

# **WEBRTC PURE SIP**

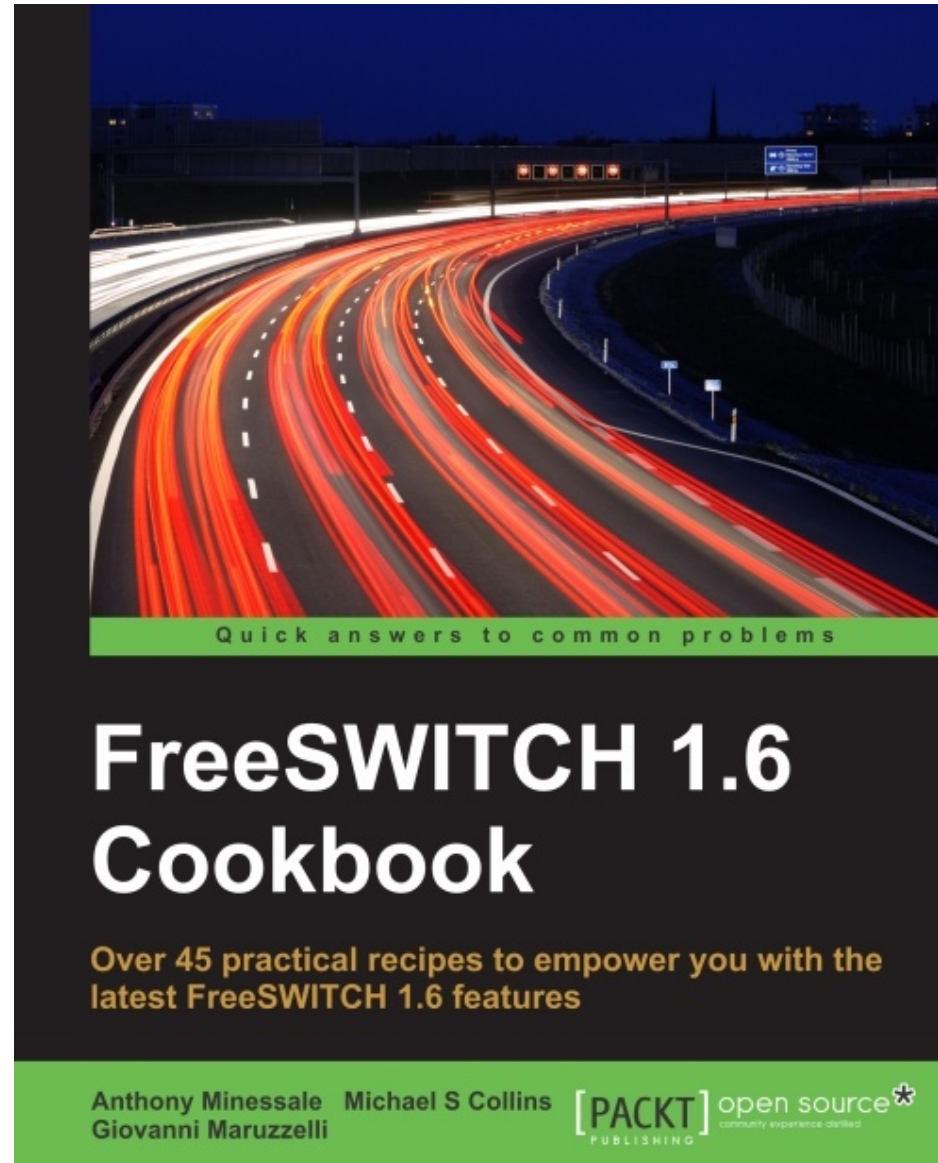
## **Video Call Conferencing and Chat**

### **with**

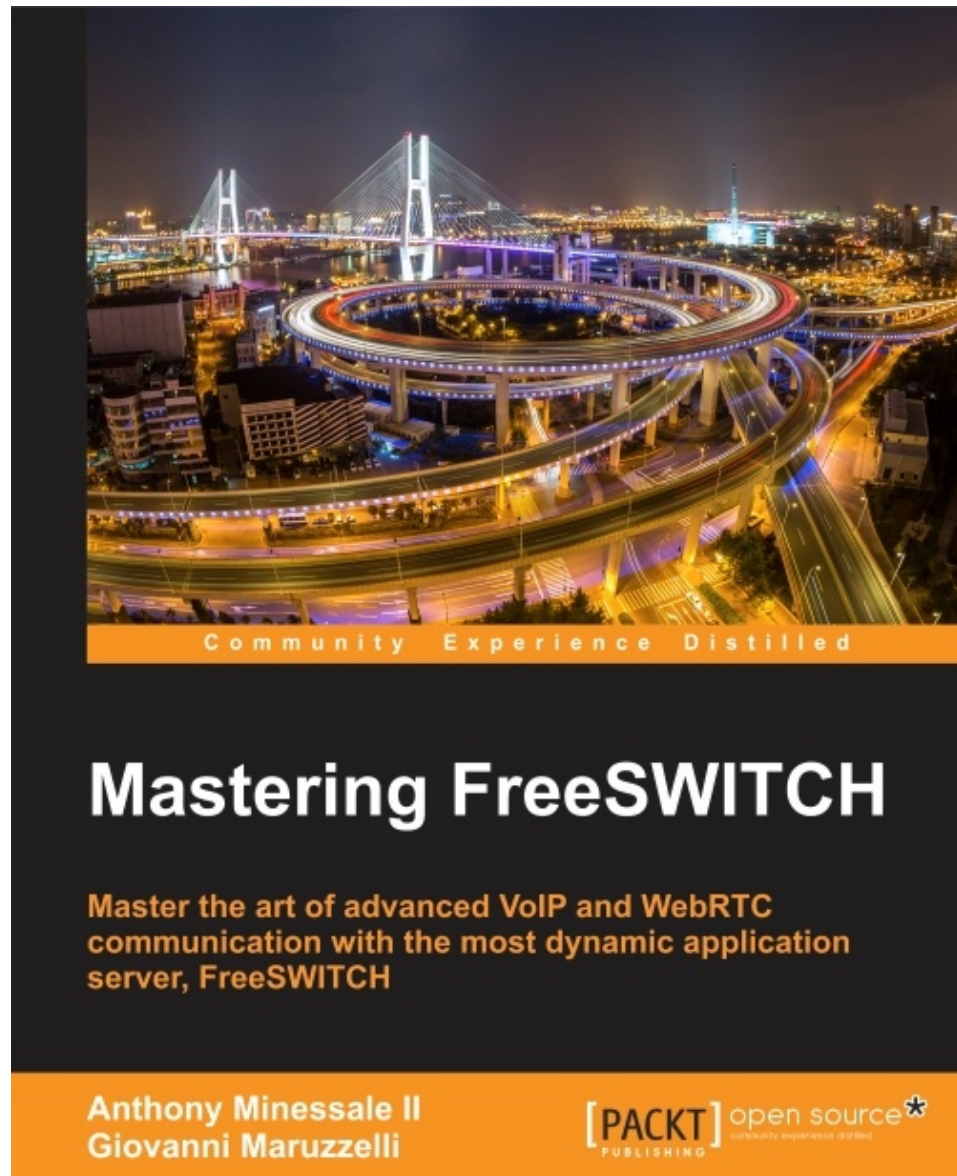
## **FreeSWITCH & OpenSIPS**

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# 1.8 Book Covers 1.6 too!

## Preface

This is the third book in a Trilogy: *FreeSWITCH 1.6 Cookbook*, *Mastering FreeSWITCH*, and yours truly, *FreeSWITCH 1.8*. Each one of them has its own role in FreeSWITCH's Universe. The call of this one you're reading is to be the reference and foundation, the exhaustive source of FreeSWITCH knowledge.

We've worked hard, we started from the efforts of our co-writers in *FreeSWITCH 1.2 book* (they were Michael S. Collins, Darren Schreiber and Raymond Chandler), we've updated, rewritten and written anew and we're now proud to introduce you to all you need to know about latest FreeSWITCH.

So much happened in the world of Real Time Communication and VoIP in the last years. And FreeSWITCH was leading innovation, implementation and interoperability in the industry: WebRTC, Video MCU, High Definition Audio, OPUS, VP8/9, JSON, Encryption and Security, you name it, FreeSWITCH was there.

We're about to announce FreeSWITCH 1.8: an evolutionary step from 1.6, that enhance performances and changes a lot of under the hood plumbing that will be the nervous system of The Next Big Things (TM) in FreeSWITCH Universe.

From an operation, administration and application programming point of view 1.8 is completely compatible with 1.6, and this book cover them both.

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## What this book covers

Chapter 1, *Architecture of FreeSWITCH*, gives a brief, but thorough introduction to the underlying architecture of FreeSWITCH.

Chapter 2, *Building and Installation*, shows how to download and install FreeSWITCH on Windows and Unix-like operating systems.

Chapter 3, *Test Driving the Example Configuration*, provides a hands-on look at the powerful and feature-rich example FreeSWITCH configuration.

Chapter 4, *User Directory, SIP, and Verto*, offers an introduction to the concept of users and the directory as well as a brief look at SIP and Verto user agents.



# Giovanni Maruzzelli

- Consulting ( Solutions and Training )
- Principal at OpenTelecom.IT
- Internet Communication since 1994 (no web then)
- Specialized in Industrial Grade Communication
- SIP - WebRTC
- FreeSWITCH - OpenSIPS - Kamailio - FusionPBX - Homer
- Load Balancing - High Availability - Geo Distribution

# AGENDA:

- SIP Video Calls vs Video Conferencing
- enter FreeSWITCH
- enter WebRTC
- ( SIP SIMPLE + SIP Proxy ) == Scalability for Free
- enter RTPEngine
