Using Real-time Audio to Replace Telephone Call-ins On-Air

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What’s the Problem?

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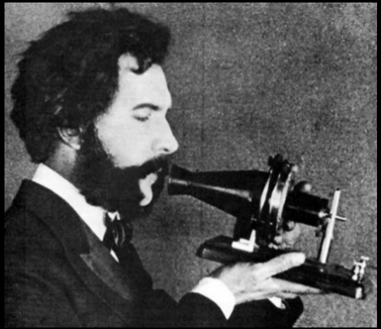
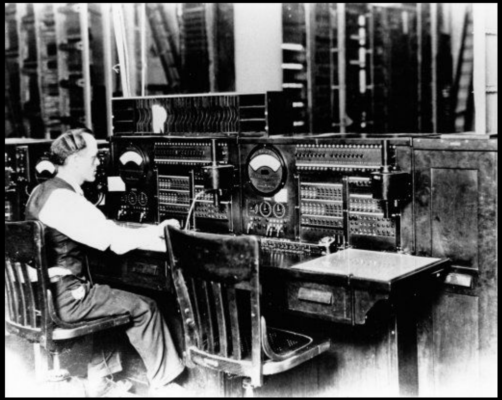
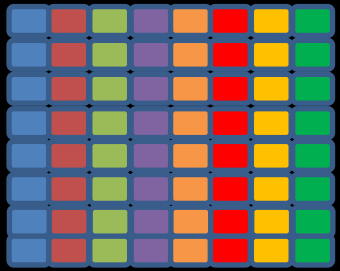


- The telephone network was not engineered for quality
- For economy and simplicity, voice was filtered
- Low frequencies were filtered to reduce hum
- Digitization of voice calls began in the 1960s



- Eight Bit Samples
- Samples taken 8000 times per second
- Voice channel = 8 x 8000 = 64 kb/s



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- The telephone network samples voice at 8KHz
- Nyquist Rate is 4 KHz
- Due to filtering required, actual bandwidth is lower
- Typical phone is 300-3000Hz

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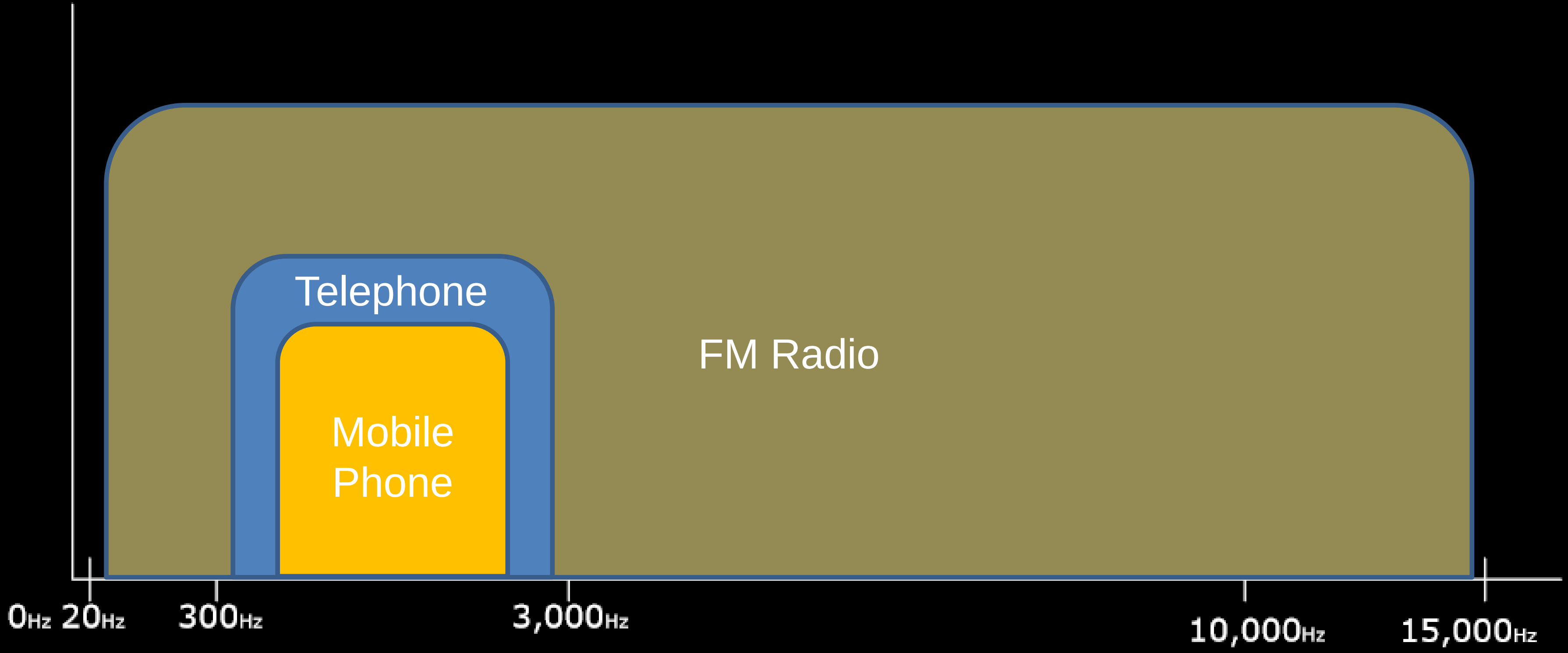

