CERN

Towards a modernisation of CERN's telephony infrastructure

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- A glimpse of SIP
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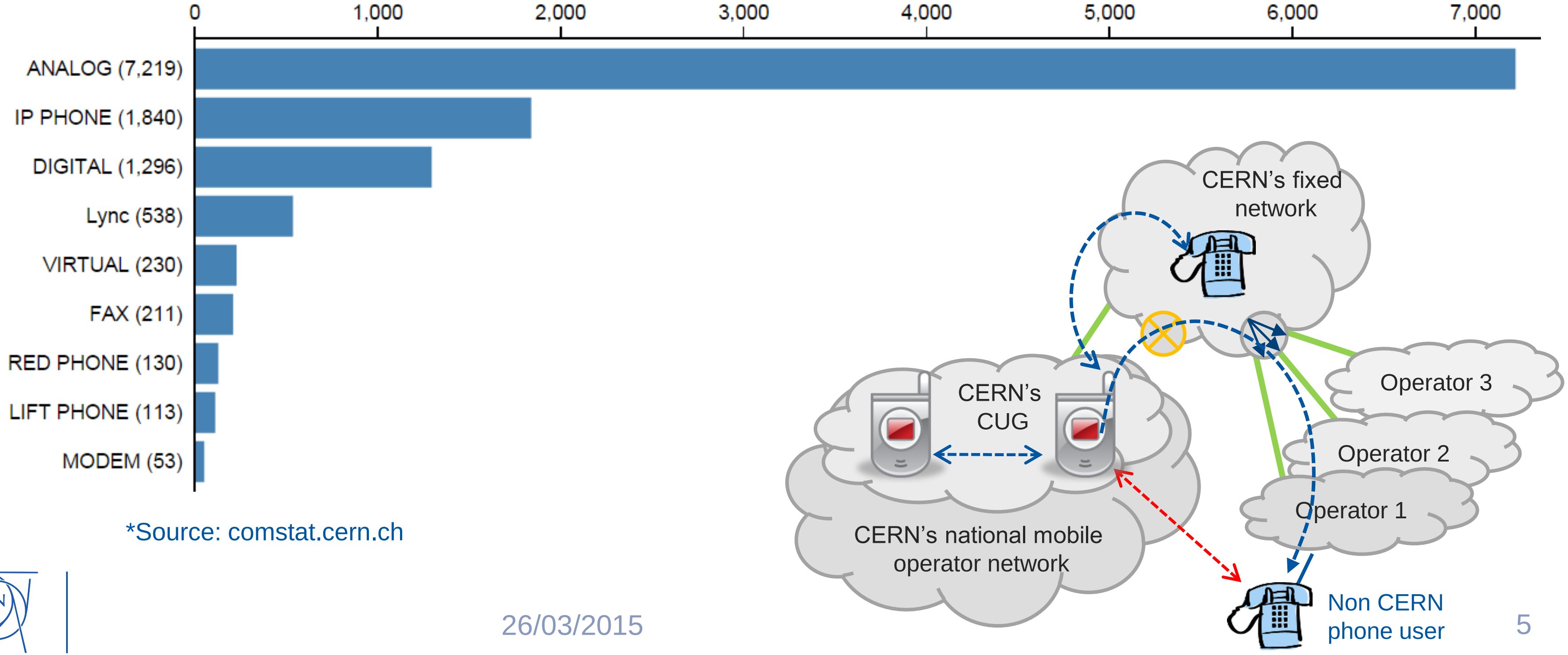
Today's network

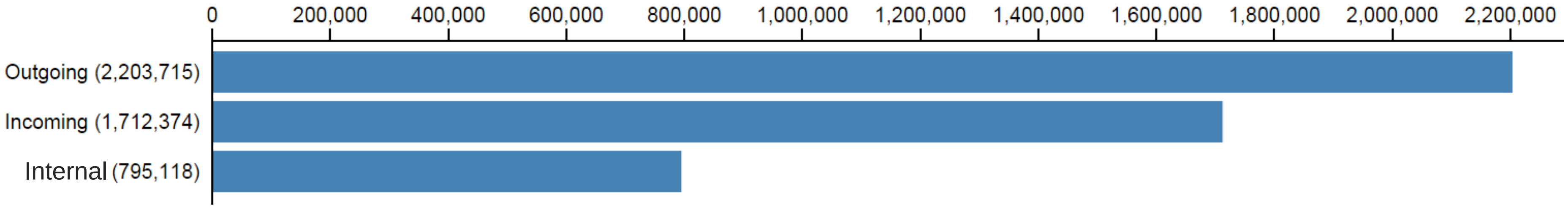
Manages 12K fixed lines + 6K mobile phones (Closed User Group)