- OpenSIPS Config
- FreeSWITCH Config
- ( HTML & JS ) => Browser => Cordova => App
- server PUSH IT !

# Aim:

- a modern video-conferencing system
- reachable from phones, browsers, apps
- with chat, video effects, screen sharing, moderation, etc
- scalable and highly available
- multitenant
- built from proven and manageable technologies
- usable from low powered, low batteries, low cpu devices

# yo, SIP got video calls since...

- but conferencing is not "video calls"
- need to convey messaging between participants
- need to allow for moderator functions:
  - mute participants
  - give floor (select speaker)
  - kick out and invite in participants
- need to allow for mixing, splitting and compositing of streams:
  - audio
  - video
  - screen share
- need to allow for system feedback:
  - who is speaking
  - wait for conference start

# FreeSWITCH

- most powerful multimedia switch
- SIP/Verto/WebRTC/TDM support
- HD audio and video transcode and mixing
- enterprise PBX features
- static dialplan / dialplan from http / scripts execution / remote call management
- Auto Attendant / IVR / fully programmable access to DBs and legacy systems
- multi language sounds management with locale smart phrases



# FreeSWITCH

- TTS and ASR
- audio/video conferencing
- enterprise video MCU and CG effects (and CPU-friendly video follow floor mode)
- fully featured enterprise grade voicemail
- callcenter / ACD / call queues management
- best fax/T38 support
- multiple SIP gateway support with failover
- complete multidimensional CDR generation

# WebRTC

- base technology for Real Time Communication (RTC)
- first developed for (billions of) web browsers
- adopted by smartphone apps
- most probable candidate for Internet of Things media
- only cares about media streaming
- does not mandate a user location or a session protocol
- you're free to invent your own ad hoc solution

# **well, so we need to...**

- accept and distribute video calls, and mix, slice, dice and combine their streams, from any kind of client
- accept and distribute chatting messages between participants
- allow for moderators to give commands and interact with system
- allow for system to scale and be highly available
- provide for security, AAA, and interconnections

# This is a Job for SIP

- you know is robust and it scales, don't you?
- OpenSIPS SIP Proxy
  - security
  - AAA, Authentication, Authorization, Accounting
  - load balancing, dispatching
- SIP SIMPLE
  - chat
  - system interaction
- FreeSWITCH SIP B2BUA and Application Server
  - Media Services (Conferencing, audio Mix, Video stream select, CG, Mux and MCU)
  - chatplan
- SIP signaling for WebRTC
  - SIP.js

