Transcoding and Call Centers with OpenSIPS

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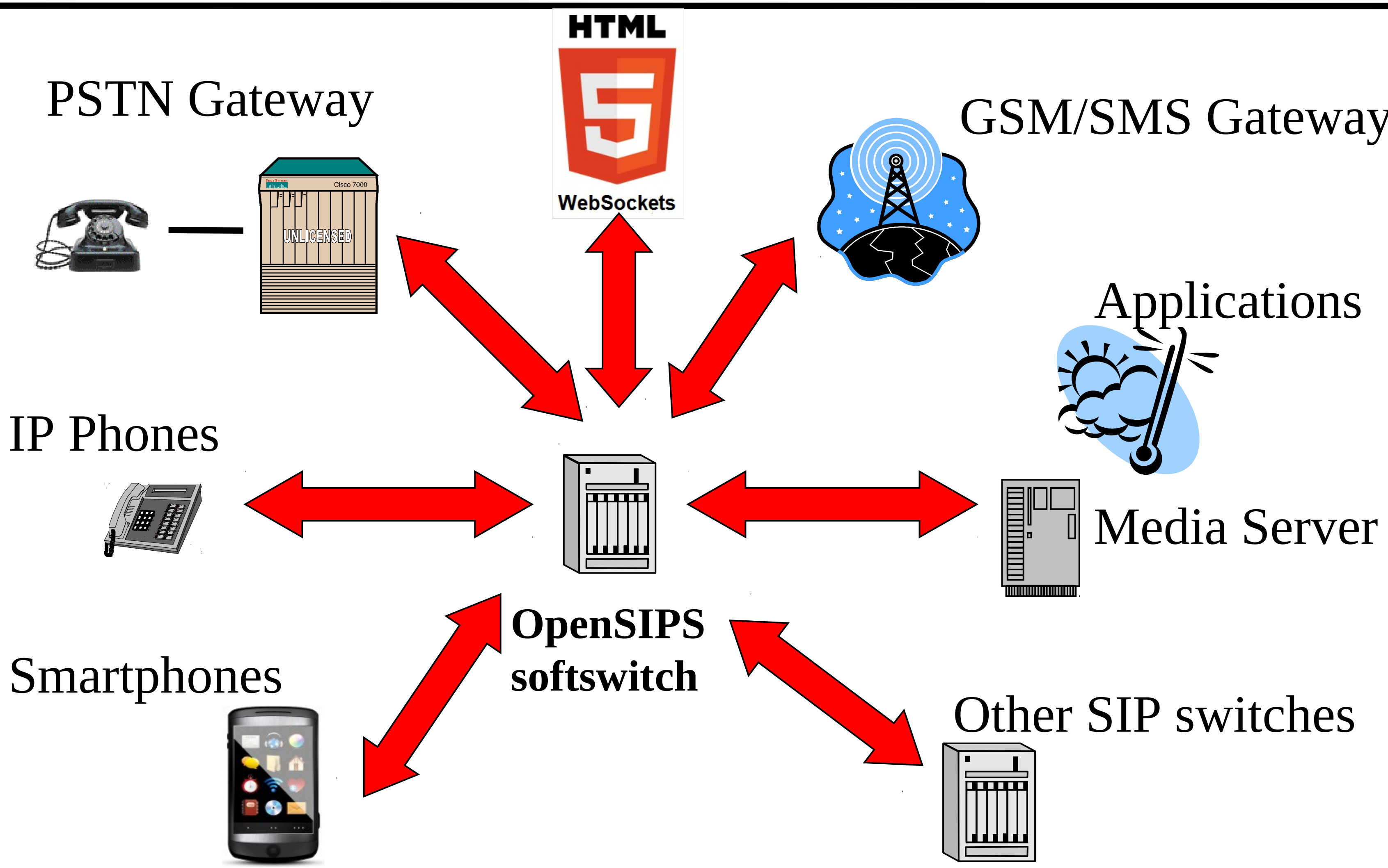
OpenSIPS Solutions

About OpenSIPS

What OpenSIPS is ?

