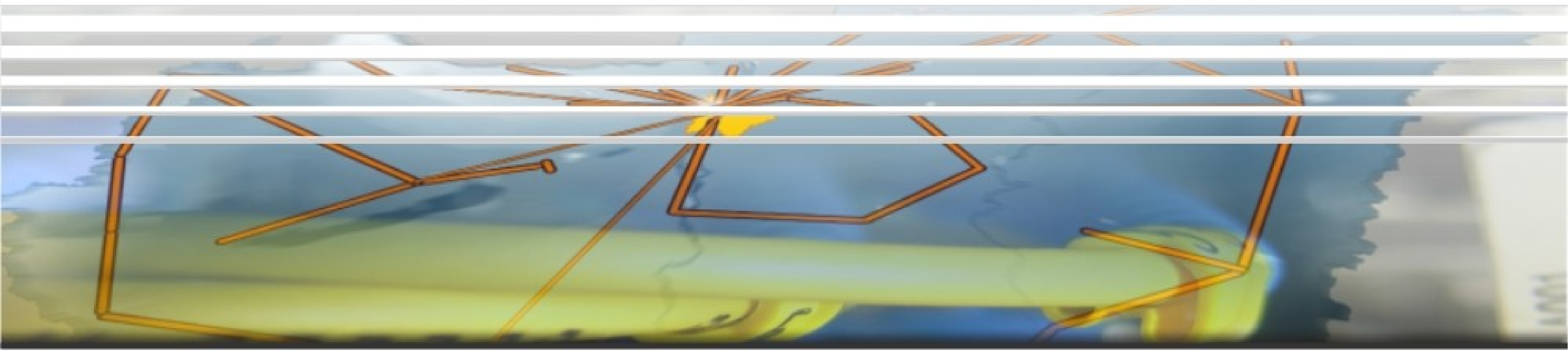
Asterisk Basics (SIP)



03/12/10
Budapest / Hungary

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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 felxible so you should know very well what you are doing, so need more knowledge.
  - Commitred to strictly follow the IETF SIP RFC-s
  - Sip register binding AoR to many SIPURI
  - Forking, or sequntial sip forwarding

- Asterisk
  - B2BUA,Media server
  - Ipv4 only
  - Transport prtocols
    - 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

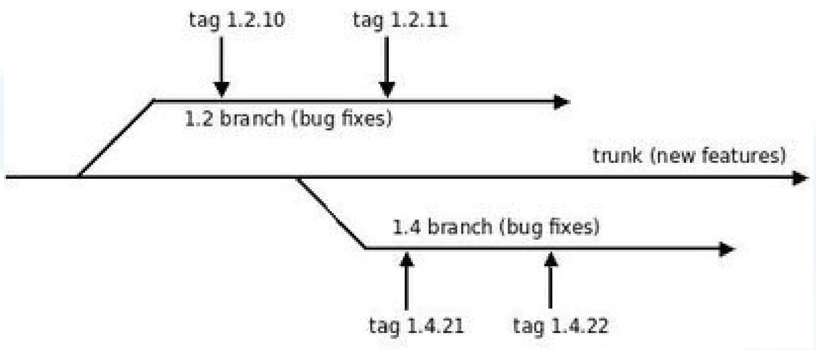
- Goal:Bridging the gap between Traditional and Network Telephony
- Asterisk is an IP PBX wit interface to other systems and protols (IAX,SIP,H.323,XMPP,MGCP,SCCP,ISDN/POTS,etc.)
- Motivation
  - Price
  - Flexibility
  - Security
  - Interoperability
- The community led by Mark Spencer of Digium
- Zapta Jim Dixon
  - CPU has enough power to software can replace DSP in some cases.
- Digium PC POTS/ISDN cards