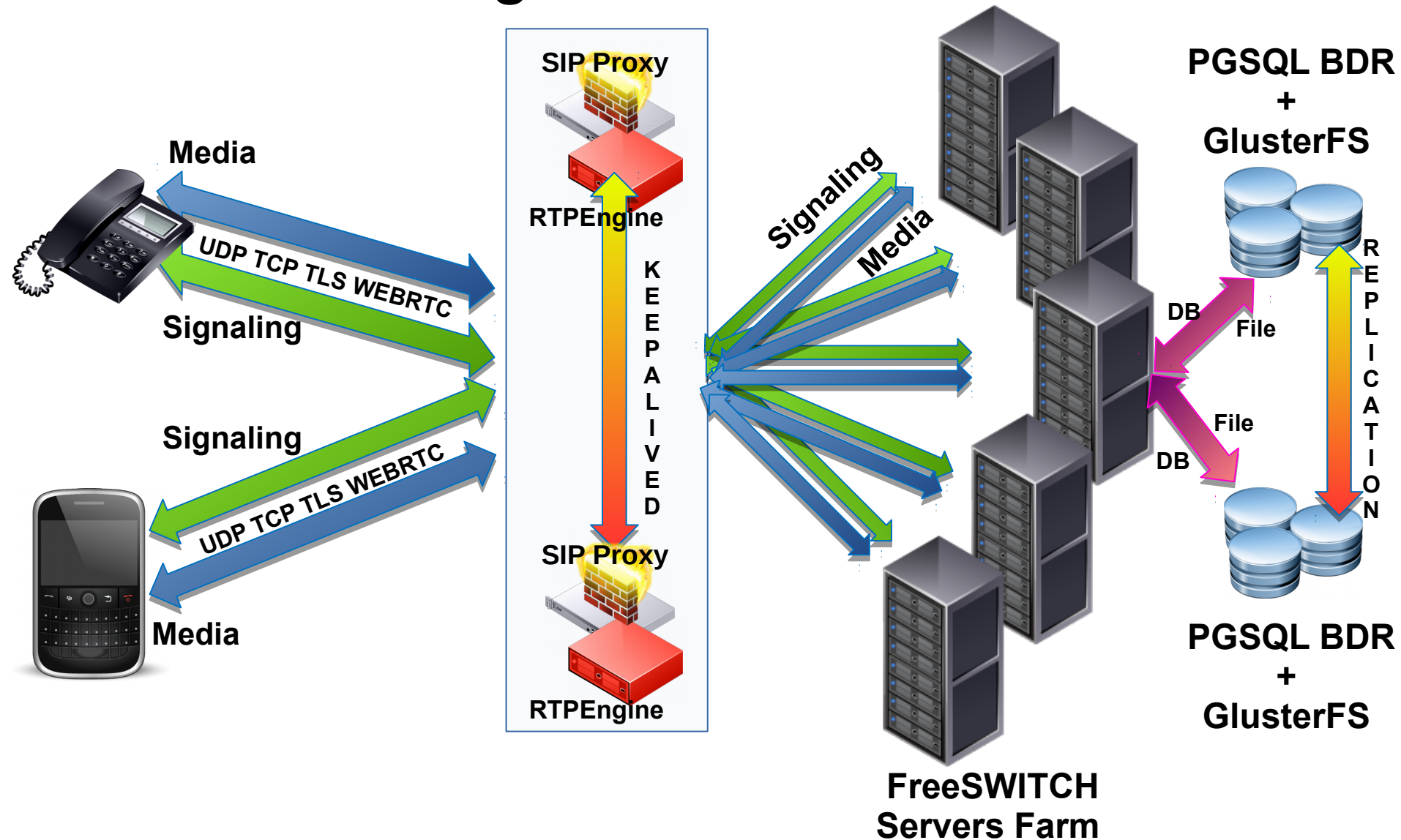
# Aaaaand

- Cordova
  - create smartphone native apps from html + JS
- Google FCM
  - push delivery
  - supports both iOS and Android
  - wakeup the app on the smartphone
  - supported by Cordova



# SIP

## Scaling Video Conferences



# OpenSIPS Configuration

- HA in active-passive, with VRRP (Keepalived)
- Pretty standard WebRTC configuration
- WebSocket
- NAT management
- RTPEngine management
- Registrations
- Dispatch

# OpenSIPS Configuration

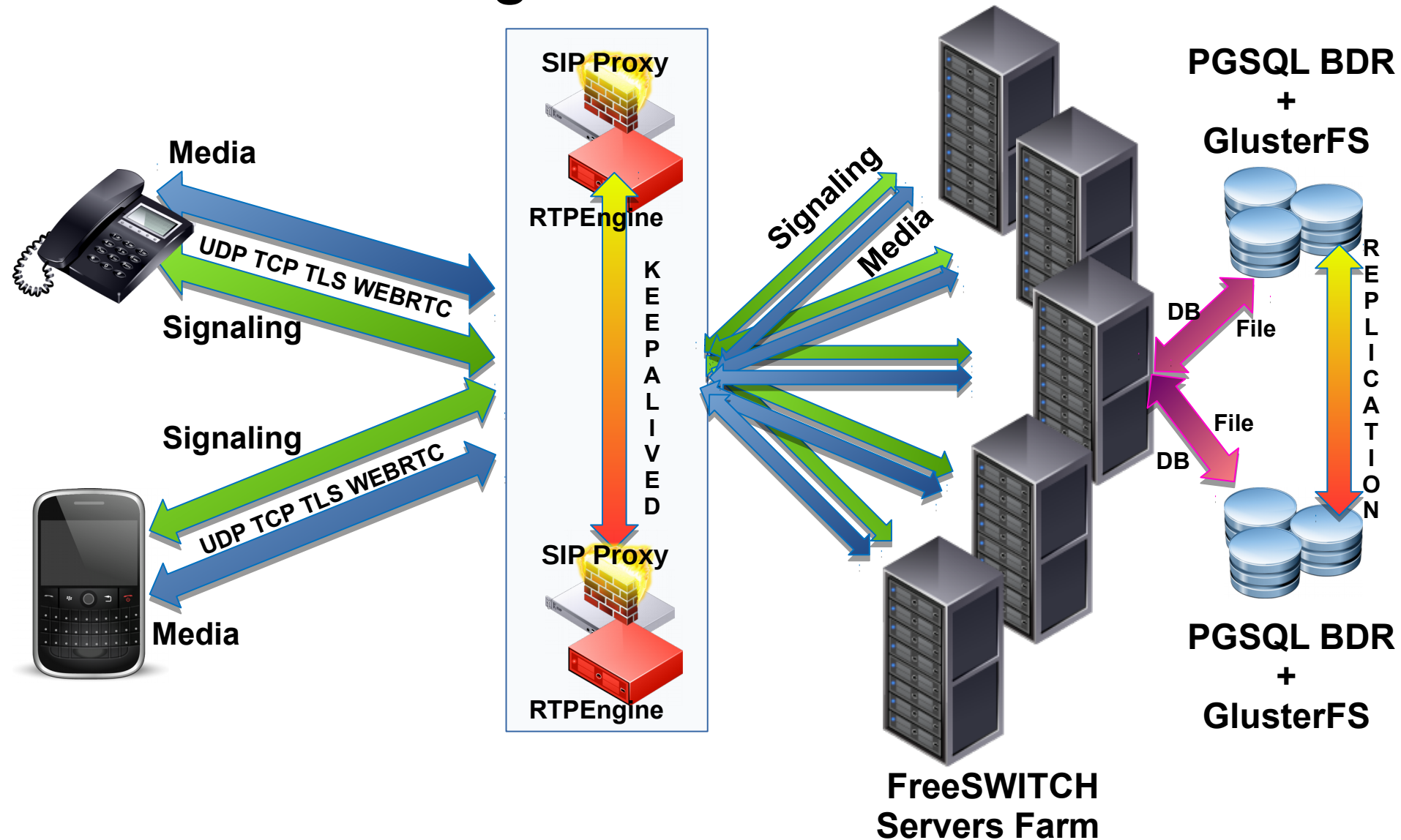
- REGISTER
- INVITE
- MESSAGE
- possibly
  - PUBLISH
  - SUBSCRIBE
  - NOTIFY

