THE 12 TASKS OF ASTERISK

Introduction

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http://www.sinologic.net/

Asterisk
PBX

Asterisk PBX

Sinologic.net

Asterisk SCF

Scalable Communications Framework

Astricon - October 2010

Introduction

Asterisk PBX

Asterisk SCF

Scalable Communications Framework

September 2012

Asterisk

Framework

Software as ideas, are better the more you can leverage

Introduction

	Stable Production	Unstable Development
2007	Asterisk 1.4	
2008		Asterisk 1.6
2010	Asterisk 1.8	
2011		Asterisk 10
2012	Asterisk 11	
2013		Asterisk 12
2014	Asterisk 13	

We will see some changes in Asterisk to become a Framework of development of voice applications

Flexibility

Flexibility

+ Options

+ Flexible

+ Powerful

+ Solutions

Flexibility

PBX

VoiceMail(account,options)
VoiceMailMain([account])

Framework

VoiceMail(account,options)
VoiceMailMain([account])
+
MinivmGreet
MinivmRecord
MinivmMWI
MinivmNotify
MinivmAccMess
MinivmDelete

ConfBridge

Asterisk 1.8

ConfBridge([confno][,options])

Asterisk 11

ConfBridge(conference[,bridge_profile[,user_profile[,menu]]])

Users templates

Bridges templates

Menus templates

Fax

Asterisk 1.8 allow:
Send faxes (SendFax)
Recieve faxes (RecieveFax)

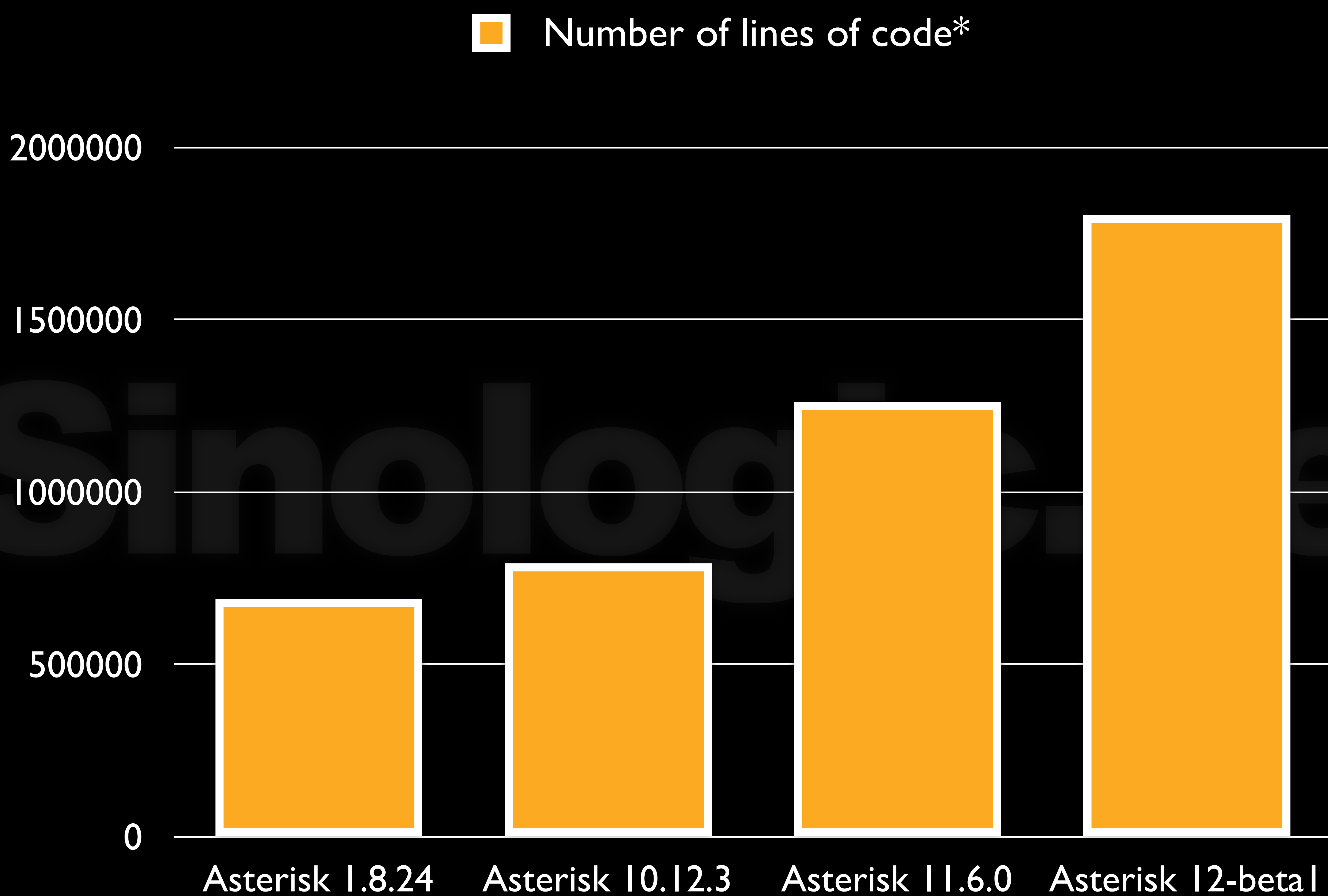
You can not relay faxes using T.38 - *FaxRelay*

We recieve a Fax using T.38 we want send to any ATA where a Fax machine is installed.

Asterisk 11

```
exten=>fax,1,FAXOPT(gateway)=yes
exten=>fax,n,Dial(SIP/FaxATA)
```

Flexibility



Más información: http://www.sinologic.net/proyectos/voip2day/2013/indice-asterisk

