音视频通话webrtc开发那些坑

介绍

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- 玩耍直播创始人
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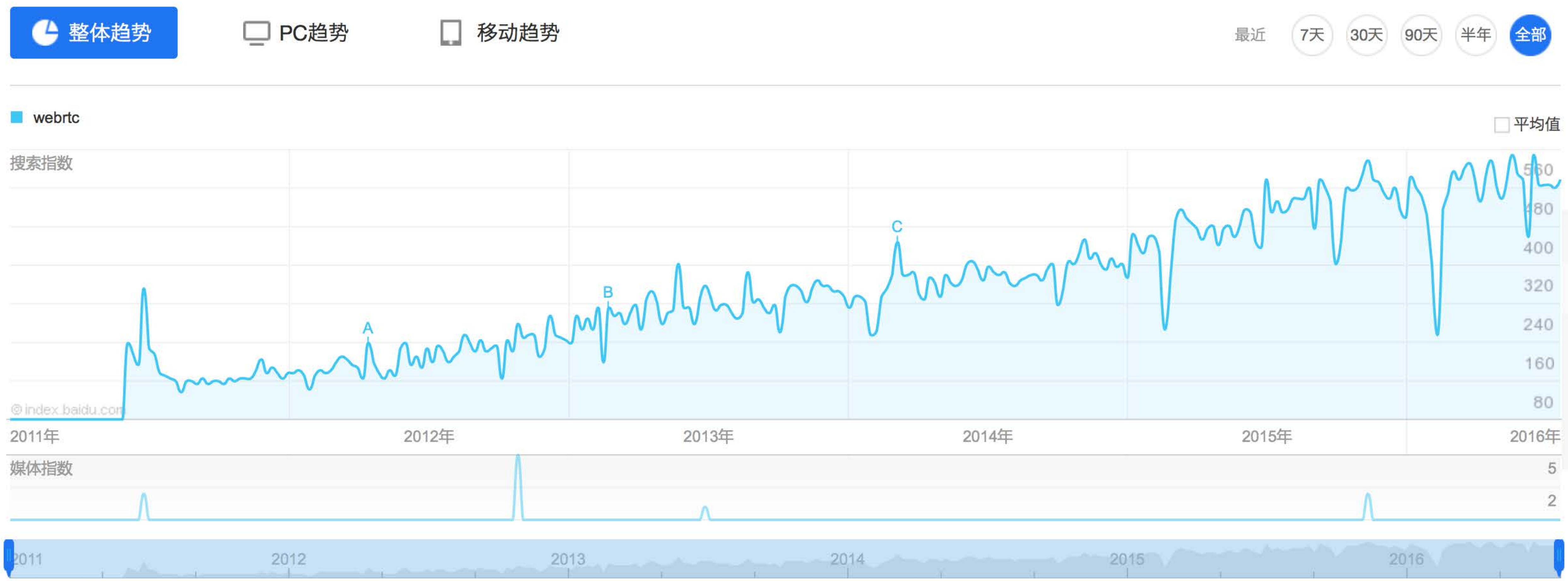
WebRTC 是什么

"WebRTC 可以让你在浏览器中 移动平台上 嵌入式设备中进行实时的音视频通话以及文件传输"