# RTPEngine Role

- take care of all ICE (eg, establish the media session with remote peer, via offer/accept of candidates)
- RTP proxy ( Internet  $\Leftrightarrow$  FreeSWITCH Farm )
- decrypt, translate SRTP AVPF to simple RTP
- decrypt, translate SRTP DTLS to simple RTP
- can transcode ( OPUS  $\Leftrightarrow$  G711  $\Leftrightarrow$  G729 )
- FreeSWITCH Farm only see "normal" media (UDP RTP)
- ( Farm already see "normal" signaling from OpenSIPS )

# SIP

## Scaling Video Conferences





# Same Conference Same FS box

- OpenSIPS by default distribute calls to all FS servers
  - dispatcher or load\_balancer
- each time OpenSIPS sends a call to an FS server, it writes down which destination "number" is going to which FS server
- while the call is up, if another call comes for the same (or related) destination then OpenSIPS sends the new call to that same FS server, bypassing default distribution algorithm

# FreeSWITCH Configuration

- Use OpenSIPS as outbound\_sip\_proxy
- Blindly Accept SIP Registrations and Calls from OpenSIPS
- 3 kind of audio/video conferences:
  - Passthrough (Lowest CPU, can't do VP8<->H264)
  - Transcode (Low CPU, can do VP8<->H264)
  - Mux (High CPU, MCU, Split Screens, CG, Transcode)
- Conference Command Profiles
  - eg: DTMF commands for participants

# FreeSWITCH ChatPlan

- Chatplan is the dialplan used by FreeSWITCH's mod\_sms
  - mod\_sms does not do Short Text Messages itself
  - it deals with all "chat" things via the chat API
    - SIP SIMPLE
    - JINGLE/XMPP CHAT
    - GSMOPEN SMS
  - it process the incoming chat message by patterns
    - then act on it
      - resend to some destination(s)
      - execute scripts
      - etc

# FreeSWITCH ChatPlan & Conferences

- FreeSWITCH Conferences have a configuration setting for
  - broadcast to all conference participants the "chat" messages that are directed to the conference extension
- Using ChatPlan is then possible to:
  - recognize especially crafted messages from participants, and execute conference management commands
  - send back to participant feedback from command execution
  - broadcast to all participants messages intended as "group chat"
  - send messages about system state (who is talking now)

# Conference Clients

- Cellphones and POTS phones
  - via SIP trunks
- SIP Phones
  - deskphones, softphones, and deskvideophones
- Browsers
  - via HTML and SIP.js JavaScript
- Smartphone Apps
  - built by Cordova from same HTML and SIP.js JavaScript



# Conference Clients

- Cellphones and POTS phones
  - will only interact via DTMF
- SIP Phones
  - will interact via DTMF and possibly via programmable keys
- Browsers
  - full interactive features, via JavaScript and SIMPLE
- Smartphone Apps
  - full interactive features, via JavaScript and SIMPLE
  - wake up via PUSH, no battery drain

