Asterisk Basics (SIP)

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OpenSIPS vs Asterisk from SIP point of view

- **Opensips**
  - Proxy, no media handling
  - IPv6 and IPv4 and multicast
  - Transport protocols
    - sctp, tcp, udp, tls
  - RFC3263
    - NAPTR, SRV
  - Very flexible so you should know very well what you are doing, so need more knowledge.
  - Committed to strictly follow the IETF SIP RFC-s
  - Sip register binding AoR to many SIPURI
  - Forking, or sequential sip forwarding

- **Asterisk**
  - B2BUA, Media server
  - IPv4 only
  - Transport protocols
    - 1.4 only udp!
    - 1.6 udp, tcp, tls
  - Easy to learn and use
  - Lazy and quick and dirty SIP implementation
  - Trancoding and MCU features
  - Sip register implementation can only handle AoR <=> ip:port binding
  - MoH is activated if there is established call with only single direction media.
  - No SRTP
Few from asterisk functions

- SIP B2BUA
- GW
  - SIP, ISDN/POTS, H.323, MGCP, XMPP etc.
- Media Server
  - IVR
  - MoH
  - Call Park
  - Call pickup
  - Voicemail
  - Monitor (Lawful interception etc.)
  - Transcoding
  - MCU
  - Call Queue
History

- Goal: Bridging the gap between Traditional and Network Telephony
- Asterisk is an IP PBX with interface to other systems and protocols (IAX, SIP, H.323, XMPP, MGCP, SCCP, ISDN/POTS, etc.)

Motivation
- Price
- Flexibility
- Security
- Interoperability

- The community led by Mark Spencer of Digium
- Zapta Jim Dixon
  - CPU has enough power to software can replace DSP in some cases.
- Digium PC POTS/ISDN cards
DAHDI

- The old name: ZAPTEL
- Drivers for Digium PC cards
  - ISDN
    - Primary rate ISDN
    - Basic Rate ISDN
  - POTS analog
    - FXO/FXS
  - DSP for media transcoding
  - DSP for echo cancellation
- Asterisk Timer
  - ztdummy
  - dahdi_dummy
  - At least meetme application is depending on it.
Versions

- **1.2.x**
  - obsoleted

- **1.4.x**
  - Old stable
  - 1.4.29
    - Branch Release Date: 2006-12-23
    - Security only fixes: 2010-12-23
    - End of Life (EOL): 2011-12-23

- **1.6.x**
  - devel

- **1.8.x**
  - LTS Long Term Support
  - Release date: TBD (estimated Q3 2010)

- **Roadmap**
• **Architecture**
  - **Core**
  - **Modules**
    - app
    - funct
    - format
    - chan
    - res
    - bridge
  - **API**
Configuration

● Static File
  ■ configuration
    □ asterisk.conf
    □ modules.conf
    □ sip.conf
    □ sip_notify.conf
    □ musiconhold.conf
    □ rtp.conf
    □ queue.conf
    □ voicemail.conf
    □ meetme.conf
    □ logger.conf
    □ cdr.conf
    □ udptl.conf (T.38)

● Static File
  ■ Runtime reloadable
  ■ Include file

● Realtime
  ■ extconfig.conf
  ■ backends
    □ SQL
    □ postgresQL
    □ MySQL
    □ odbc
    □ Ldap
    □ etc.
Config templating (sip.conf)

```
[basic-options](!)
  dtmfmode=rfc2833
  context=from-office
  type=friend

[natted-phone](!,basic-options)
  nat=yes
  canreinvite=no
  host=dynamic

[public-phone](!,basic-options)
  nat=no
  canreinvite=yes

[my-codecs](!)
  disallow=all
  allow=ilbc
  allow=g729
  allow=gsm
  allow=g723
  allow=ulaw

[ulaw-phone](!)
  disallow=all
  allow=ulaw

; and finally instantiate a few phones
[2133](natted-phone,my-codecs)
  secret = peekaboo

[2134](natted-phone,ulaw-phone)
  secret = not_very_secret

[2136](public-phone,ulaw-phone)
  secret = not_very_secret_either
```
Logger/Asterisk CLI

- **Logger**
  - `logger.conf`
    - `[general]`
    - `[logfiles]`
      - `console => notice, warning, error`
      - `messages => notice, warning, error`
  - **Type**
    - `debug`
    - `verbose`
    - `notice`
    - `warning`
    - `error`

- **asterisk -r**
  - Asterisk `-vvvvvvr`
    - increase verbosity level to 6

- **asterisk -rx "logger reload"**

- **Commands**
  - **Core commands**
    - `debug verbosity level`
      - `Core set debug`
      - `Core set verbosity`
    - `codecs`
      - `core show codecs`
      - `core show translation`
    - **Help**
      - `core show applications`
      - `core show function`
    - **Core**
      - `core show uptime`
      - `core show settings`
    - **etc.**
  - Auto-complete `<tab>`, `<?>`
General dialplan

- Static or dynamic (extensions.conf)
- Dialplan reload from CLI
- Dialplan broken into section called “context” for example context “blabla” is seems like:
  [blabla]
- “,” and “|” as separator char. When Asterisk parses the dialplan, it converts any commas in the application arguments to pipes.