90%的音视频通话是基于webrtc技术搭建，图为百度搜索指数变

WebRTC涉及到的模块

client

signaling

stun/turn

mcu/sfu

WebRTC client

brower

native client

custom client

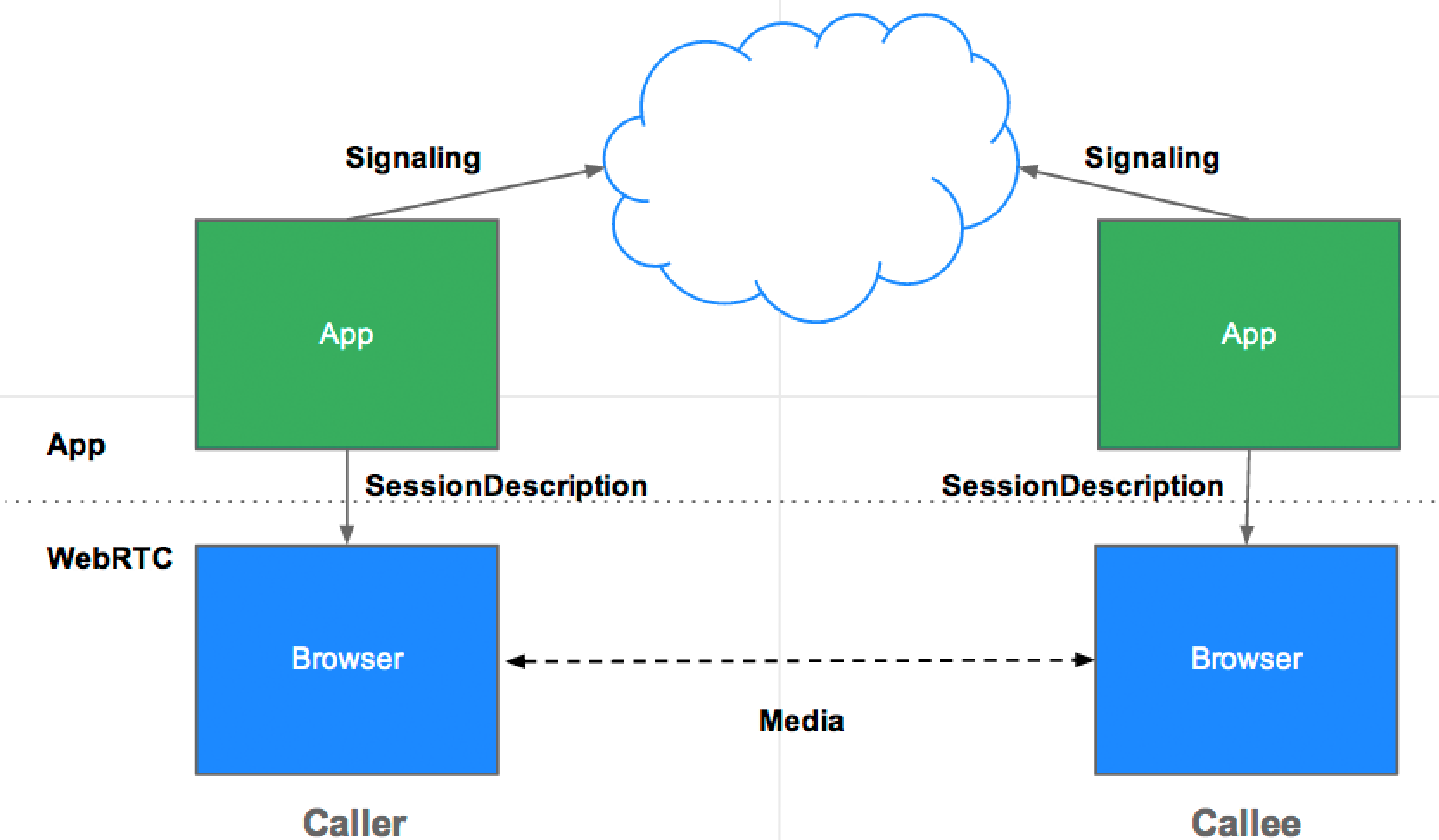
Signaling

- 用来交换各自的能力
