Intro to WebRTC and VozDigital

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whoami

- From Bilbao
- Enjoying Tuenti since 2010
- Member of Voice Team
- @ivmos
- @TuentiEng


WebRTC
VOZDIGITAL

WebRTC

- May 2011
- “To enable rich, high quality, RTC applications to be developed in the browser via simple JavaScript APIs and HTML5.”

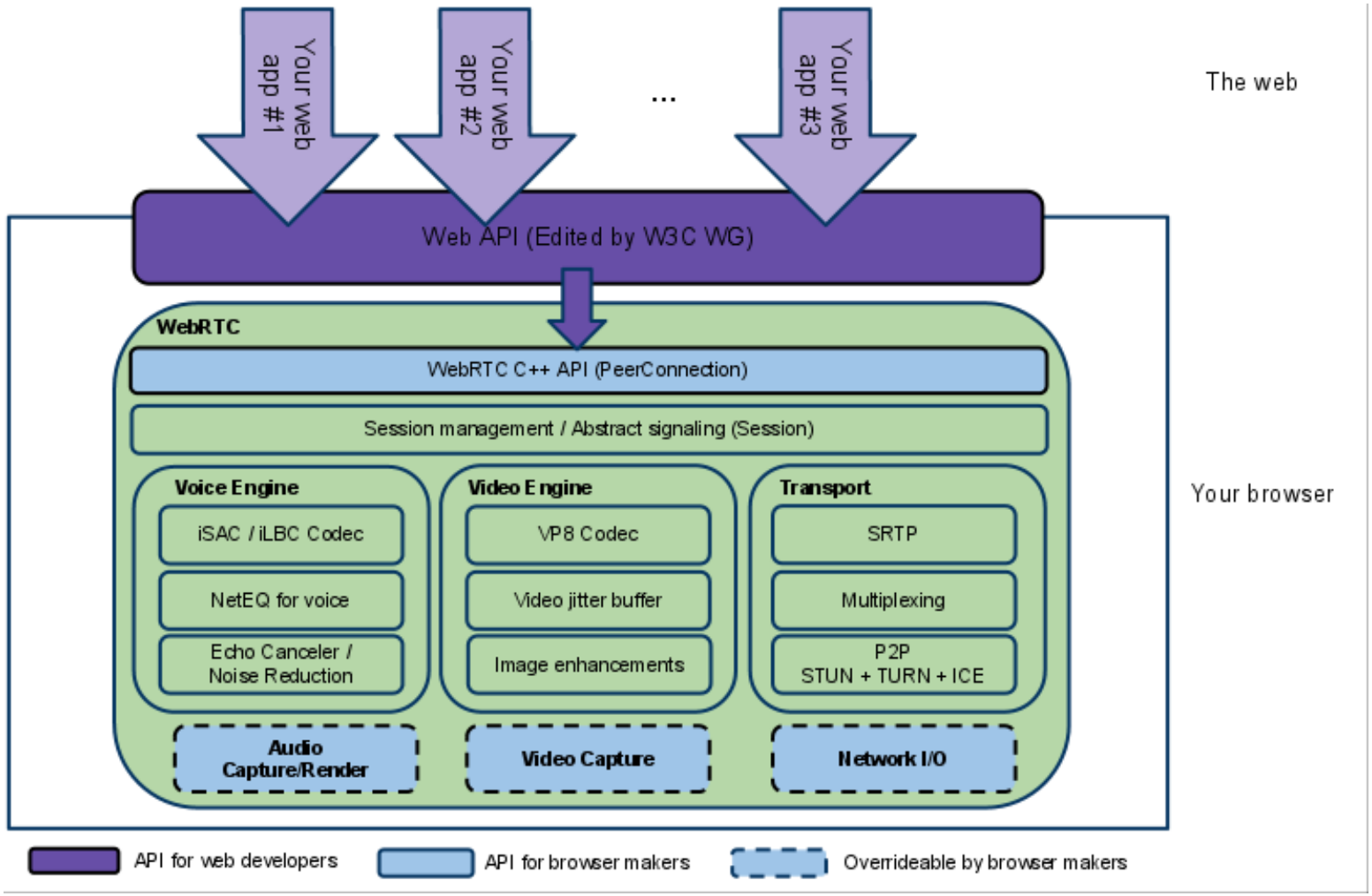
- Plugin-free
- Javascript API
- Native API available too

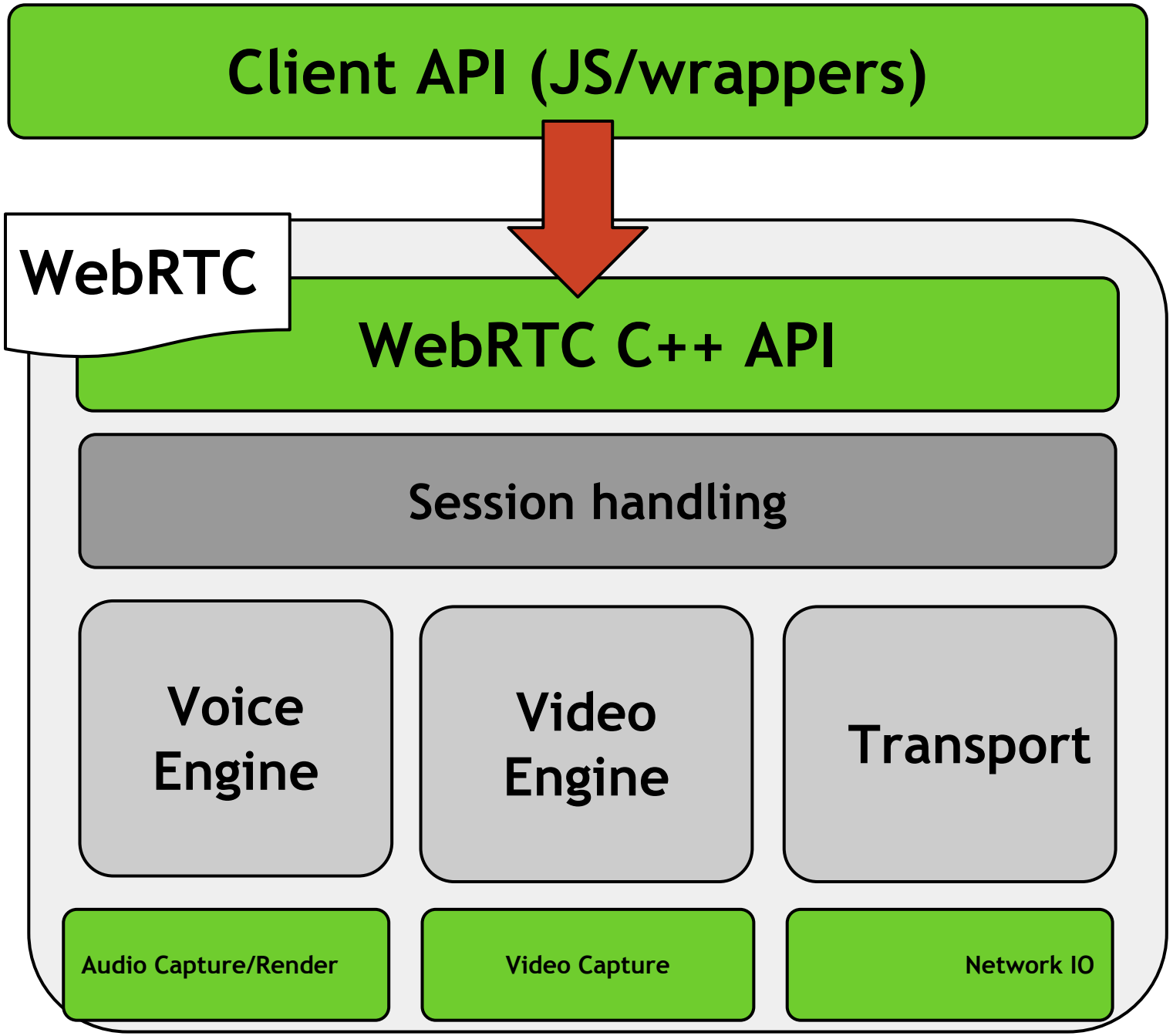
Before WebRTC?

