# Using OpenSIPS as a PBX

Lessons Learned

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*“Anyone who has never made a mistake has never tried anything new.”*

*ALBERT EINSTEIN*

- Telephony and VoIP solution provider
  - Started in 2007 in Brazil
  - Focused in the ISP/ITSP Market
- More than 200 customers for
  - Wholesale
  - Residential
  - PBX/Hosted PBX
  - LNP Redirect Services
  - Anti Fraud Services

# Is OpenSIPS suitable as a PBX?

- No media capabilities
- No built-in PBX features
- Tremendous effort to implement even simple features

# Why to use OpenSIPS?

- SIP Compliance
  - SIP Phones and Gateways expect to have a proxy in the middle.
  - Simple (Presence and Instant Messaging)
  - Easier to implement advanced SIP features
  - Built in security TLS
  - Codec Agnostic
  - No Audio interference, no extra latency or jitter
- Scalability
  - 50.000 or more users per server
  - Multi-domain
  - Easily scalable to  thousands of calls per second
- Excellent Platform for Centralized Hosted PBX

# Case Study: CMA

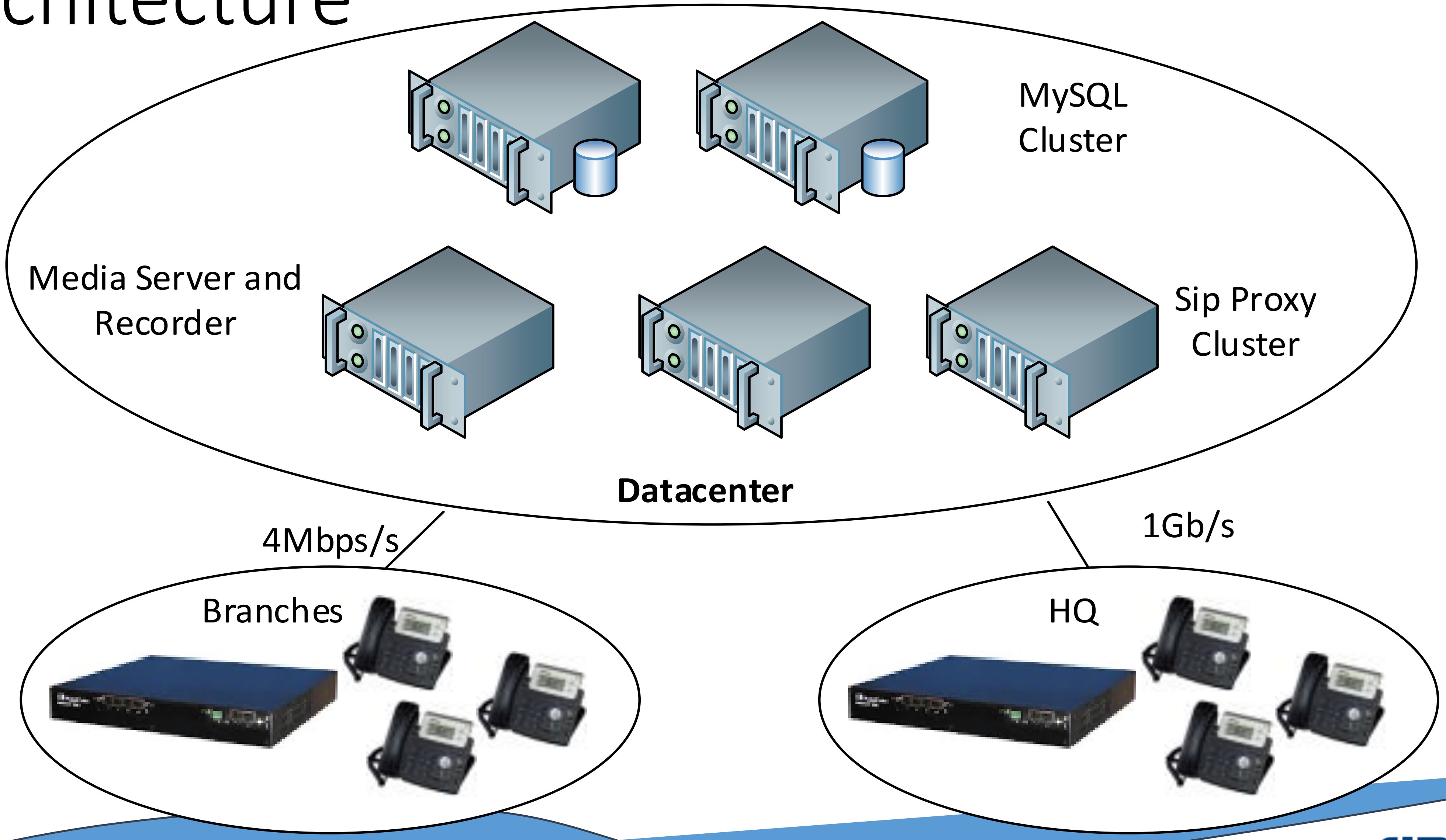
- Internet Provider for the Financial Sector
- Units in São Paulo, Porto Alegre, Fortaleza, Rio de Janeiro, Argentina, Europe, US, Uruguai, Chile e Colombia.
- Deployment:
  - Phase 1: Deploy Internally
  - Phase 2: Deploy to customers in different domains

