Audiosync Final Presentation

by Simon Grätzer & Jan Garcia

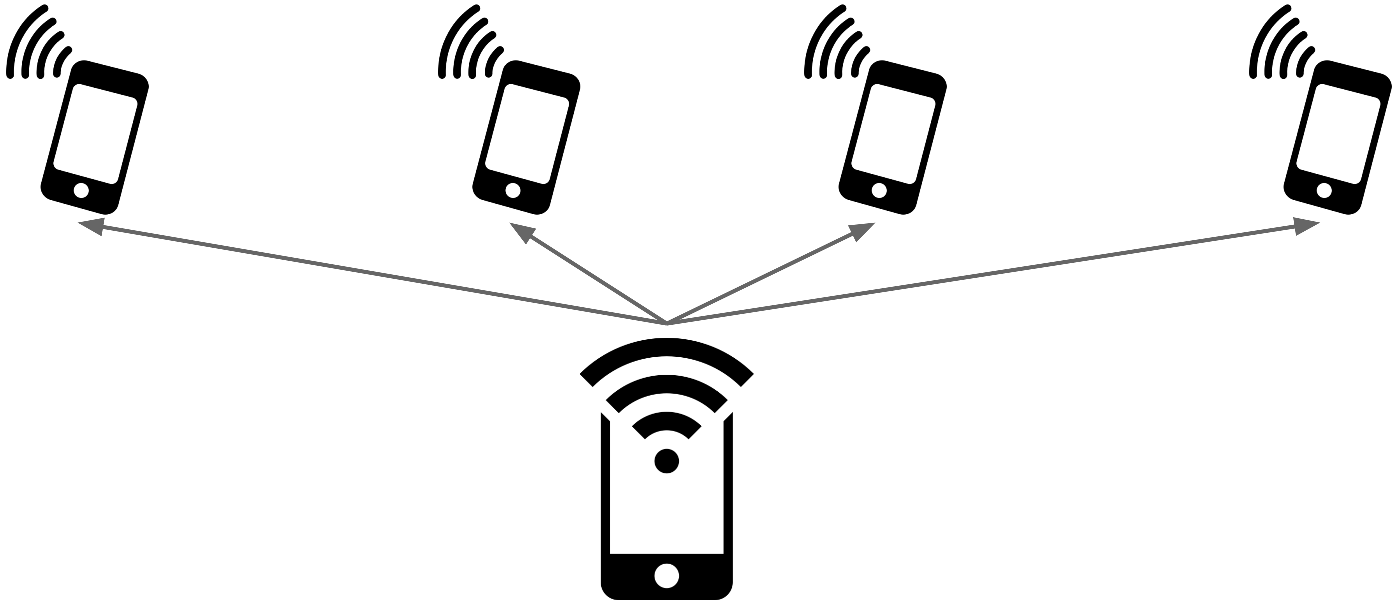
Organization

- Goals
- Sync. audio streaming easy?
- Development timeline
- Software architecture
- Network protocol
- Future work
- Demo

Goals

- Synchronous audio playback on multiple devices
- Wifi-Streaming
- One sender, many receivers
- Support for heterogeneous receivers
- No setup required

Sync. audio streaming easy?



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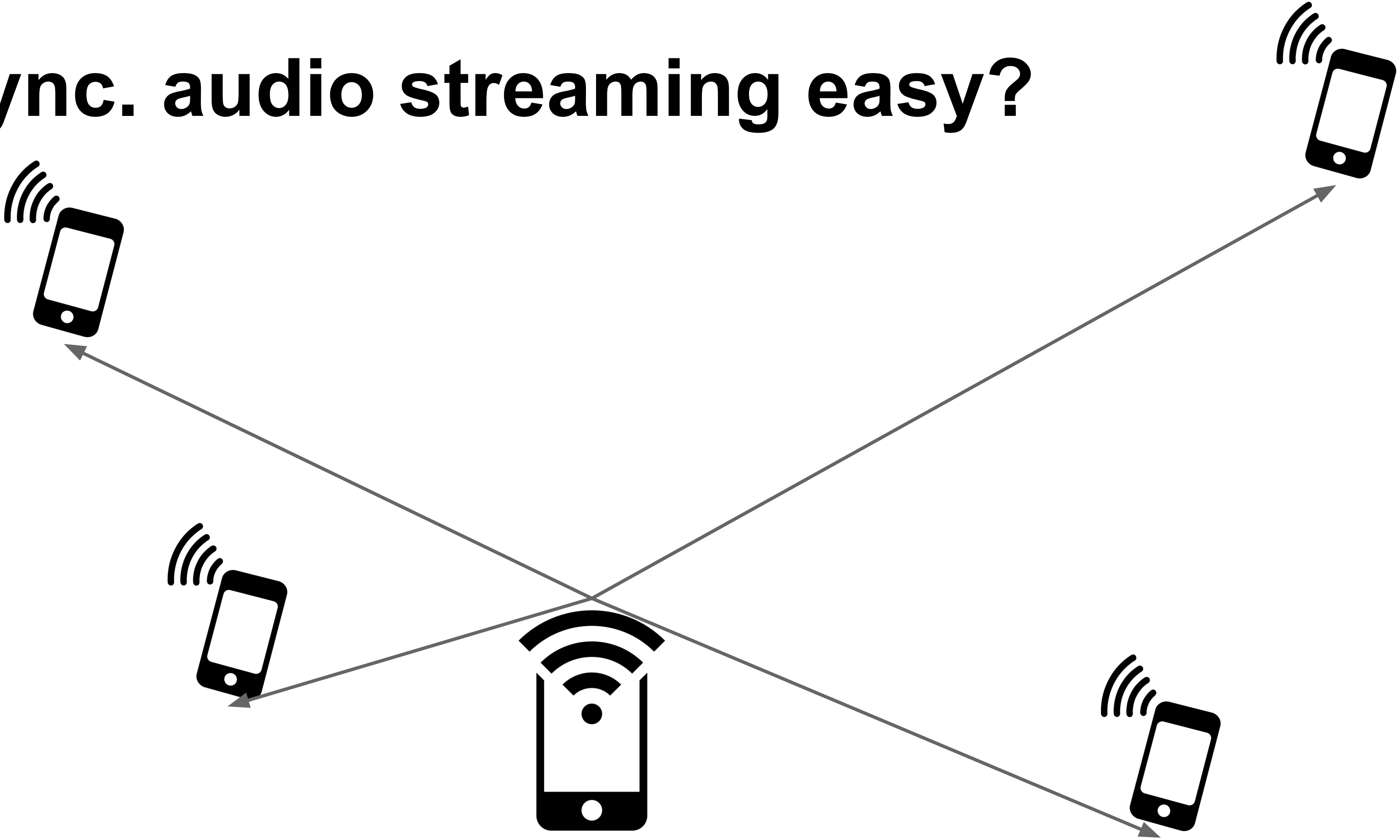

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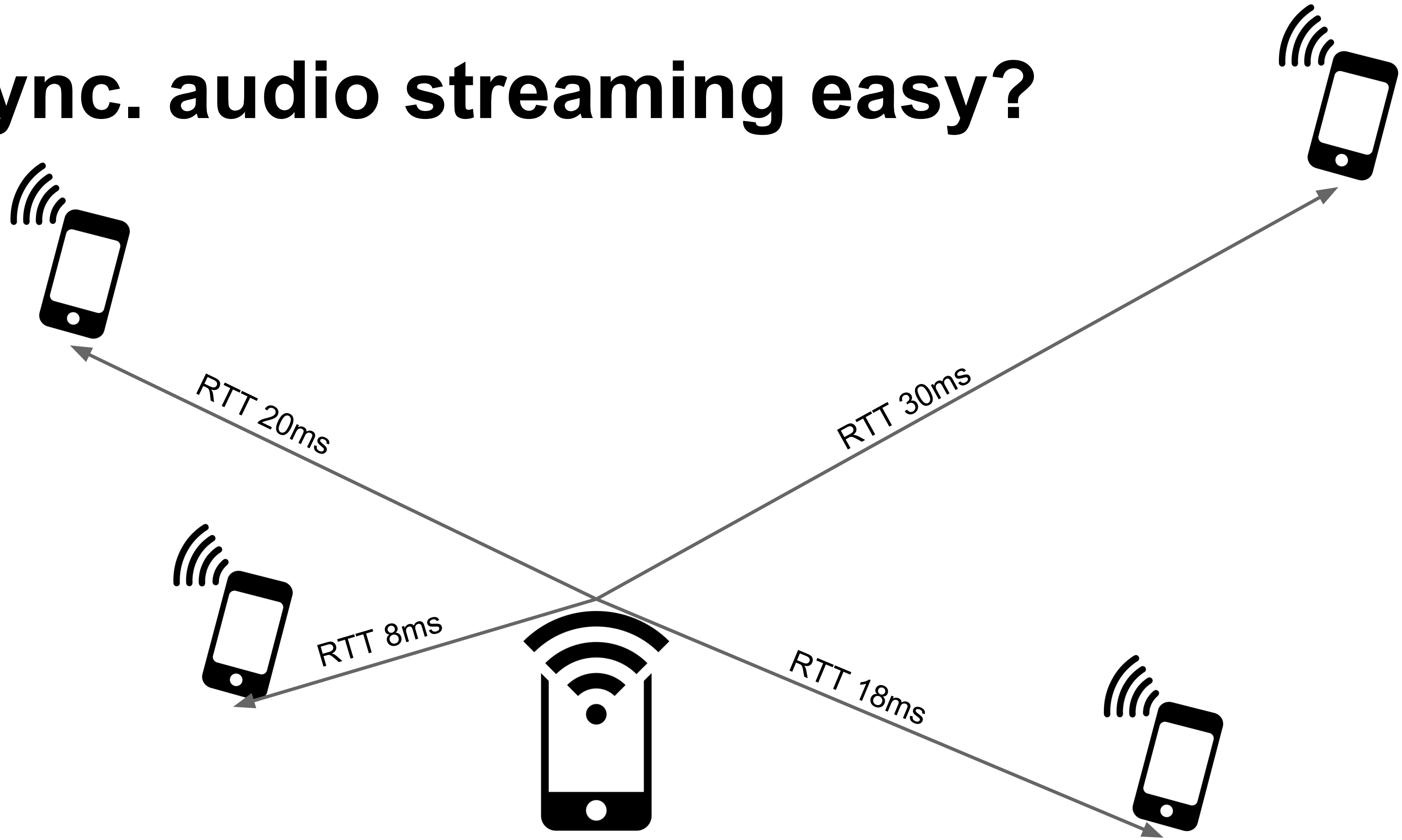