- primarily a SIP proxy
- multi purpose proxy
- doing voice, video, presence, IM and other
- signaling only, no media

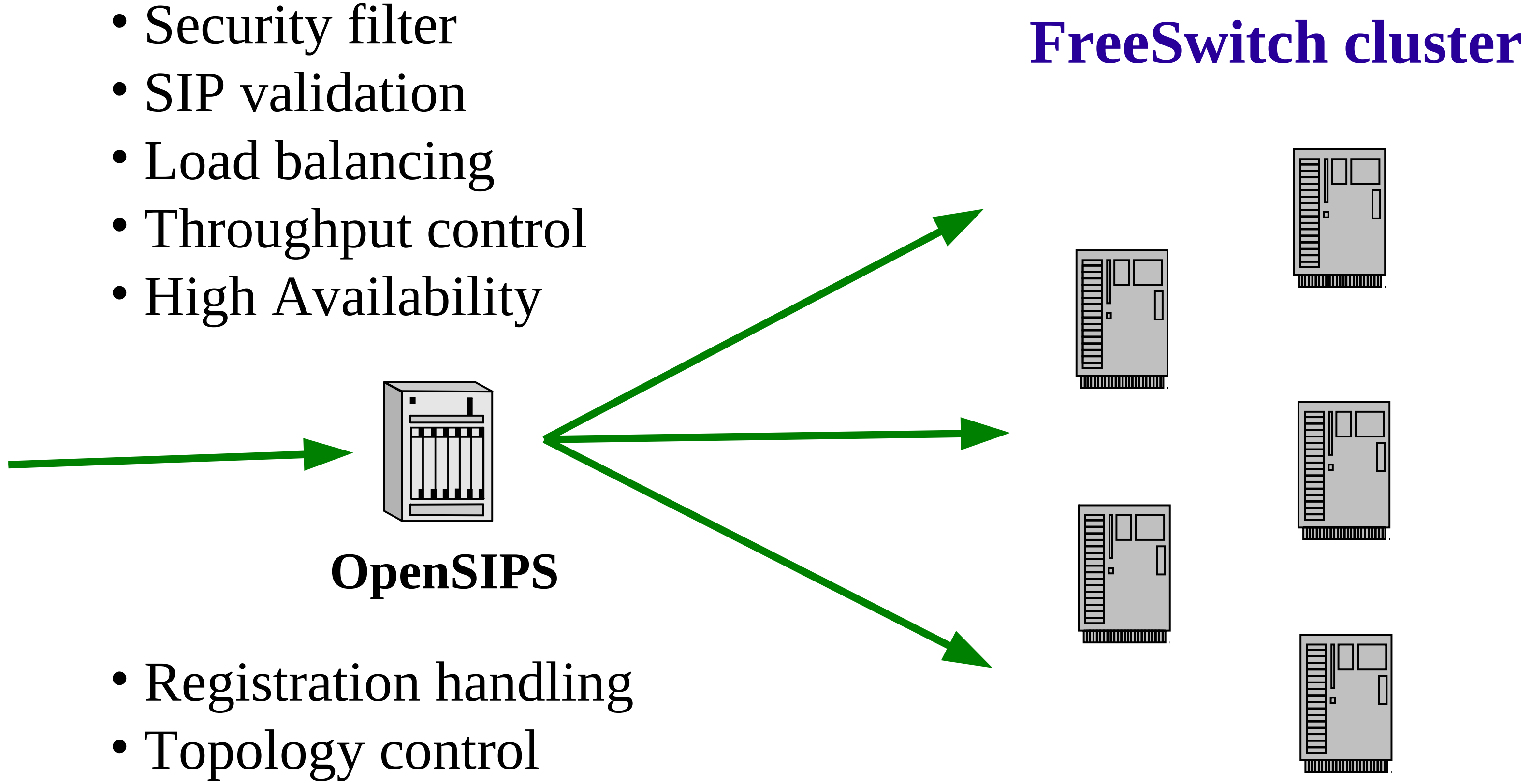

PSTN Gateway
Cisco 7000
UNLICENSED
HTML
5
WebSockets
GSM/SMS Gateway
Applications
IP Phones
Media Server
OpenSIPS
softswitch
Smartphones
Other SIP switches

Why OpenSIPS ?