# Choice Criteria

- Possibility to provide PBX services to clientes
- Possibility to provide Recording and Secure Communications
- Multidomain (1 per client)
- Embedded Billing (Sale and Cost)
- Scalability
- Fault Tolerance
- Security

SIP Platform

# Architecture



# Challenge #1 Documentation

- Where are the documentation to build a PBX using OpenSIPS?

The Guide

RFC5359 - SIP Services

# Challenge #2 - Redundancy

- We've chosen to work with RFC3263, DNS redundancy
  - Implemented in Yealink Phones
  - Implemented in AudioCodes Gateways

| | order | pref | flags | service | regexp | replacement |
|---|---|---|---|---|---|---|
| IN NAPTR | 50 | 50 | "s" | "SIPS+D2T" | "" | _sips._tcp.example.com |
| IN NAPTR | 90 | 50 | "s" | "SIP+D2T" | "" | _sip._tcp.example.com |
| IN NAPTR | 100 | 50 | "s" | "SIP+D2U" | "" | _sip._udp.example.com |

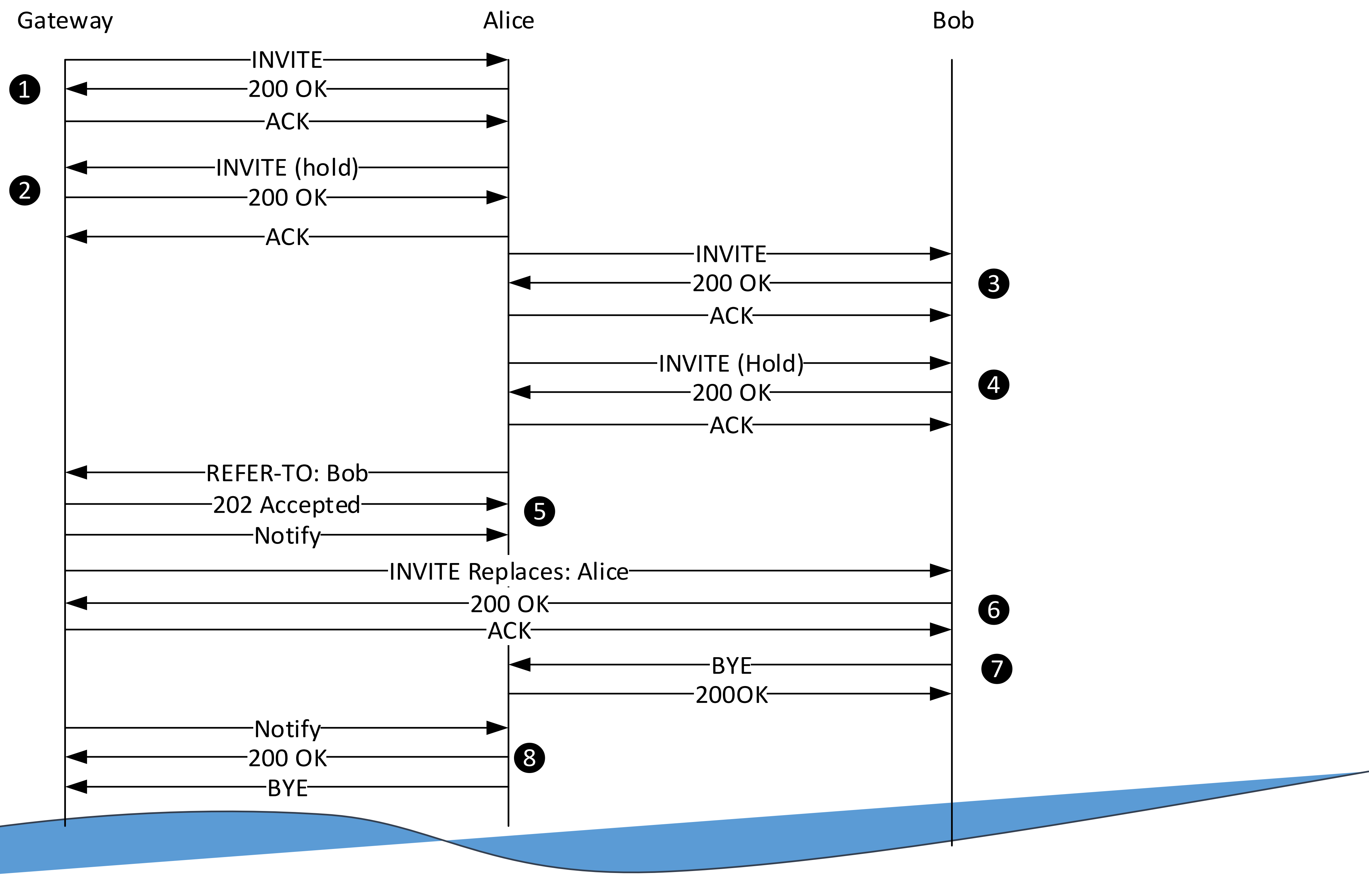
| | Priority | Weight | Port | Target |
|---|---|---|---|---|
| IN SRV | 0 | 1 | 5060 | server1.example.com |
| IN SRV | 0 | 2 | 5060 | server1.example.com |

