SIP Express Media Server
SBC

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Contents

- The SIP Express Media Server
- SEMS SBC

……………………. snip …………………….

- #MoreCrypto

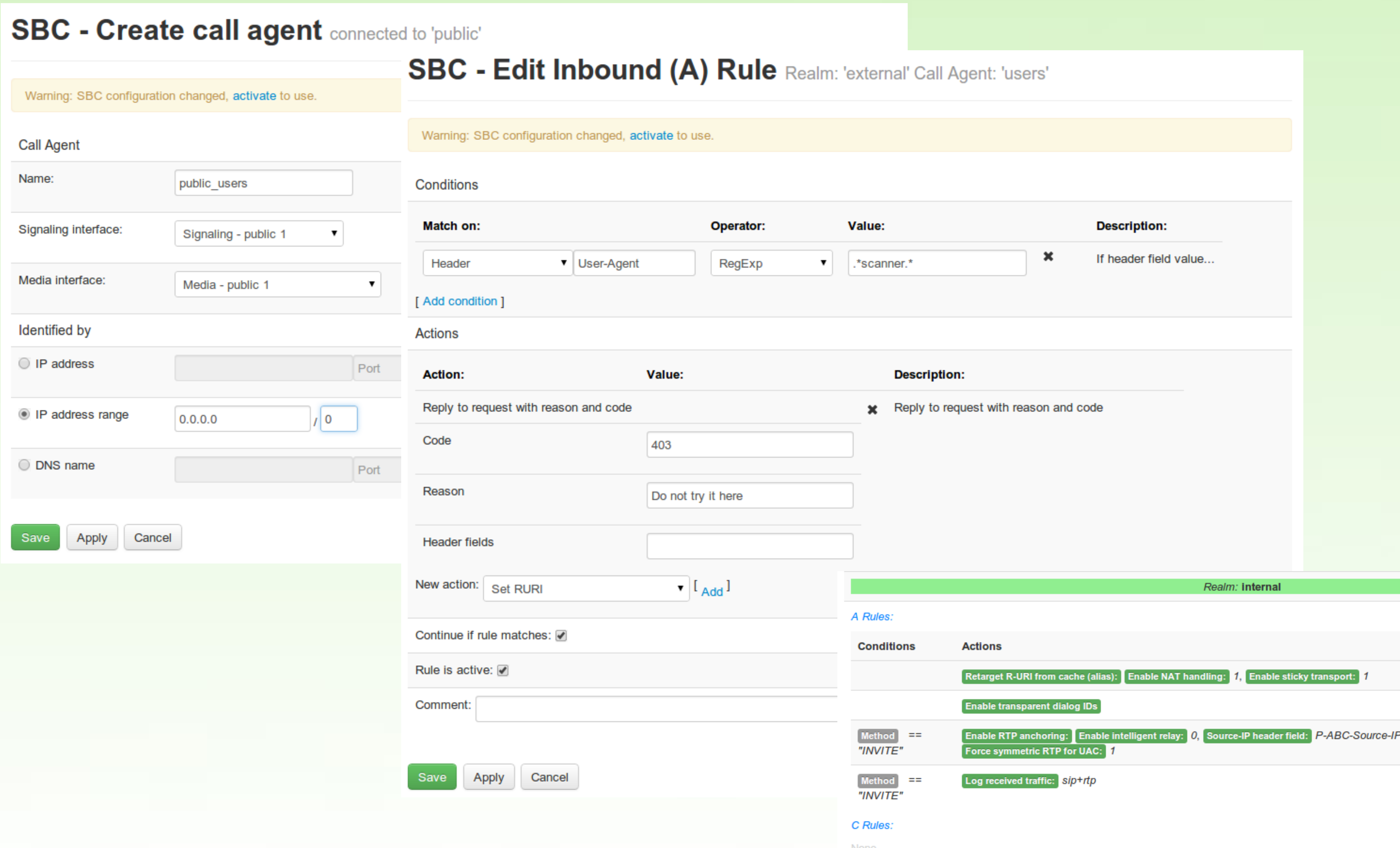
SEMS

- Originates from the same team as SER (Kamailio/OpenSER/...) at Fraunhofer FOKUS
- SIP Media and Application Server
- Developed at various related companies (iptelorg, IPTEGO, ...)
- Since 2010 mainly at FRAFOS
- Open Source community since 2003