*Source: comstat.cern.ch

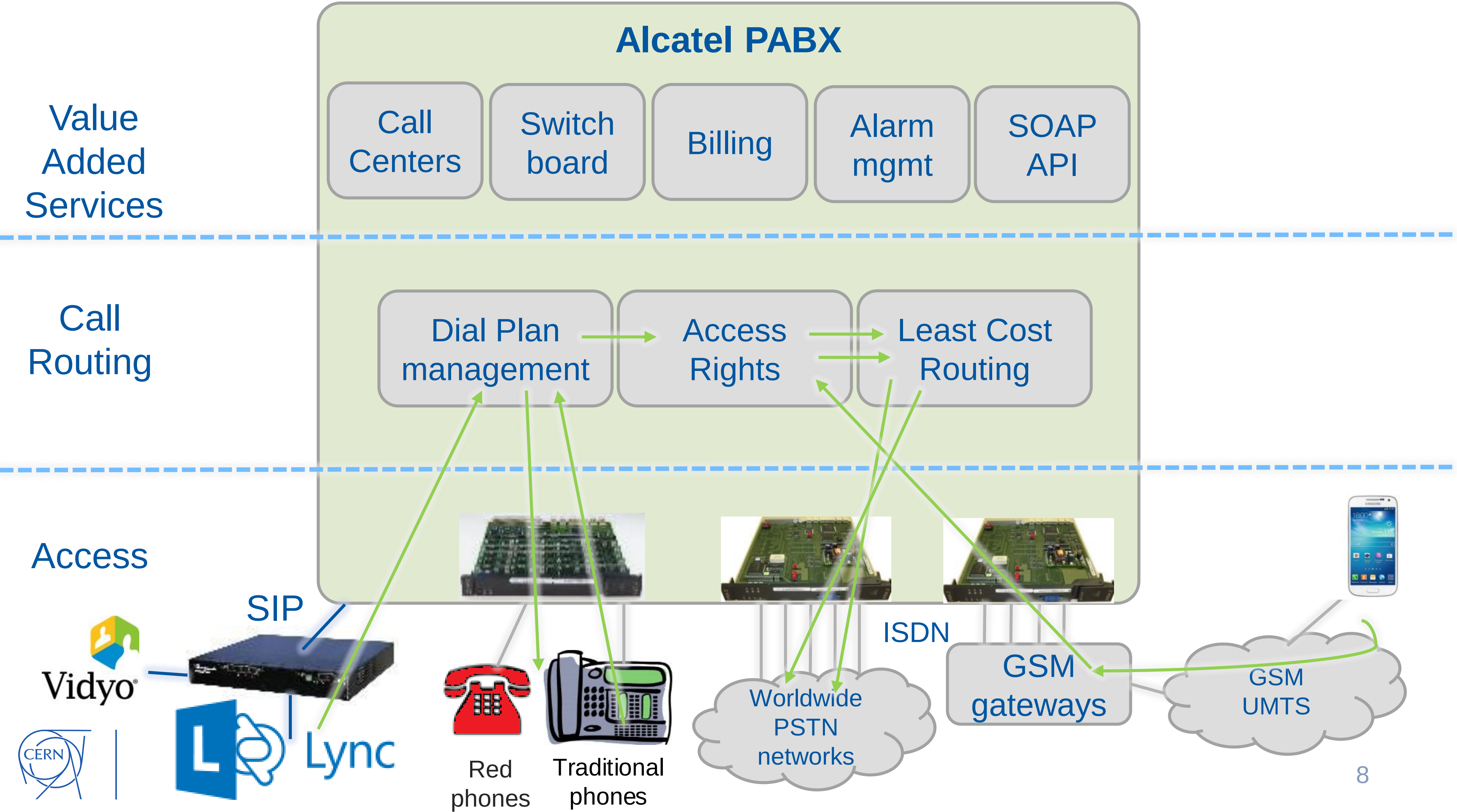
Today’s network

- Least Cost Routing (LCR) for outgoing calls
- Worldwide numbering plan with 800 destinations
- Local extensions with different external access rights
- Around 5M calls/year



