REAL-TIME HIGHT-QUALITY AUDIO STREAMING

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Technical and Artistic Strategies to Perform Around the Globe
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SOUNDWire

Sound Waves on the Internet from Real-time Echoes

Perform Music

On-Line

in Real-Time

with the Highest Audio Quality Possible

Goals of High-Quality Audio over Networks

Maximize Audio Quality for available networks conditions

Minimize Latency

But more important, **Minimize Jitter**

Adjustable **Number of Channels**

Audio routing flexibility

Multiple peers

→ Keep Delay **Constant**
Maximize Audio Quality

Why Latency Matters

Saint Lawrence String Quartet (Quintet)

25 ms One-way Delay

Banff Centre, Alberta, Canada
Quartet

Stanford, Anechoic Room
Viola

Why Latency Matters

Saint Lawrence String Quartet (Quintet)

25 ms One-way Delay

- What Happens Naturally with Delay?
- Now, they are conscious...
- ...and they try to be stable
- The whole Quintet

Some Historical/Technical Foundations

Basic Principle: *Uncompressed Audio*

Year 2000

McGill Xu & Cooperstock

Stanford Chafe, Wilson, Leistikow, Chisholm, Scavone

UDP and TCP revisited

UDP

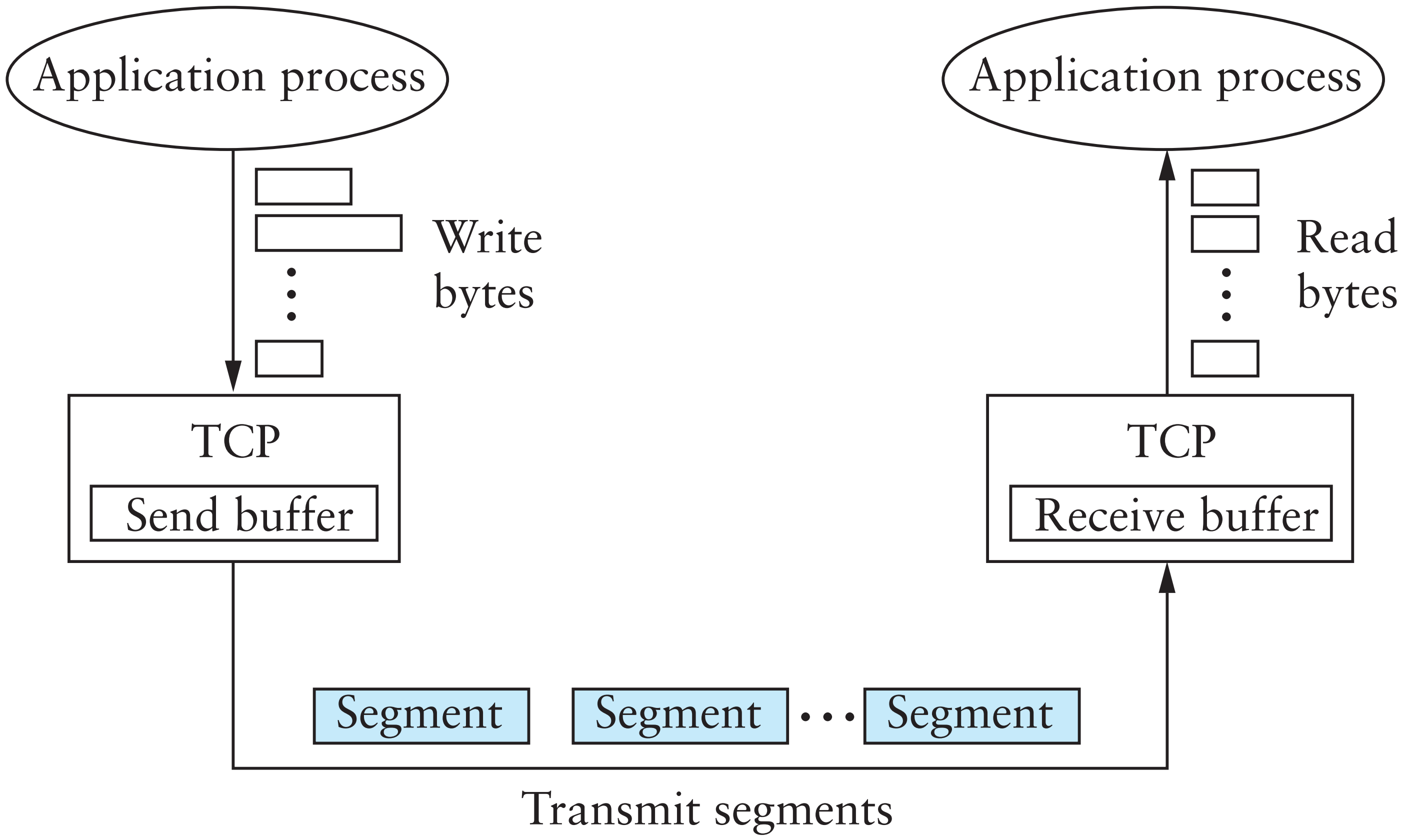
- Unreliable
- Connections-less
- Datagram Oriented