Multi-tenant

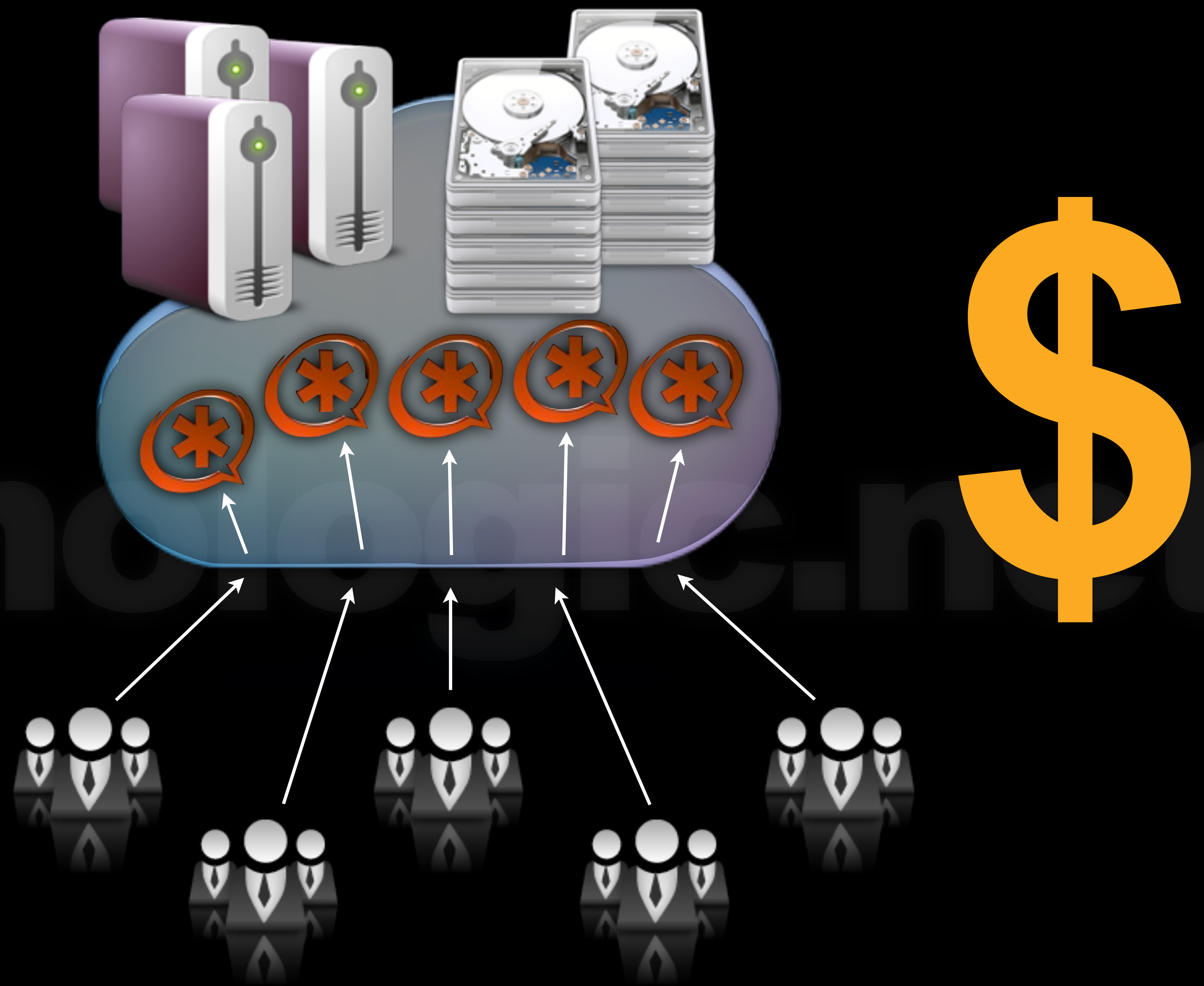
Multi-Tenant

Multitenancy:

Principle in software architecture where a single instance of the software runs on a server, serving multiple client-organizations (tenants).

Multi-Tenant

Multi-Tenant



Most common structure

Multi-Tenant

Multi-tenant system

How it could be implemented?

...some of code...

extensions.conf

```
...
exten=>912345678,1,Goto(company1,s,1)
...
exten=>987654321,1,Goto(companyN,s,1)
...

[company1]
#include "extensions_company1.conf"

...

[companyN]
#include "extensions_companyN.conf"

...
```

```
exten => s,1,Set(CHANNEL(namedcallgroup)=company1)
exten => s,n,Playback(company1_bienvenida)
exten => s,n,Goto(company1_IVR,s,1)

[company1_IVR]
exten => s,1,Read(opcion,company1_menuyopciones)
exten => s,n,Goto(company1_IVR,${opcion},1)

exten => 1,1,Dial(SIP/company1_300)
exten => 2,1,Dial(SIP/company1_301)
exten => 3,1,Queue(company1_comercial)
exten => 4,1,Queue(company1_soporte)
...


[company1_outgoing]
exten=>_3XX,1,Dial(SIP/company1_${EXTEN})
exten=>_[67]XXXXXXXX,1,Macro(outboundCall,"company1")
exten=>_[89]XXXXXXXX,1,Macro(outboundCall,"company1")
...
exten=>*89,1,Answer()
exten=>*89,n,VoiceMailMain(company1_${CALLERID(name)}@company1)
...
```

Multi-Tenant

Parkings

features.conf (Asterisk 1.8, 10, 11)
res_parking.conf (Asterisk 12)

```
[company]
context => company1_park
parkpos => 800-850
findslot => next
comebacktoorigin = no
comebackdialtime = 90
comebackcontext = company1_outgoing
parkedmusicclass = company1_moh

Command: Park(company1[,options])
```

Users / Phones

sip.conf

```
[default](!)
type=friend
secret=c82j34r9c82j398c9jh9438cj
host=dynamic
label=cuenta
nat=force_rport
context=none

[company1](!,default)
context=company1_outgoing
namedcallgroup=company1
namedpickupgroup=company1

[company1_ext300](company1)
callerid=300 <300>
secret=rcn2398rjc92834jc92
…
[company1_ext399](company1)
callerid=399 <399>
secret=icj984j92834jc928984c
…
```

extensions.conf

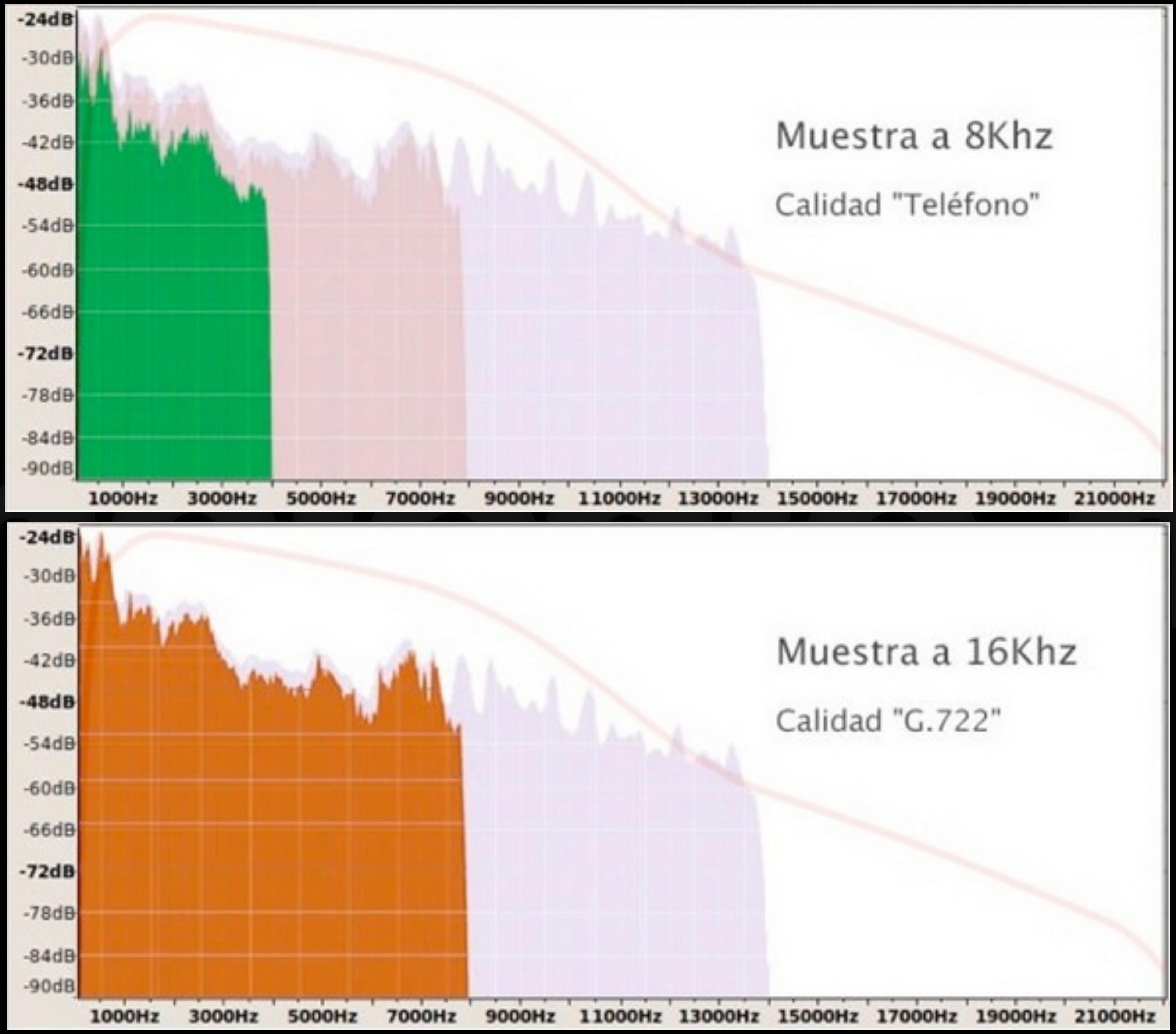
```
[company1_default]
exten => s,1,Set(CHANNEL(namedcallgroup)=company1)
exten => s,n,Playback(company1_welcome)
exten => s,n,Goto(company1_IVR,s,1)

[company1_IVR]
exten => s,1,Read(option,company1_menuandoptiones)
exten => s,n,Goto(company1_IVR,${option},1)

exten => 1,1,Dial(SIP/company1_300)
exten => 2,1,Dial(SIP/company1_301)
exten => 3,1,Queue(company1_commercial)
exten => 4,1,Queue(company1_support)
...
```