Encoders

- G.711—traditional telephone
- GSM, AMR– Mobile Phones
- G.722--- First widely used wideband encoder
- MP3, AAC
- HE-AAC, AAC-ELD

Opus

- No patent obligations
- Can be provided in free apps
- Voice and non-Voice sub-modes
- Low delay
- Wide network operating range

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Smart Phones/apps

The Browser

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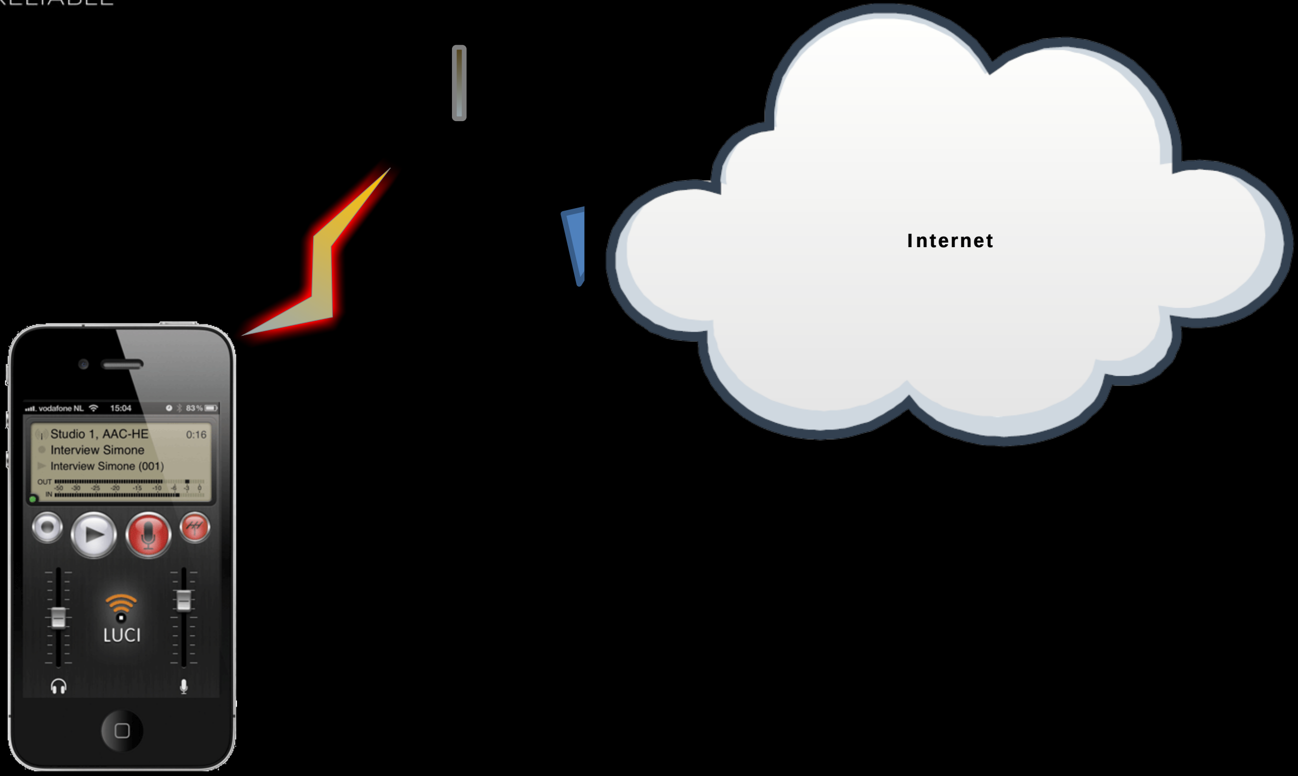

LUCI Live

- Pro Grade Softphone
- Real Support
- HE-AAC
- Android and iOS

Session Initialization Protocol

- Common Protocol for VoIP
- Based on RFC standards
- Flexible and expandable
- Basis for broadcast interoperability

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Non-pro-grade option?