Today's network

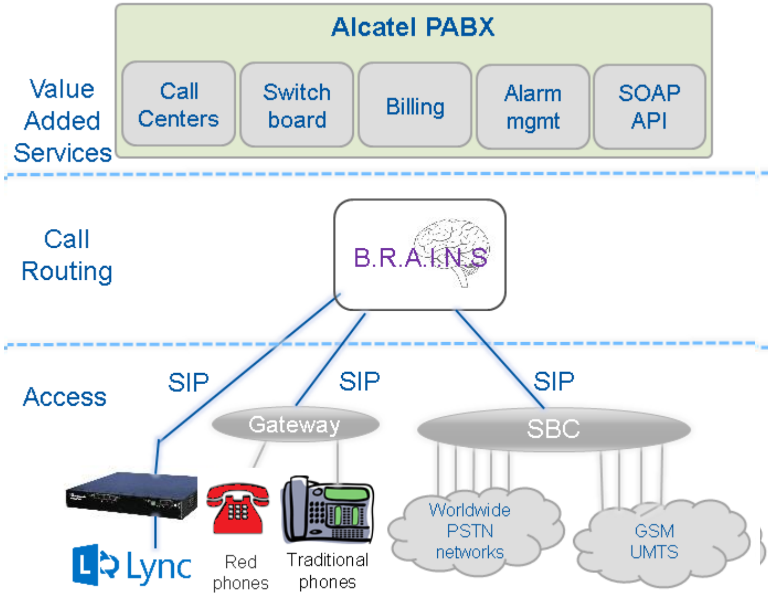
- Critical and safety services
  - Switchboard
  - Call Centers: Fire Brigade, CCC, Service Desk, IT Helpdesk
  - TETRA interconnection
  - Special analog lines: Red Phones & Lift Phones
- Integration with Mobile Telephony – CUG
- Integration with Unified Communications – Lync
- Integration with conferencing systems - Vidyo



Alcatel PABX
Value Added Services
Call Centers
Switch board
Billing
Alarm mgmt
SOAP API
Call Routing
Dial Plan management
Access Rights
Least Cost Routing
Access
SIP
Vidyo
Lync
CERN
Red phones
Traditional phones
ISDN
Worldwide PSTN networks
GSM gateways
GSM UMTS

Project goals

- Replace the PBX by a software-based solution
  - Hardware/license costs
  - Avoid vendor lock-in
  - TETRA for critical communications
- Decouple call routing function to a new entity
- Capability to support non-Lync softphones
  - Today Lync is the only option for office phones
- Use SIP trunking with the external operators
- SIP core Network
  - Using open-source solutions
  - Fosters the introduction of new VoIP services



A glimpse of SIP

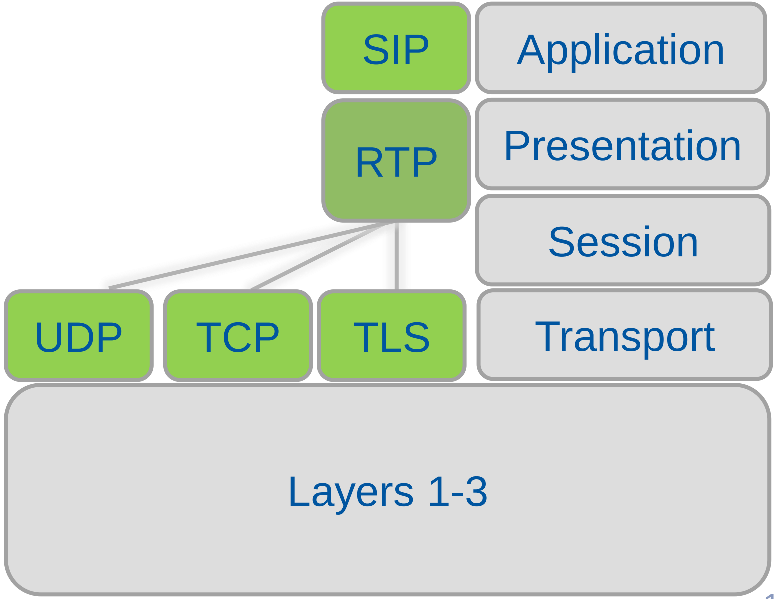
SESSION INITIATION PROTOCOL

Defined in 1996. RFC2543 in 1999 replaced by SIP v2 in 2002 (RFC3261)

Uses the HTTP request/response model.