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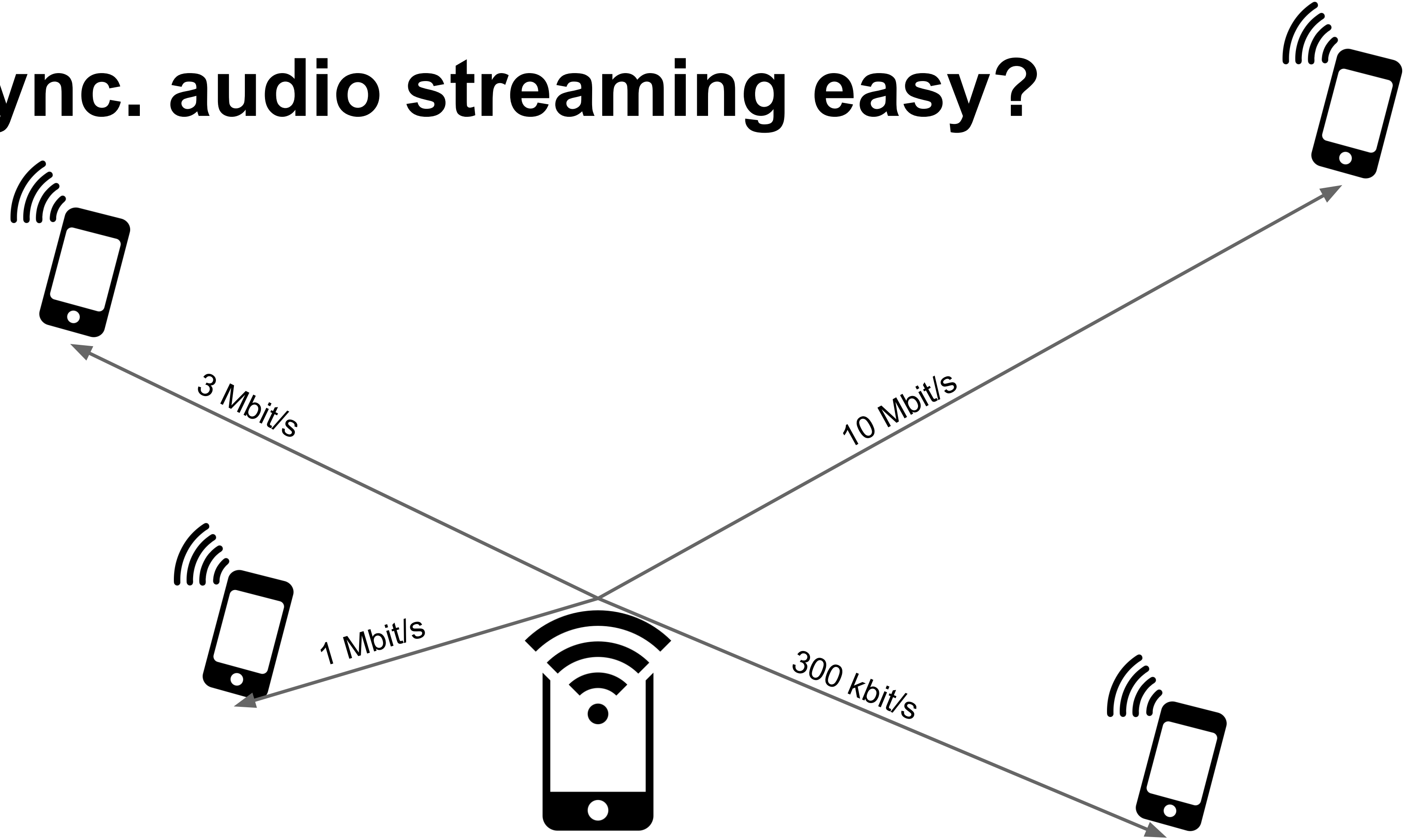

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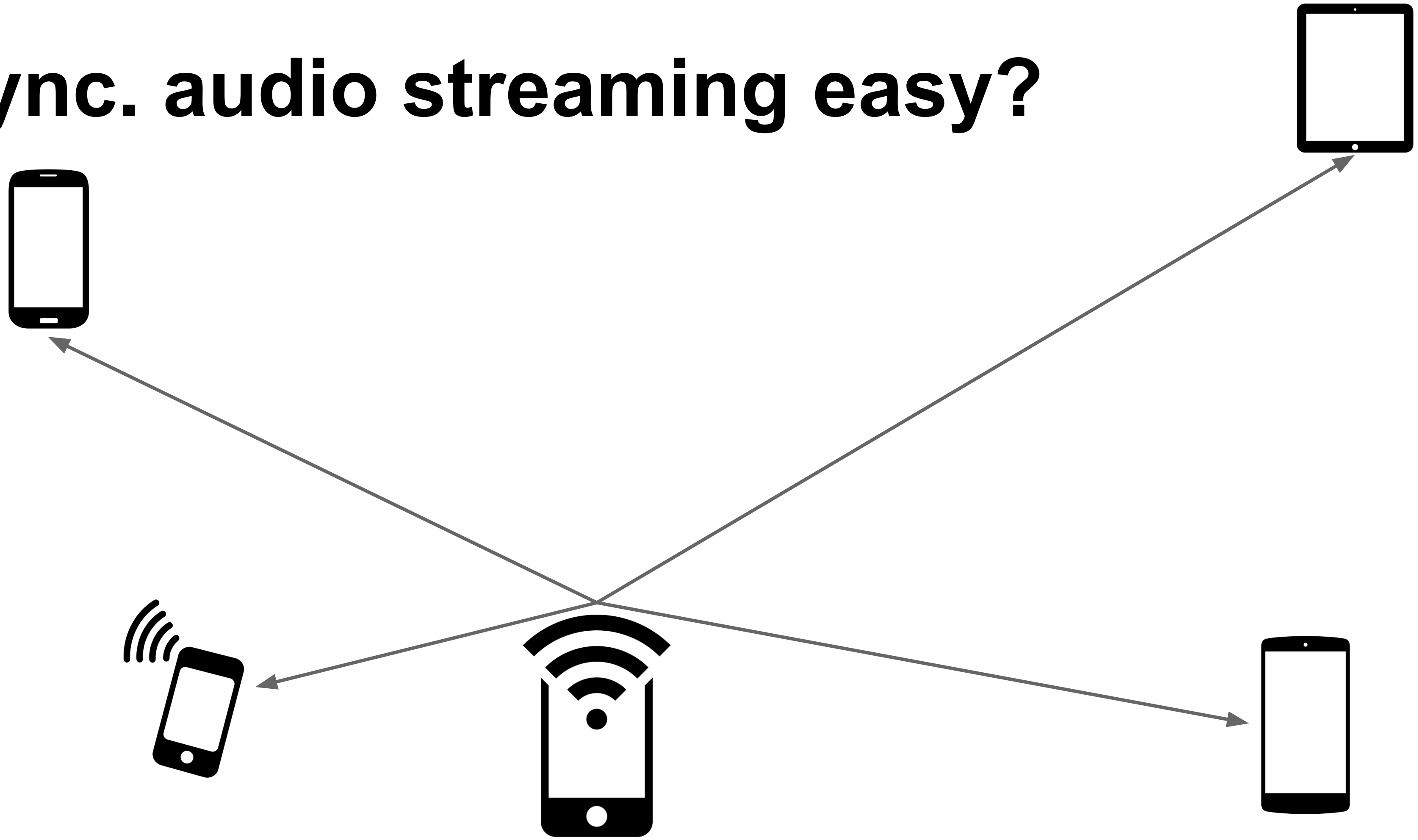