Codecs

Asterisk 1.8 supports:

Alaw / G.711a / PCMA
Ulaw / G.711u / PCMU
G.729a
iLBC
Speex
G.722
G.726
GSM
LPC10
SLIN

G.722 marked a milestone in quality of the sound



From then, every IP phones support G.722

Codecs Asterisk 10

Alaw / G.711a / PCMA
Ulaw / G.711u / PCMU
G.729a
iLBC
Speex 8/**16/32**
G.722
G.726
GSM
LPC10
SLIN 8/**12/16/24/32/44/48/96/192**
CELT 32/44/48
SILK 8/12/16/24

Codecs HD

Codecs

March 3, 2009:

Skype release SILK

September 11, 2012:

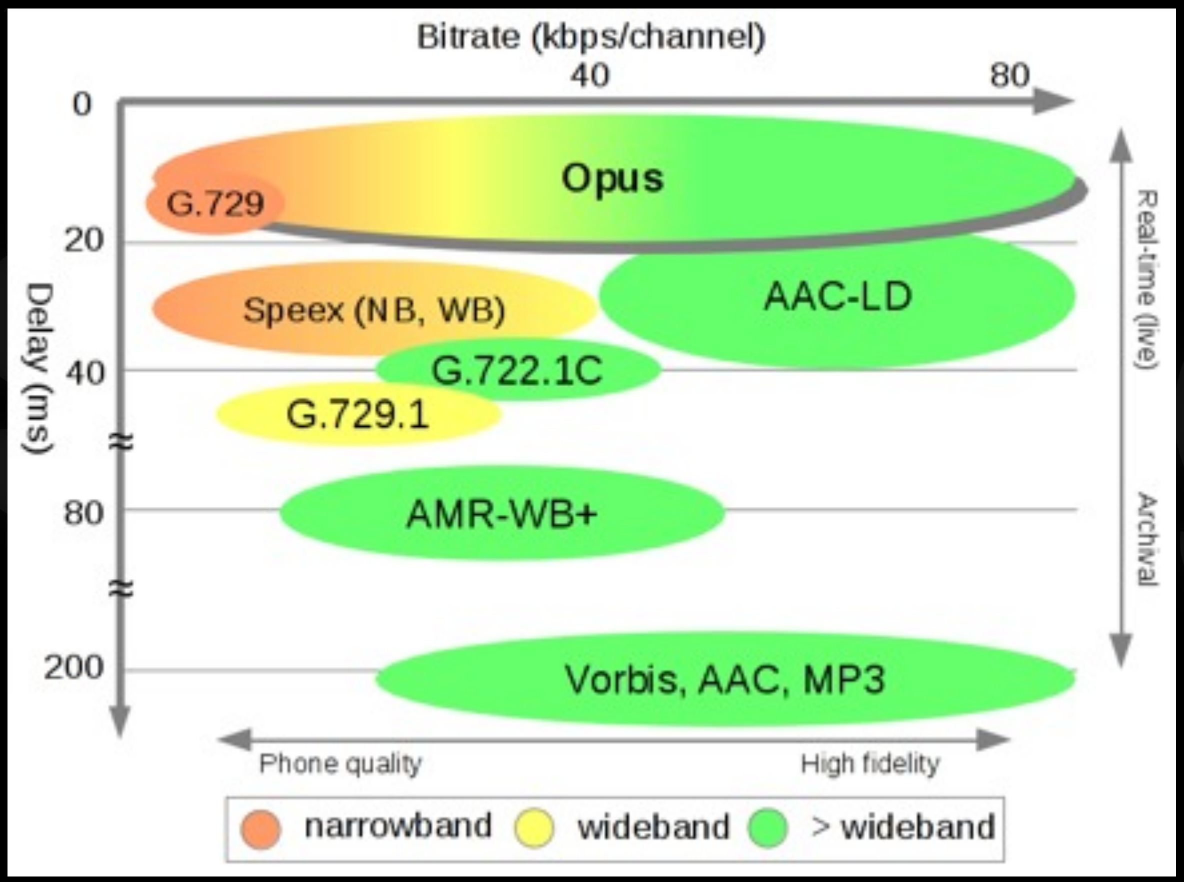
Xiph *(creator of Speex and CELT)* uses SILK and release:

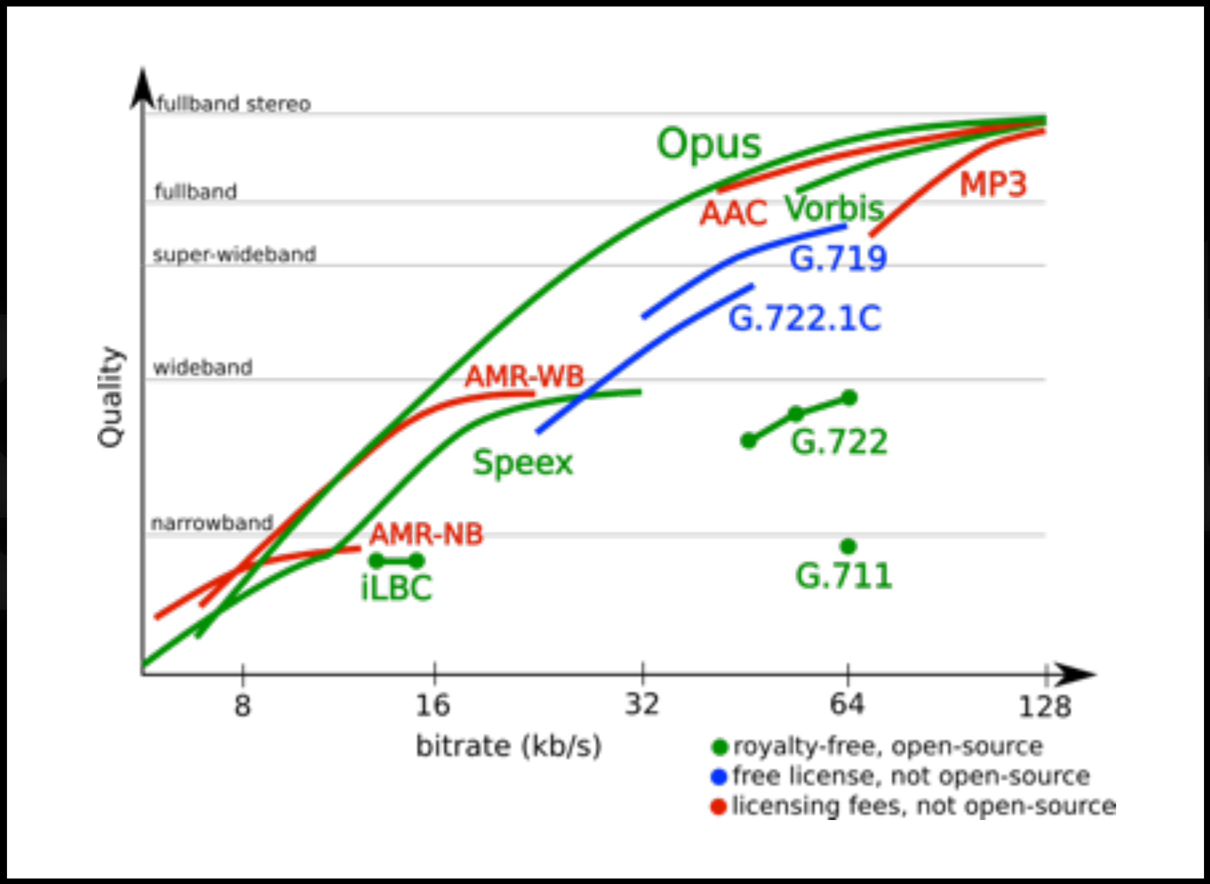
OPUS

"The definitive codec"

http://www.sinologic.net/blog/2012-09/ha-nacido-un-nuevo-codec-opus.html

OPUS


Bitrate (kbps/channel)
0
40
80
20
Delay (ms)
200
Opus
G.729
Speex (NB, WB)
AAC-LD
G.722.1C
G.729.1
AMR-WB+
Vorbis, AAC, MP3
Real-time (live)
Archival
Phone quality
High fidelity
narrowband
wideband
> wideband