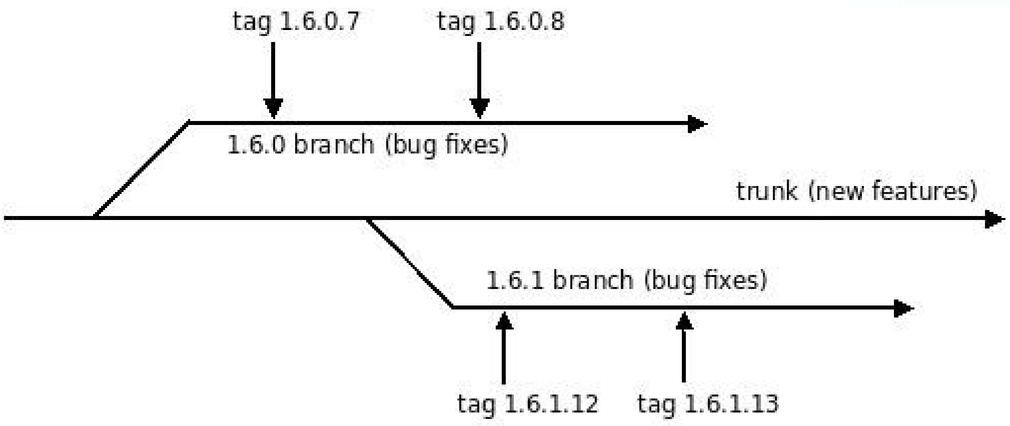
DAHDI

- 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

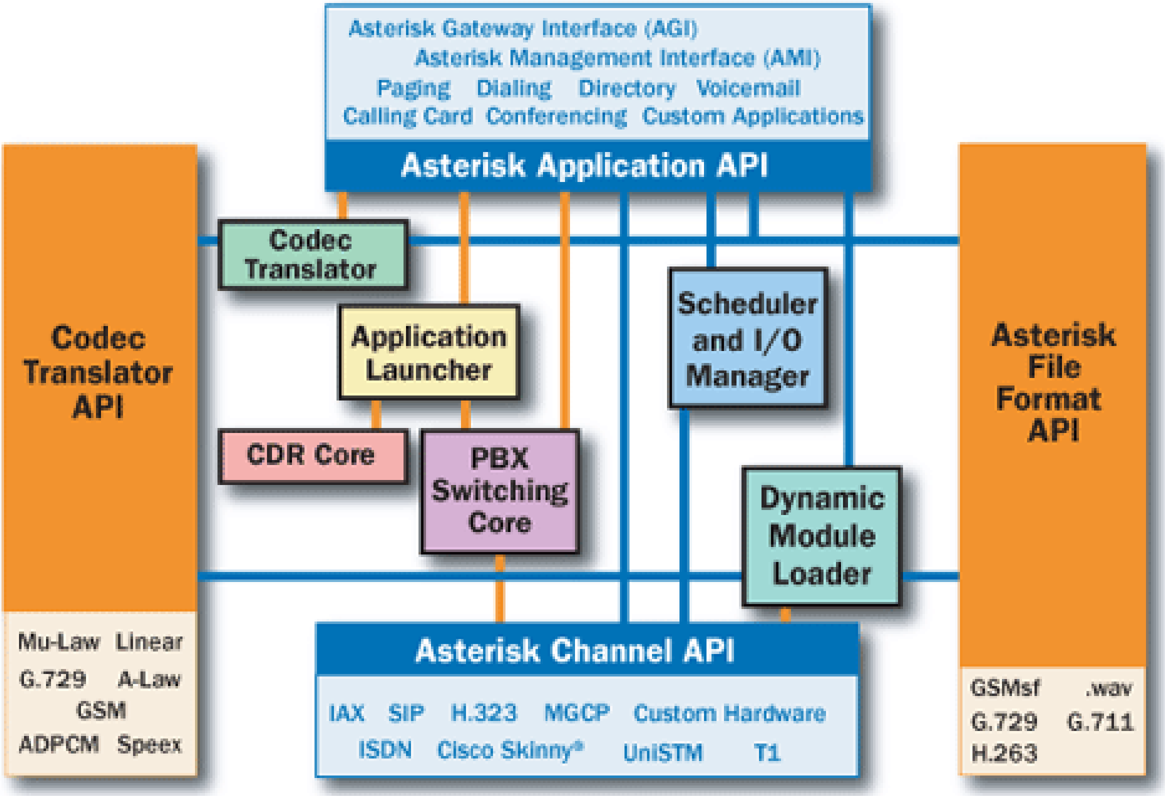
- 1.2.x
  - obsoleted
- 1.4.x
  - Old stable
  - 1.4.29
    - Branch Release Date: 2006-12-23
    - Security only fixes: 2010-12-23
    - End of Life (EOL): 2011-12-23
- 1.6.x
  - devel
- 1.8.x
  - LTS Long Term Support
  - Release date: TBD (estimated Q3 2010)
- Roadmap
  - https://issues.asterisk.org/roadmap_page.php





Architecture

- 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](!)                ; a template
        dtmfmode=rfc2833
        context=from-office
        type=friend

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

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

[my-codecs](!)                    ; a template for my preferred codecs
        disallow=all
        allow=ilbc
        allow=g729
        allow=gsm
        allow=g723
        allow=ulaw

[ulaw-phone](!)                   ; and another one for ulaw-only
        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"
- Auto-complete <tab>, <?>

- Commands
  - Core commands
    - debug verosity 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.