- 对传输层,传输协议没有要求
- sip/websocket+自定义协议
- 重连

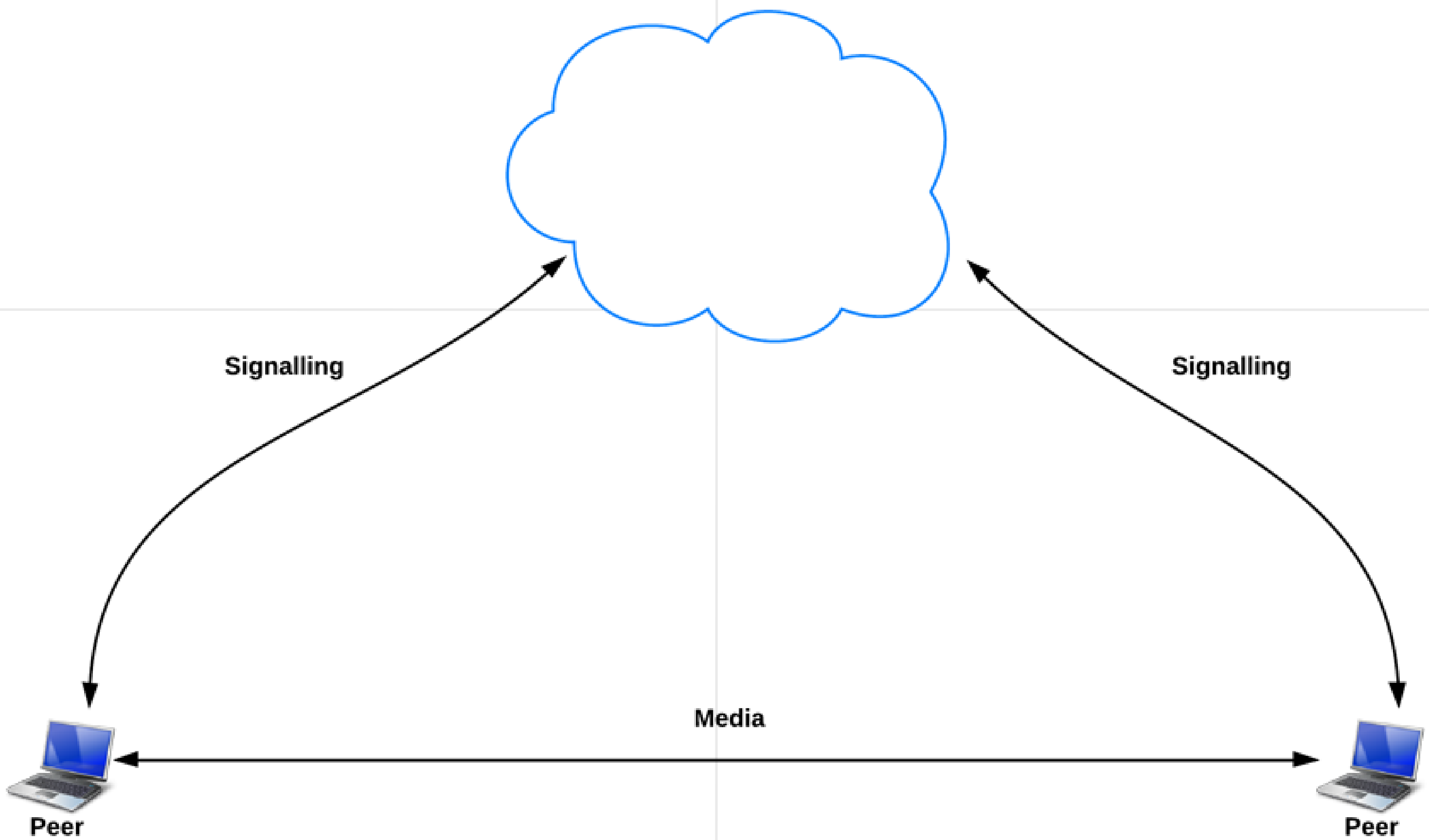
视频编码的选择

- vp8是默认编码
- vp9已经可用,同样质量码率可以比vp8小30%
- h264硬件支持,还有很多问题
- 苹果只支持h264
- Baseline Profile/ High Profile