Quality
fullband stereo
fullband
super-wideband
wideband
narrowband
Opus
AAC
Vorbis
MP3
G.719
G.722.1C
AMR-WB
Speex
G.722
AMR-NB
iLBC
G.711
8
16
32
64
128
bitrate (kb/s)
royalty-free, open-source
free license, not open-source
licensing fees, not open-source

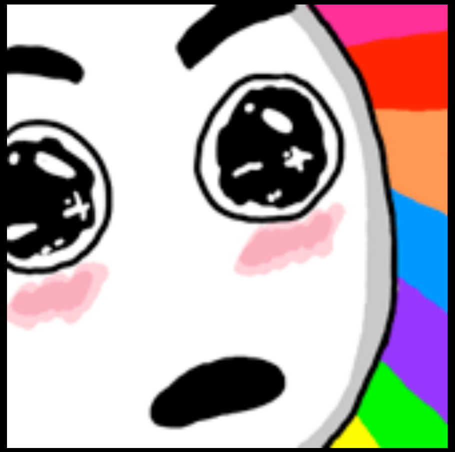
Asterisk 11 supports:

Alaw / G.711a / PCMA
Ulaw / G.711u / PCMU
G.729a
iLBC
~~Speex~~
G.722
G.726
GSM
LPC10
SLIN 8/12/16/24/32/44/48/96/192
~~CELT~~
~~SILK~~
OPUS

Codecs

Opus is the codec that everyone have dreamed to have:

- **Quality equal or better than MP3**
- **Bandwidth equal or less than G.729**
- **Auto-adjust quality to bandwidth available**
- **Standard of other tecnologies: WebRTC**
- **Consumption of processor awesome**
- **Really a free codec (free** like **free**dom**)**



OPUS

Asterisk 11 supports Opus and VP8 using a patch

https://github.com/meetecho/asterisk-opus

Asterisk 12 supports Opus and VP8 native but passthrough

(for the moment)

http://lists.digium.com/pipermail/asterisk-dev/2013-May/060421.html

Asterisk 12 supports:

Alaw / G.711a / PCMA
Ulaw / G.711u / PCMU
G.729a
iLBC

G.722
G.726
GSM
LPC10
SLIN 8/12/16/24/32/44/48/96/192

OPUS

Databases

Databases

Asterisk uses database...

to store **configuration**.

to store **calls** CDR/CEL.

to store **dialplan**.

to store **ASTDB**.

PostgreSQL

ODBC

SQLite3

TDS

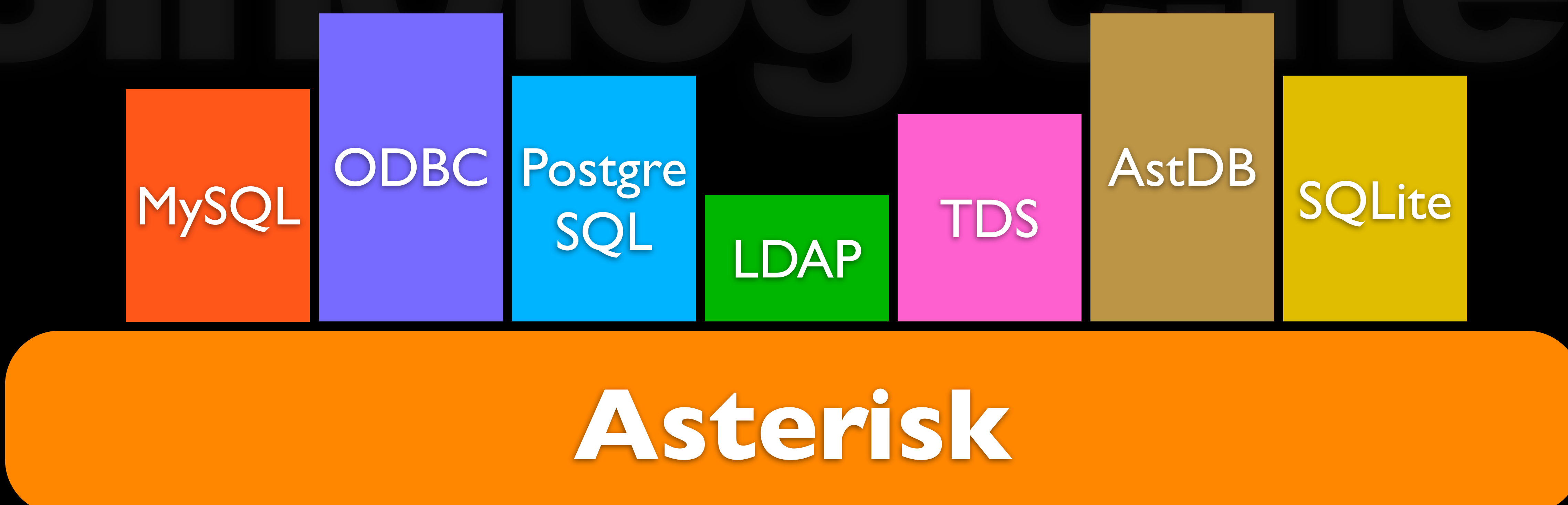
LDAP

AstDB = SQLite3

```
# sqlite3 /var/lib/asterisk/astdb.sqlite3
SQLite version 3.7.13 2012-06-11 02:05:22
Enter ".help" for instructions
Enter SQL statements terminated with a ";"
sqlite> .tables
astdb
sqlite> select * from astdb;
/SIP/Registry/erojano|90.126.122.53:54468:3600:erojano:sip:erojano@192.168.0.1:1024
/SIP/Registry/ratienza|90.126.122.53:54468:3600:ratienza:sip:ratienza@192.168.0.2:1024
/dundi/secret|LeQyTgBhJzpYUtUZgDL+Iw==;HhDfflzf+gPSLgpstOZTaQ==
/dundi/secretexpiry|1383599365
```

Databases

Each database supported is a different module with its own support.

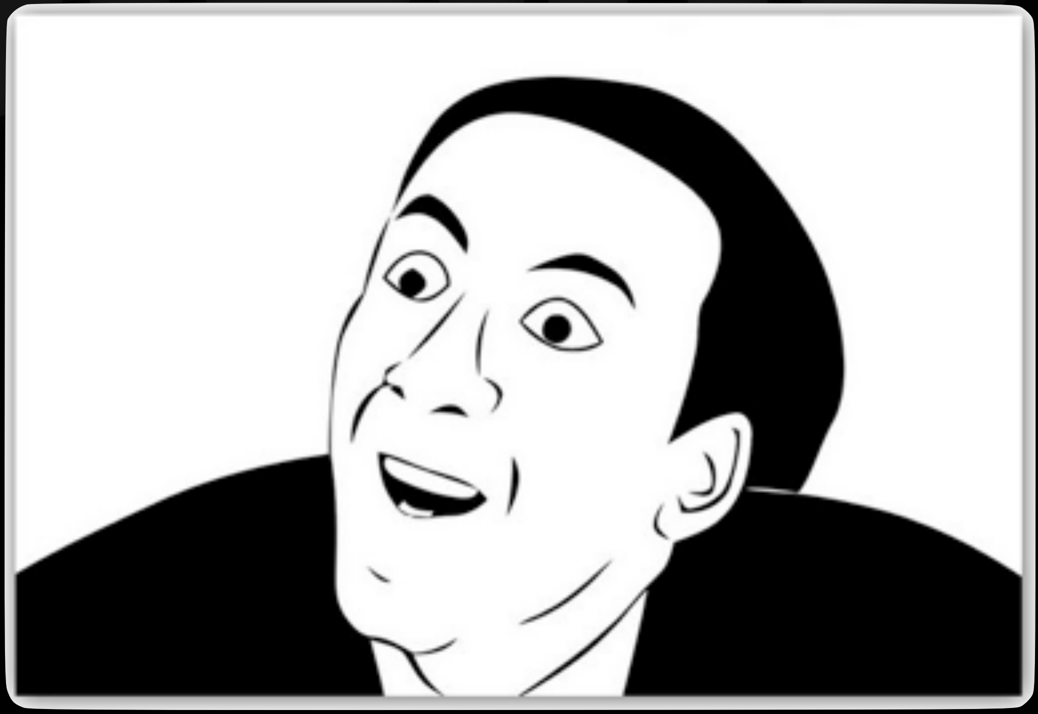


Asterisk 12
Sorcery

Abstract layers will be very important from now.