- Android and iOS
- SIP-based
- Free
- Actively maintained
- Low delay wideband audio (Opus)

Linphone

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The Mobile Phone

The Browser

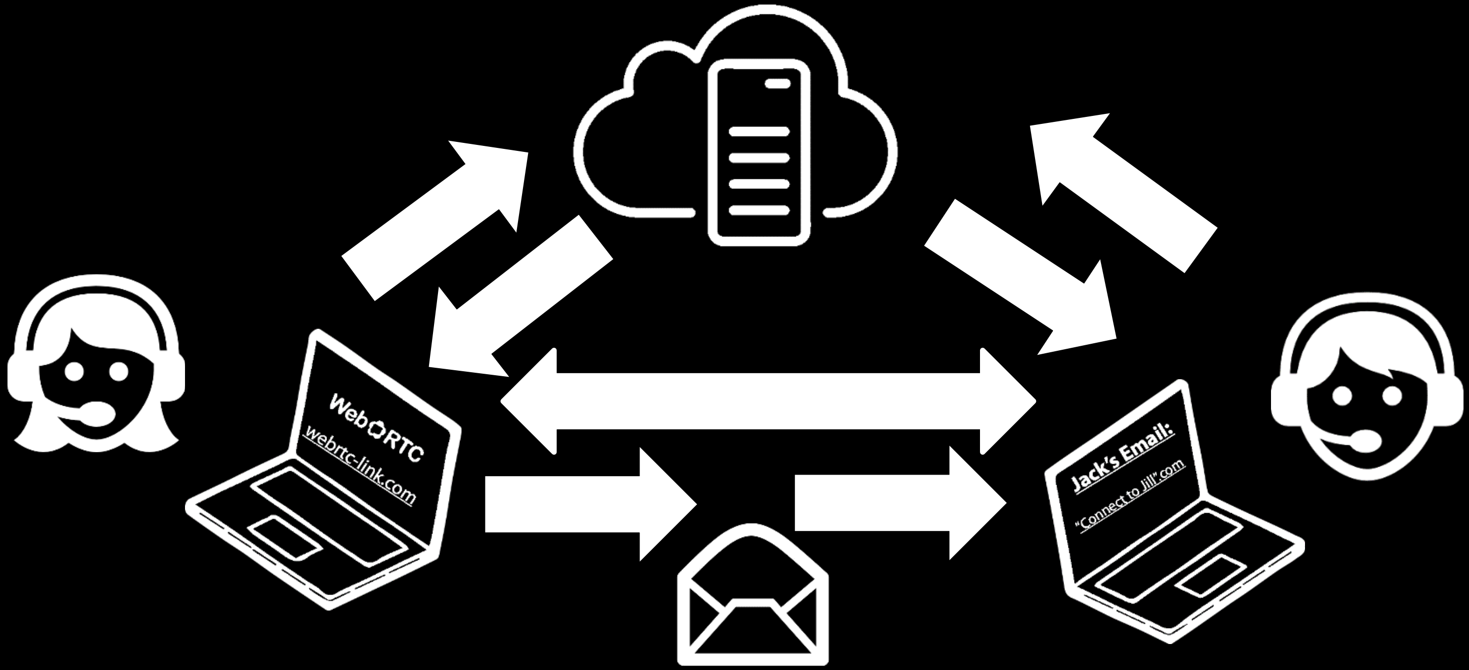
WebRTC Basics

- Designed for video conferencing
- Based on open standards
- Encoders built in to web browsers
- Connections done via javascript applet

WebRTC Basics (cont)

- Support growing
- Chrome, Firefox, Opera, Edge
- Safari (announced)
- Supports a/v or audio only
- Uses Opus encoder

WebRTC Connection



Free WebRTC Servers

- DIY
- Airtime.com (formerly Vline.com)
- Source Connect Now
- ipDTL
- Mozilla Firefox Hello (depreciated)
- Talky: Provides both video and screen sharing using WebRTC.
- Cisco Spark: Create rooms for video calling, group messaging and sharing.
- Appear.in: Group video calling for up to eight people.
- Jitsi Meet: Group video calling and screen sharing using WebRTC.