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
- $[expression]
  exmple:
  exten => 321,1,Set(COUNT=3)
  exten => 321,n,Set(NEWCOUNT=$[${COUNT} + 1])
  exten => 321,n,SayNumber(${NEWCOUNT})
- Most important predefined channel variables
  - ${CALLERID(all)}
    - ${CALLERID(num)}
    - ${CALLERID(name)}
  - ${DIALSTATUS}
    - CHANUNAVAIL, CONGESTION, BUSY, NOANSWER, ANSWER, CANCEL, HANGUP
  - ${CONTEXT}
  - ${CHANNEL}
  - ${EXTEN}

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)

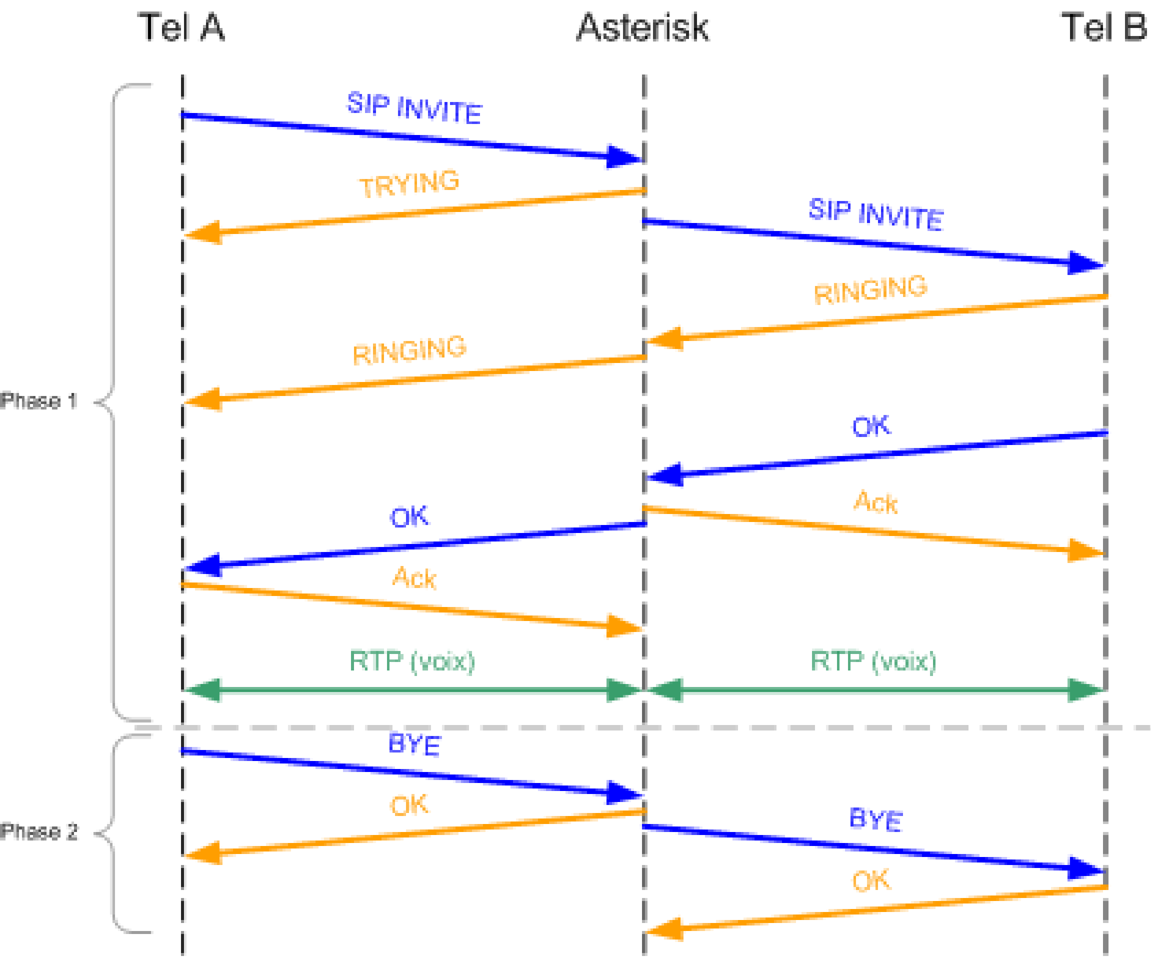
  - Macro definition is similar to context
  - xyz is the name of the macro
  - Can triggered from any context with
    [default]
    exten => 6601,1,Macro(xyz,SIP/1000,10)
  - Where SIP/100 is ARG1 and 10 is Arg2
- In a macro context, extra channel variables are available.
  - ${ARG1}: The first argument 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