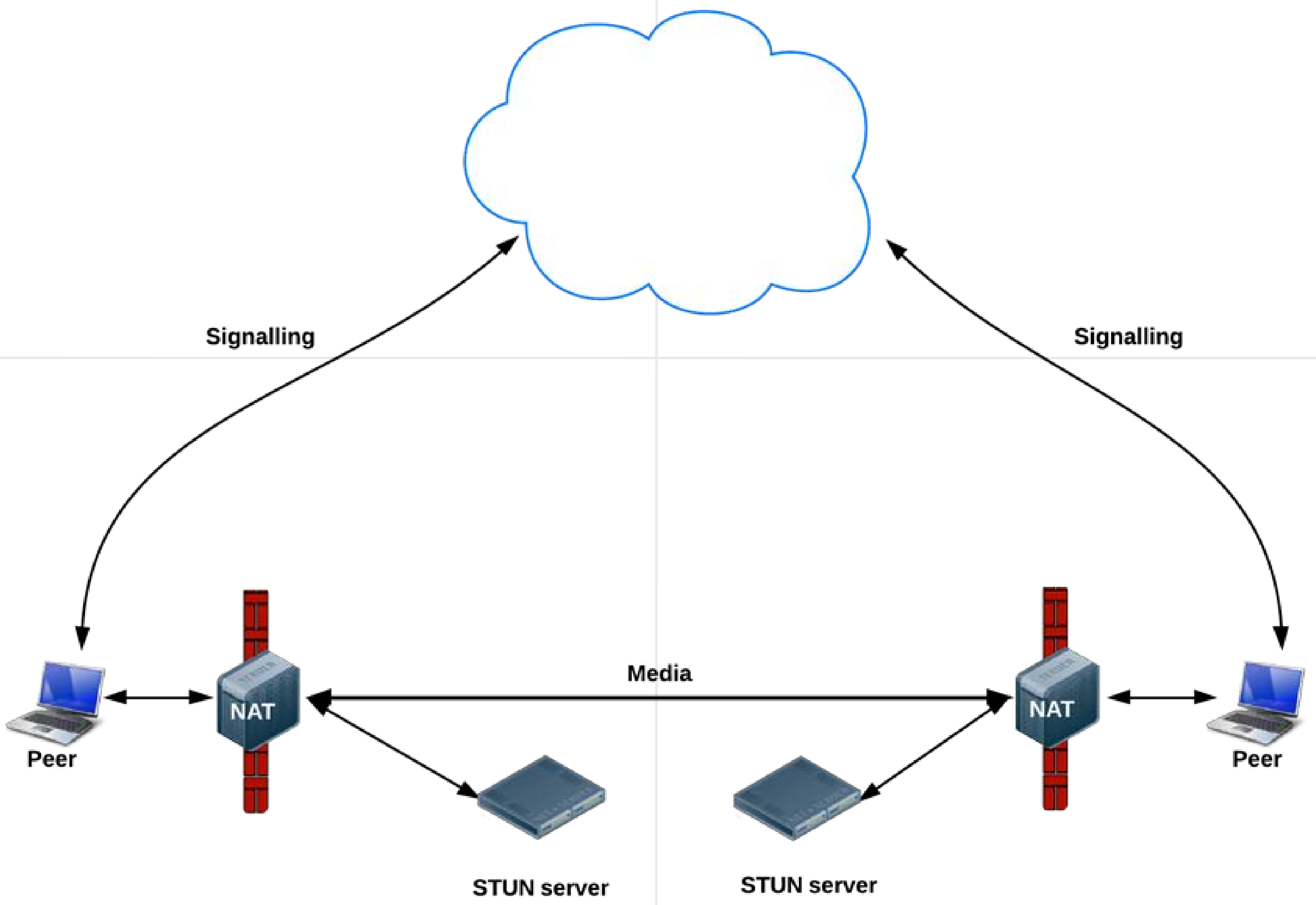
Signaling


Signaling
Signaling
App
App
App
SessionDescription
SessionDescription
WebRTC
Browser
Browser
Media
Caller
Callee