WebRTC

Studio Side

- Capable of taking calls via VoIP (Bridge to PSTN)
- Capable of taking SIP wideband calls from apps (Opus)
- Capable of taking calls from WebRTC

Studio Side


SEND
AUX/NEXT
DROP ALL
CHANGE PRIORITY
AUTO ATTENDANT
COMREX
STAC VIP
ANSWER HOLD
ON AIR
DROP
COMREX
HYBRID 1
SEND
CALLER
HYBRID 2
STAC VIP
RESET

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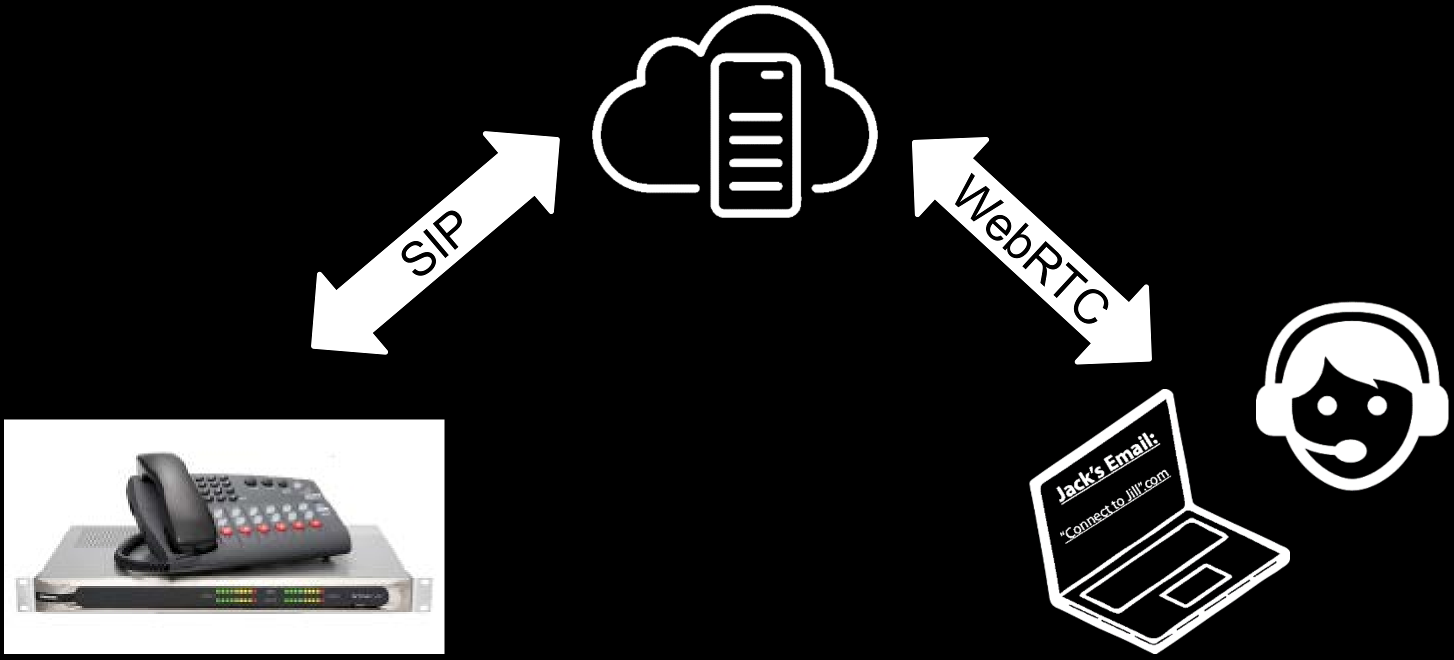
WebRTC/SIP Connection



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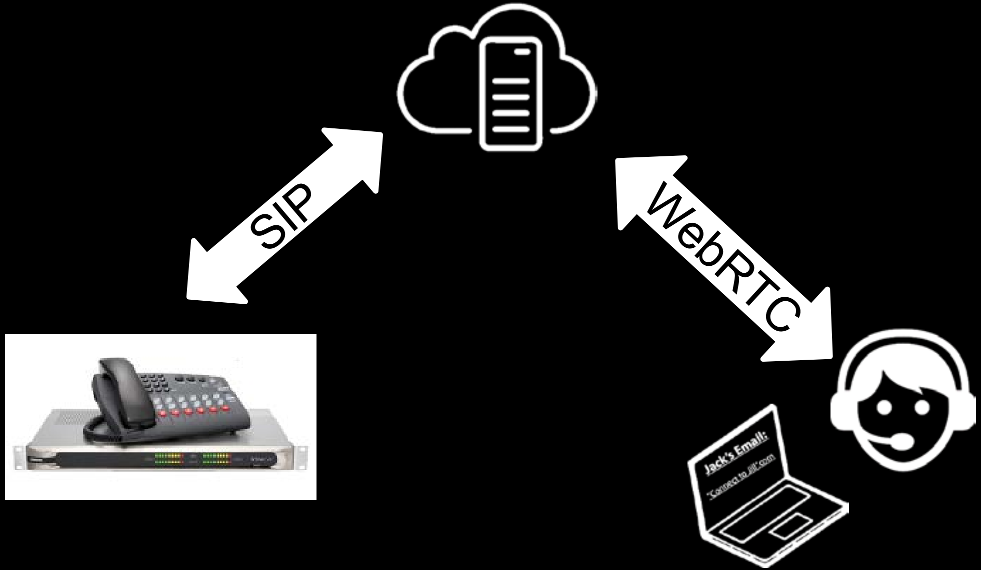
WebRTC/SIP Bridging