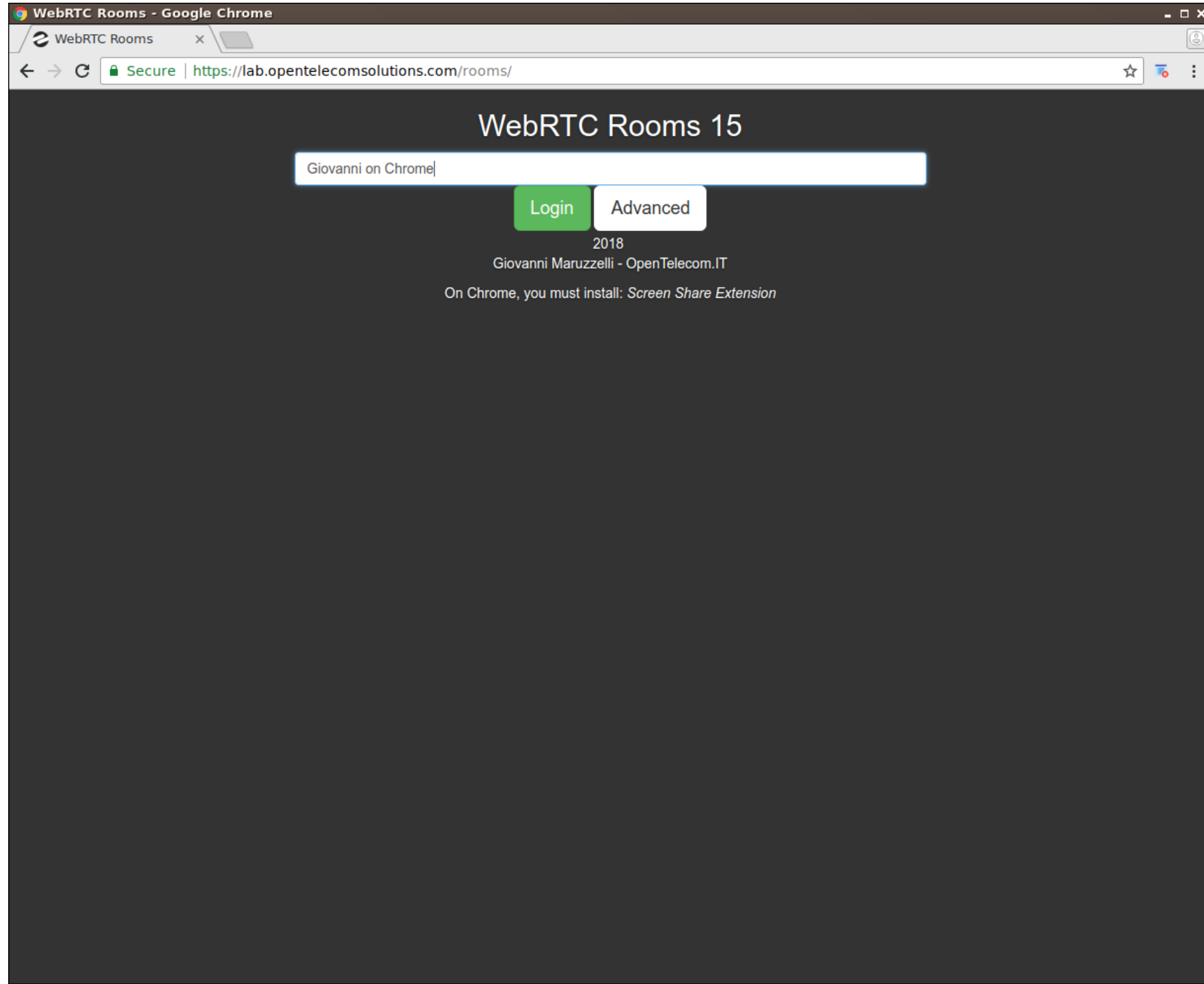
# Oh Baby



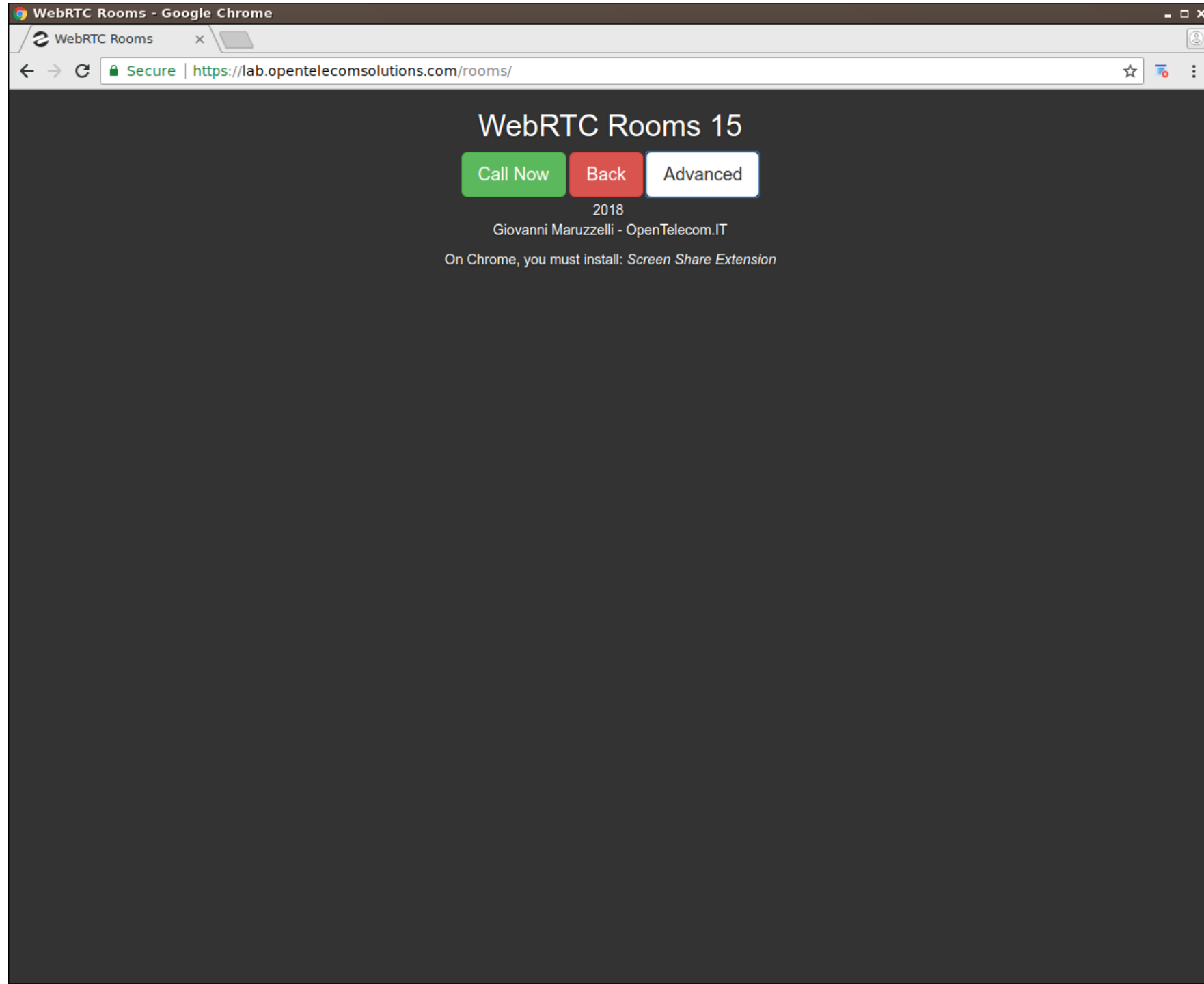
# it's Demo Time ?