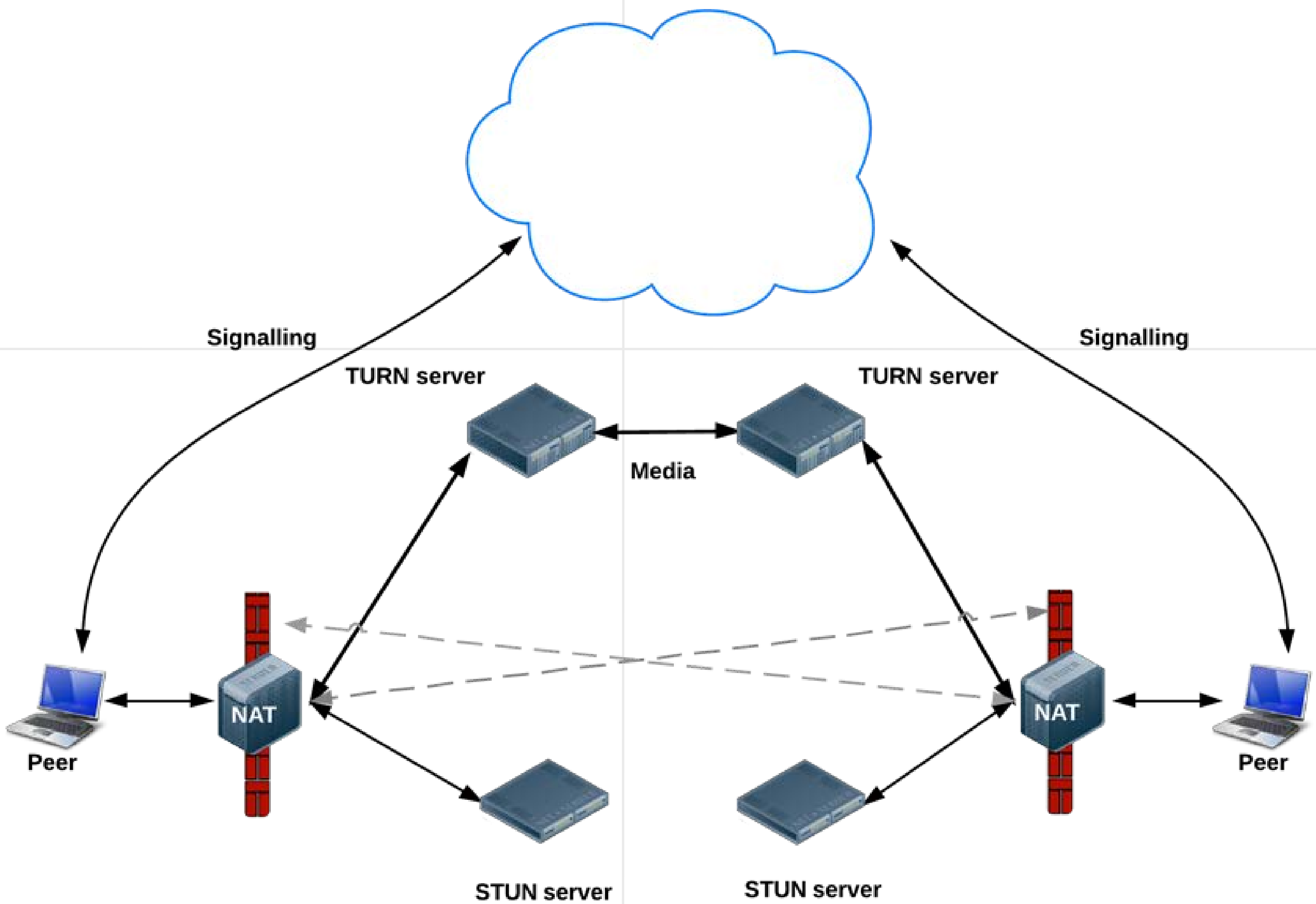
STUN And TURN



STUN

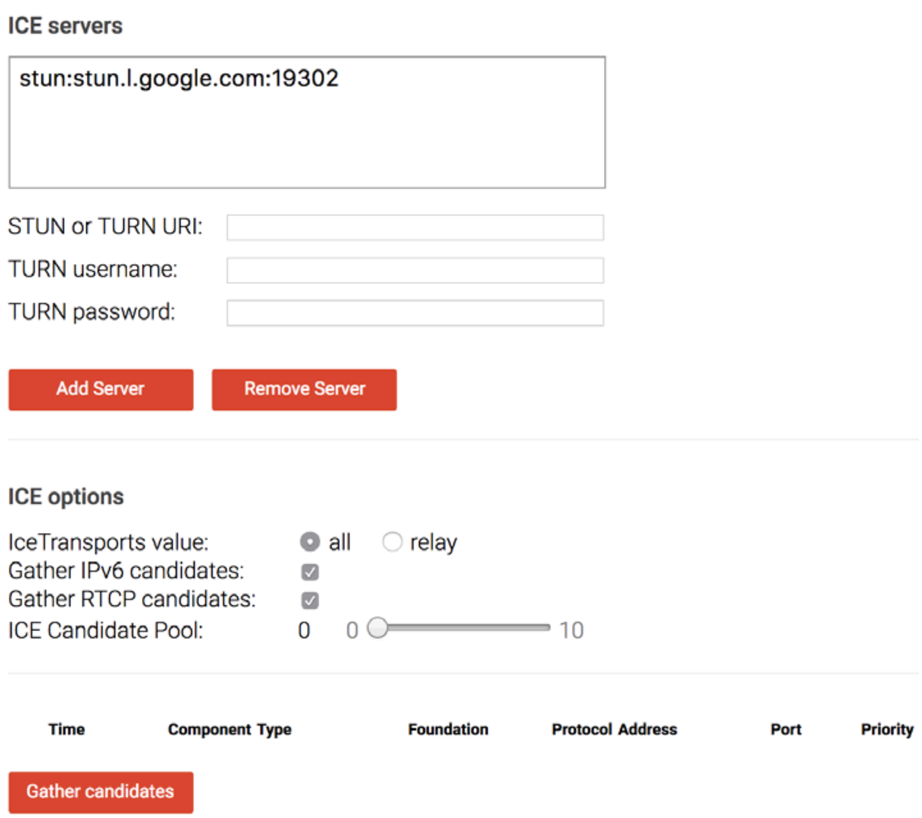

Signalling
Signalling
Media
NAT
NAT
Peer
Peer
STUN server
STUN server

TURN



Signalling
Signalling
TURN server
TURN server
Media
NAT
NAT
Peer
Peer
STUN server
STUN server