Enhanced management of NAT

NAT is a problem



NAT

IAX2 isn't the solution of NAT

Mapping ports neither

Using DMZ even less

Exists some real solutions and Asterisk already supports

ICE

Interactive Connectivity Establishment protocol

More information: http://www.slideshare.net/saghul/ice-4414037

ICE is a protocol to discover what kind of NAT we are behind.

If there is some way to establish a communication, ICE will find it.

ICE is supporting in STUN servers

If it can't transmit audio directly, then we must to use TURN to relay the audio.

ICE must be supported on the client side too. (softphones / IP phones)

List of STUN servers available: https://gist.github.com/hellc2/7290201

sip.conf

```
[general]
...
icesupport = yes
stunaddr=servidor.stun.org
...
```

rtp.conf

```
[general]
...
icesupport = yes
stunaddr=servidor.stun.org
turnaddr=servidor.turn.org
turnusername=USUARIO
turnpassword=PASSWORD
...
```

WebRTC

You already must to know
what WebRTC is

Requirements of Asterisk:

- Support of **WebSocket** ✔
- Support of **OPUS** ✔
- Support of **VP8** ✔
- Support of **ICE** ✔
- Support of **SRTP** ✔

So we can use WebRTC in Asterisk?

YES

Configuring WebSocket support

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088              ;; Port where Websocket will listen to
prefix=asterisk            ;; http://SERVER:8088/asterisk/ws
```

http.conf

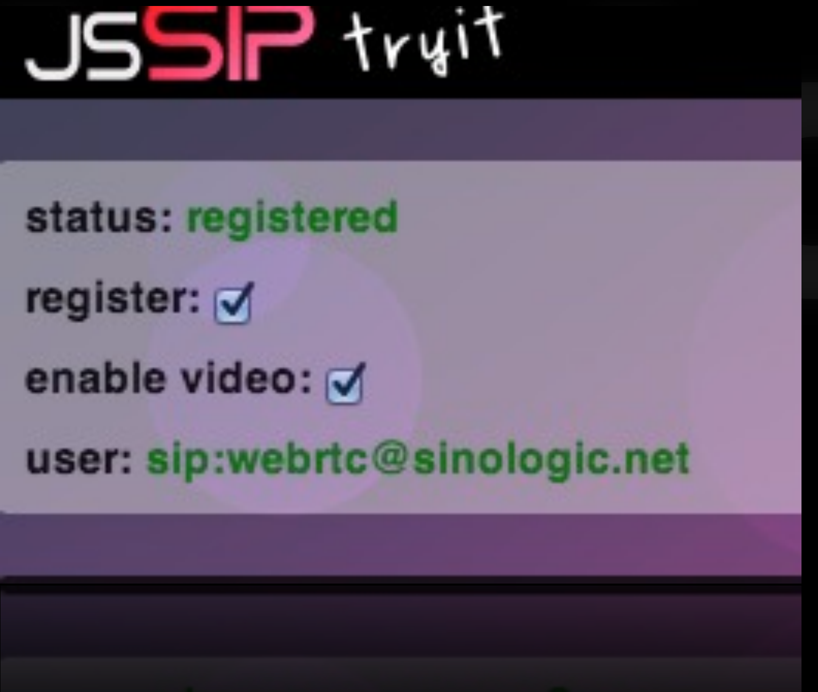
Configuring SIP accounts with the requirements of WebRTC

```
[webrtc_user1]
type=friend
host=dynamic
context=webrtc
secret=user1@webrtcpass1
transport=ws,wss         ;; Enabling of WebSocket support and WebSocketSecure
encryption=yes           ;; Enabling of SRTP
avpf=yes                 ;; Kind of stream we are working with
nat=yes,force_rport      ;; Compatibility with Asterisk 11
disabled=all
allow=opus,vp8
```

sip.conf

WebRTC

```
SSL certificate ok
[Nov  3 19:44:58] WARNING[5922]: sip/config_parser.c:812 sip_parse_nat_option: nat=yes is deprecated, u
  == Parsing '/etc/asterisk/sip_notify.conf': Found
  == WebSocket connection from '79.158.116.133:61543' closed
  == WebSocket connection from '79.158.116.133:61614' for protocol 'sip' accepted using version '13'
    -- Registered SIP 'webrtc' at 79.158.116.133:61614
  == Using SIP RTP CoS mark 5
```



https://github.com/versatica/JsSIP

Enhance Security

Security

SIP crypted using **TLS**

RTP crypted using **SRTP**

Password crypted using **MD5**

Access Control Lists - **ACL**

Special log system for **Security**

Authenticate, VMAutenticate applications

Control of records and attempts of calls

Management of contexts based in permissions of SIP accounts

Applications to manage ports like portsentry, port-knocking,...

Posibility of using internal and external firewalls

Use of crypted VPN

Slap to who touch

...

Security

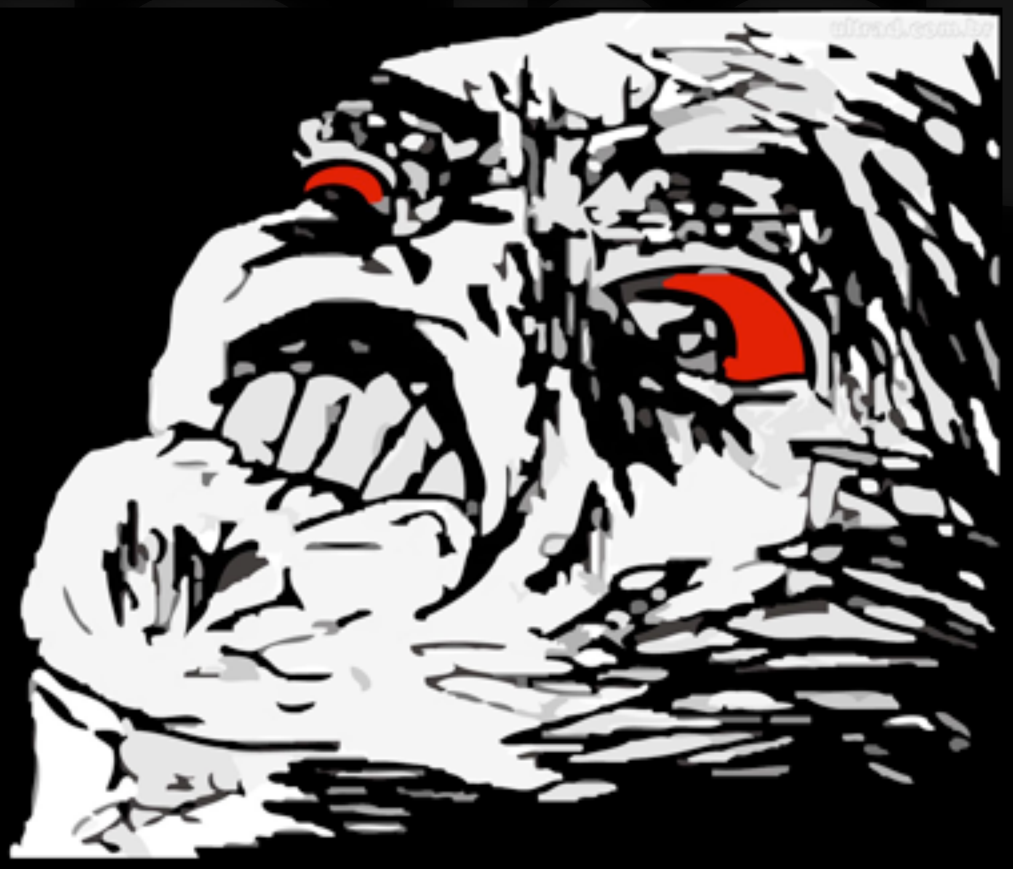
Very serious topic

Lots of different kind of attacks

Lots of different kind of defense

You only need to use it

1234 isn't a real password *pass100* neither



Security

Too much information = No information

logger.conf