image courtesy  
Michael Mavroudis



- First screen
- Web




- after name
- web

WebRTC Rooms - Google Chrome

WebRTC Rooms Secure | <https://lab.opentelecomsolutions.com/rooms/>

## WebRTC Rooms 15

Giovanni on Chrome (id: 19) is TALKING



Hangup Controls Share Unshare FullScr

type here your chat msg

Send Msg

2018  
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On Chrome, you must install: *Screen Share Extension*

- click call
- first in conference
- yes I am talking

WebRTC Rooms - Google Chrome


WebRTC Rooms Secure | <https://lab.opentelecomsolutions.com/rooms/>

## WebRTC Rooms 15


Giovanni on Chrome ▼

Refresh List Floor Mute VMute Hup

overlaps ▼ Layout



Giovanni on Chrome (id: 19) is TALKING



Hangup Controls Share Unshare FullScr

23:38 Giovanni on Chrome> ciao bellissimo

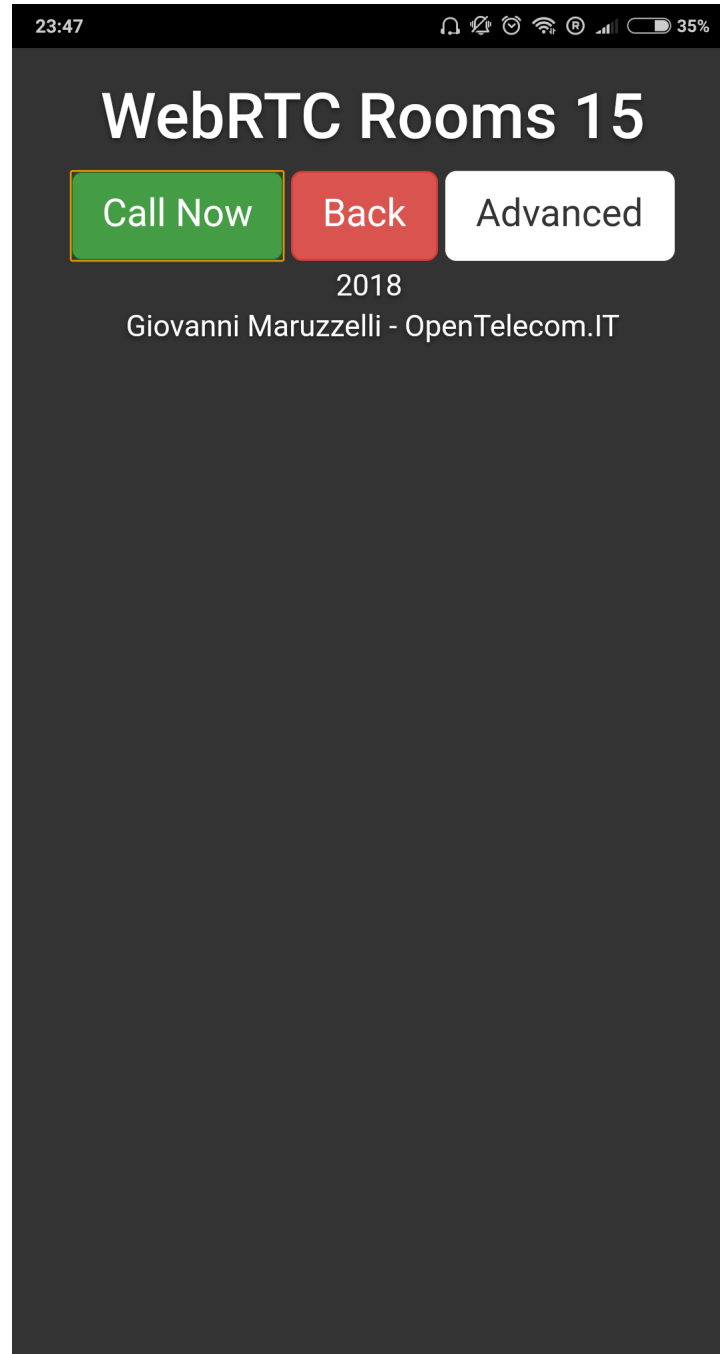
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Send Msg

2018

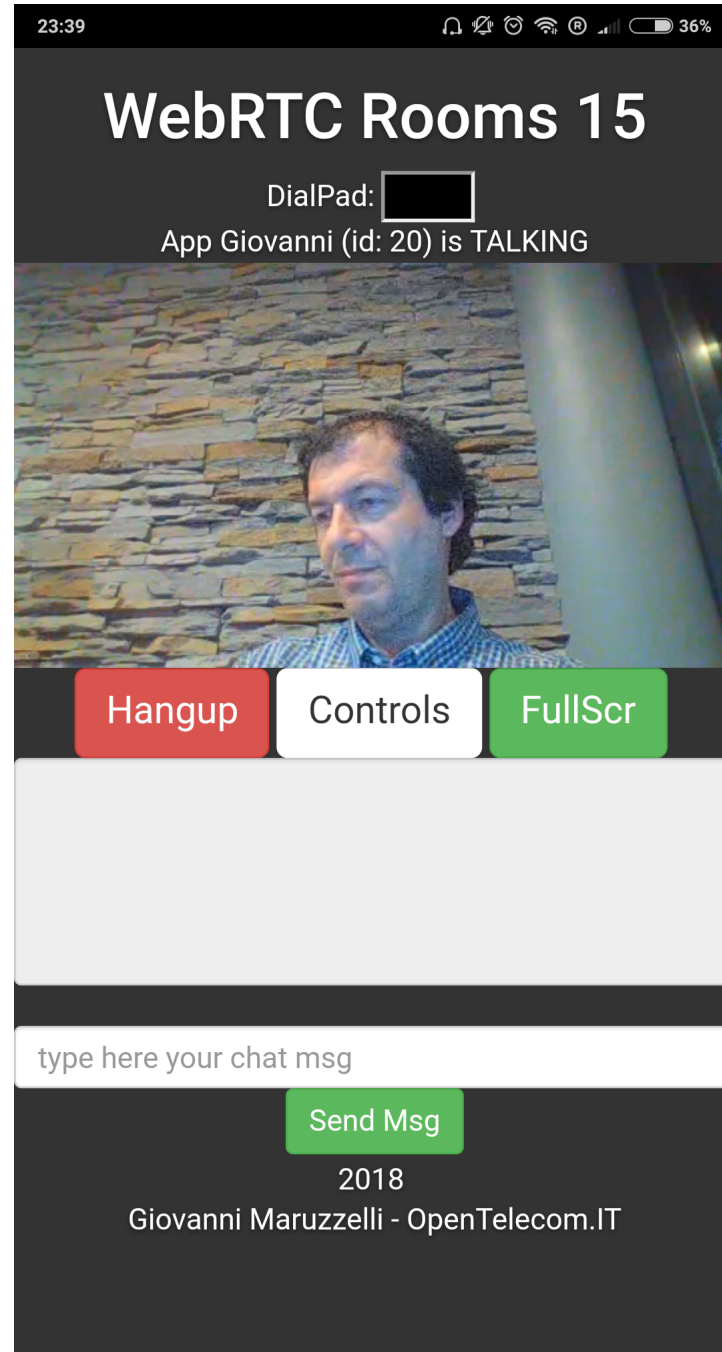
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- and
- chatting
- and using controls
- moderator
- choose other peers in list
- to mute, kick, floor, etc