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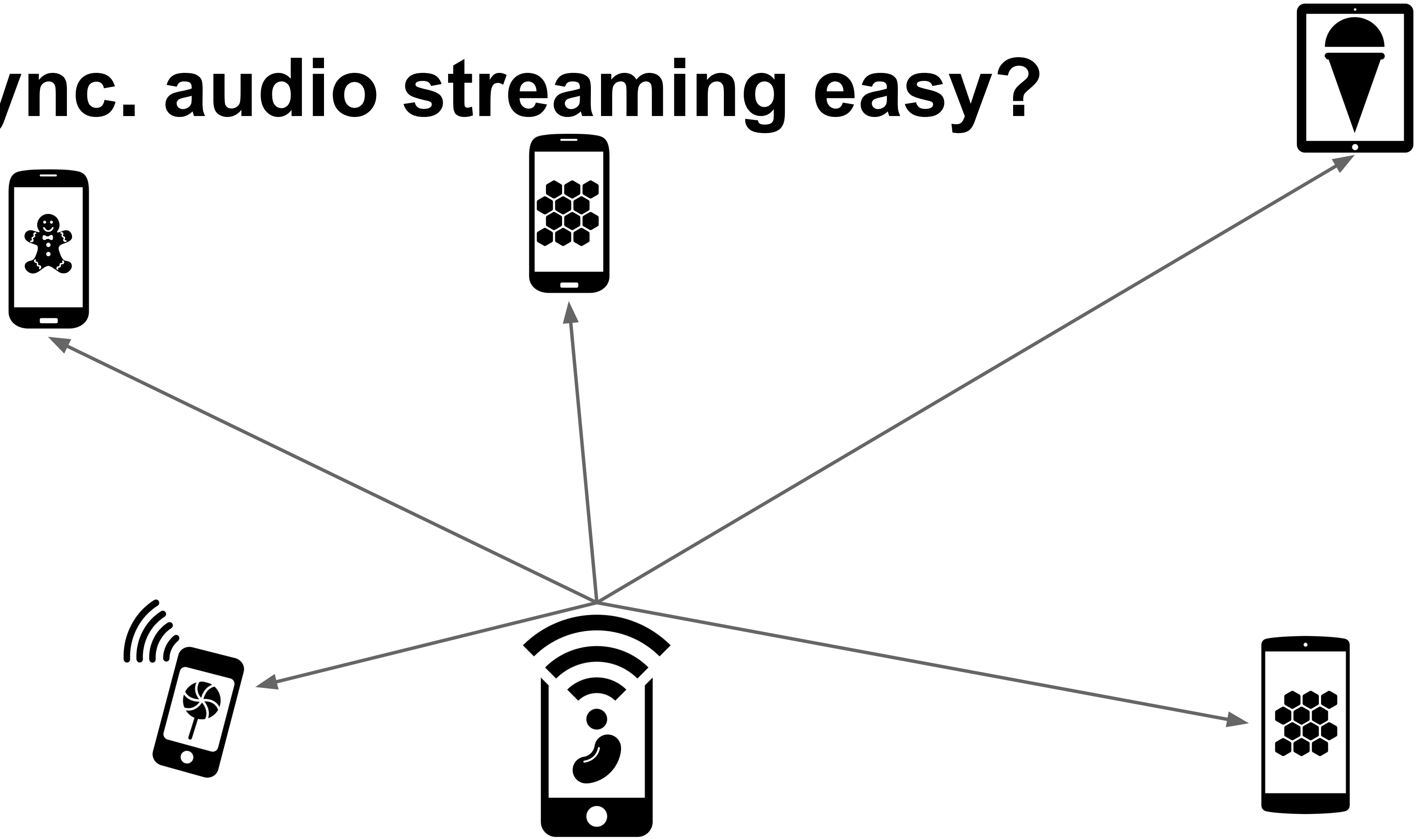

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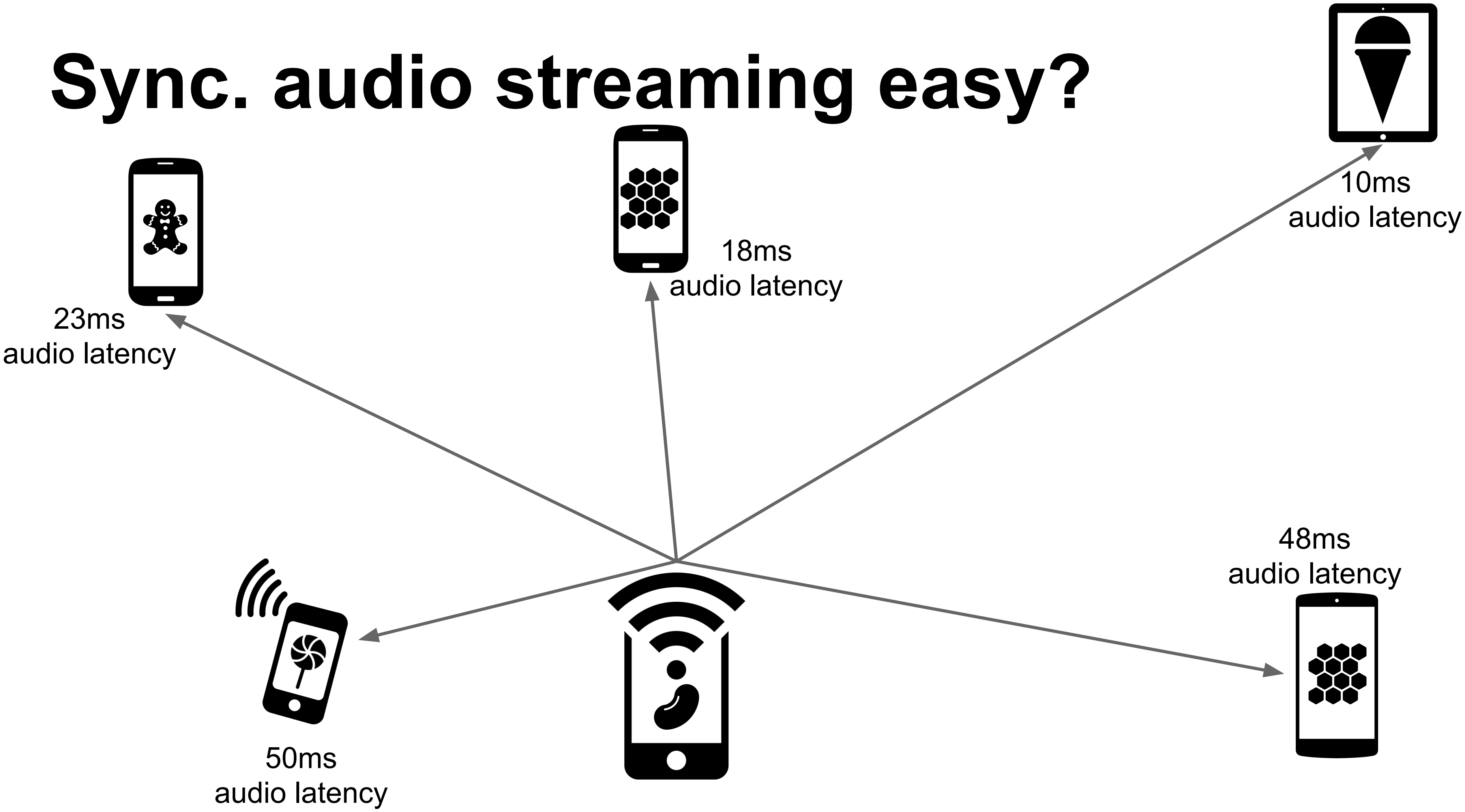

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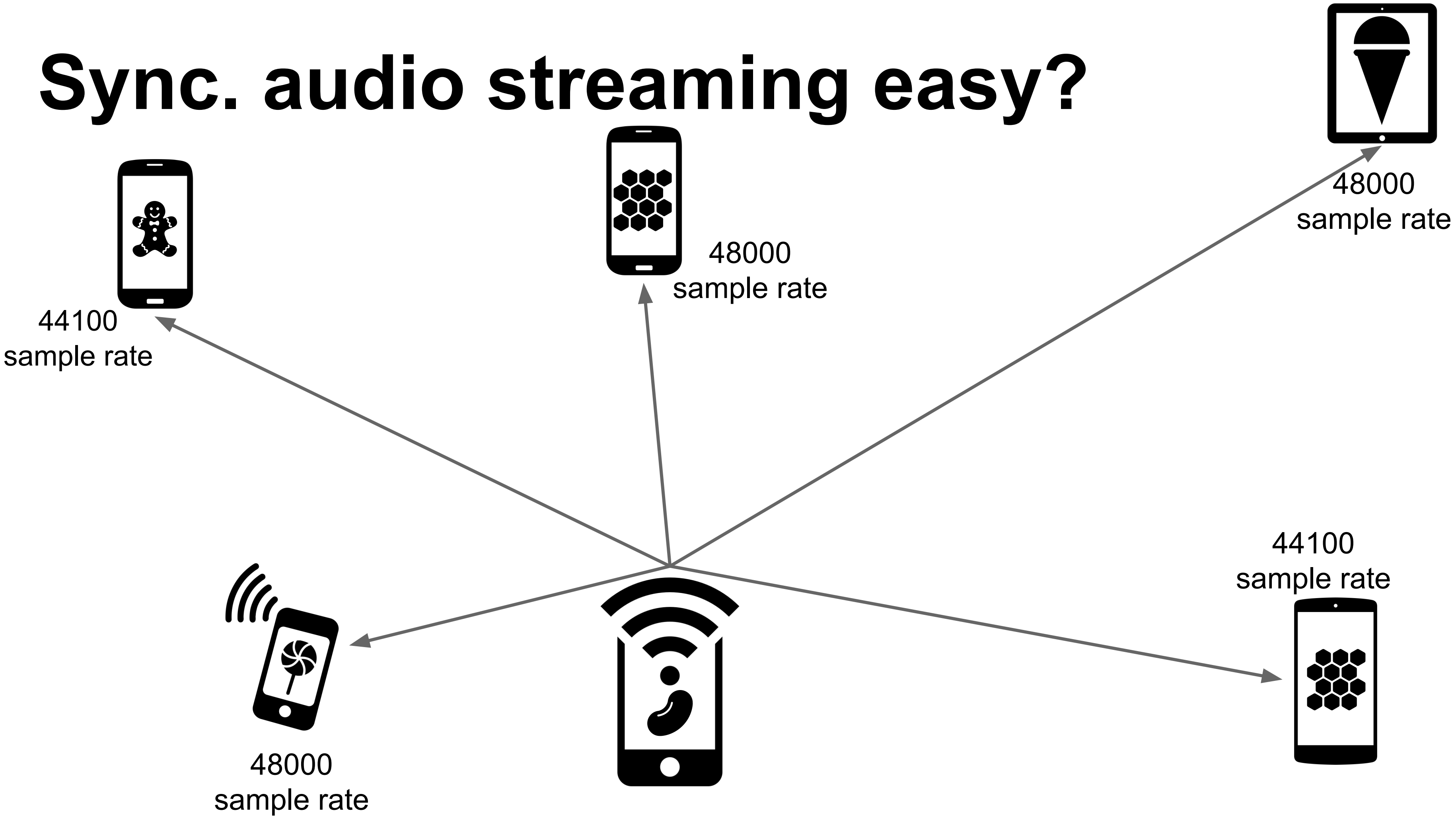
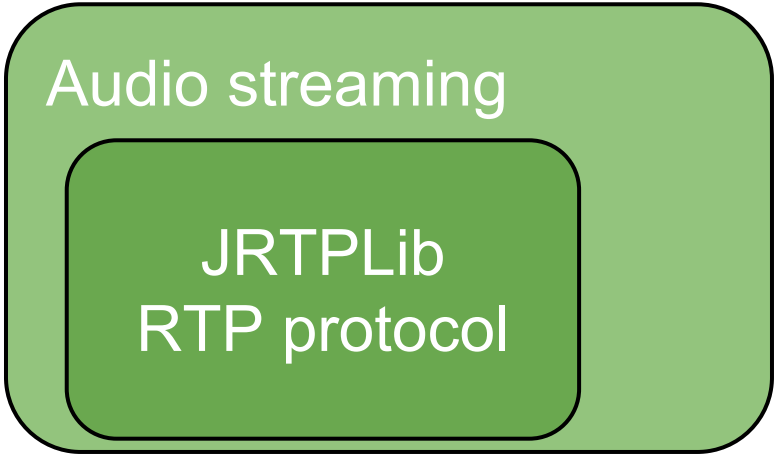
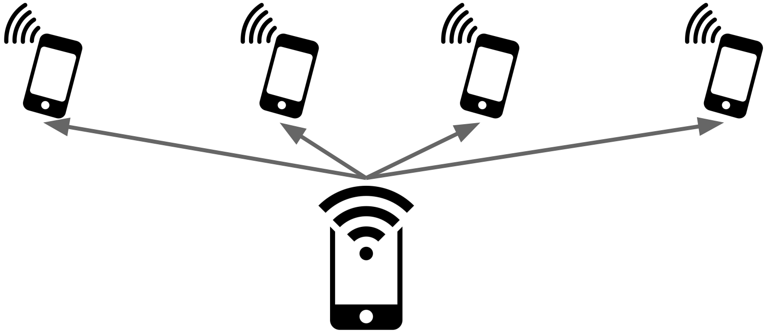