FRAFOS ABC SBC

- Full-fledged SBC, turn-key solution
- Border security, monitoring, SIP control and mediation, registration offload, transcoding etc
- Software only, on FRAFOS-provided hardware or virtualized deployment (incl EC2)
- HA with active-hot standby (SIP+RTP)
- 100% rule based administration through GUI
- **Application offloading and integration through open APIs and programming platform**
- **WebRTC gateway**

ABC SBC - GUI

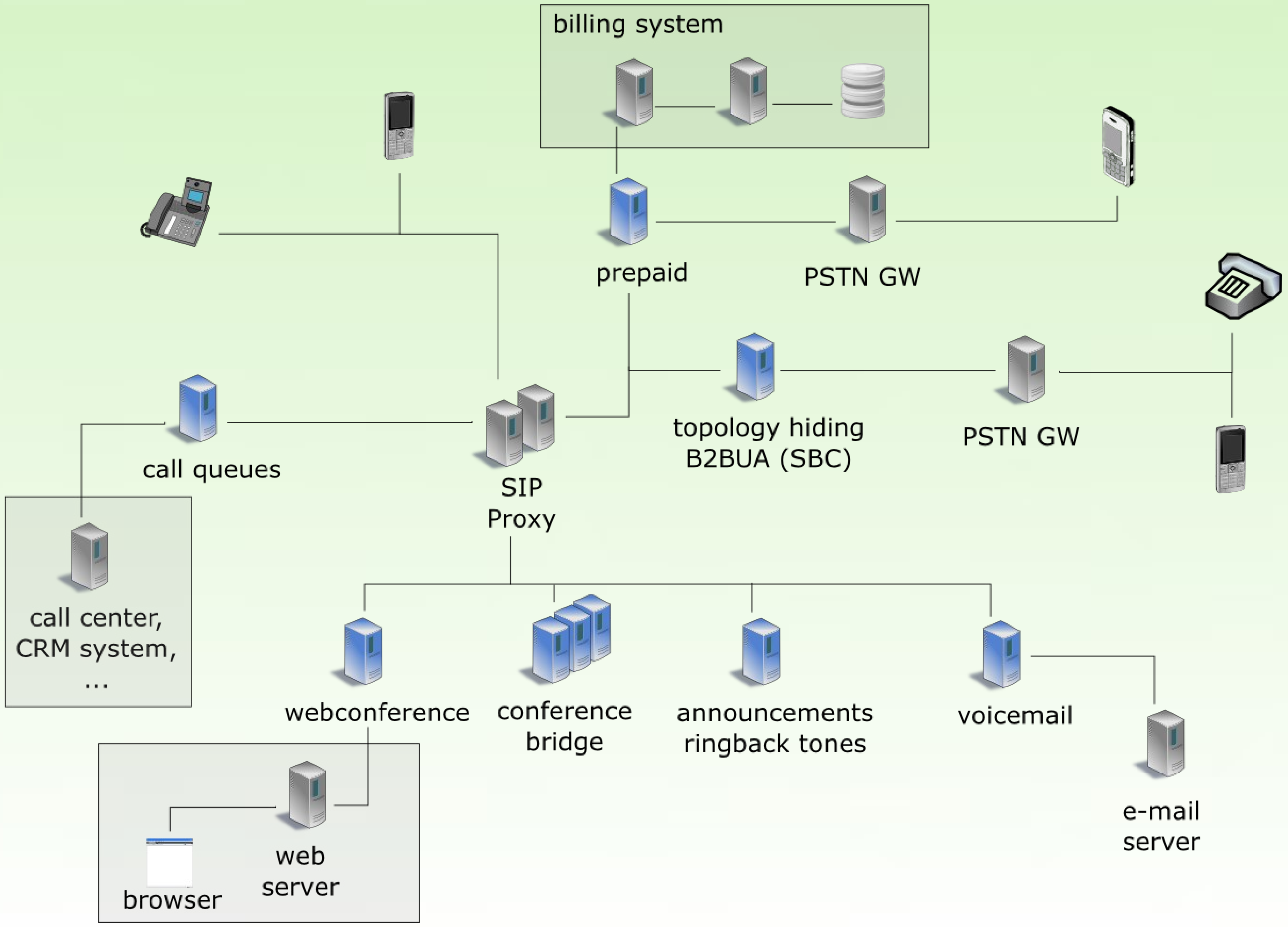

SBC - Create call agent connected to 'public'
Warning: SBC configuration changed, activate to use.
Call Agent
Name: public_users
Signaling interface: Signaling - public 1
Media interface: Media - public 1
Identified by
IP address
Port
IP address range
0.0.0.0
/ 0
DNS name
Port
Save
Apply
Cancel
SBC - Edit Inbound (A) Rule Realm: 'external' Call Agent: 'users'
Warning: SBC configuration changed, activate to use.
Conditions
Match on:
Operator:
Value:
Description:
Header
User-Agent
RegExp
.*scanner.*
If header field value...
[Add condition]
Actions
Action:
Value:
Description:
Reply to request with reason and code
Reply to request with reason and code
Code
403
Reason
Do not try it here
Header fields
New action: Set RURI
[Add]
Continue if rule matches:
Rule is active:
Comment:
Save
Apply
Cancel
Realm: internal
A Rules:
Conditions
Actions
Retarget R-URI from cache (alias):
Enable NAT handling: 1,
Enable sticky transport: 1
Enable transparent dialog IDs
Method == "INVITE"
Enable RTP anchoring:
Enable intelligent relay: 0,
Source-IP header field: P-ABC-Source-IP,
Force symmetric RTP for UAC: 1
