log (missed) calls

visualize in CRM
CDR

log all calls

different output modules

  csv

  mysql

  etc.
[global]

hostname = localhost
dbname=asteriskcdrdb
password = amp109
user = asteriskuser
userfield=1
calldate: 2010-05-29 11:26:18
duration: 22
clid: "Extern: 123" <123>
billsec: 11
cid: 123
disposition: ANSWERED
src: 123
amaflags: 3
dst: 31
accountcode:
dcontext: from-internal
uniqueid: 1275125178.20185
channel: SIP/9-09bde8f8
userfield:
dstchannel: SIP/31-09bb7550
lastapp: Dial
lastdata: SIP/31|20
userfield:
AGI

executables

chmod +x & shebang

receive variables through STDIN

  just like HTTP-headers, Key:Value

send commands through STDOUT

  fwrite(STDOUT,"EXEC Playback tt-allbusy \\
        \n");

PHPAGI
A SIMPLE AGI

#!/usr/bin/env php
<?php
require 'phpagi/phpagi.php';
$agi = new AGI();
$agi->text2wav('Please enter the PIN.');
$pin = $agi->get_data( 'beep', 5000, 4 );
if( $pin['result'] != '2342' ) {
    $agi->text2wav('The entered pin was wrong.');
} else {
    start_servers($agi);
}
function start_servers($agi) {
    $agi->text2wav('Which server should be started?');
    $server = $agi->menu(array(
        '1' => '*Press 1 for CRM',
        '2' => '*Press 2 for ERP',
    ));
    $agi->text2wav('Waking up server ' . $server);
    WakeOnLan($server);
}
WHO’S CALLING?

#!/usr/bin/env php
<?php
require 'phpagi/phpagi.php';
$agi = new AGI();
$cid = $agi->getVariable('callerid(name)');
if(!$cid['result'])
    exit;
$agi->set_callerid(sprintf('%s<%d>',
    lookupNameByNumber($cid['data']),
    $cid['data']
));
WHO’S CALLING?

exten => 11,1,AGI(callerid_lookup)
exten => 11,2,Dial(SIP/alice)
PHP & PHONES

XML browser

directory lookup

missed call list

busy lamp field

„action buttons“
BUSY LAMP FIELD

indicates status of phones using „hints“

available

ringing

busy/unavailable

can indicate „devstate“
DEVSTATE

can be controlled using
dialplan

`exten => 23,n,Set(DEVSTATE(Custom:foo) = RINGING)`

cli command
devstate change Custom:foo RINGING

AMI
DEVSTATE: USECASES

presence

non-phone indicators

  escalating support ticket

system status (nagios?)

build failures? ;-)
MISSED CALLS

common asterisk issue: missed calls

possible solution: CDR / AGI and XML application / webgui
CONCLUSION

PHP can …

originate calls

interact with calls

interact with caller

interact with callee

analyze logs

enrich phones
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FURTHER REFERENCES

http://das-asterisk-buch.de/ (german, source for some examples)

http://www.the-asterisk-book.com/

http://eder.us/projects/phpagi/

http://www.voip-info.org/

http://www.asterisk.org/