- Plugins: flash, java, native addons..
- Codecs..

Let´s go!

- We need some theory.
- APIs:
  - MediaStream/getUserMedia
  - RTC PeerConnection
  - RTC Data channel (remember, not only a/v!)


Your web app #1
Your web app #2
...
Your web app #3
The web
Web API (Edited by W3C WG)
WebRTC
WebRTC C++ API (PeerConnection)
Session management / Abstract signaling (Session)
Voice Engine
iSAC / iLBC Codec
NetEQ for voice
Echo Canceler / Noise Reduction
Video Engine
VP8 Codec
Video jitter buffer
Image enhancements
Transport
SRTP
Multiplexing
P2P STUN + TURN + ICE
Audio Capture/Render
Video Capture
Network I/O
Your browser
API for web developers
API for browser makers
Overrideable by browser makers


Client API (JS/wrappers)
WebRTC
WebRTC C++ API
Session handling
Voice Engine
Video Engine
Transport
Audio Capture/Render
Video Capture
Network IO

What WebRTC does not provide...

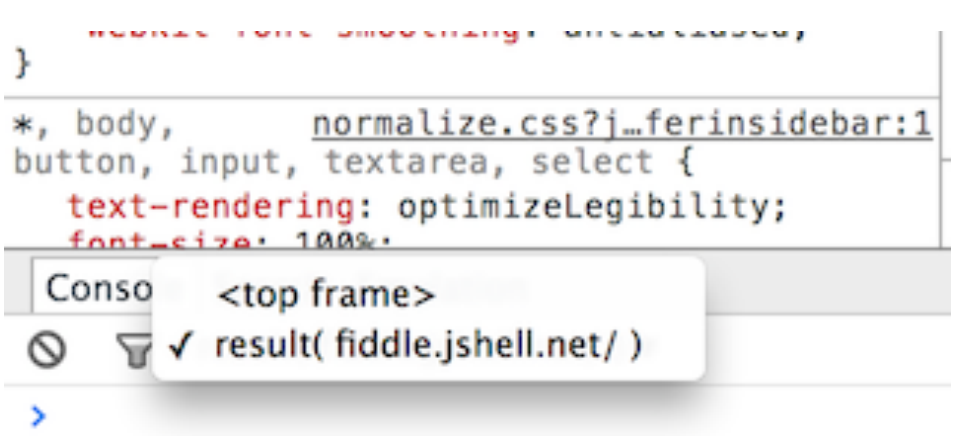
Signaling!!!

- WebRTC solves the media transport problem, not the high level signaling one.
- NAT traversal problem: ICE

Workshop hints

- https://gist.github.com/ivmos/HASH
- http://jsfiddle.net/gh/gist/lib/V/HASH

Open Developer Tools and bring focus to the console	Ctrl + Shift + J	Cmd + Opt + J



GO GO GO: let´s get media!

http://jsfiddle.net/gh/gist/library/pure/7fb79f59ab1f2a36ff27/
Action: check console

- Audio&Video source with MediaStream API.
- Chrome snippets (interop) (polyfill)
- navigator.webkitGetUserMedia

let’s hype a bit

- You can mix with other Javascript APIs
  - ASCII camera
  - Photobooth app

- We´re not studying it but RtcDataChannel API is awesome too
  - http://www.peer-server.com/
  - https://rtccopy.com/
  - https://github.com/feross/webtorrent

We have the video, let´s offer it

http://jsfiddle.net/gh/gist/library/pure/8ce6340f8870f6d91f0a/2b1596f3124037dedf7ea443413c8a0352599606/

Action: check offer SDP

- **pc:** RTCPeerConnection
- Once we have the video, with the **pc** we need to **offer** it with a **description**
- pc.**addStream**
- pc.**createOffer**

RTCSessionDescription

- **pc.setLocalDescription (offer)**
- RTCSessionDescription
- offer, answer..
- **sdp** **(1998, RTP, RTSP…)**
- check offer.sdp :)
- sending that sdp… (no signaling defined)

```
v=0
o=- 4420499408415283528 2 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE audio video
a=msid-semantic: WMS
VNGaWbmub1HsBjaxybQCE91gqOW5RFN3zt2Z
m=audio 1 RTP/SAVPF 111 103 104 0 8 106 105 13 126
c=IN IP4 0.0.0.0
a=rtcp:1 IN IP4 0.0.0.0
a=ice-ufrag:rTPGidiJq18Xsr2l
a=ice-pwd:xrvmCOYkTGlIcqNMuCFr9yrf
a=ice-options:google-ice
a=fingerprint:sha-256 A0:15:3C:15:22:98:E9:D1:4C:60:
D0:0D:F7:34:A8:DA:9F:52:E0:D1:D0:16:43:B3:5D:95:EC:
EB:B4:9C:90:D5
a=setup:actpass
a=mid:audio
a=extmap:1 urn:ietf:params:rtp-hdrext:ssrc-audio-
level
a=extmap:3 http://www.webrtc.org/experiments/rtp-
hdrext/abs-send-time
```

http://jsfiddle.net/gh/gist/library/pure/6272bd4f5c626ab7a2bc/68170b303b509d9a17d9c968a5447aaee117cbb3/
Action: check how we're "sending" the offer SDP
http://webrtchacks.com/sdp-anatomy/

Let´s answer it

http://jsfiddle.net/gh/gist/library/pure/7fbe699afc6462f22cd1/a418431770c29a00a252d48730e4613500c196df/

Actions:

- Send offer sdp to callee
- Send answer sdp to caller

- isCaller flag
- pc.**setRemoteDescription**
- pc.**createAnswer**
- **pc.setLocalDescription(answer)**

… and receiving the answer

http://jsfiddle.net/gh/gist/library/pure/8d0c1e1bd09fe27761ff/

Action:

- Offer
- Answer

- Caller does not know about callee yet
- Receiving the answer the same way the callee received the offer (signaling)
- pc.setRemoteDescription with type.answer
- check pc state..

pc state until now, finished?

http://jsfiddle.net/gh/gist/library/pure/8d0c1e1bd09fe27761ff/

Same code, but this time pay attention to the pc state

- pc.localDescription OK
- pc.remoteDescription OK
- pc.onicecandidate ???
- pc.onaddstream ???