- Headers
- Status codes
- Dialog vs. Transaction
- SIP URIs

sip:username@host:port



A glimpse of SIP

Voice packetization is the key enabler of VoIP.

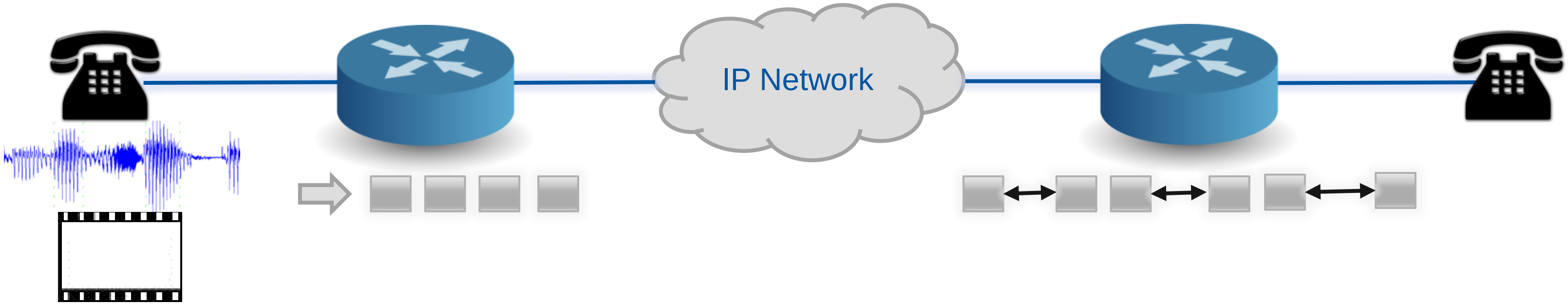
Time Division Multiplexing

- End-to-end physical channel reserved
- Constant bandwidth
- Fixed number of channels 1 E1 = 30 channels
- High infrastructure costs