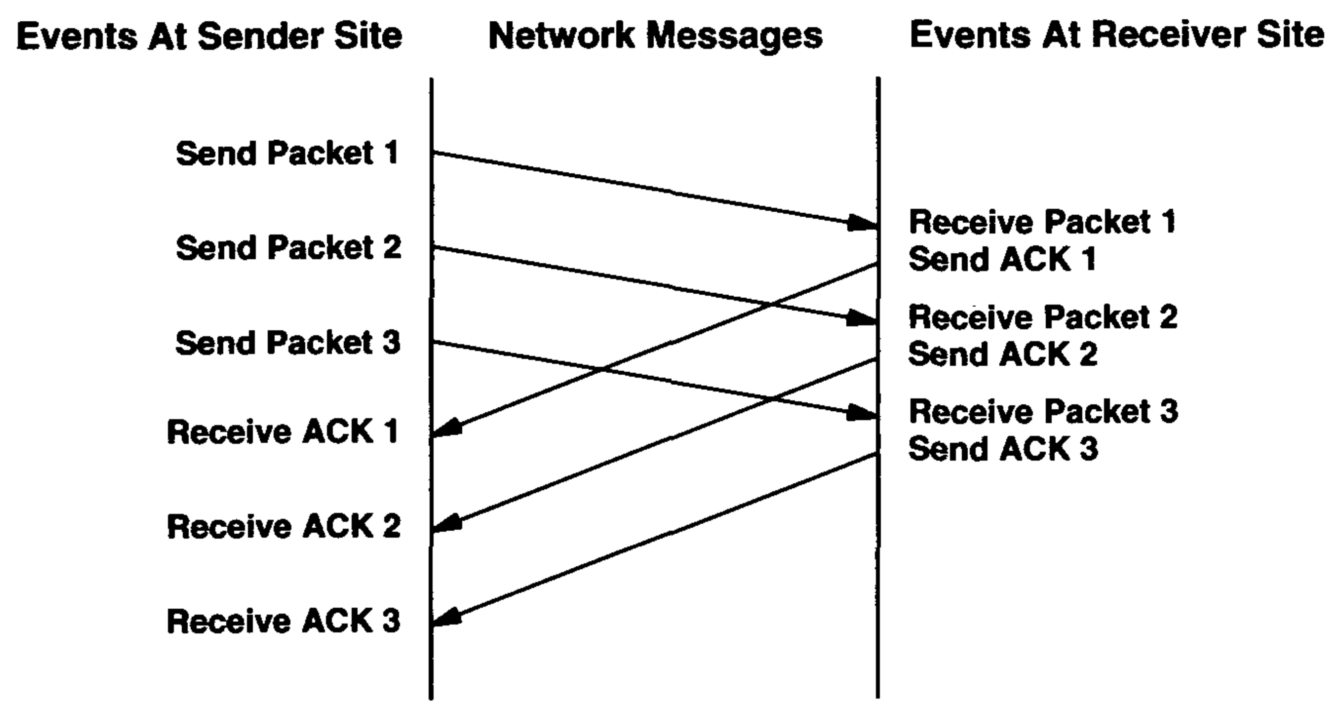
TCP

- Reliable
- Connected (virtual circuit)
- Byte-Stream Oriented

Delay Constant?