Step 1: Streaming some audio

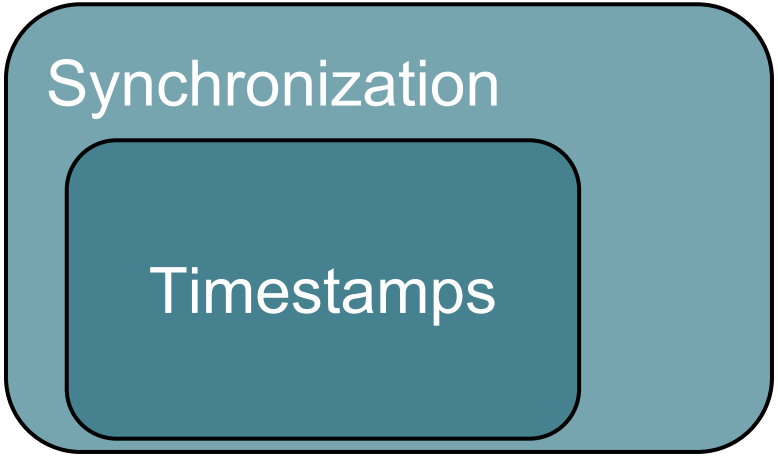
- Wifi broadcast
  - Too slow, 1 MBit/s
- Unicast, stream simultaneously
  - Existing protocol: RTP

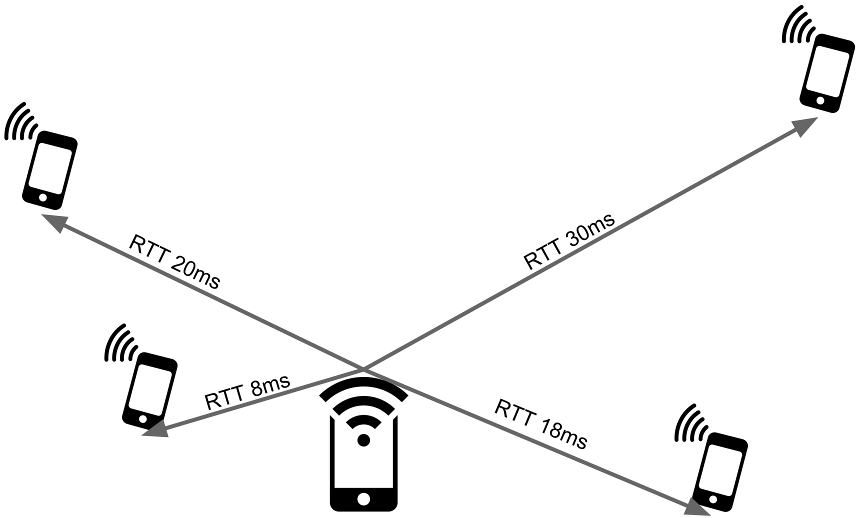




Step 2: Synchronize playback times

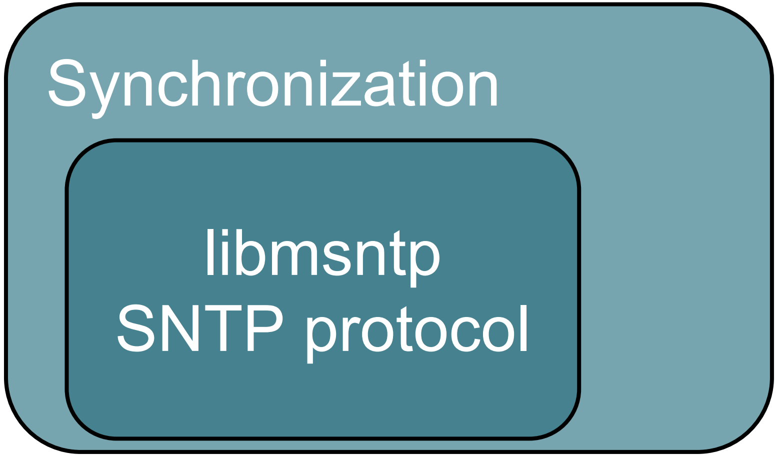
- Extend RTP protocol (extension header)
- Audio packets have timestamps
    - Time when to start playing packet

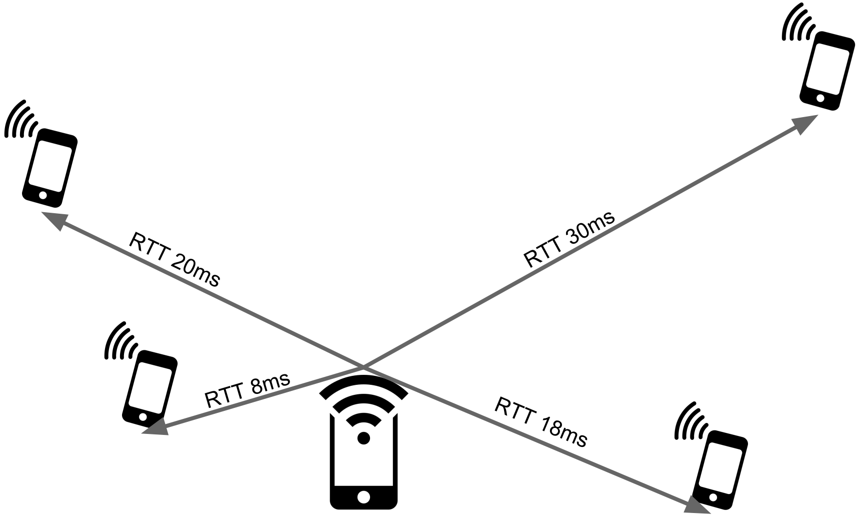




Step 3: Synchronize device times

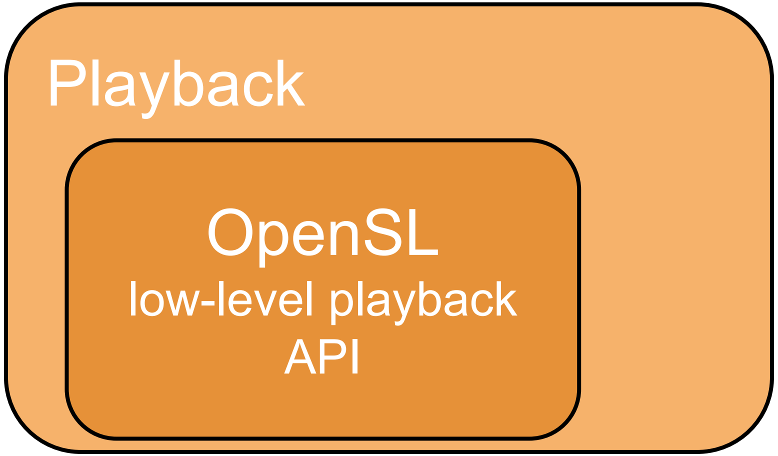
- Existing solution: NTP protocol
- We use a simplified version
  - SNTP