- Extensions (number and/or letters)
  exten => 3rt45
  exten => _361X.

- Applications
  - For example application: Dial(), WaitExten(), Goto(), Macro(), NoOp(), Set(), etc.
Dialplan syntax

- **Pattern prefixed by “_”**
  - X
    any digit from 0-9
  - Z
    any digit from 1-9
  - N
    any digit from 2-9
  - [1235-9]
    any digit in the brackets (in this example, 1,2,3,5,6,7,8,9)
  - .
    wildcard, matches anything remaining (e.g. _9011. matches anything starting with 9011 excluding 9011 itself)
  - !
    wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible

- **Priority**
  - 1
    integer
  - n
    next previous+1 regardless of whether the previous priority was associated with the current extension or not
  - s
    same
  - + int
    integer n+2 or s+2
  - () label/alias
    jump to this with goto()

- **Most important standard extensions**
  - s
    start: no extension
  - t
    timeout
  - h
    hangup: clean up the call
  - i
    invalid: unknown extension
dialplan / extensions example

[default]
exten => 6245,hint,SIP/Grandstream1&SIP/Xlite1(Joe Schmoe); Channel hints for presence
exten => 6245,1,Dial(SIP/Grandstream1,20,rt) ; permit transfer
exten => 6245,n(dial),Dial(${HINT},20,rtT) ; Use hint as listed
exten => 6245,n,Voicemail(6245,u) ; Voicemail (unavailable)
exten => 6245,s+1,Hangup ; s+1, same as n
exten => 6245,dial+101,Voicemail(6245,b) ; Voicemail (busy)
exten => 6361,1,Dial(IAX2/JaneDoe,,rm) ; ring without time limit
exten => 6389,1,Dial(MGCP/aaln/1@192.168.0.14)
include => context1

[context1]
exten => 6391,1,Dial(JINGLE/asterisk@digium.com/mogorman@astjab.org); Dial via jingle
exten => 6390,1,Dial(JINGLE/caller/callee) ; Dial via jingle using labels
Dialplan syntax

- Special Contexts
  - general
    general settings like: static, writeprotect, autofallthrough, priorityjumping
  - globals
    global variables definition and initiation
  - "regcontext" see sip.conf

- example
  ```
  [context]
  exten => someexten,{priority|label{+-}offset}{[(alias)}],application(arg1,arg2,...)
  ```

- Contexts contain several lines, one for each step of each extension. One may include another context in the current one as well, optionally with a date and time. Included contexts are included in the order they are listed. Switches may also be included within a context. The order of matching within a context is always exact extensions, pattern match extensions, includes, and switches. Includes are always processed depth-first. So for example, if you would like a switch "A" to match before context "B", simply put switch "A" in an included context "C", where "C" is included in your original context before "B".

- Location and mapping
  - ENUM E164 Number to URI mapping
  - DuNDi Distributed Universal Number Discovery
Variables

- **Types**
  - Global
    - SetGlobalVar()
  - Shared
    - Function SHARED()
  - Channel
    - Set()
  - Environment
    - ${ENV(foo)}

- **String Handling Functions**
  - ${LEN(foo)}
    - String length
  - ${foo:offset:length}
    - Substring
  - ${foo}${bar}
    - Concat

- **Variables**
  - ${CALLERID(all)}
  - ${CALLERID(num)}
  - ${CALLERID(name)}
  - ${DIALSTATUS}
    - CHANUNAVAIL, CONGESTION, BUSY, NOANSWER, ANSWER, CANCEL, HANGUP
  - ${CONTEXT}
  - ${CHANNEL}
  - ${EXTEN}

- **$[expression]**
  - exmple:
    - exten => 321,1,Set(COUNT=3)
    - exten => 321,n,Set(NEWCOUNT=${[COUNT] + 1})
    - exten => 321,n,SayNumber(${NEWCOUNT})
Operators and function

- **boolean / logical**
  - `expr1 | expr2`
  - `expr1 & expr2`
  - `!expr1`

- **comparison**
  - `expr1 = expr2`
  - `expr1 != expr2`
  - `expr1 < expr2`
  - `expr1 <= expr2`
  - `expr1 > expr2`
  - `expr1 >= expr2`

- **Arithmetic**
  - `expr1 + expr2`
  - `expr1 - expr2`
  - `- expr (unary negation operator)`
  - `expr1 * expr2`
  - `expr1 / expr2`
  - `expr1 % expr2`

- **Regular expression**
  - `expr1 : regexp2`
    - The regular expression is anchored to the beginning of the string with an implicit `'^'`.
  - `expr1 =~ expr2`
    - The match is not anchored to the beginning of the string.