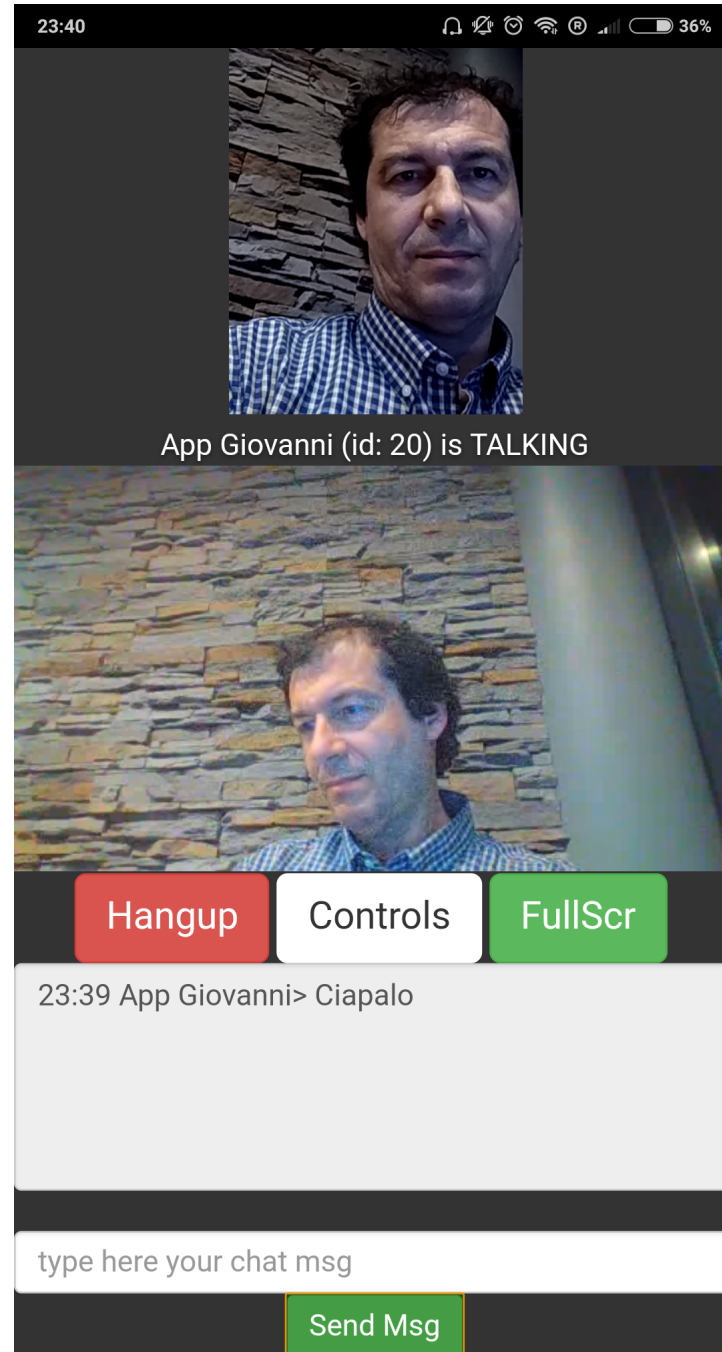


- this is
- the app





- click call
- I am in the
- conference



- and I get
- the same controls
- and the chat
- windows

WebRTC Rooms - Google Chrome


WebRTC Rooms Secure | <https://lab.opentelecomsolutions.com/rooms/>

## WebRTC Rooms 15


App Giovanni

Refresh List Floor Mute VMute Hup

overlaps Layout



App Giovanni (id: 20) is TALKING



Hangup Controls Share Unshare FullScr

23:38 Giovanni on Chrome> ciao bellissimo  
23:39 App Giovanni> Ciapalo

type here your chat msg

Send Msg

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- yes, we
- actually communicate

WebRTC Rooms - Google Chrome


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## WebRTC Rooms 15


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Refresh List Floor Mute VMute Hup