- DIY
- ipDTL
- Callme.fm
- GetOnsip.com



Opal IP Audio Gateway

Cutting Out The Middle Man

1. Reduces Delay - Improves Connectivity
2. Provides SSL security transparent to user
3. Eliminates need for guest configuration or S/W install
4. Free Apps and common web browsers

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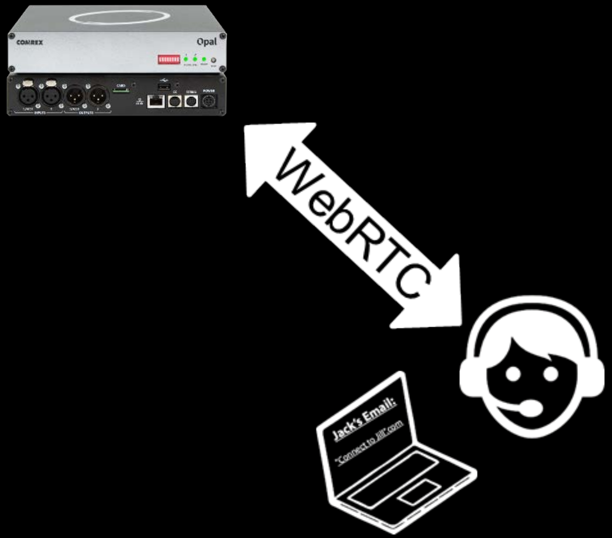
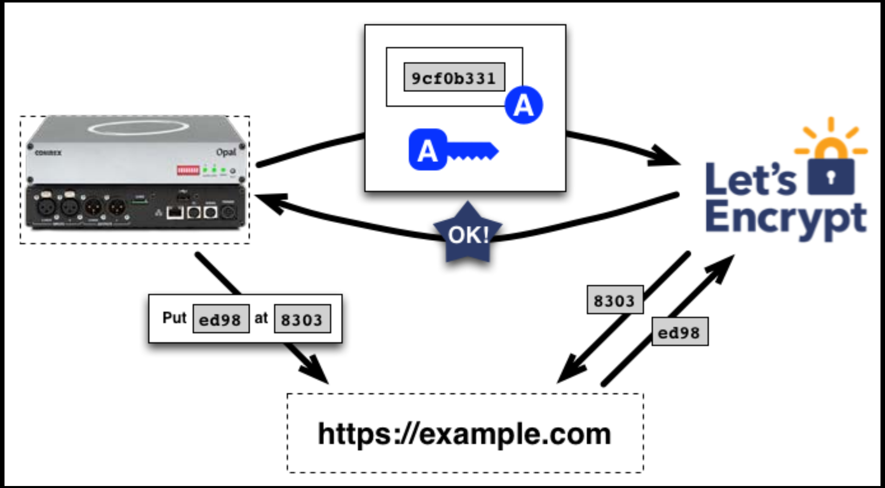
Cutting Out The Middle Man

1. **Reduces Delay - Improves Connectivity**



Cutting Out The Middle Man

2. **Provides SSL security transparent to user**



Cutting Out The Middle Man

3. **Eliminates need for guest configuration or S/W install**

Cutting Out The Middle Man

4. **Free Apps and common web browsers**

https://opalatl.com/caller/?t=n9mfGas1zR0GSZ92

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- It’s time to reduce telephone audio on the radio

- Opus offers new ways to integrate web audio

- New Free apps allow easy mobile phone connections

- WebRTC allows leverage of audio codecs built into browsers
- The Opus Codec
- WebRTC for ACCESS & BRIC-Link
- WebRTC Primer