- **Conditional operator**
  - `expr1 ? expr2 :: expr3`

- **Operator Precedence**
  1. Parentheses: (, )
  2. Unary operators !, -
  3. Regular expression comparison: :, =~
  4. Multiplicative arithmetic operators: *, /, %
  5. Additive arithmetic operators: +, -
  6. Comparison operators: =, !=, <, >, <=, >=
  7. Logical operators: |, &
  8. Conditional operator: ? :

- **Function**
  - `FUNCTION_NAME(argument)`
  - `${FUNCTION_NAME(${FUNCTION_NAME(argument)})}`
Macro

- **Example**
  ```
  [macro-xyz]
  exten => s,1,Dial(${ARG1},${ARG2},t)
  ```

  - Macro definition is similar to context
  - xyz is the name of the macro
  - Can triggered from any context with
    ```
    [default]
    exten => 6601,1,Macro(xyz,SIP/1000,10)
    ```
    Where SIP/100 is ARG1 and 10 is Arg2

- **In a macro context, extra channel variables are available.**
  - `${ARG1}`: The first argument passed to the macro
  - `${ARG2}`: The second argument passed to the macro (and so on)
  - `${MACRO_CONTEXT}`: The context of the extension that triggered this macro
  - `${MACRO_EXTEN}`: The extension that triggered this macro
  - `${MACRO_OFFSET}`: Set by a macro to influence the priority where execution will continue after exiting the macro
  - `${MACRO_PRIORITY}`: The priority in the extension where this macro was triggered
AstDB database

- Berkley DB version 1
- family, key, value
- Used to store for example SIP registration
- Functions what can be used from dialplan to manipulate data in this database
  - DB(family/key)
    Read from or write to the Asterisk database.
  - DB_DELETE(family/key)
    Return a value from the database and delete it.
  - DB_EXISTS(family/key)
    Check to see if a key exists in the Asterisk database.
- Example
  exten => 456,1,Set(DB(test/count)=1)
  exten => 456,n,Set(COUNT=${DB(test/count)})
  exten => 456,n,SayNumber(${COUNT})
Asterisk as B2BUA vs. reINVITE direct media
Sip dial strings

- Sip dial strings
  - SIP/devicename
  - SIP/username@domain (SIP uri)
  - SIP/username[:password[:md5secret[:authname[:transport]]]]@host[:port]
  - SIP/devicename/extension

- Device is a "UA" [ ]
[general]
- Fax pass trough T.38
- Outgoing registration
- Sip domain
- Realtime
- NAT
- Media
- Subscribe
- Session-timer
  - Check call keep-alive sending sip messages
- RTP timer
  - Check call keep-alive using rtp timeout
- SIP timer

[authentication]
- Realm,user,pass to auth outgoing requests it requested by UAS or by proxy.

Sip devices

[devicename]
- type
  - User
    - search incoming message with the From: header user part
  - Peer
    - Search incoming message peer ip addr
  - Friend= User and Peer
    - search incoming message with the From: header user part and after if no match found then search by src ip.
- canreinvite
  - bridged media
  - direct media
- qualify
  - SIP Option “ping”
- dtmfmode
- host
voicemail.conf

- imap or disk storage
  - Default file storage /var/spool/asterisk/voicemail/
    - Voicemail files are inside this folder are grouped by context/voicemailbox

- Sections
  - [general]
    - format
    - Email options
  - [zonemessages]
    - Time zone greetings and date/time format
  - Voicemail contexts
    - [default]
      - Format: mailbox => password,name[,email[,pager_email[,options]]]
      - Example:
        4200 => 9855,Mark Spencer,markster@linux-support.net,mypager@digium.com,attach=no|serveremail=myaddy@digium.com|tz=central|maxmsg=10
    - [other]
    - [etc]
Call parking / MeetMe

- Parking
  - features.conf
    - parkext
    - parkpos
    - context
    - parkingtime
  - Parking lot
  - Context parkedcalls
  - Applications
    - ParkAndAnnounce: Park and Announce
    - ParkedCall: Answer a parked call.