Method == "INVITE"
Log received traffic: sip+rtp
C Rules:
None

SEMS project focus

- Telecoms applications, carrier environment
  - High volume prompts, voicemail, conferencing, …
  - B2BUA / SBC
- Speed and reliability
- Only SIP, not multi-protocol (almost)
- Versatile and easy to use app server for SIP networks

- *Built for purpose*

SEMS use cases



SEMS SBC application

- B2BUA, completely transparent to fully opaque
- Handles SIP and (optional) RTP
- Flexible and programmable
- *”The Swiss Army Knife of call stateful SIP processing”*

SEMS SBC features

- B2BUA, network separation
- SIP message manipulation & mediation, header/message filter
- SIP NAT handling, TCP/UDP, DNS SRV w/failover
- SST enforcement
- Registration Caching
- SIP client auth
- CDR generation, call timer, parallel call limits, prepaid, ...

SBC: media features

- RTP relaying
- Near & far end NAT traversal
- Codec filter, SDP filter
- Transcoding

SBC: Profile based control

set_fromto.sbcprofile.conf

```
URI=$tU@sbc1.mypeer.net
From=<$fU@mynet.net>
To=<sip:$tU@mypeer.net>
Call-ID=$ci_leg2
enable_rtprelay=yes
```

known
SER
pseudo-variables

SEMS SBC

```
#
U 210.13.3.122:5080 -> 210.13.3.100:5060
INVITE sip:+49123@osbc1.mynet.net SIP/2.0
From: "John" <sip:+431556221@mynet.net>;tag=12
To: "Clara" <+49123@mynet.net>
Call-ID: 3cde5d1a960a-dez6oz34llo4
...
```

```
#
U 210.13.3.100:5060 -> 213.192.59.75:5060
INVITE sip:+49123@sbc1.mypeer.net SIP/2.0
From: <+431556221@mynet.net>;tag=3213
To: <sip:+49123@mypeer.net>
Call-ID: 3cde5d1a960a-dez6oz34llo4_leg2
...
```