ICE

- Purpose
- Candidate
- Types:
  - local
  - reflexive/stun
  - relay/turn
- pc.onicecandidate
- pc.iceconnectionstate

http://jsfiddle.net/gh/gist/library/pure/0ea6abc8a6f7064895a6/f1086199fbb3296a39172ac2119c05e9fd851a29/

Action: check refactor

http://jsfiddle.net/gh/gist/library/pure/0ea6abc8a6f7064895a6/1d4a6676e86824353b4b17aabad02031690b28f5/

Action:

- check the candidates
- check pc.iceGatheringState

http://jsfiddle.net/gh/gist/library/pure/0ea6abc8a6f7064895a6/c9d54da712a7d69d44dafa4e1339fea3764ca01a/

Action:

- Check candidates again

http://jsfiddle.net/gh/gist/library/pure/0ea6abc8a6f7064895a6/798c3aba9bf521f508ad31f9d0e93e6d81ccb61f/

Actions: do the whole flow again, offer, answer and candidates

no video yet, what´s missing?

- pc.onaddstream

http://jsfiddle.net/gh/gist/jquery/1.11.0/0ea6abc8a6f7064895a6/05f7404850c1933e07fc4b72faa07841180c320f/

Action: try again the whole flow, you should now have video in both tabs

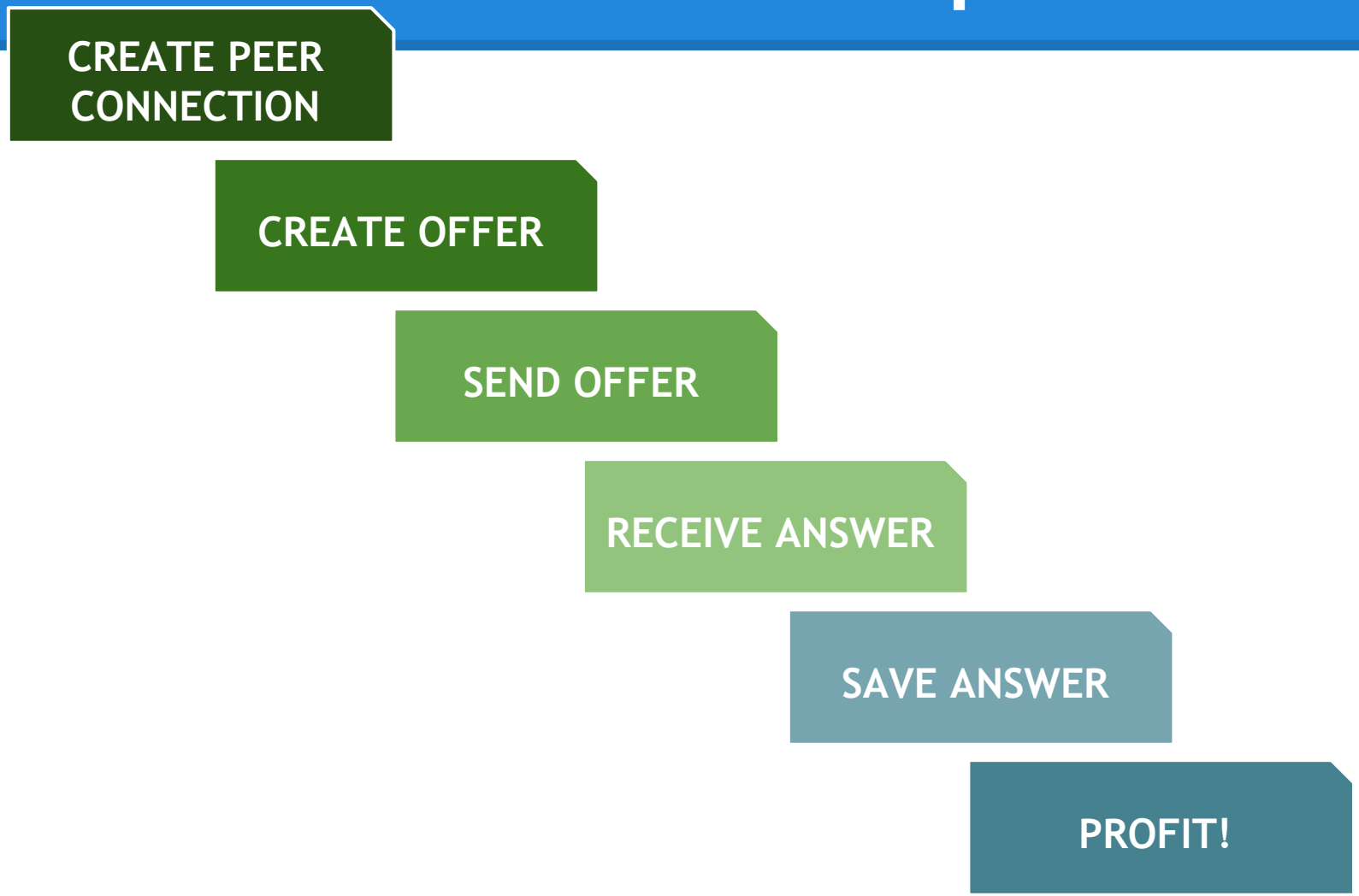
:)

https://github.com/ivmos/webrtc-workshop/tree/master/cpJsApp

Recap

- create pc
- create offer, pc.setLocalDescription(offer)
- send offer
- receive offer
- pc.setRemoteDescription(offer)
- pc.createAnswer, pc.setLocalDescr(answer)
- send answer
- receive answer
- pc.setRemoteDescription(answer)
- ICE!!

Recap: caller



Recap: callee

BOTH


SEND ICE CANDIDATE
SEND ICE CANDIDATE
SEND ICE CANDIDATE
SEND ICE CANDIDATE
SEND ICE CANDIDATE
SEND ICE CANDIDATE

Server signaling iteration

- XHR polling example
- API:
  - startSession(sid, success)
  - getParticipants(sid, success)
  - getMessages(sid, success)
  - sendOffer(sid, to, offerSdp, success)
  - sendAnswer(sid, to, answerSdp, success)
  - sendIceCandidate(sid, to, candSdp, success)
  - sendTerminate(sid, to, success)
- MySql backend:
  - sessions
  - requests

Base snippet:

http://jsfiddle.net/gh/gist/jquery/1.11.0/9a93a110efb9cf0ae46f/7d03c3c617753d2029950b05181d557925a4a7d4/

1. Use real signaling

- Instead of copy-paste signaling, use the provided API. “s/console.log/Api.*/g”
- mySid, otherSid hardcoded by now, we still assume we’re only working with 2 tabs.
- Candidates to be sent one by one
- Api.startSession needed

2. Listen for messages

- We need to listen for offer, answer and candidate type messages
- setInterval loop

3. adding candidates fix

- Check pc.remoteDescription before pc.addIceCandidate. Enqueue if needed.
- Do the pc.addIceCandidate after pc.setRemoteDescription
- Not a problem for caller as he starts the signaling
- Profit!