Manual Switching → Mechanical Switching → Automatic PBX

A glimpse of SIP

IP packetization and independent routing

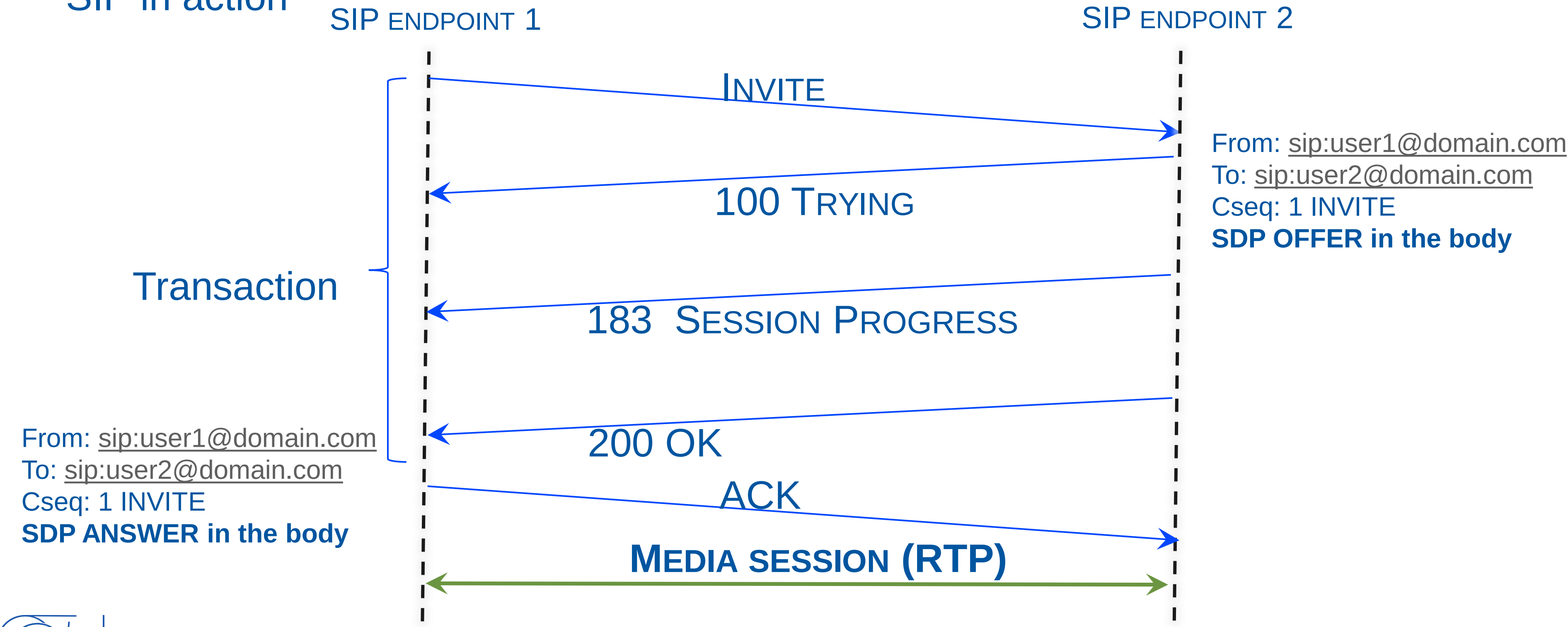


- Fewer infrastructure costs
- Packet loss and/or variable delay or ***Jitter***
- Changing packetization time and codec may help
- QoS mechanisms needed