- High throughput (calls, cps, registrations)
- Flexibility for routing and integration
- Effective application building (120 modules)

FreeSwitch FrontEnd

- Security filter
- SIP validation
- Load balancing
- Throughput control
- High Availability



- Registration handling
- Topology control
- Carrier interfacing

OpenSIPS 1.10

1.10 major release

- 1.10 beta released on 5th of August
- 1.10 it is not an LTS, 1.11 will be !
- 1.9 to still be maintained for ~ 6 months (until 1.11)
- As LTS, 1.8 still has ~ one year to go

Generic

- New MATH module (billing scenarios)
- Easy scripting with more script ops/functions; goes into combination with named flags, routes with parameters, script tracing
- Embedded REST client

TCP support

- Asynchronous read/write on TCP connections
- Optimized for large number of active connections
- Better detection of TCP based attacks (via blocking, fragmentation)

NoSQL support

- SQL to noSQL wrapping module
- Bulk removal on Local Cache

WebSockets

- Support for routing WS and WSS transport protocol
- SIP WEB client on opensips.org service

SCA support

- Shared Call Appearance support as defined by BroadWorks SIP Access Side Extensions Interface
- Integrate with dialog module to automatically generate the Call-Info data
- Self sufficient SCA implementation (generate and distribute data)

Dialog replication

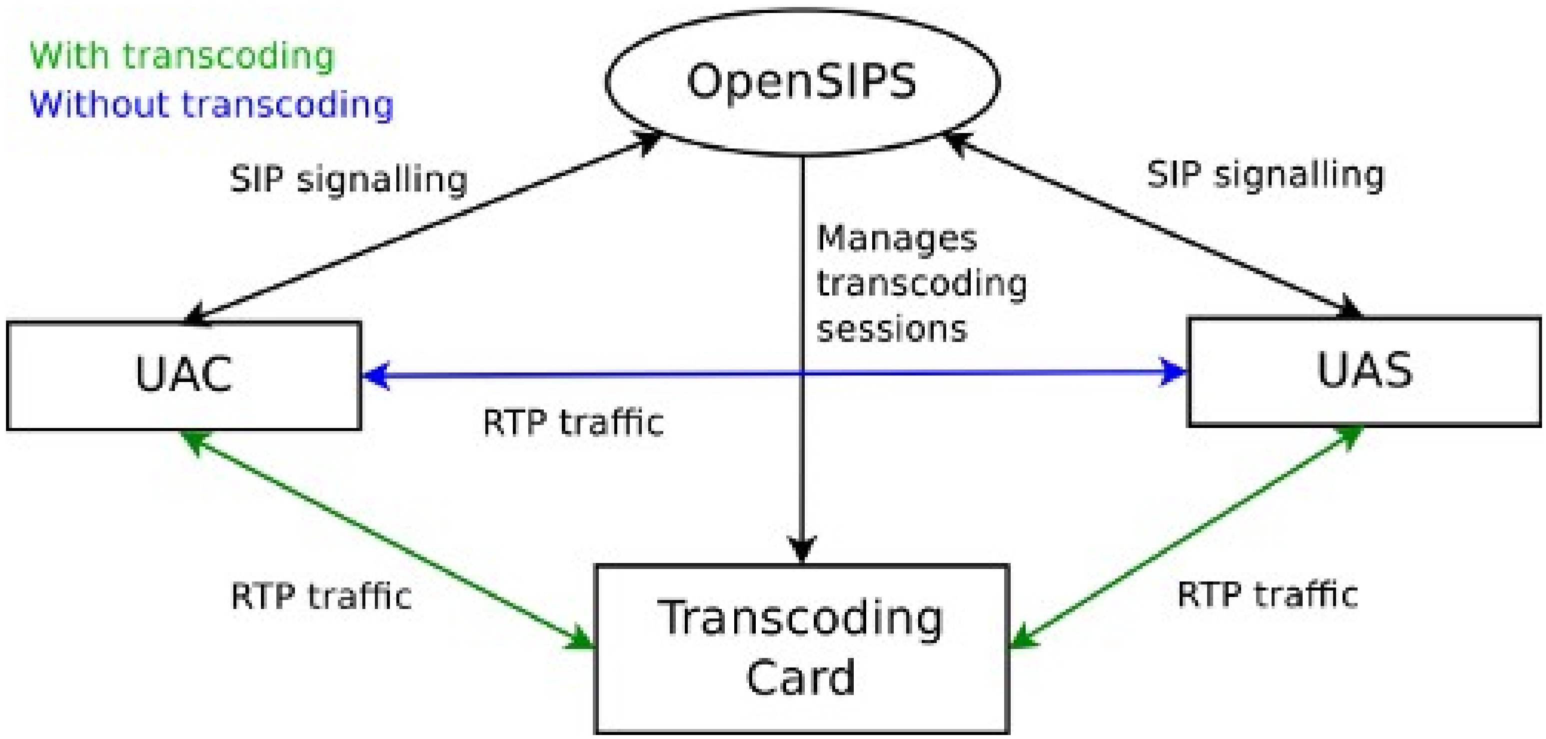
- New Binary INternal Interface (BIN) for inter OpenSIPS communication
- Dialog module replicates in realtime the dialogs and their state to other OpenSIPS instances
- Full dialog recovery on other OpenSIPS instances