# Lessons Learned

- Fill all the records, NAPTR and SRV. The lack of NAPTR record increases the Post Dial Delay.
- Keep the second server running.
- It works fine where no NAT is involved.
- At the end, works like a charm.

# Challenge #3 Implement Call Transfer

- OpenSIPS by itself does not handle transfers
- OpenSIPS can route REFERs and RE-INVITEs
- We've chosen to implement Call Transfer as indicated at RFC5359


Gateway
Alice
Bob
INVITE
200 OK
ACK
INVITE (hold)
200 OK
ACK
INVITE
200 OK
ACK
INVITE (Hold)
200 OK
ACK
REFER-TO: Bob
202 Accepted
Notify
INVITE Replaces: Alice
200 OK
ACK
BYE
200OK
Notify
200 OK
BYE
1
2
3
4
5
6
7
8

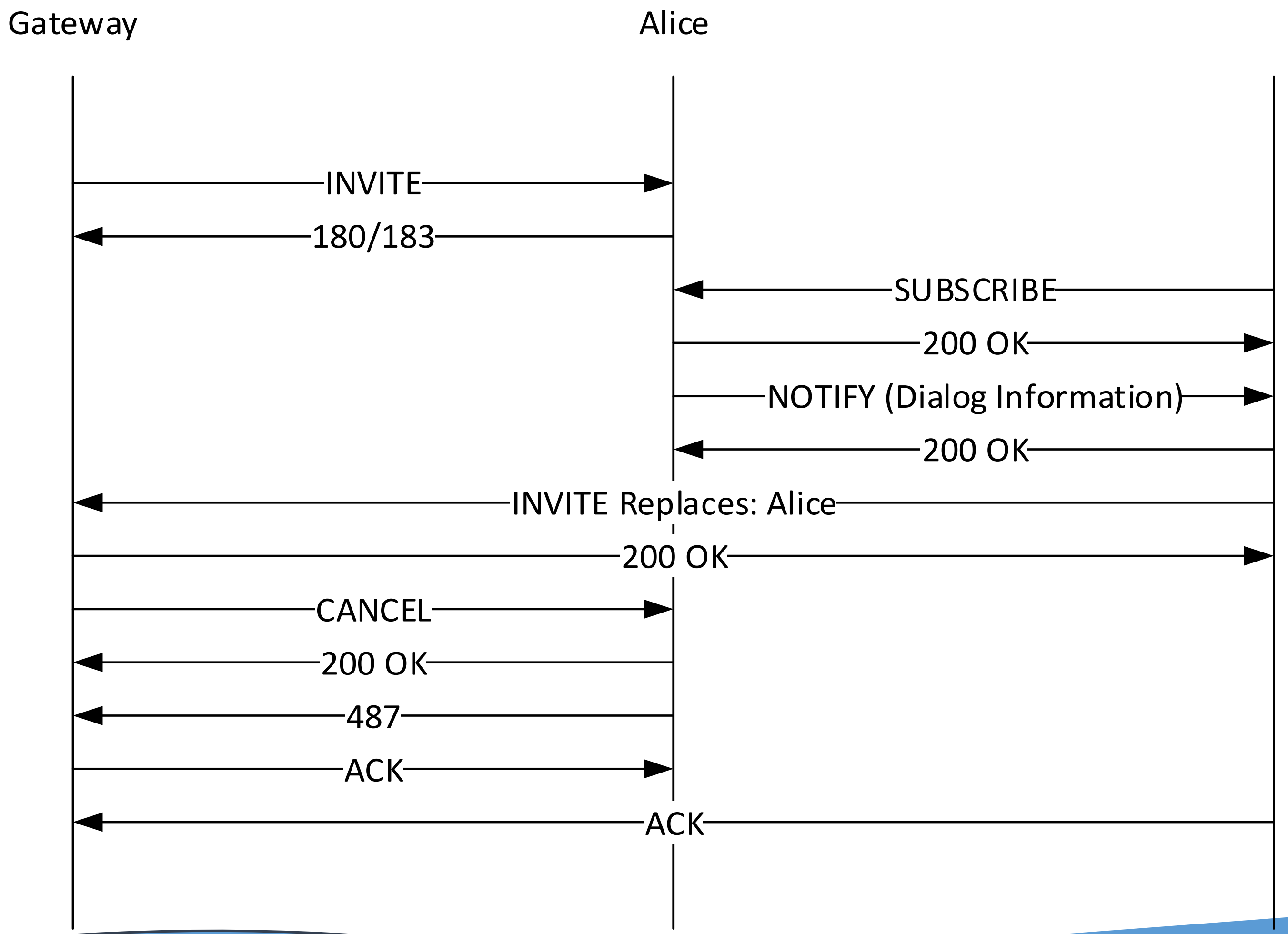
# Lessons Learned

- Configure gateways properly for transfers
  - Return Routes.
  - Refers are handled in the same way as Invites
- You need to take care of the CDRs and Authorizations
  - If the call started with a privileged user, when transferred, the authorization of the referrer should be used instead of the referee
- Semi-Attended-Transfers lead to Ghost Calls in Voicemail

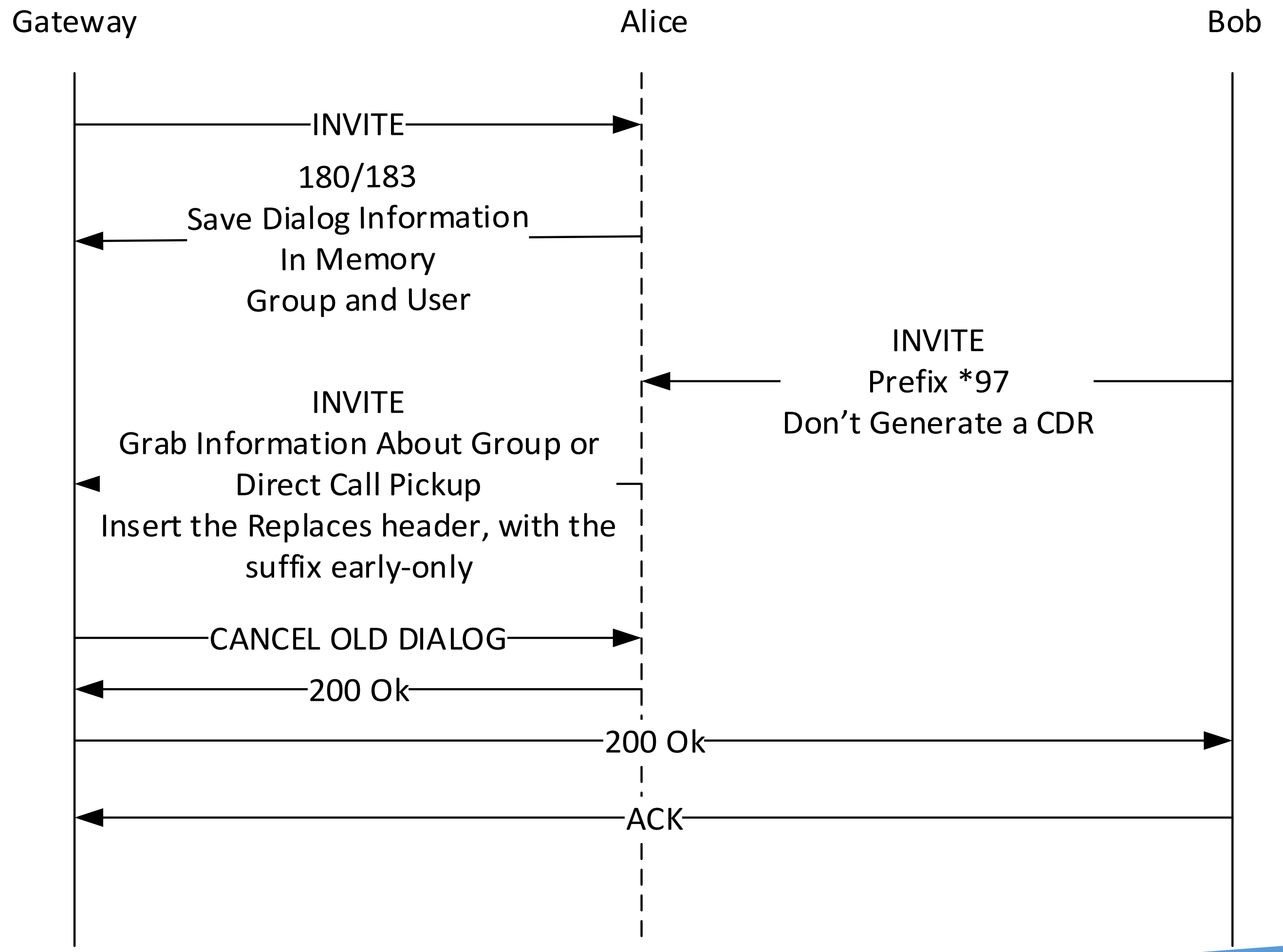
# Challenge #4 – Call Pickup/Group Call Pickup