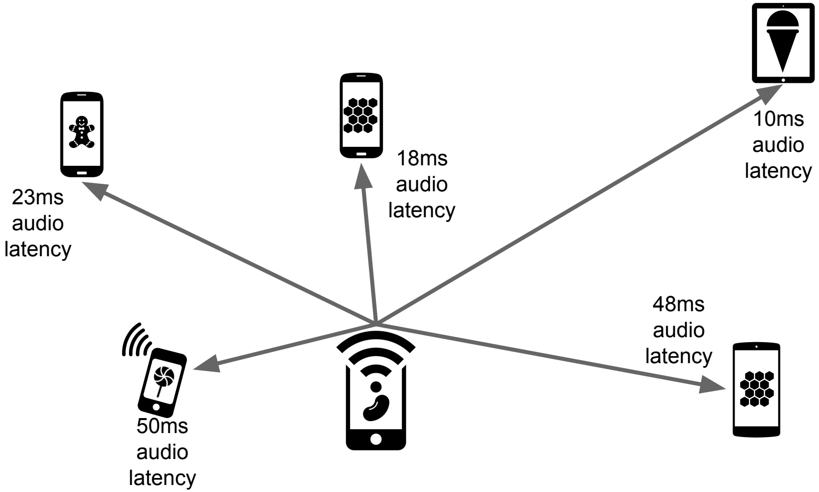




Step 4: Reduce playback latency

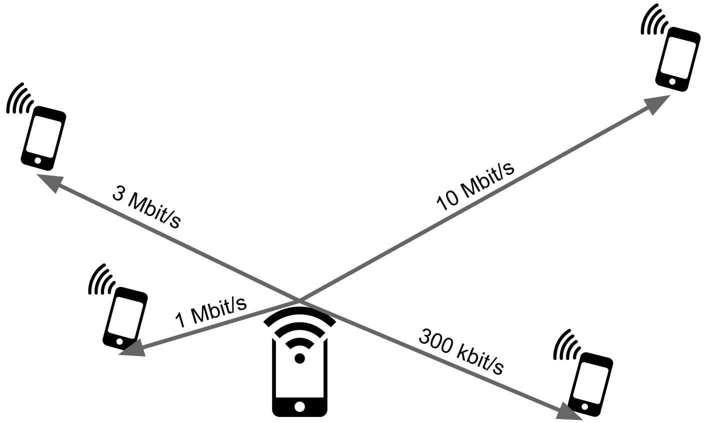
- High level APIs
  - easy to use, higher latency
- Low level APIs
  - harder to use, low latency



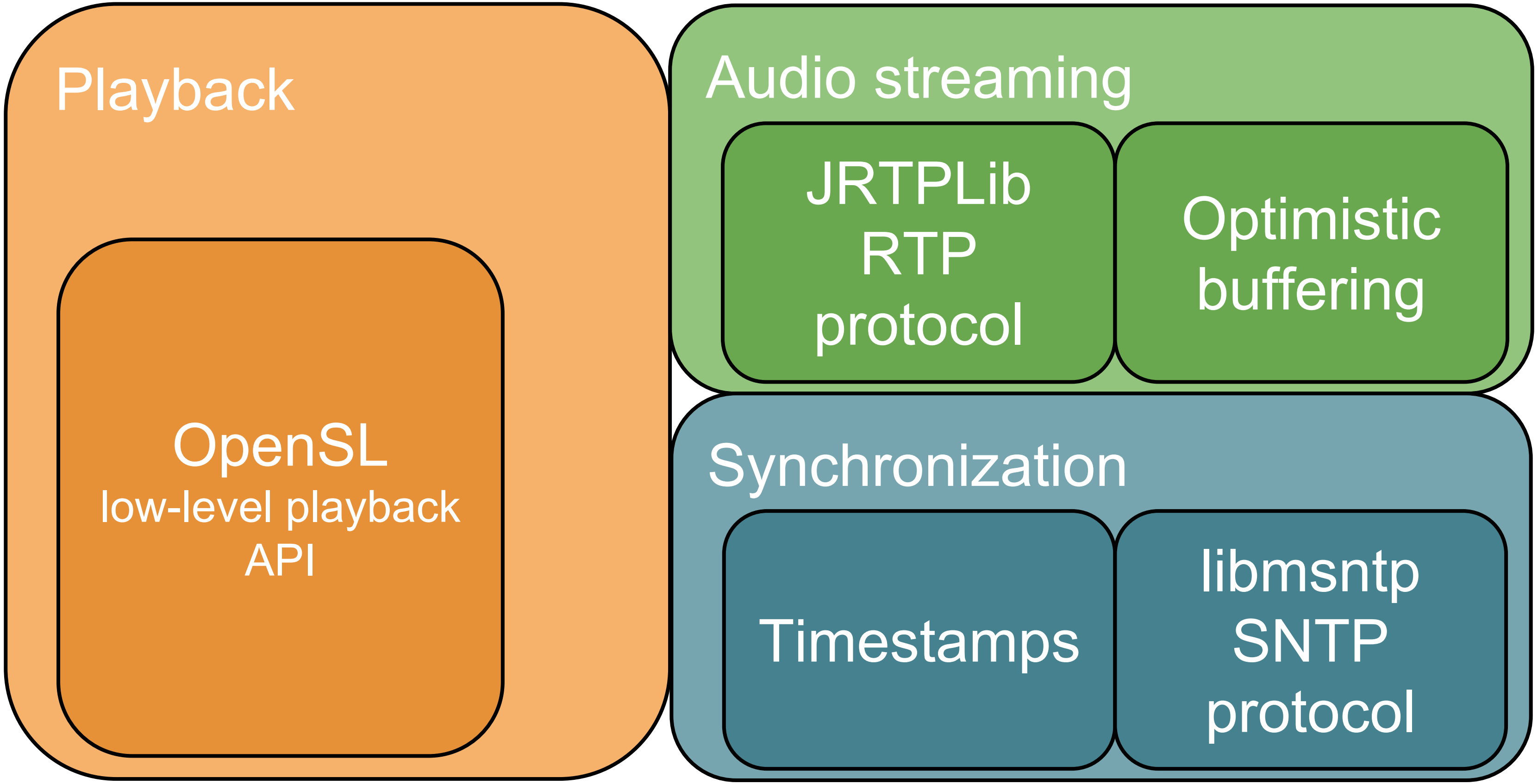


Step 5: Regard individual data rates

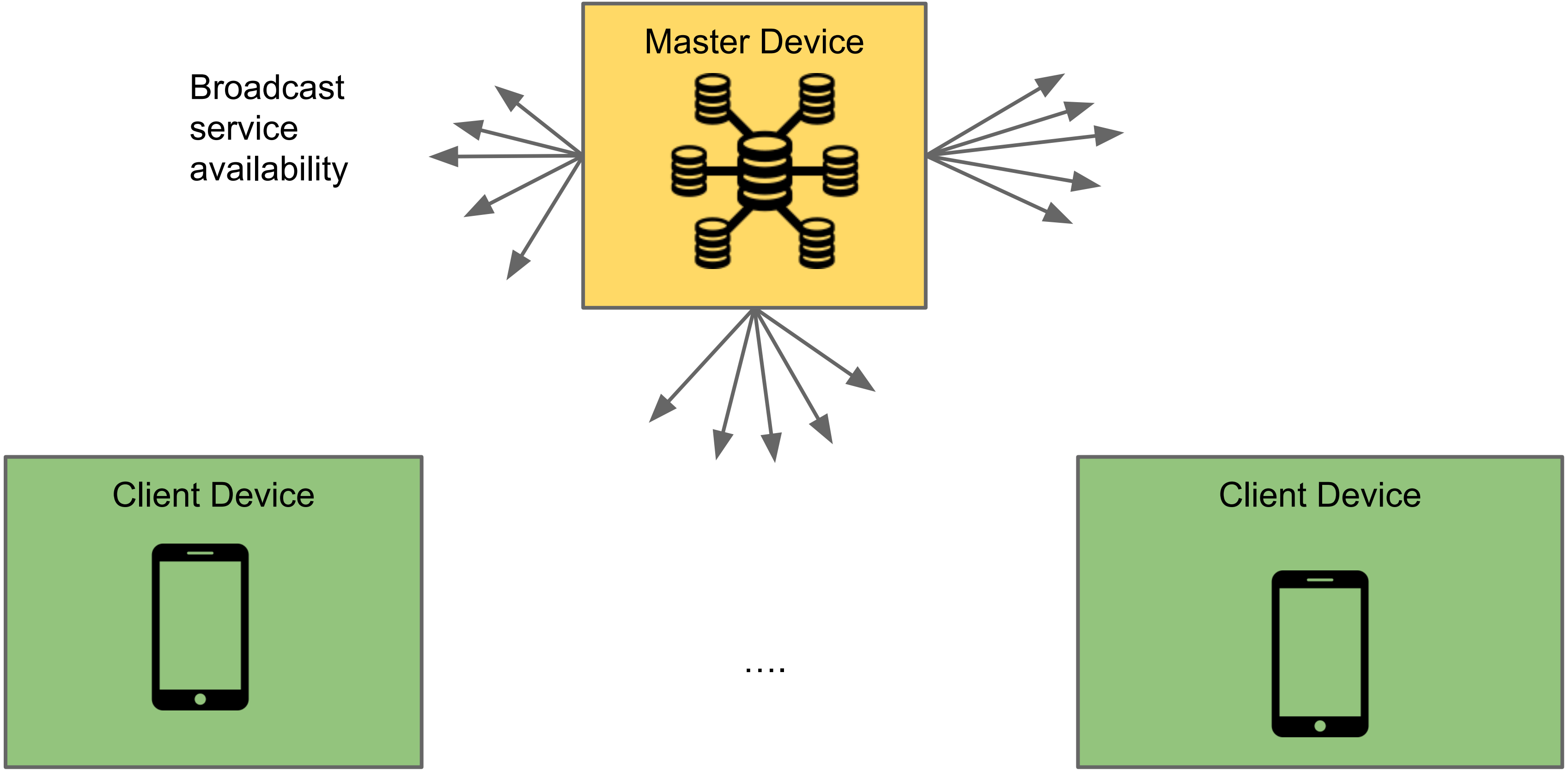
- Our solution
  - optimistic buffering
- Future work
  - report data rate, adjust it in the sender