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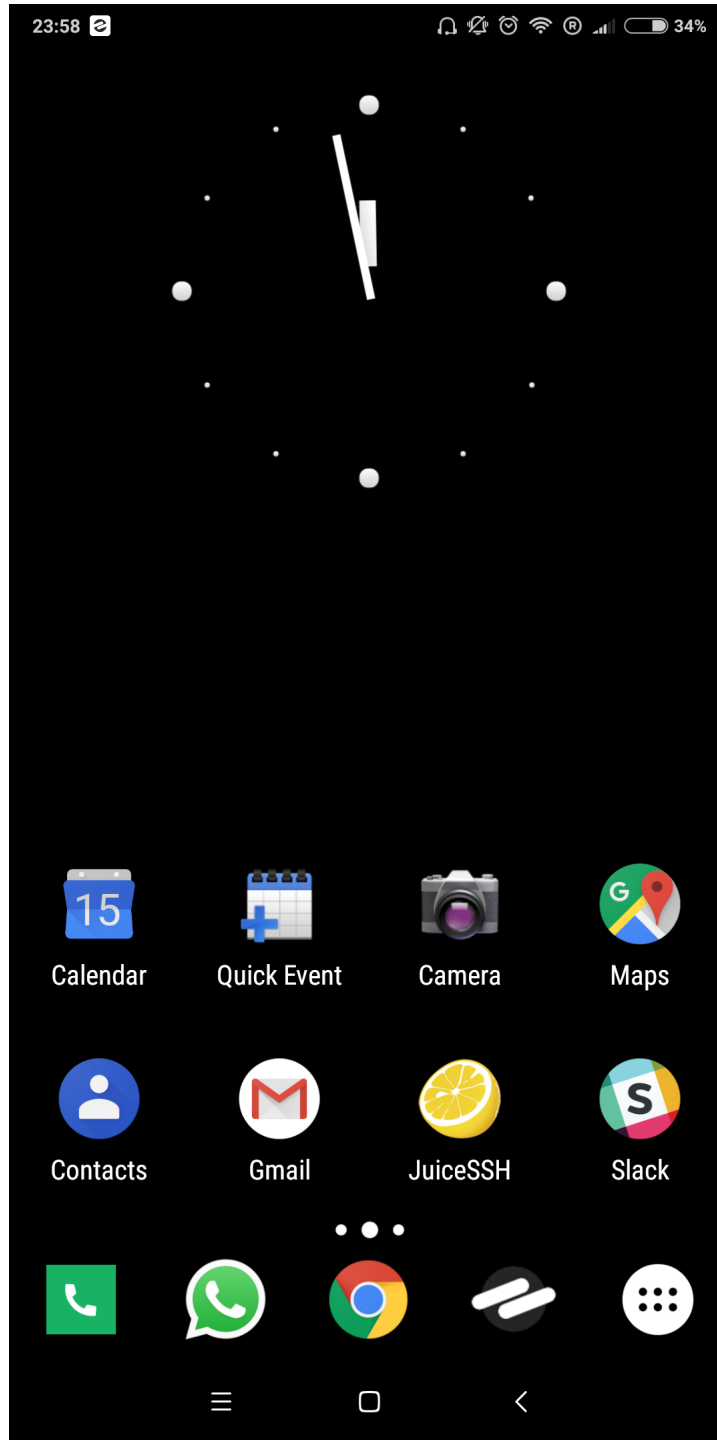
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type here your chat msg

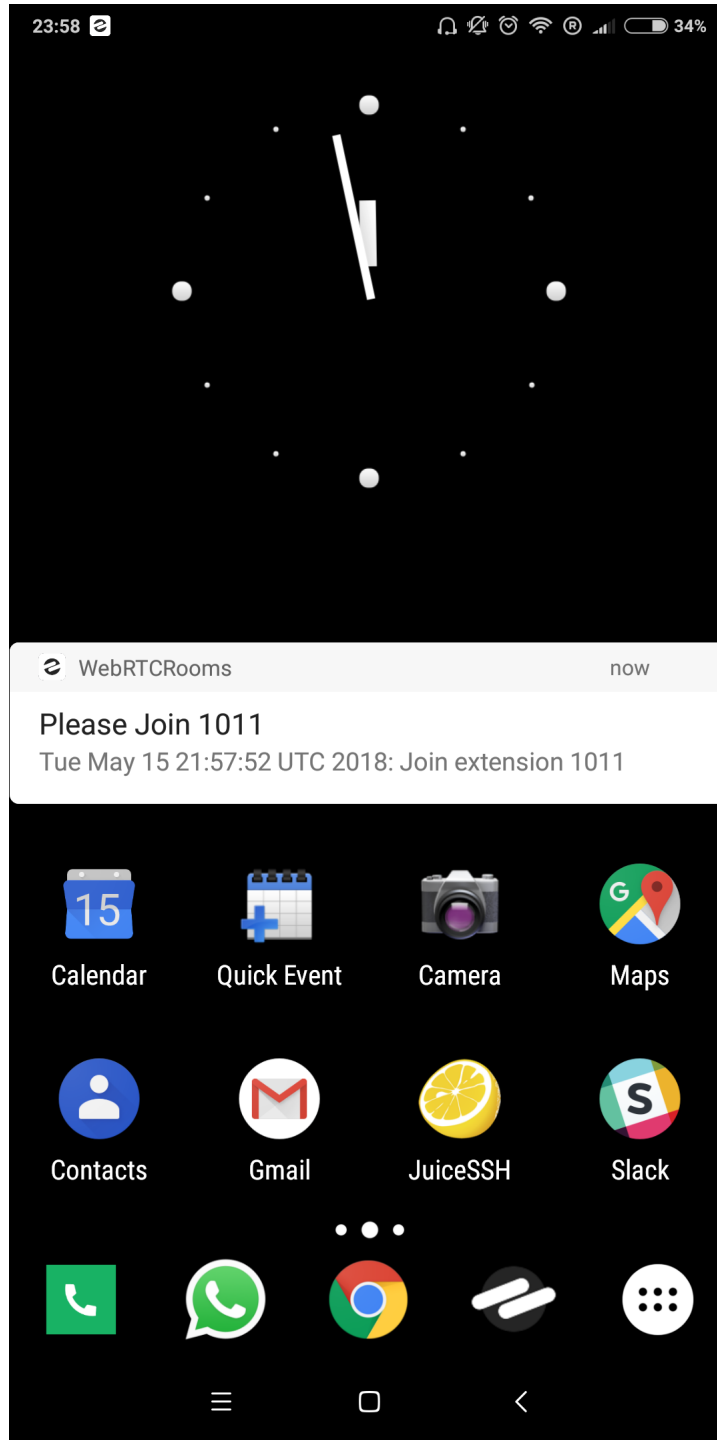
Send Msg

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- let's change
- who has the floor



- now I shut down
- the app
- nothing is running
- but I send a call
- from another client
- and a push notification
- is coming



- here we are
- click on push message
- and the app connect
- the call



# it's actually, Demo Time for real ?



image courtesy  
Michael Mavroudis

# Demo Link

<https://bit.ly/2lhGCts>



Thank You

**QUESTIONS ?**

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*consulting & training*