Transcoding with OpenSIPS

Sangoma transcoding

- Using (remotely) Sangoma transcoding D cards
- Signaling stays in OpenSIPS, RTP on the cards
- Multiple cards controlled by same OpenSIPS instance
- Control over the codec selection too
- The transcoding is SIP transparently done !

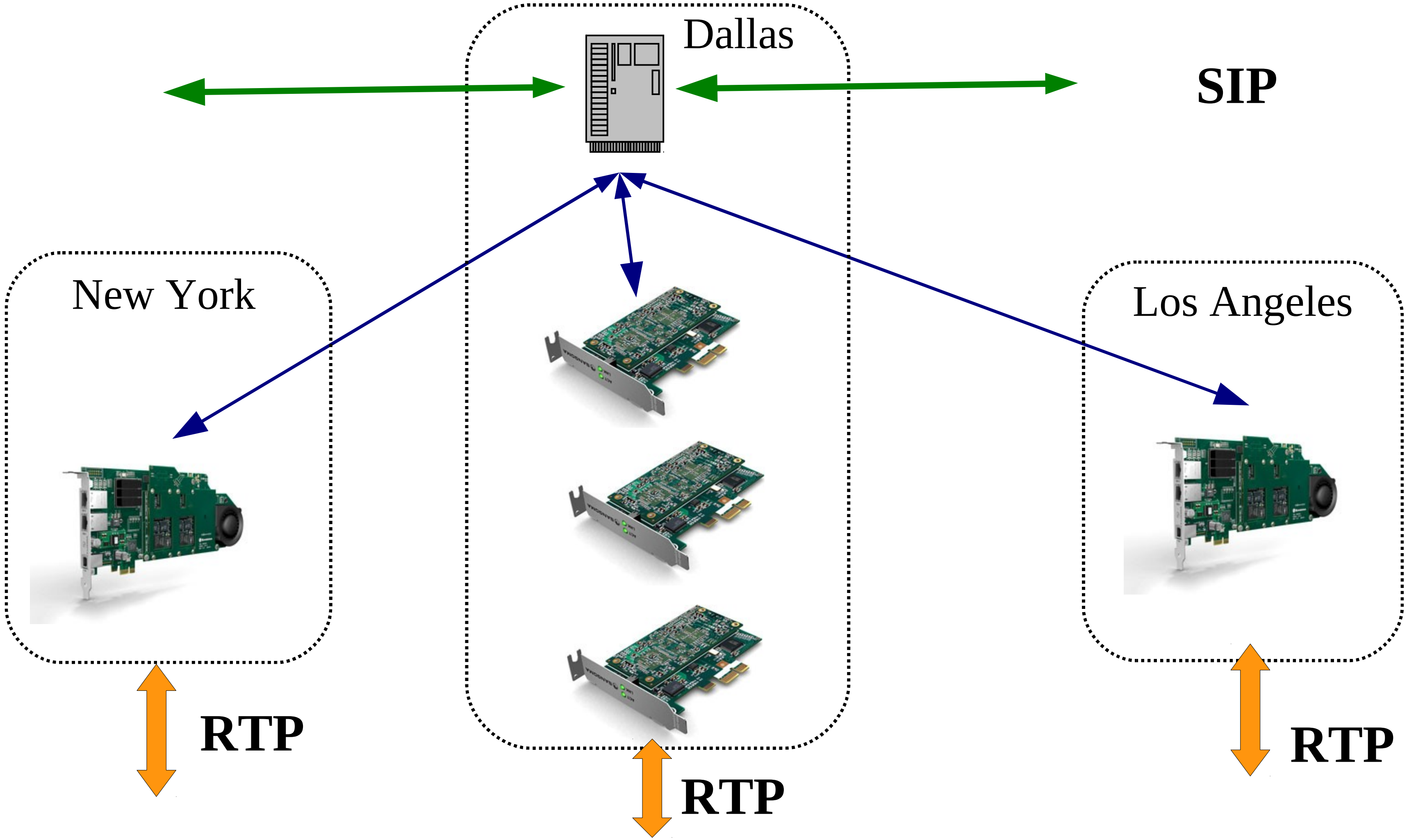
Sangoma Transcoding



Advantages

- Complete decoupling of SIP and RTP
- No need to add another SIP entity in your network
- Transcoding can scale and distribute with no change over the SIP signaling part
- Better control over transcoding as done directly from the routing logic