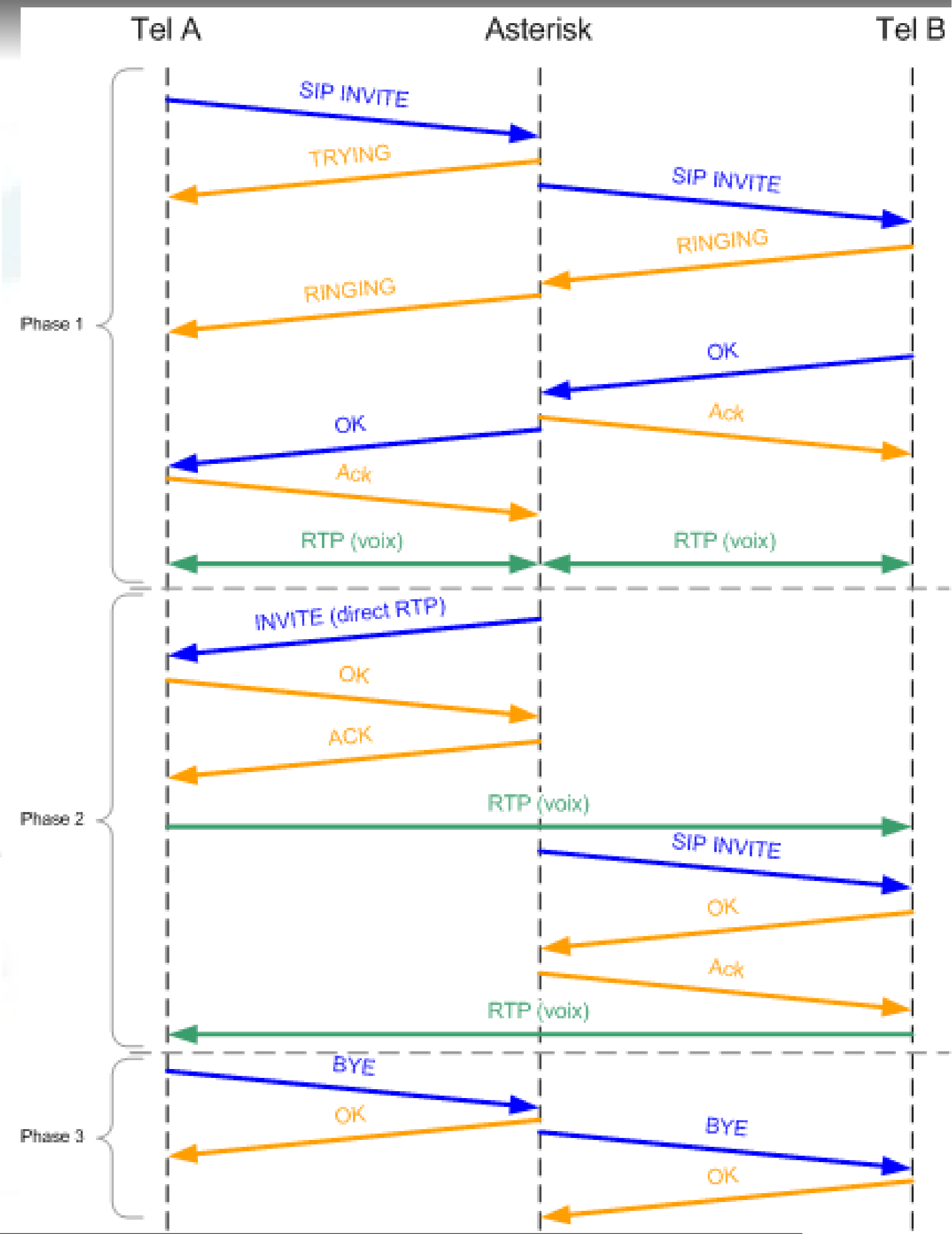
AstDB database

- Berkley DB version 1
- familly, 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” []

sip.conf

- [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 timout
  - 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]]]
      - Eample:
        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-respose
  - 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 packagae

- MySQL

```
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
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- http://downloads.oreilly.com/books/9780596510480.pdf
- http://astbook.asteriskdocs.org
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- http://www.voip-info.org/wiki/view/Asterisk+tips+openhours
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- http://areski.net/asterisk-stat-v2/about.php
- extensions.conf

Thank You!