Improvements

- unharcode otherSid
  - <select> for to (callee)
    - getParticipants loop
- Call button: isCaller and offer
    - call method
- unharcode mySid
  - <input> for sessionId (caller)
- Add your own video too

http://jsfiddle.net/gh/gist/jquery/1.11.0/5692b5b7c0fa4d855d2b/

https://github.com/ivmos/webrtc-workshop/tree/master/xhrJsApp

Security

- In our implementation, signaling security concerns?
- In media channel?
  - WebRTC security

A few links

- WebRTC 1.0 RFC
- Understanding why ORTC effort (Iñaki Baz interview)
- JS Lib using SIP signaling: http://jssip.net
- http://webrtchacks.com/
- https://www.webrtc-experiment.com/ (Muaz Khan interview)
- http://corporate.tuenti.com/en/dev/blog/Building-a-VoIP-Service-using-WebRTC


WebRTC


VOZDIGITAL

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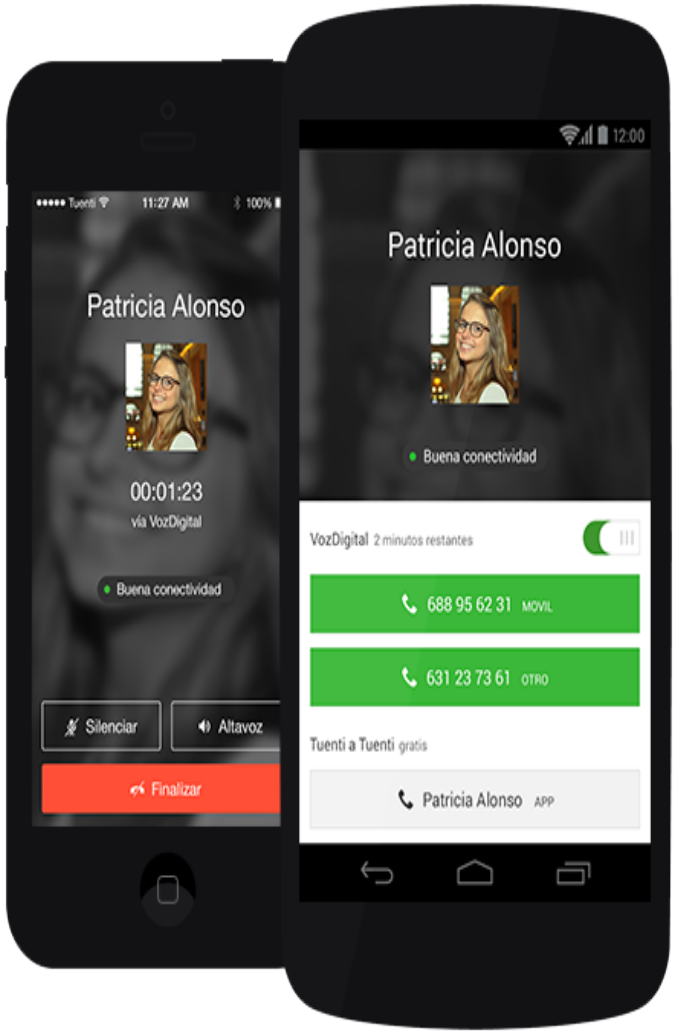
An inbound & outbound VoIP service

...using the customer's GSM number!

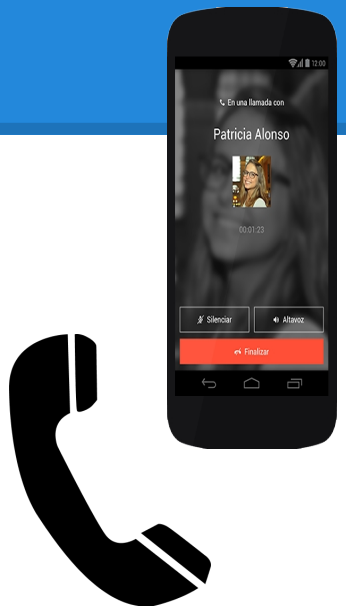
...with no additional data charges!