SBC example: auth_b2b

- Identity change
- SIP auth upstream
- Set e.g. In headers
  - $P(*name*) selects *name* from P-App-Param

auth_b2b.sbcprofile.conf

```
RURI=sip:$rU@$P(d)
From="\"$P(u)\" <sip:$P(u)@$P(d)>"
To="\"$rU\" <sip:$rU@$P(d)>"

enable_auth=yes
auth_user=$P(u)
auth_pwd=$P(p)
```

Test:

```
        if (uri=~"^sip:\+49.*") {
                remove_hf("P-App-Name");
                remove_hf("P-App-Param");
                append_hf("P-App-Name: sbc\r\n");
                append_hf("P-App-Param: u=8708138;d=sipgate.de;p=mypasswd\r\n");
                force_send_socket(192.168.2.32:5060);
                t_relay_to_udp("192.168.2.34","5060");
                exit;
        }
```

Some profile options

```
RURI=$r
From=$f
To=$t
Contact=<sip:$Ri>
Call-ID=$ci_leg2

outbound_proxy=sip:192.168.5.106:5060
force_outbound_proxy=yes
next_hop=192.168.5.106:5060
outbound_interface=extern

enable_reg_caching=yes
min_reg_expires=3600
max_ua_expires=60

dlg_nat_handling=yes

enable_rtprelay=yes
rtprelay_force_symmetric_rtp=yes
aleg_rtprelay_interface=intern
rtprelay_interface=default
```

```
header_filter=blacklist
header_list=P-App-Param,P-App-Name
sdp_filter=whitelist
sdpfilter_list=g729,g723,ilbc,speex,gsm

append_headers="P-Src-IP: $si\r\n"

enable_session_timer=yes
session_expires=120
minimum_timer=90

enable_auth=yes
auth_user=$P(u)
auth_pwd=$P(p)

...
```

SBC: programmability

- Modules included e.g.
  - Blacklist from REDIS: *bl_redis*
  - SIP/feature control from http (REST) API: *rest*
- Simple Call Control API - start()/connect()/end()
- Extended Call Control API
  - Control each message in detail
  - Switch call legs PBX style, e.g. Mid-call prompts
  - Program also with DSM script

SBC programmability example

switch
B2B
to
local media
processing

```
transition "state changed" RUN - legStateChange / logParams(3) -> RUN;

transition "timer hit" RUN - timer(#id == 1) / {
  -- save other leg's ltag
  dlg.getOtherId($b_ltag);

  -- don't send hold, keep media session
  sbc.disconnect(false, true);

  -- instruct other leg to hang up
  set($cmd="hangup");
  set($call_id=@local_tag);
  postEvent($b_ltag, cmd;call_id);

  setInputPlaylist();
  connectMedia();
  playFile("wav/default_en.wav");
  sbc.streamsSetReceiving(false, false);

} -> PLAYING_FILE;

state PLAYING_FILE;
```

E stands for Express?

- *E*xcellent signaling performance
- RTP: fills 2x1 GbE to ~55% line rate (G711)
  - Limit: high PPS (loss NIC-kernel)
  - Perf testing without packet loss detection is meaningless!
- tuning:

• HT on/off

Makefile.defs:

```
USE_THREADPOOL=yes
MAX_RTP_SESSIONS=...
```

/etc/init.d/sems:

```
ulimit -n 100000
```

/etc/sems/sems.conf:

```
session_processor_threads=32
media_processor_threads=32
rtp_receiver_threads=32
sip_server_threads=16
```

start with cores x 2

#MoreCrypto - Motivation

- Too much centralization of power is dangerous
  - e.g. see Joseph Nacchio case
- Who is going to participate in society and politics in a 100% controlled Orwellian state with ubiquitous surveillance?