Sangoma Transcoding

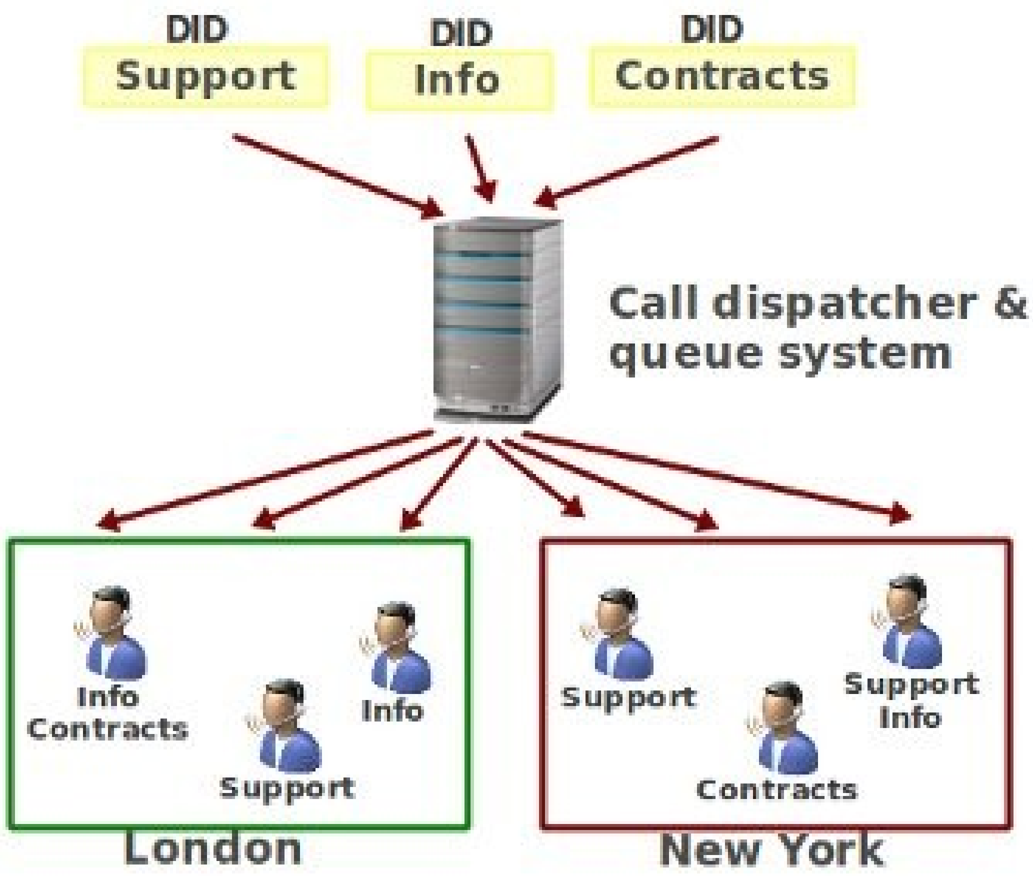


Call Queuing with OpenSIPS

Call Center

- Call queuing in OpenSIPS (signaling only)
- The media part is provided by a external Media Server
- Based on the B2BUA engine from OpenSIPS
- Inbound call center – multiple queues, sets of agents, skills, priorities
- Thousands of agents and queued calls

OpenSIPS Call Queuing



Inbound call center

- DID mapping to queues (via Dynamic Routing)
- Can be frontended by external IVRs for queue selection
- Auto login on agent SIP registration
- re-Routing of outbound calls from agents (redirects)
- Realtime statistics and reports

Outbound call center

- OpenSIPS can initiate calls (via B2BUA) as dialer
- Large amount of calls can be triggered and handled
- Use external Media Servers for audio support
- Accepted calls to be pushed in queues – sync the dialing rate with the load on the queues.
- No support for detecting Fax or VM via RTP

Thank you for your attention
You can find out more at www.opensips.org
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Questions are welcome