...integrated with existing Tuenti chat infrastructure

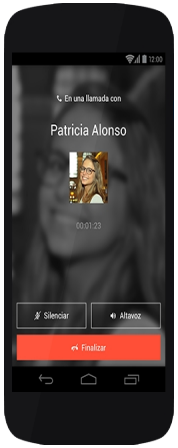
WebRTC for App2App calls

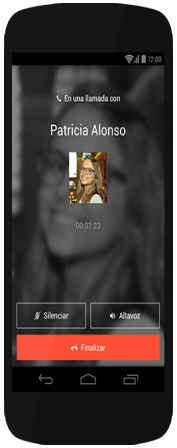


Multiple resources

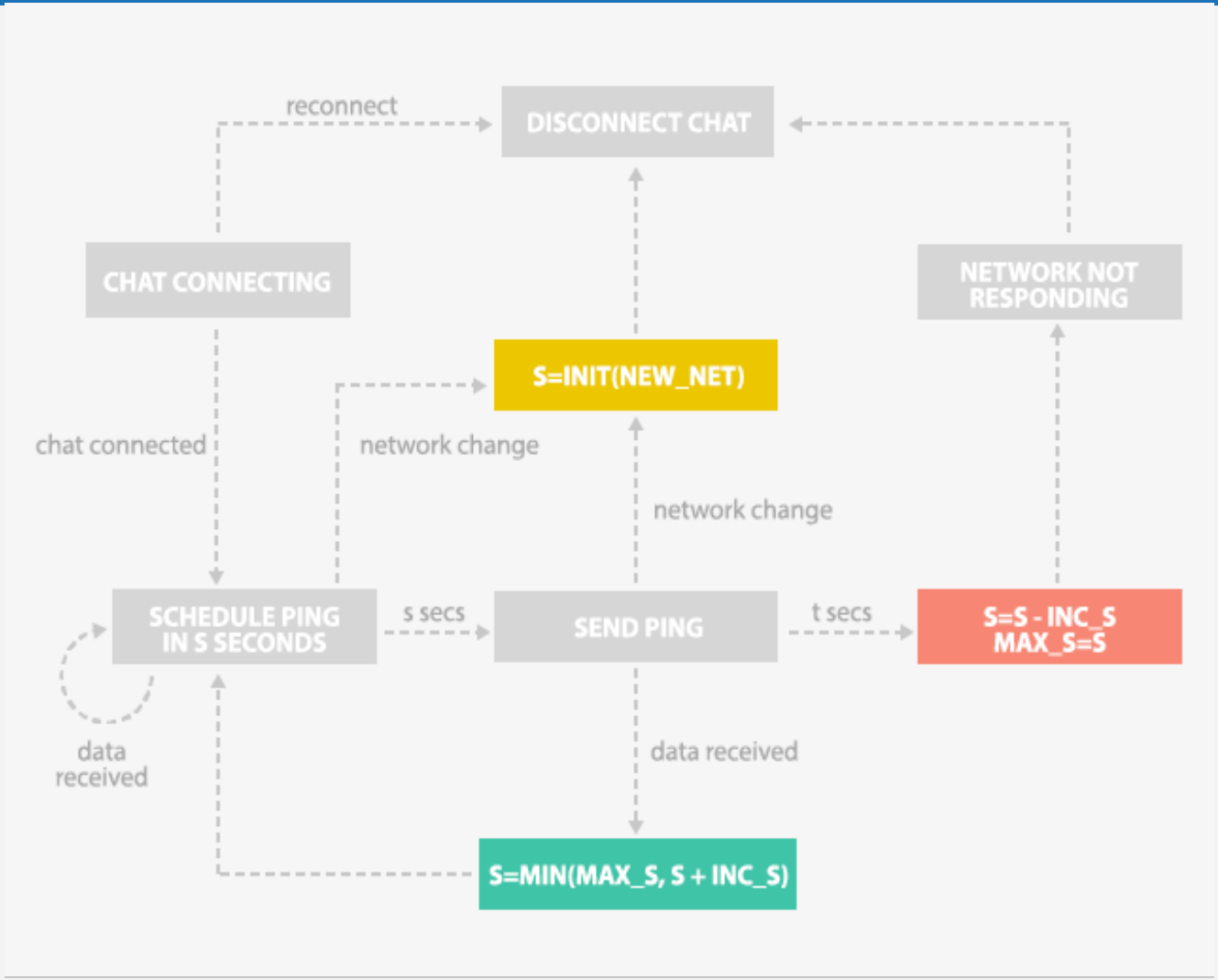


Will he be available??



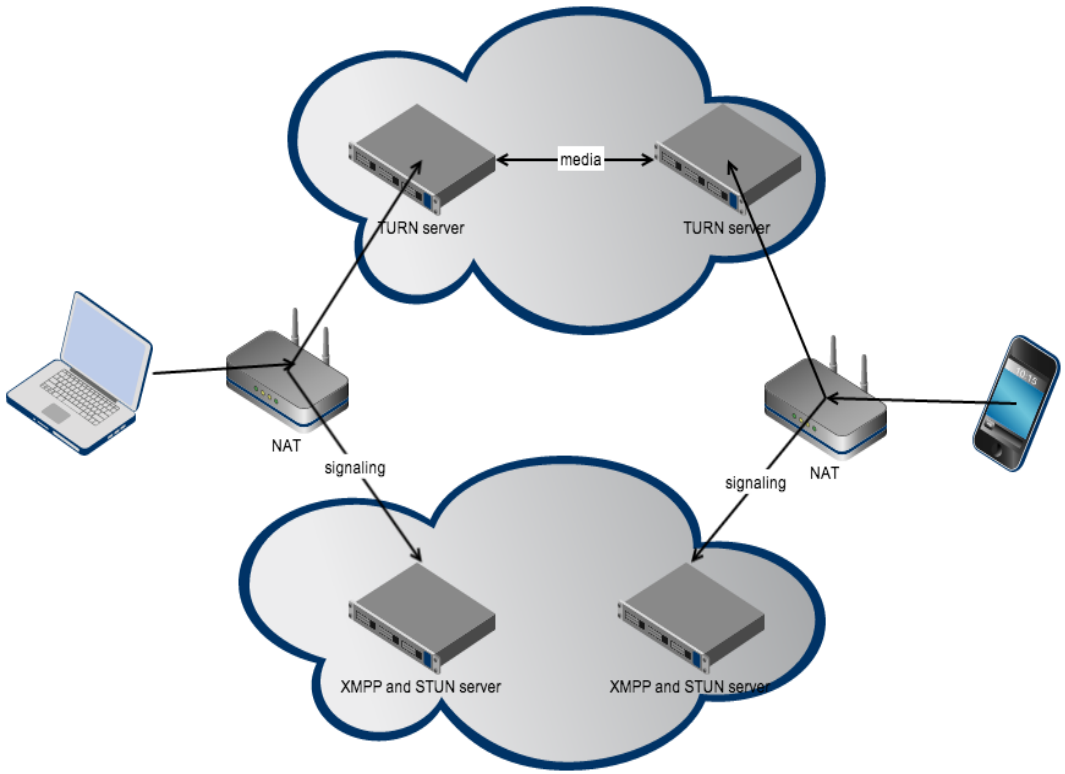


Always Connected Application




ZEROLÍMITES
LLAMA, CHATEA Y COMPARTE CON TUENTI MÓVIL Y LA NUEVA APP
INCLUSO SIN SALDO, SIN CONSUMIR DATOS
TOTALMENTE GRATIS Y SIN LÍMITES
ZEROSALDO
Ahora, hasta sin saldo, también estás conectado
ZEROCONSUMO
Mantente conectado y olvídate de contar megas
ZEROBARRERAS
Llama gratis desde nuestra app a tus amigos sean del operador que sean