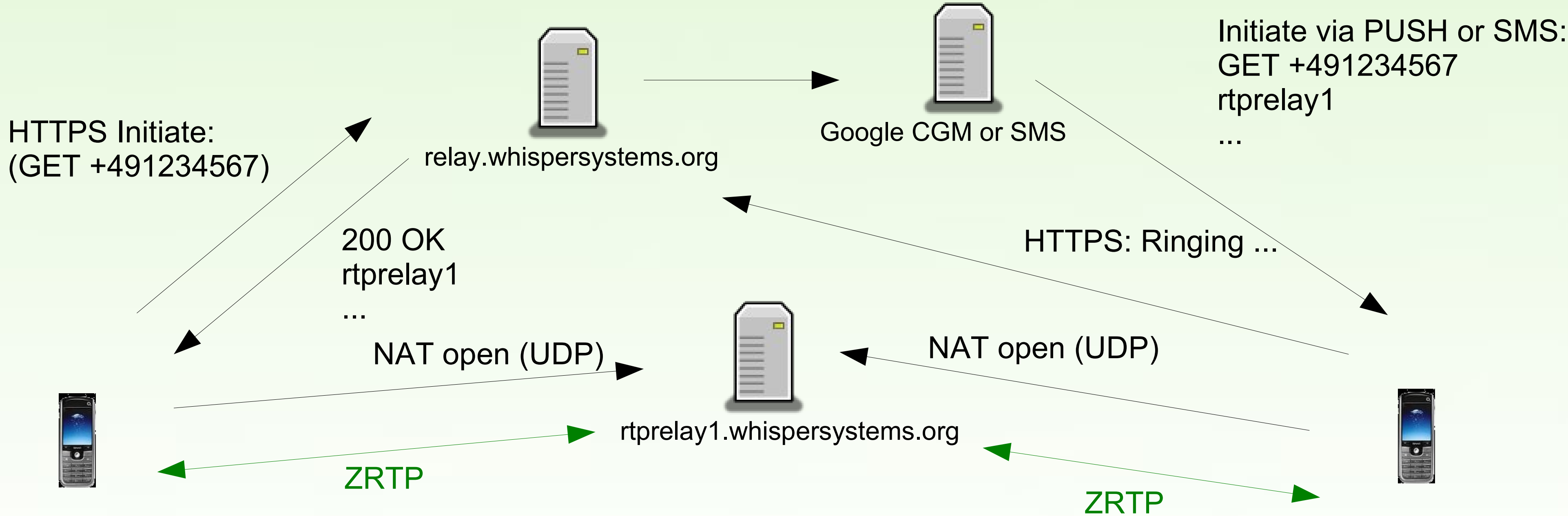
- I want to live in a *free society under rule of law*
  - Secret laws with secret courts are NOT rule of law
- Where people also contribute to *common good*
  - Not only to the interests of rich & powerful few

#MoreCrypto - WebRTC

- Widespread consumer use of encryption with DTLS-SRTP
- Great VoIP UA stack in browser and mobile
  - e.g. webrtc for android app anyone?
- FRAFOS ABC SBC
  - WebRTC-gateway (to vanilla-SIP)
  - TLS, SDES/SRTP & DTLS-SRTP, ICE in SEMS

#MoreCrypto - RedPhone

- Android VoIP app with ZRTP from Open Whisper Systems (makers of TextSecure)
- Elegant app, doesn't get in your way
- Signaling: HTTP-websocket-ish



#MoreCrypto - RedPhone-SIP-GW

- Based on SEMS, DSM, mod_httpd
- Challenges
  - Extend libmicrohttpd with websockets
  - Testing on real Android instead of simulator
  - Will have to implement codec (PT) negotiation

- WIP – need help!
  - Join OWS ML, join dev @github/sanchi/, PM

#MoreCrypto - #redecentralize

- Need to decentralize signaling (as in p2psip)
  - Each user her own DNS domain too complex
  - Location DB on P2P overlay (MaidSafe?)
- Distributed NAT handling (ICE, TURN)
  - Use friend's, or FOAF's server as turn server?
- Call hash(pubkey) instead of name/telnr
- Keys from namecoin, DNS, keyserver, webfinger, QR-code, NFC …
- Add to Freedombox, ArkOS?

Questions?

Thanks for your attention.

Links and References

- SEMS homepage: http://iptel.org/sems
- Code: sems repo at git.sip-router.org
- DSM documentation

  http://git.sip-router.org/cgi-bin/gitweb.cgi?p=sems;a=tree;f=doc/dsm
- FRAFOS website: www.frafos.com