```
security => security
console => notice,warning,error,debug
messages => notice,warning,error
full => notice,warning,error,debug,verbose,dtmf,fax
```

```
tail -f /var/log/asterisk/security
```

```
[Nov  3 20:02:53] SECURITY[5995] res_security_log.c:
SecurityEvent="InvalidPassword",
EventTV="2013-11-03T20:02:53.585+0100",
Severity="Error",
Service="SIP",
EventVersion="2",
AccountID="1001",
SessionID="0x7f4cf4dcc788",
LocalAddress="IPV4/UDP/178.60.201.227/5060",
RemoteAddress="IPV4/UDP/90.126.102.129/1024",
Challenge="7b331afe",
ReceivedChallenge="7b331afe",ReceivedHash="3bd127dc9f06e7f7d6a2fb2000364bf8"
```

ACL : Access Control List

```
sip.conf

[erojano]
type=friend
host=dynamic
nat=force_rport
secret=TH1S1SMyFxCk3DP4SsW0rD
deny=0.0.0.0/0.0.0.0
permit=192.168.0.0/255.255.255.0
permit=80.33.84.32/255.255.255.255
permit=10.0.0.0/255.255.0.0
permit=195.245.135.53/255.255.255.255
```

ACL : Access Control List

```
sip.conf

[erojano]
type=friend
host=dynamic
nat=force_rport
secret=TH1S1SMyFxCk3DP4SsW0rD
acl=erojano_sites
```

ACL : Access Control List

```
[erojano_sites]
deny=0.0.0.0
permit=192.168.0.0/255.255.255.0
permit=80.33.84.32/255.255.255.255
permit=10.0.0.0/255.255.0.0
permit=195.245.135.53/255.255.255.255
```

acl.conf

ACL : Access Control List

http://www.sinologic.net/proyectos/asterisk/acl/

```
wget http://www.sinologic.net/proyectos/asterisk/acl/acl_ES.conf -O /etc/asterisk/

acl.conf

[ES]
#include “acl_ES.conf”
```

Warning!
Using ACL don’t avoid attacks ;)

ACL : Access Control List

```
[erojano]
type=friend
host=dynamic
nat=force_rport
secret=TH1S1SMyFxCk3DP4SsW0rD
acl=ES    ;; Only connections from Spain
acl=DE    ;; Only connections from Germany
```

sip.conf

Security

Basic test of security:

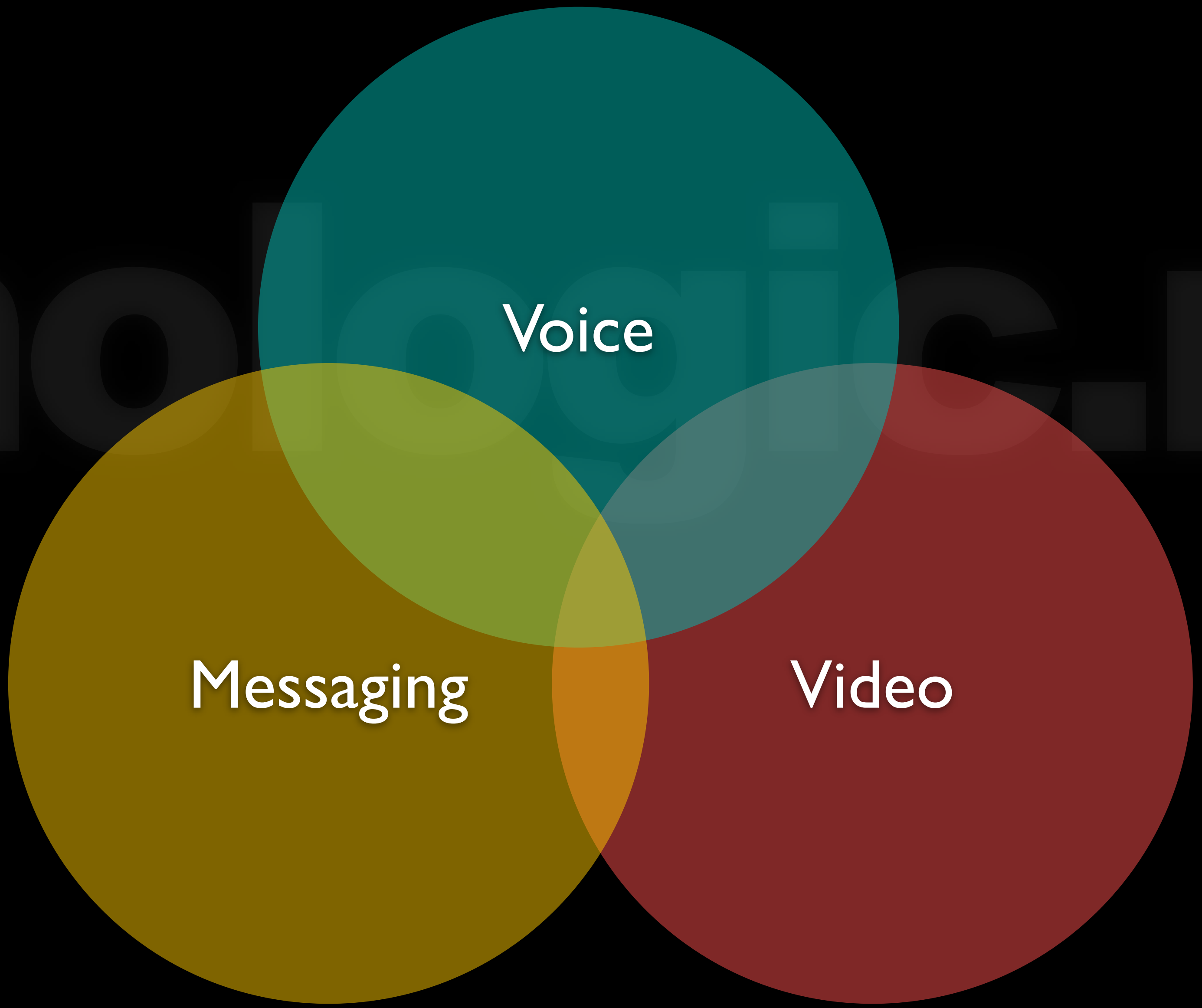
- Do you allow anonymous calls to PSTN?
- Do you have any user without password?
- Are there some numeric password?
- Are there any password that includes the username?
- Are there some users with the same password?
- Do you allow users from any IP?
- Can some user pretend to be other?

If answer YES to some of this questions, then you should review your security.

Repeat this test each 7 days

Instant Messaging

VoIP >> Voice over IP



Instant Messaging

XMPP

gtalk.conf

motif.conf

JINGLE

XMPP

SIP

sip.conf

MESSAGE

WebRTC

????

DATA CHANNEL API

Instant Messaging

Google every day complicates the connection with its messaging services

¿¿¿ May 2014 ???

Ideally for messaging is to use our own XMPP/Jabber Server

How many companies has got its own messaging XMPP/Jabber server?


Instant Messaging
SIP
Vs
XMPP

JABBER/XMPP

Asterisk supports functions of messaging compatible with Jabber and XMPP