Solution Overview



Signaling over XMPP

- Jingle protocol

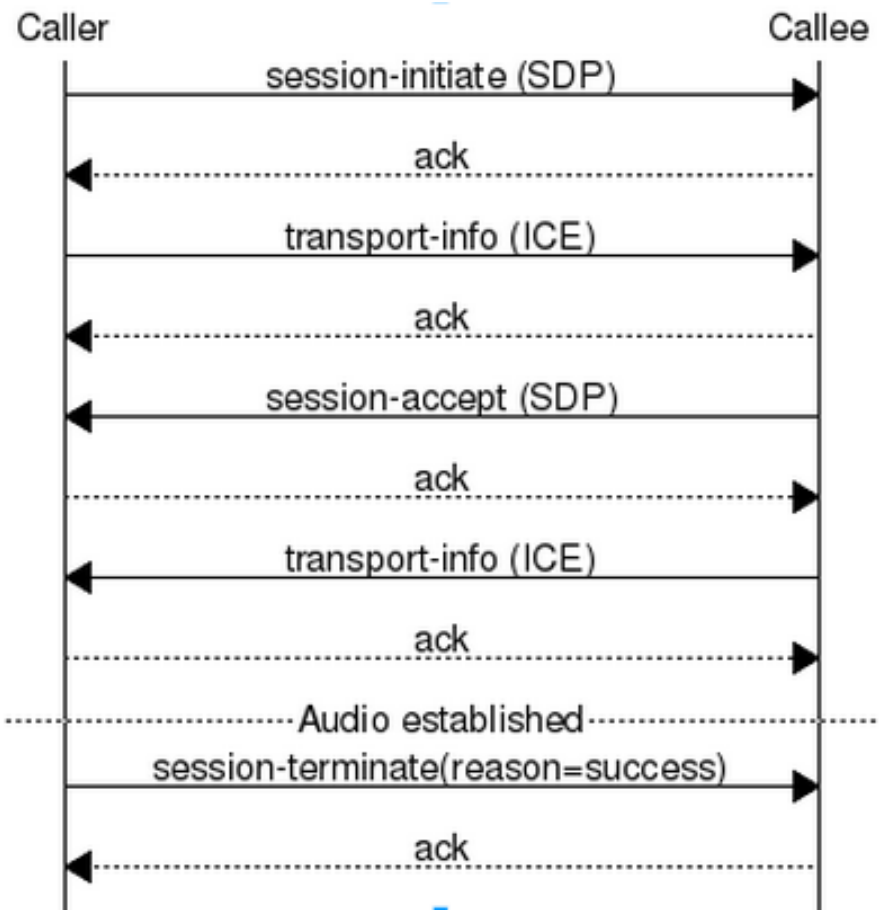
```
<jingle xmlns='urn:xmpp:jingle:1'
        action='session-initiate'
        initiator='romeo@montague.lit/orchard'
        sid='a73sjjvkla37jfea'>
  <content creator='initiator' name='voice'>
   <description xmlns='urn:xmpp:jingle:apps:rtp:1' media='audio'>
    <payload-type id='96' name='speex' clockrate='16000'/>
    <payload-type id='97' name='speex' clockrate='8000'/>
    <payload-type id='18' name='G729'/>
    <payload-type id='0' name='PCMU' />
    <payload-type id='103' name='L16' clockrate='16000' channels='2'/>
    <payload-type id='98' name='x-ISAC' clockrate='8000'/>
   </description>
  </content>
 </jingle>
```

SDP over XMPP in tuenti

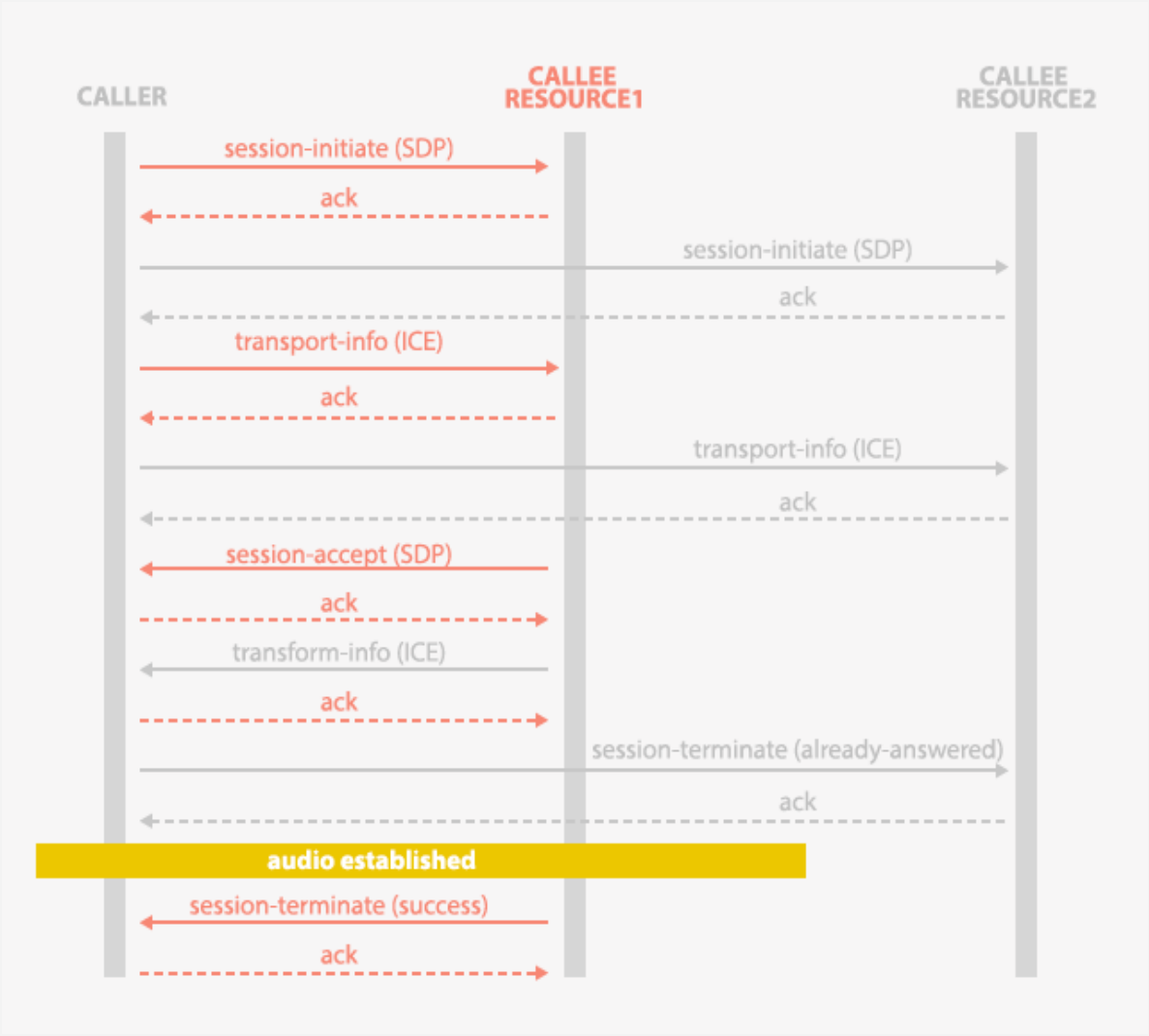
- Tangle protocol

```
<tangle xmlns='urn:tuenti:tangle:1'
    action='session-initiate'
    initiator='javierf@tuenti.com'
    sid='a73sjjvkla37jfea'>
    <sdp><![CDATA[SDP]]></sdp>
</tangle>
```