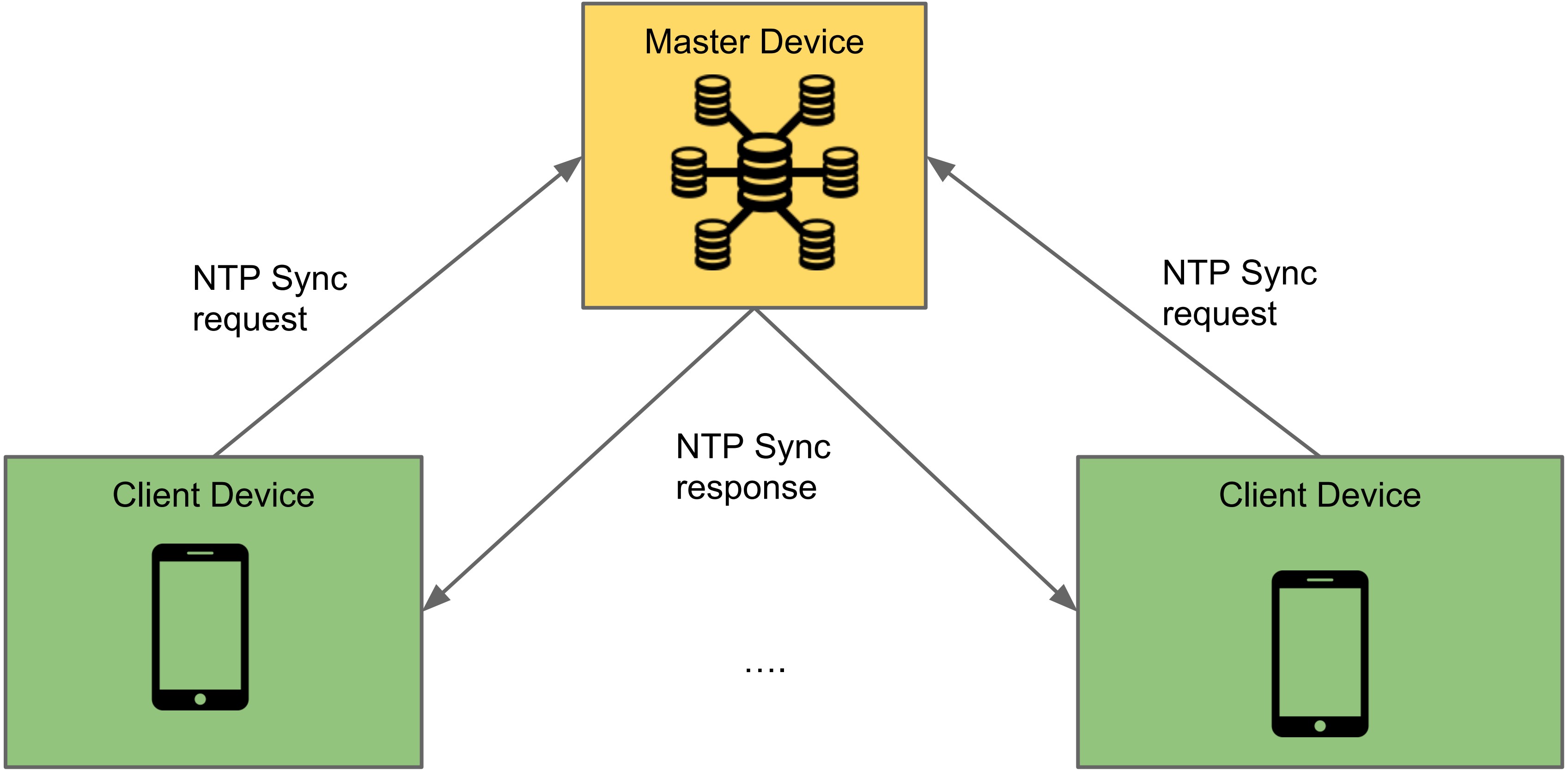
Audiosync architecture



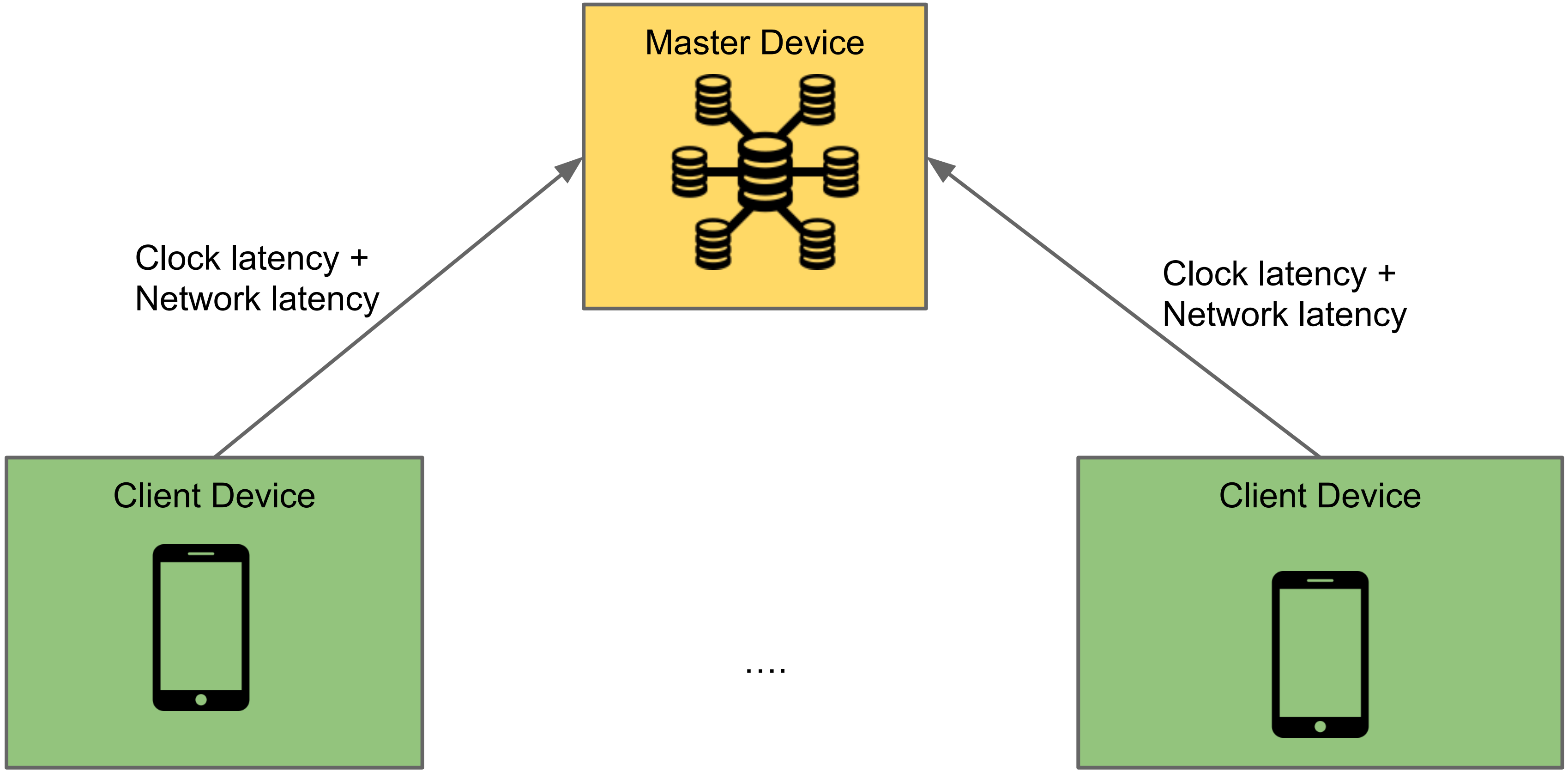
Network protocol step 1



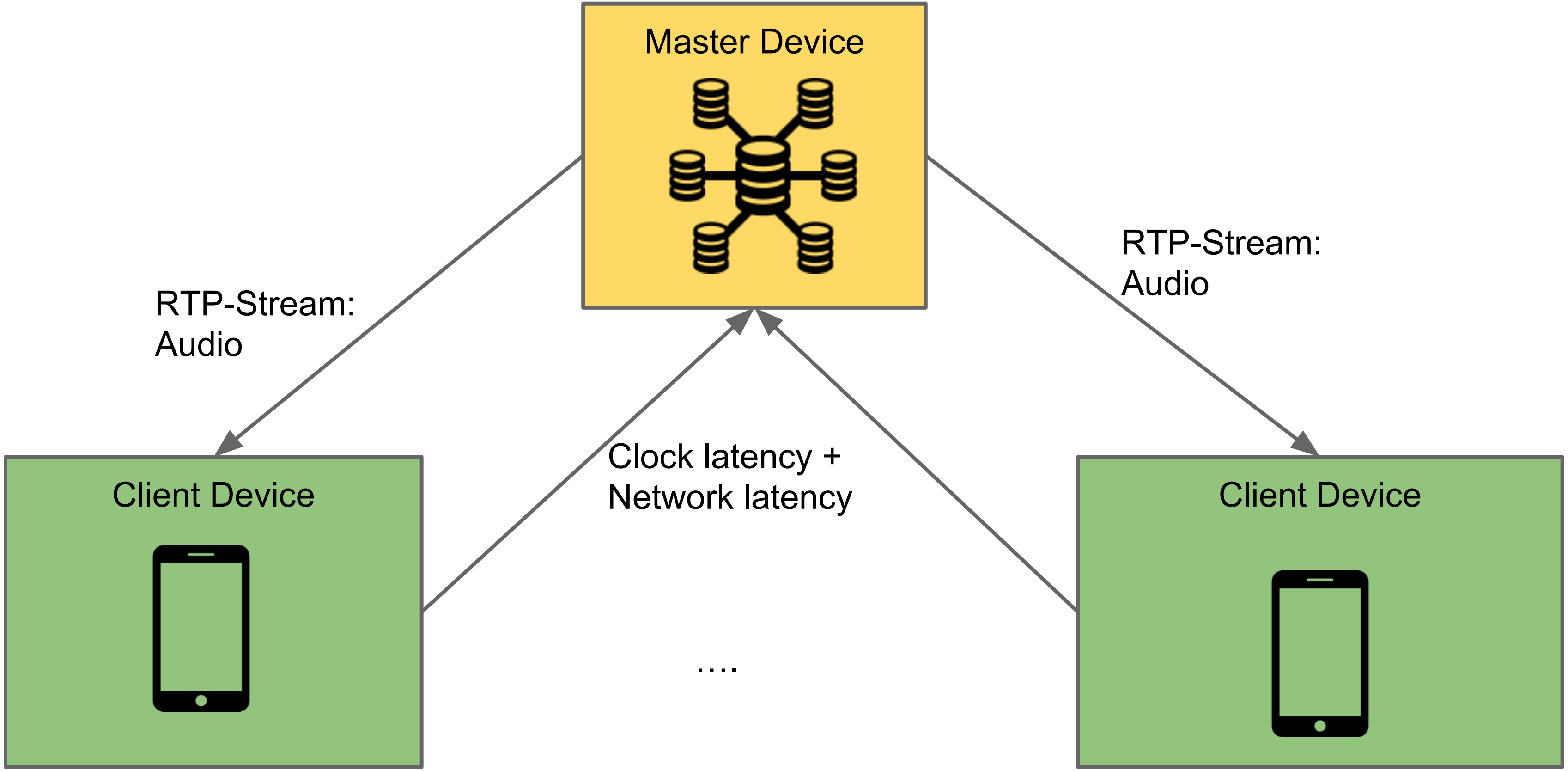
Network protocol step 2



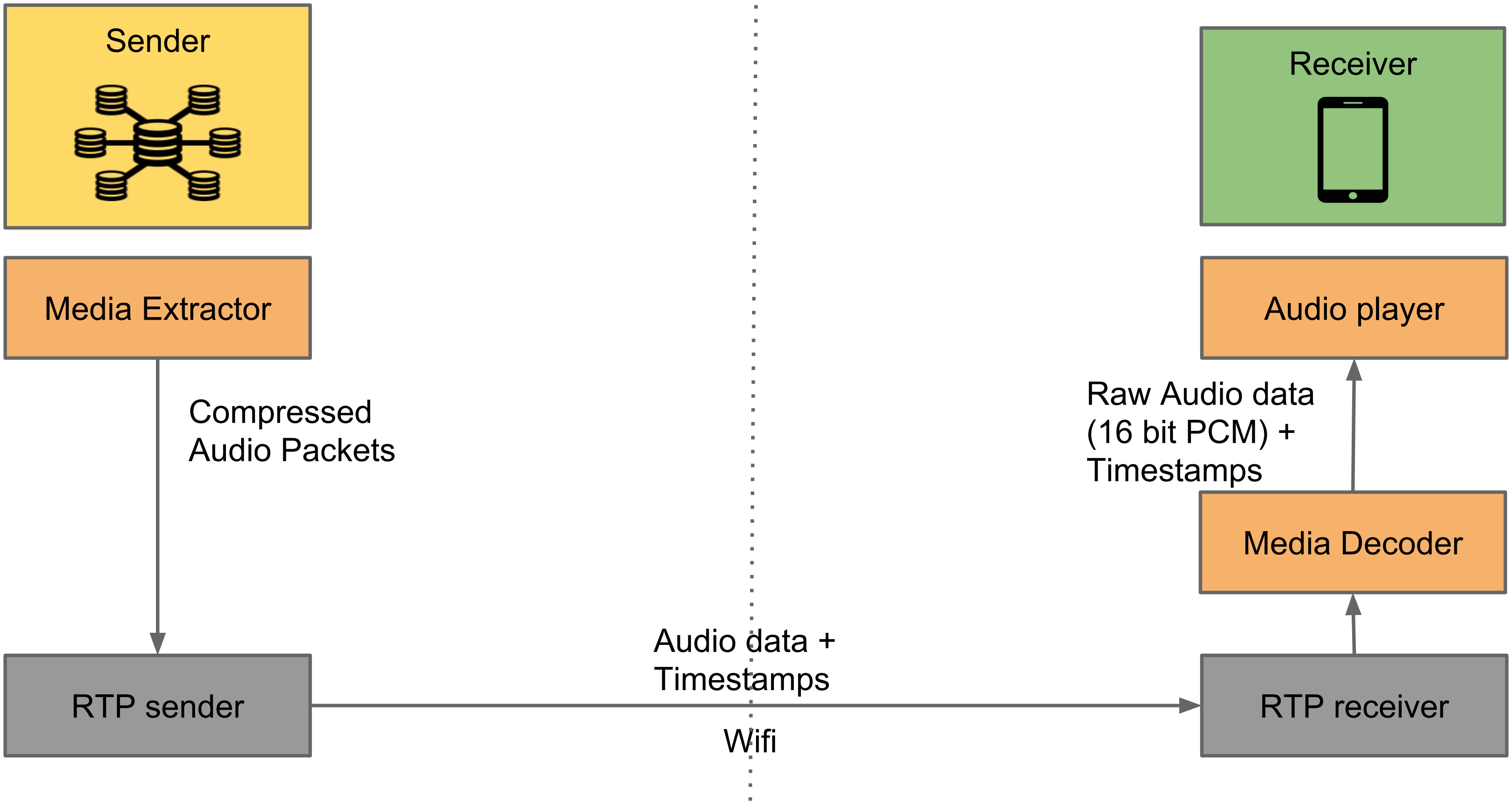
Network protocol step 3



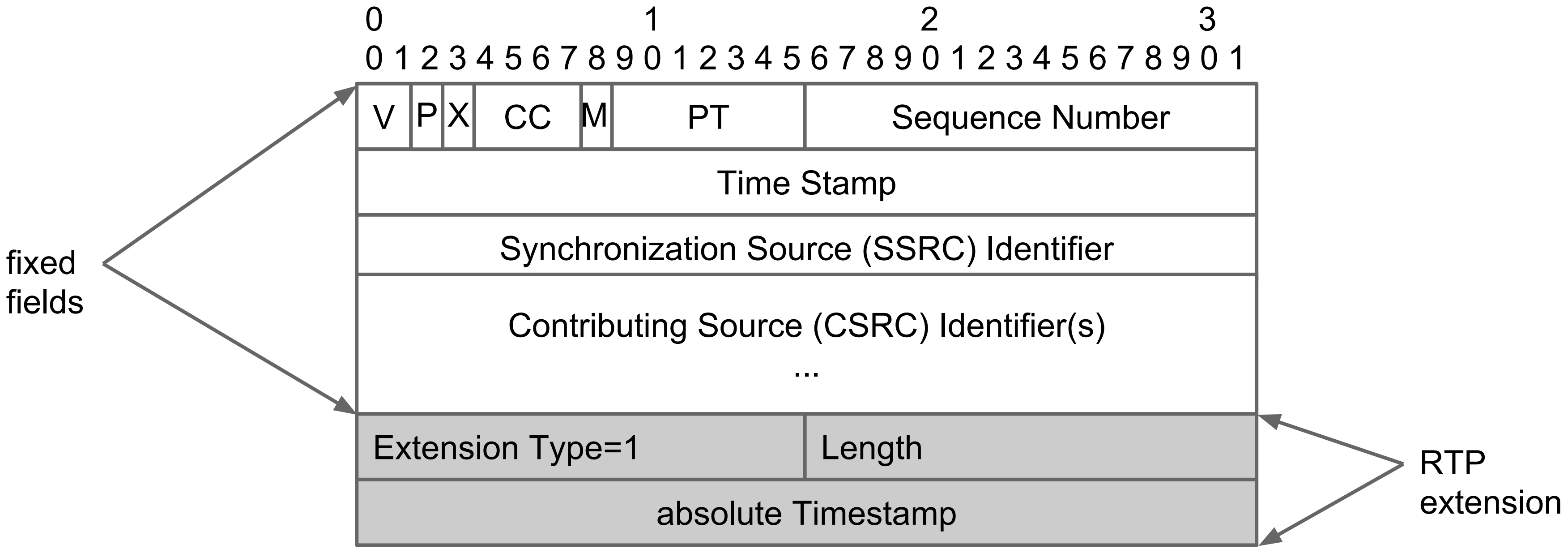
Network protocol step 4



Audio Data Streaming Architecture



RTP Protocol Timestamp-Extension



- RTP packets are timestamped
- Relative Timestamp for position in audiofile
- Absolute time of when the content should be played

Audio Player Implementation

- Key piece of the entire system
- Responsible for matching the playback time with the timestamps