- Softphones and Softswitches – but with specific hardware for media handling

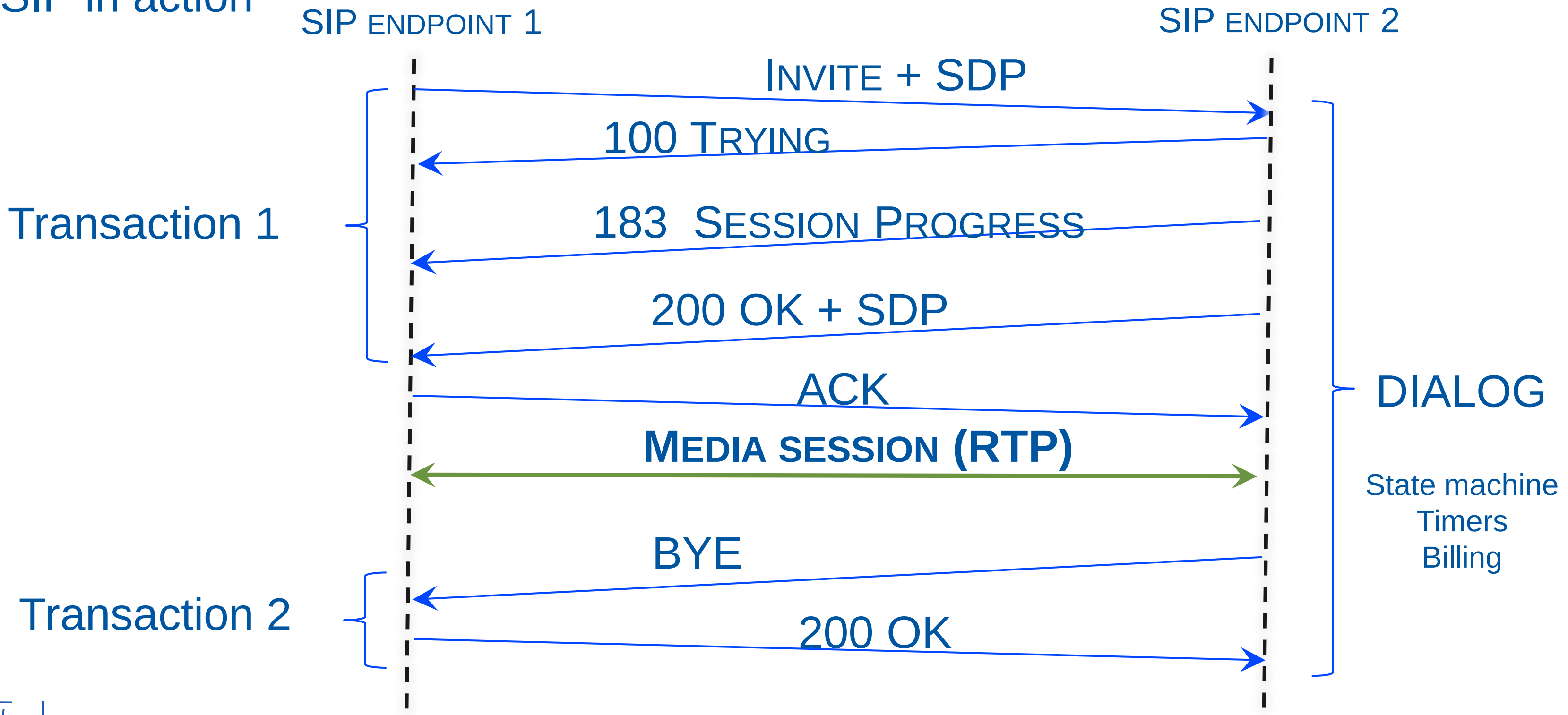
A glimpse of SIP

SIP in action



A glimpse of SIP

SIP in action

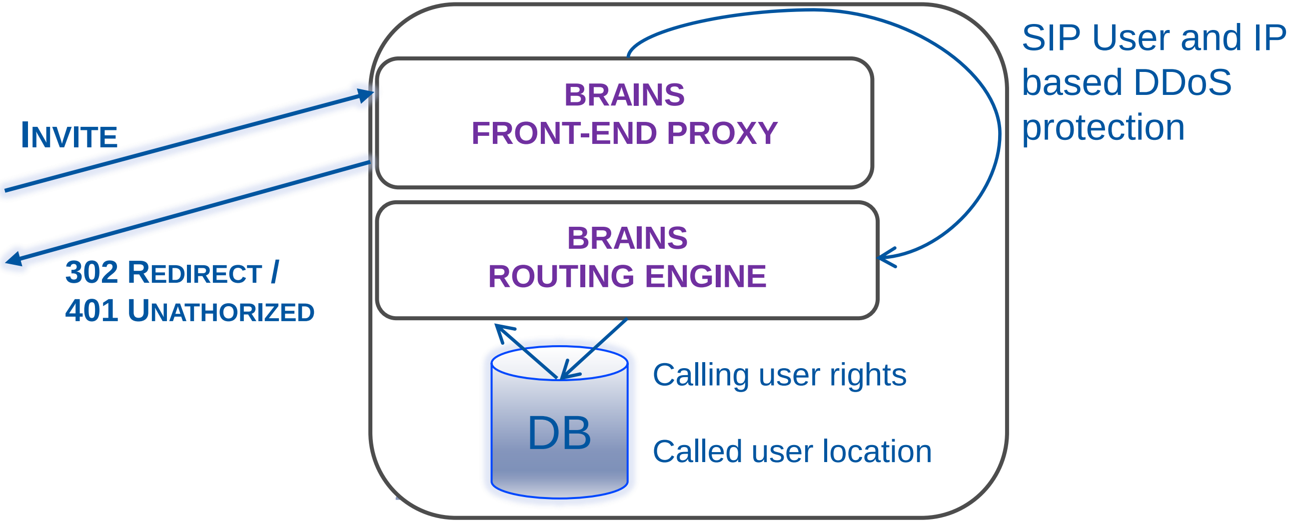


BRAINS

Boîte pour **R**eduire l'**A**lcatel-PBX et **I**ntroduire des **N**ouveaux **S**ervices

Box for **R**educing the **A**lcatel-PBX and **I**ntroducing **N**ew **S**ervices

- Center of all routing decissions for all real-time media sessions.
- SIP transaction-aware redirect server