VoIP message sequence chart

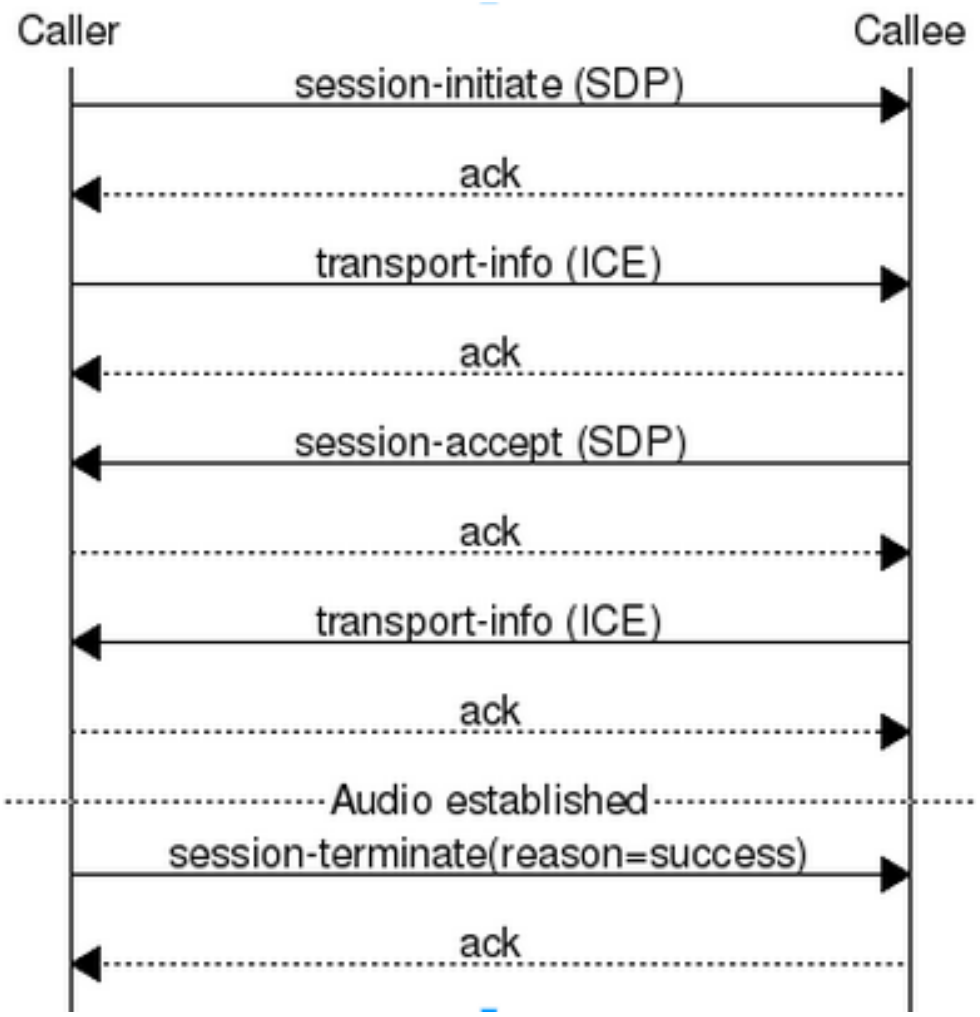


Tangle multiple resources

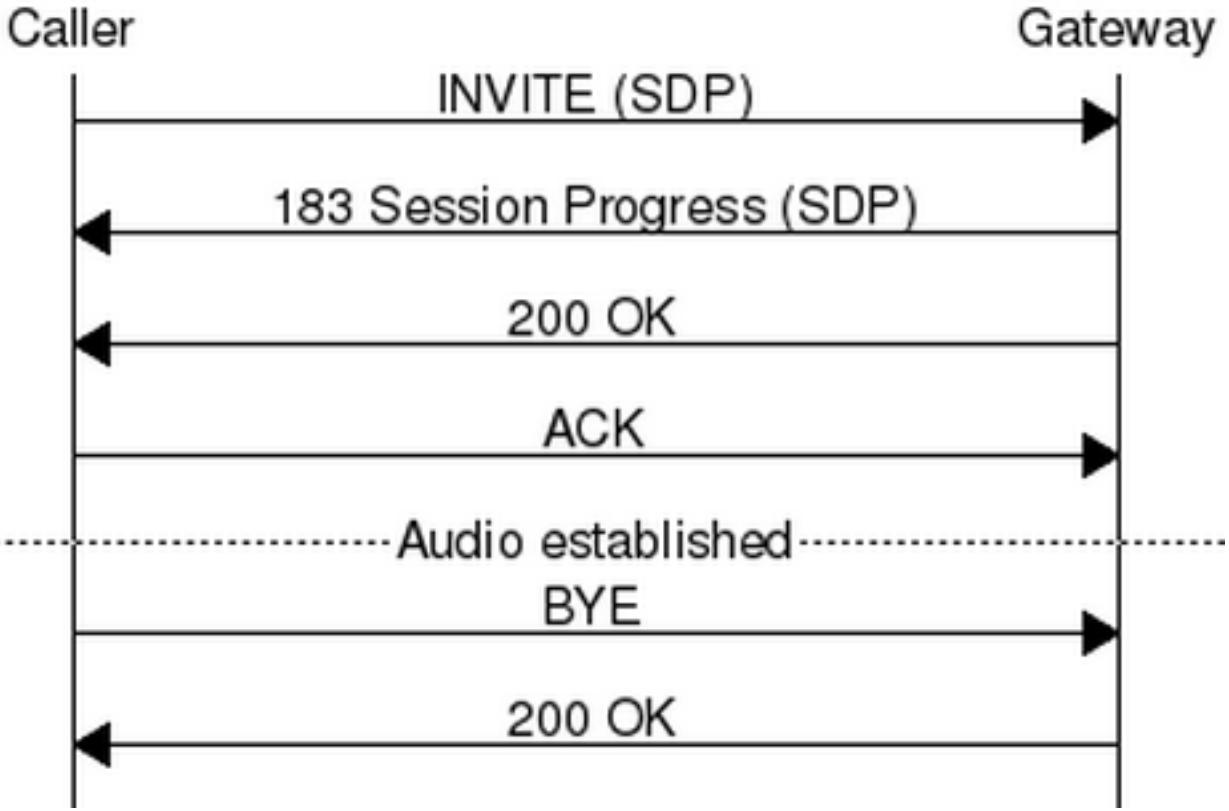


We need a SIP stack!