→ Sometimes easier said then done

Assumptions about audio playback

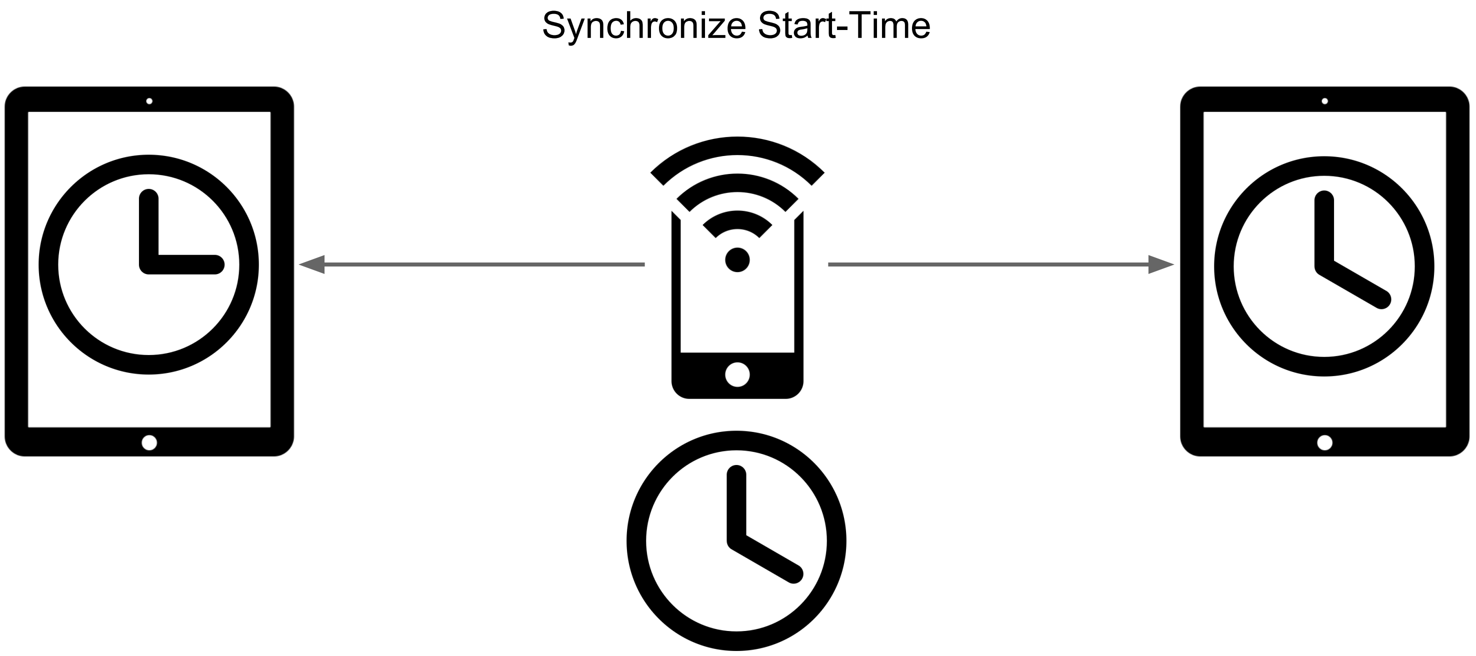
- Starting playback will have no big latency
- Playback speed will remain approximately constant
- Playback speed is the same for all songs
- Same device model’s will have the same playback rate and latencies

→ Nothing of this holds true

Evaluation Setup

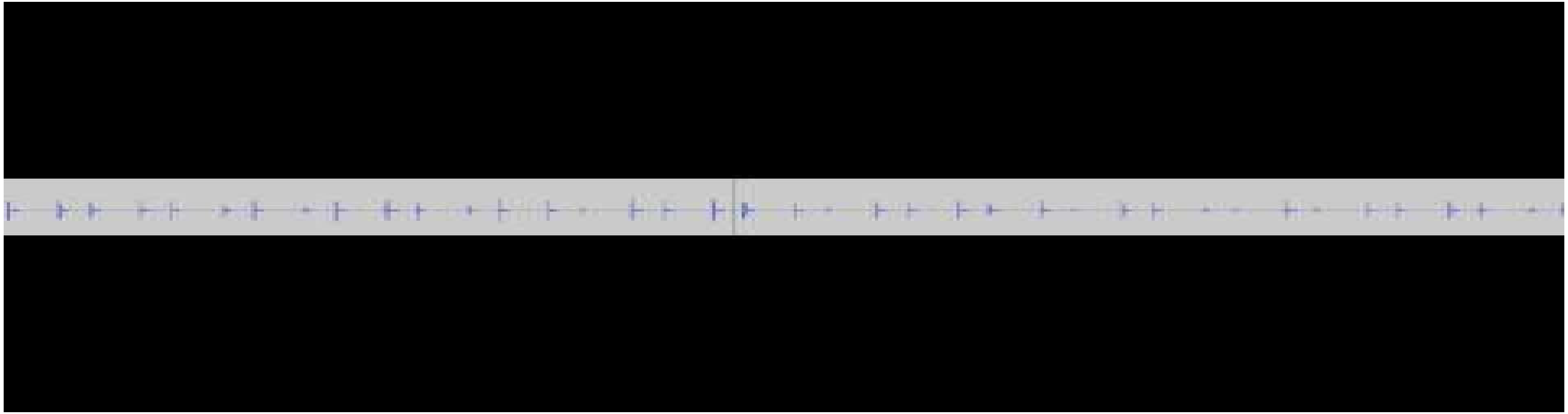
Two Nexus 7 (Model 2013) Tablets

Audio demo time



Synchronizing Playback Start and Skip or Pause playback accordingly

- Try to use NTP clock offsets to control exact start moment
- Reduce differences by skipping or pausing

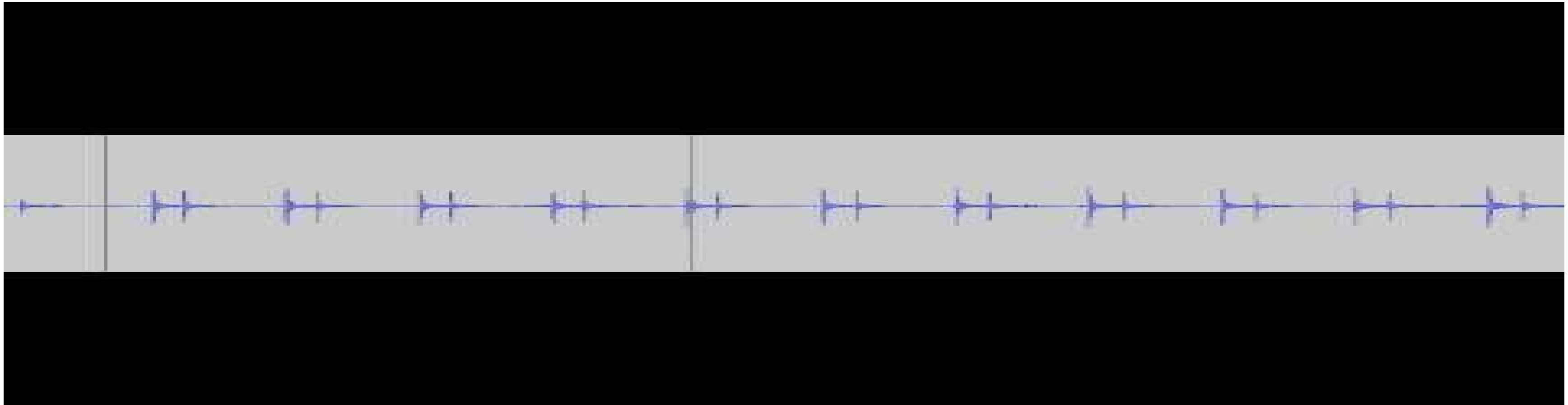


Audio demo time

Playback-Rate compensation

Control Playback Rate

- For 4 seconds disable skipping / pausing and measure playback speed
- Then enable skipping / pausing for the accumulated time difference
- Playback is only off by a constant amount (~200ms)

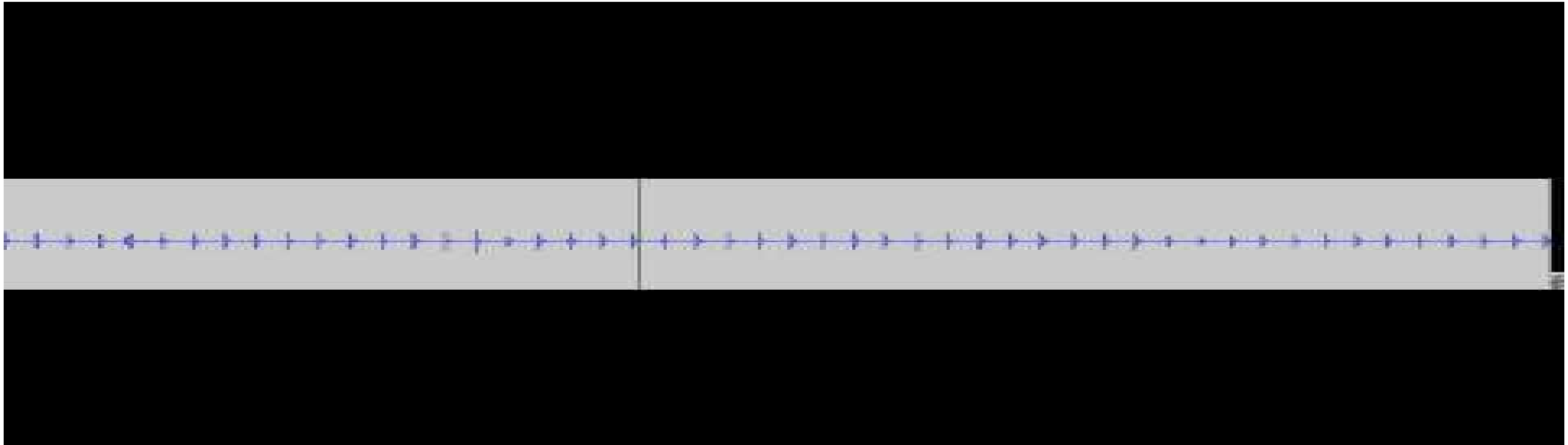


Take Device Latency into Account

Add some static latency to the lagging device

Sometimes the playback jumps

After 4 seconds the rate is re-adjusted



Future work

- Using OpenSL is not enough
  - take device-specific playback latencies into account
  - Measure system latency while using all our audio effects
- QOS
  - Measure data rate and adjust audio quality
  - E.g. decrease playback rate on all devices if we don’t have enough bandwidth

References

- **An Internet Protocol Sound System** (2004): Bob Atkinson, Tom Blank, Michael Isard, James D (JJ) Johnston, and Kirk Olynyk
- Iconography: Aaron K. Kim, Creative Stall, Pantelis Gkavos, Kevin Kwok, Roberto Chiaveri, Martin Jordan, David Lopez, Alessandro Suraci, Edward Boatman, Mario Bieh from Noun Project

Thank you for your attention!