BRAINS

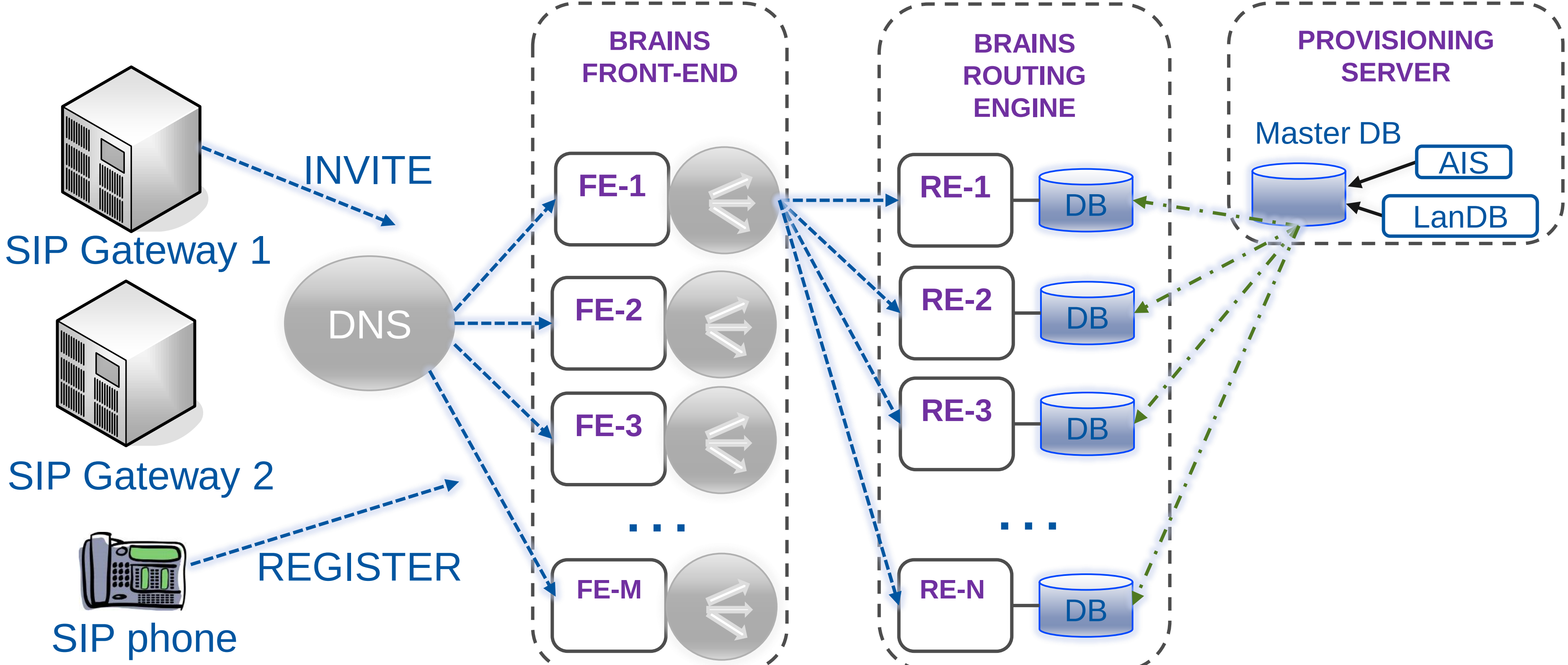
Open-source alternatives

- Call Routing Engine: *Asterisk (PJSIP) vs FreeSwitch (Sofia SIP)*
- Front End (Proxy): *Kamailio(OpenSER) vs OpenSIPs*

System architecture:

- Cluster of CentOS 7 machines (OpenStack + Puppet)
- Front end cluster reachable by incoming SIP trunks
- Routing engine with local cached database
- Provisioning and monitoring servers
- DNS load balancing + SIP Options

BRAINS



SIP Gateway 1
INVITE
SIP Gateway 2
DNS
SIP phone
REGISTER
BRAINS FRONT-END
FE-1
FE-2
FE-3
...
FE-M
BRAINS ROUTING ENGINE
RE-1
DB
RE-2
DB
RE-3
DB
...
RE-N
DB
PROVISIONING SERVER
Master DB
AIS
LanDB

Summary

Progress:

- Technology review + shortlist of open source solutions
- Concept validation
- Architecture and roadmap proposal

Next steps:

- SIP trunking with external operators
- BRAINS Beta service: Q3 2015
- Key issues to be addressed:
  - Evolution of Value Added Services
  - Solution for special analog lines

Thank you!

Questions?

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