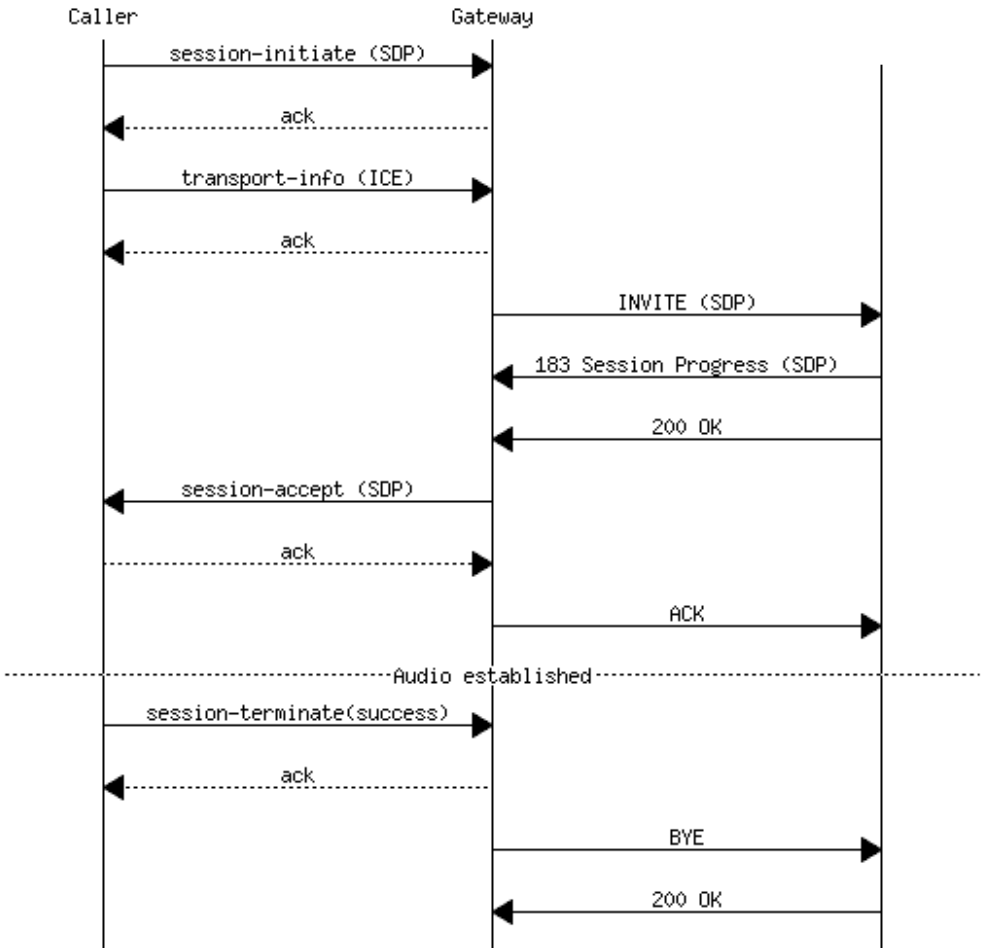
VoIP message sequence chart



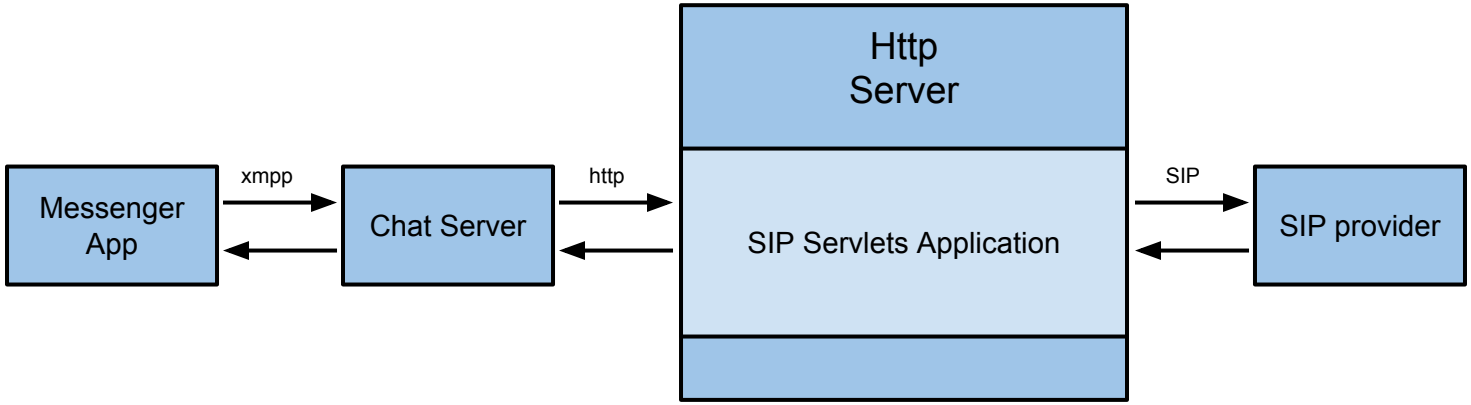
SIP message sequence chart

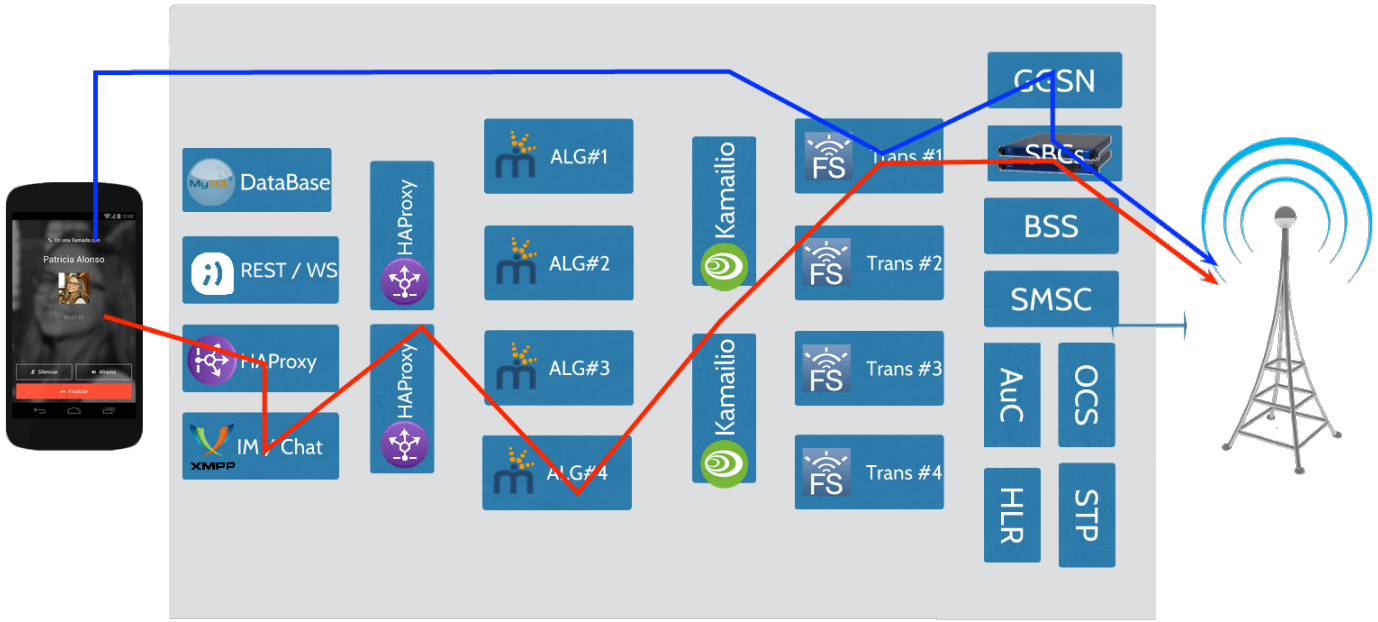


Merging the message sequence charts



Voice Gateway Overview




Patricia Alonso
DataBase
REST / WS
HAProxy
IM / Chat
XMPP
HAProxy
HAProxy
ALG#1
ALG#2
ALG#3
ALG#4
Kamailio
Kamailio
FS
Trans #2
FS
Trans #3
FS
Trans #4
GGSN
BSS
SMSC
AuC
OCS
HLR
STP

Questions?