```
JabberJoin: Join a chat room
JabberLeave: Leave a chat room
JabberSend: Sends an XMPP message to a buddy.
JabberSendGroup: Send a Jabber Message to a specified chat room
JabberStatus: Retrieve the status of a jabber list member
```

SIP SIMPLE

```
exten=>1,1,Set(MESSAGE(body)="Prueba de Mensaje”)
exten=>1,n,MessageSend(sip:erojano@sinologic.net)
```

```
<--- SIP read from UDP:178.60.201.227:5060 --->
MESSAGE sip:erojano@sinologic.net SIP/2.0
Via: SIP/2.0/UDP 178.60.201.227:5060;branch=z9hG4bK0bb64867
Max-Forwards: 70
From: "asterisk" <sip:asterisk@178.60.201.227>;tag=as72ef437d
To: <sip:erojano@sinologic.net>
Contact: <sip:asterisk@178.60.201.227:5060>
Call-ID: 053bbbb50a2cbe414a258a1e7c769c17@178.60.201.227:5060
CSeq: 102 MESSAGE
User-Agent: Asterisk PBX 12.0.0-beta1
Content-Type: text/plain;charset=UTF-8
Content-Length: 19

"Prueba de Mensaje"
```

You don't need send the message to an open channel

PRESENCE MANAGEMENT

`not_set`: Presence is UNDEFINED

`unavailable`: Marked as UNAVAILABLE.

`available`: Available for communication.

`away`: It isn’t near from system and surely it will not answer.

`xa`: It is out, and we don't know when he comes.

`chat`: Only avail to chat message (no phone).

`dnd`: Do not disturb.

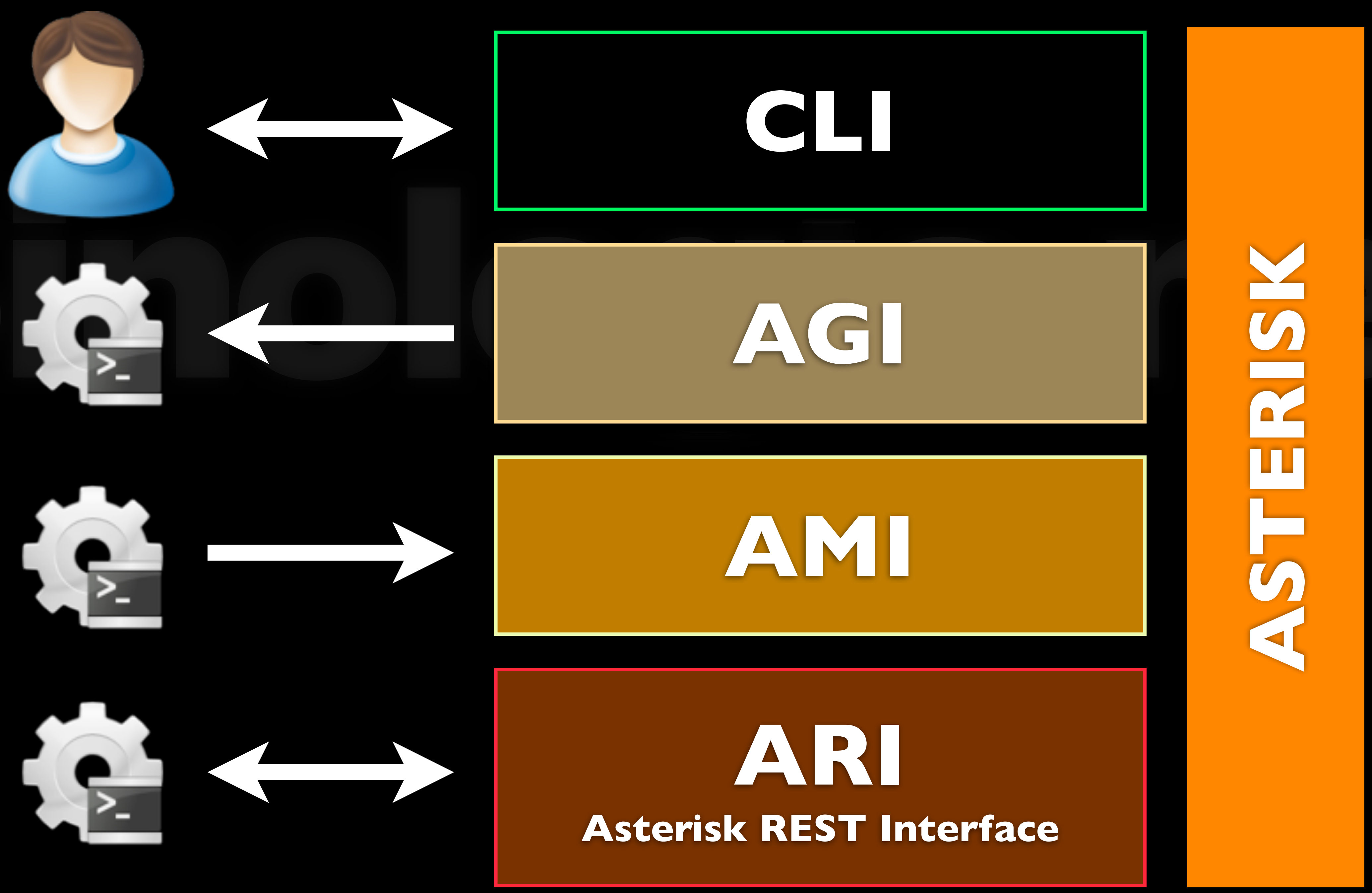

```
exten=>1234,hint,CustomPresence:led1
exten=>_XXXXXXXXX,1,Set(PRESENCE_STATE(CustomPresence:led1)=dnd,,On the phone)
exten=>_XXXXXXXXX,n,Dial(SIP/Operador/${EXTEN})

exten=>h,1,Set(PRESENCE_STATE(CustomPresence:led1)=available,,)
```

API development

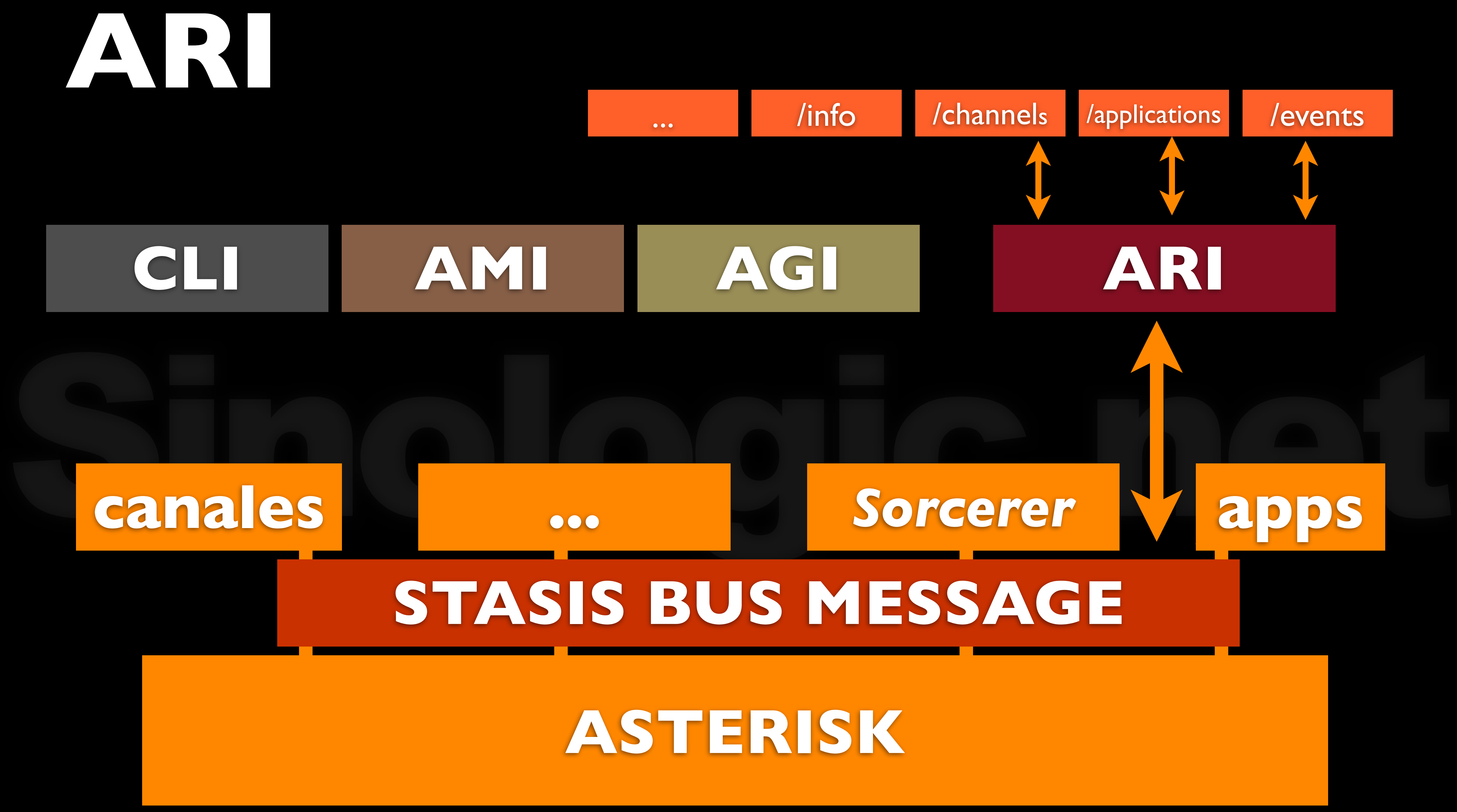
API development

A must for any Framework

API = Application Programming Interface



API development



API development

ARI allow realize lots of things that actually we do with Manager, although it isn't a substitute of Manager.

You can create 'services' that can been launched via REST requests

STASIS is other of the features of Asterisk 12 oriented to the development and a very important change in the form of work with Asterisk.

API development

We can get some information like manager does.