- MeetMe (MCU)
  - meetme.conf
    - Example
      - [rooms]
        - conf => 600
  - Room
    - Auth /Pin
  - Applications
    - MeetMe: MeetMe conference bridge.
    - MeetMeAdmin: MeetMe conference administration.
    - MeetMeCount: MeetMe participant count.
Programing interfaces AGI / AMI

- Asterisk Gateway Interface
  - Programing inerface
  - AGI, EAGI, DeadAGI, FastAGI
  - STDIN,STDOUT,STDERR
  - Interface implementation
    perl,php,python etc.
  - extension.conf
    - exten=>123,1,Answer()
    - exten=>123,2,AGI(agi-test.agi)

- Asterisk Manager Interface
  - manager.conf
    - Directory manager.d
  - Request-response
  - Example

- Call files
  - /var/spool/asterisk/outgoing/
  - Click-to-call
  - !Move file instead of copy!

Action: GetConfig
Filename: users.conf
ActionID: 9873497149817
Response: Success
ActionID: 987397149817
Category-000000: general
Line-000000-000000: fullname=New User
Line-000000-000001: userbase=6000
Line-000000-000002: hasvoicemail=yes
Line-000000-000003: hassip=yes
Line-000000-000004: hasiax=yes
Line-000000-000005: hasmanager=no
Line-000000-000006: callwaiting=yes
Line-000000-000007: threewaycalling=yes
Line-000000-000008: callwaitingcallerid=yes
Line-000000-000009: transfer=yes
Line-000000-000010: canpark=yes
Line-000000-000011: cancallforward=yes
Line-000000-000012: callreturn=yes
Line-000000-000013: callgroup=1
Line-000000-000014: pickupgroup=1
Line-000000-000015: host=dynamic
CDR

- calldate: datetime of the started call
- clid: Caller*ID with text (80 characters)
- src: Caller*ID number (string, 80 characters)
- dst: Destination extension (string, 80 characters)
- dcontext: Destination context (string, 80 characters)
- channel: Channel used (80 characters)
- dstchannel: Destination channel if appropriate (80 characters)
- lastapp: Last application if appropriate (80 characters)
- lastdata: Last application data (arguments) (80 characters)
- duration: Total time in system, in seconds (integer), from dial to hangup
- billsec: Total time call is up, in seconds (integer), from answer to hangup
- disposition: What happened to the call: ANSWERED, NO ANSWER, BUSY, FAILED
- amaflags: What flags to use: see amaflags: DOCUMENTATION, BILLING, IGNORE etc.
- accountcode: What account number to use: Asterisk billing account, (string, 20 characters)
- uniqueid: Unique Channel Identifier (32 characters)
  (In some cases, uniqueid is appended)
- user field: A user-defined field, maximum 255 characters
Accounting / CDR

- CDR
  - asterisk-stat

- Billing

- Backends
  - CSV
  - custom(file)
  - Manager interface
  - ODBC
  - PostgreSQL
  - radius
  - sqlite
  - Tds
  - MySQL
    - not installed by default
    - Can be installed from asterisk addons package

MySQL

```sql
CREATE TABLE `cdr` (  `calldate` datetime NOT NULL default '0000-00-00 00:00:00',  `clid` varchar(80) NOT NULL default '',  `src` varchar(80) NOT NULL default '',  `dst` varchar(80) NOT NULL default '',  `dcontext` varchar(80) NOT NULL default '',  `channel` varchar(80) NOT NULL default '',  `dstchannel` varchar(80) NOT NULL default '',  `lastapp` varchar(80) NOT NULL default '',  `lastdata` varchar(80) NOT NULL default '',  `duration` int(11) NOT NULL default '0',  `billsec` int(11) NOT NULL default '0',  `disposition` varchar(45) NOT NULL default '',  `amaflags` int(11) NOT NULL default '0',  `accountcode` varchar(20) NOT NULL default '',  `userfield` varchar(255) NOT NULL default '' );
```
References

- http://www.asterisk.org/
- http://www.digium.com
- http://www.voip-info.org
- http://astbook.asteriskdocs.org
- http://www.asteriskguru.com
- http://www.amoocon.de
- http://www.voip-info.org/wiki/view/Asterisk+variables
- http://www.voip-info.org/wiki/view/Asterisk+Dialplan+Patterns
- http://www.voip-info.org/wiki/view/Asterisk+priorities
- http://areski.net/asterisk-stat-v2/about.php
- extensions.conf
Thank You!