- Call Pickup is defined in the same RFC5359
- Strictly dependent on the Phone Support
- Endpoints and Gateways need to support Replaces.
- We use a creative way to support non-compliant phones.
- We had to fix CDRs
- We had to fix Authorizations

# Standard way to Pickup a Call



# Creative way to support unsupported phones



# Lessons Learned

- Firmware Updates
  - Make sure to use the latest firmware
  - Make sure that you and your users know how to use the phone
- Sometimes, it is easier to use a FreeSwitch as an SBC to normalize everything before the gateway.

# Challenge #5

- Media Related Services
  - IVR
  - Conference
  - Queueing
  - Voicemail
  - Messages and TTS
  - Multi-Tenant Parking
- Asterisk was the tool of choice

  #1 Criteria: Voicemail Internationalization

  #2 Criteria: Queuemetrics for the Queues

SIP Platform

# Lessons Learned

- TTS is very handy when dealing with messages
  - Google TTS works like a charm.
- IVR call flows and combinations are a nightmare
- Things start to get really complex here.

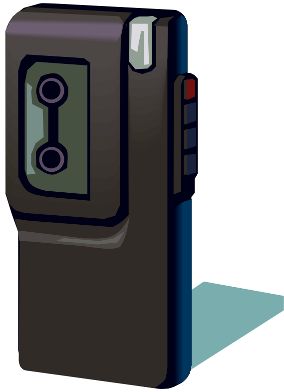
# Inbound Routing Services

- DIDs
- Huntgroups
- Time Routing
- All implemented on OpenSIPS

# Lessons Learned

- Combinations will kill your time and nerves with testing and fixing
  - IVR with Queues, Call Forward, Transfer and Redirect
  - DID, Time Routing, Hunt Group, IVR, combined with Call Forwarding all together is mental endurance challenge.
- Combinations with phone features such as Call Forwarding implemented on the phone (Redirect) also create challenges for the implementation.

# Recorder



- RTPPROXY is the tool
  - We've spent a few months to create an app to decode the recordings with quality
  - RTPBREAK is not the right tool, fix it or don't waste your time.
  - Use MP3 to save space
- Easily Integrated with the phone for on-demand recording

# Final Result

# Conclusions

- Still learning and fixing things.
- Single product for wholesale, residential and hosted PBX.
- All features are implemented in a standard way.
  - Less rework and better interoperability
- Next in Pipeline.
  - Hot Desking, Call Completion, VideoConference, WebRTC, Distinctive Ringing
- The final product is a marriage between Phone, Gateway and Proxy.

# OpenSIPS Panel

- Easisness to code
  - What can be improved?
- OpenSER Migration
  - Does it make sense to provide migration services/utilities?
- Training
  - Advanced Training Modules (Asynchronous Online)
  - Developer Training
  - eBootcamp/Bootcamp
  - University Program
- Bug Tracking/Reporting/Github
- Monitoring/Troubleshooting
- Open to the Audience
  - General Suggestions