```
curl http://USER:PASS@www.sinologic.net:8088/ari/asterisk/info
```


```
{
 "status":{
    "startup_time":"2013-10-25T22:09:25.241+0200",
    "last_reload_time":"2013-10-25T22:09:25.241+0200"
  },
  "build":{
    "user":"root",
    "options":"LOADABLE_MODULES, BUILD_NATIVE, OPTIONAL_API",
    "machine":"x86_64",
    "os":"Linux",
    "kernel":"2.6.32-19-pve",
    "date":"2013-10-10 14:55:47 UTC"
  },
  "system":{
    "version":"12.0.0-beta1",
    "entity_id":"00:00:00:00:00:00"
  },
  "config":{
    "default_language":"en",
    "name":"",
    "setid":{"user":"","group":""}
  }
}
```

API development

```
curl http://USER:PASS@www.sinologic.net:8088/ari/channels
```


```
[
 {
   "id":"1383507920.1",
   "state":"Up",
   "name":"SIP/webrtc",
   "caller":{
              "name":"",
              "number":""
             },
   "connected":{
              "name":"",
              "number":""
             },
   "accountcode":"",
   "dialplan":{
             "context":"webrtc",
             "exten":"1",
             "priority":2
             },
   "creationtime":"2013-11-03T20:45:20.508+0100"
  }
]
```

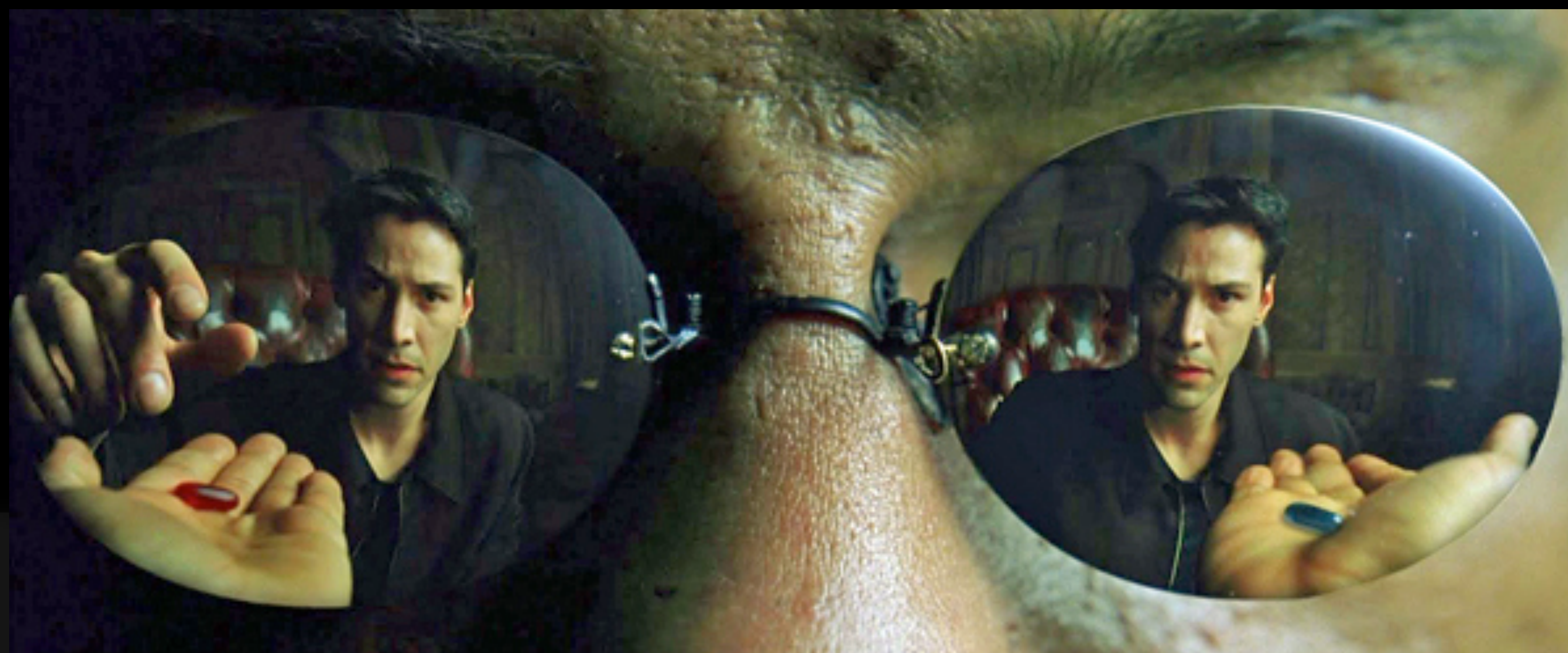
New SIP channel

New SIP channel

SIP channel needs a deep change

Difficult to change
Easy to broke
Only a little team knows how really it works

New SIP channel



Use other SIP stack

Rewrite chan_sip from scratch

PJSIP

Asterisk 12

It have to compile separately.
Don't be excited: chan_sip continues working.

Basic example: pjsip.conf

Transport

```
[transport-udp]
type=transport
protocol=udp ;,tcp,tls,ws,wss
bind=0.0.0.0
localnet=192.168.1.0/24
external_media_address=178.60.101.227
external_signaling_address=178.60.101.227
```

Trunks

```
[mytrunk]
type=registration
transport=transport-udp
outbound_auth=mytrunk_auth
server_uri=sip:sip.example.com
client_uri=sip:1234567890@sip.example.com
contact_user=1234567890
retry_interval=60
forbidden_retry_interval=600
expiration=3600
```

```
[mytrunk_auth]
type=auth
auth_type=userpass
password=1234567890
username=1234567890
realm=sip.example.com
```

Users

```
[erojano]
transport=transport-udp
type=endpoint
context=outgoing
mailbox=erojano
disallow=all
allow=alaw
auth=auth-erojano
aors=erojano
```

```
[auth-erojano]
type=auth
auth_type=userpass
username=erojano
password=m1gr4ns3cr3t0
```

```
[erojano]
type=aor
max_contacts=3
minimum_expiration=60
```

Questions?

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