STUN/TURN 验证

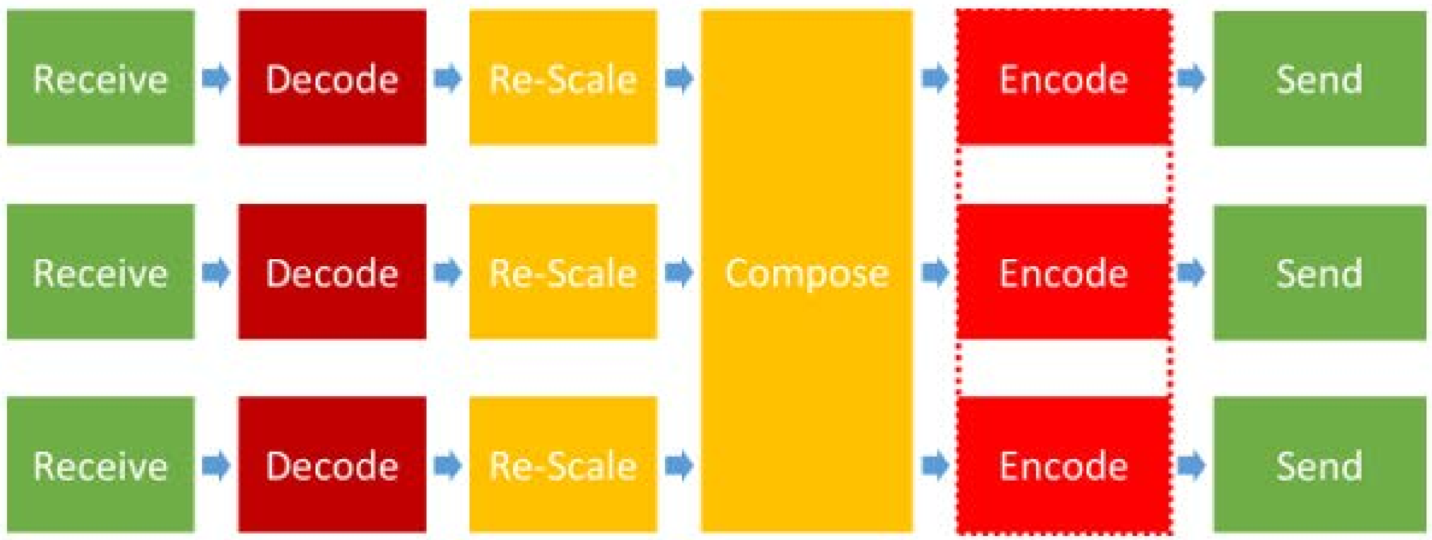


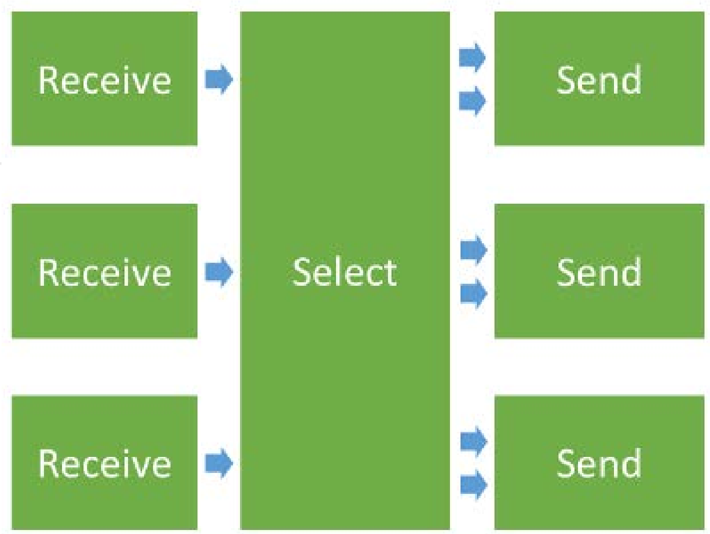
ps://webrtc.github.io/samples/src/content/peerconnection/trickle-[redacted]e/

STUN/TURN REST

MCU OR SFU

MCU VS SFU


Receive
Decode
Re-Scale
Compose
Encode
Send
Receive
Decode
Re-Scale
Encode
Send
Receive
Decode
Re-Scale
Encode
Send


Receive
Select
Send
Receive
Send
Receive
Send

SFU FIRST

开源的服务端方案

- licode(https://github.com/lynckia/licode)
- kurento(https://github.com/Kurento/kurento-media-server)
- janus-gateway(https://github.com/meetecho/janus-gateway)
- media-server(https://github.com/medooze/media-server)
- jitsi-videobridge(https://github.com/jitsi/jitsi-videobridge.git)

一些建议

- 不要使用最新的代码
- 开发阶段自己搭建stun server 和 turn server
- 视频h264编码
- 硬编软解

dot Engine

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扫一扫上面的二维码图案，加我微信