Why TCP is problematic

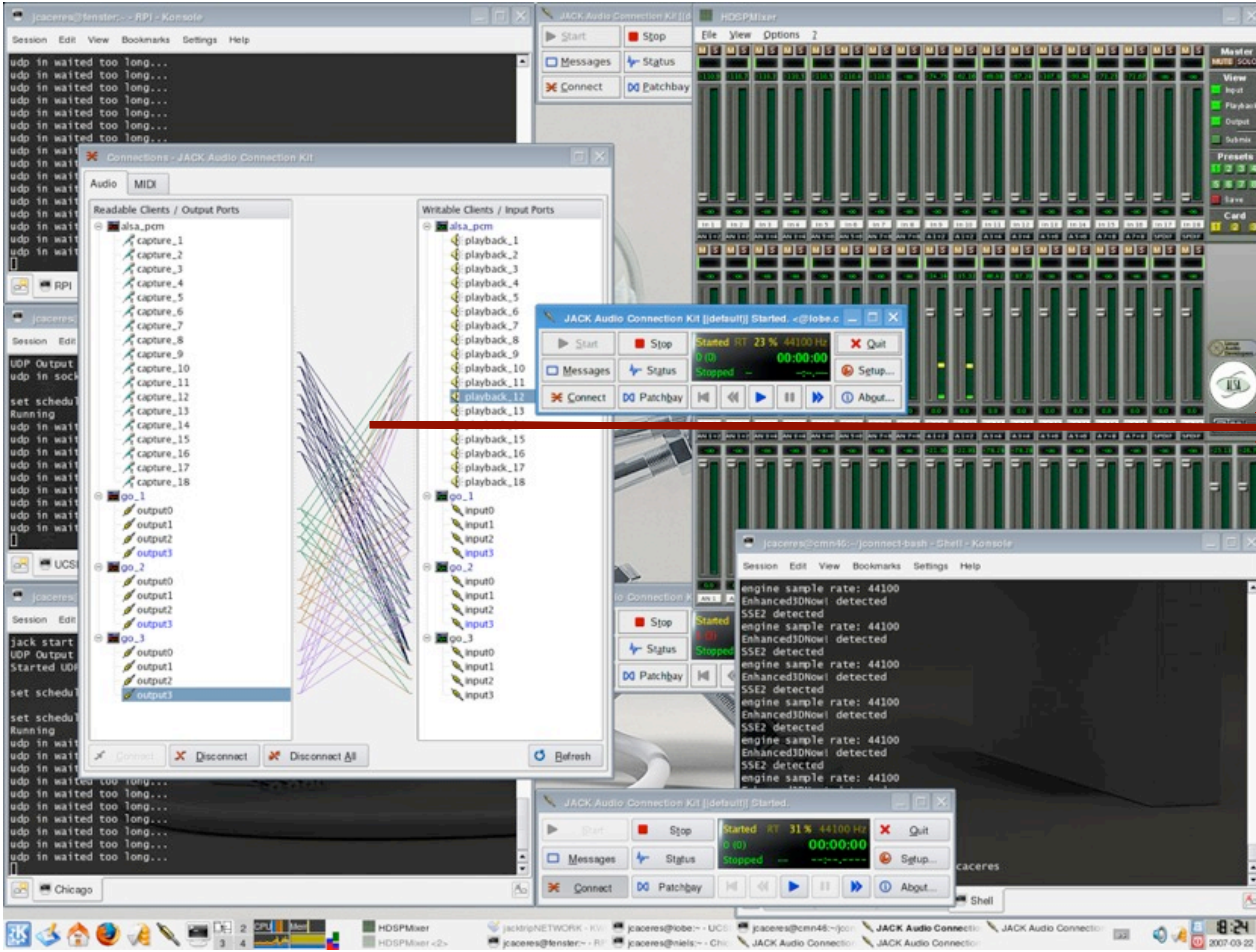




TCP makes the underlying delay elastic and ever-increasing

→ Use UDP

A JackTrip Session



Jmess

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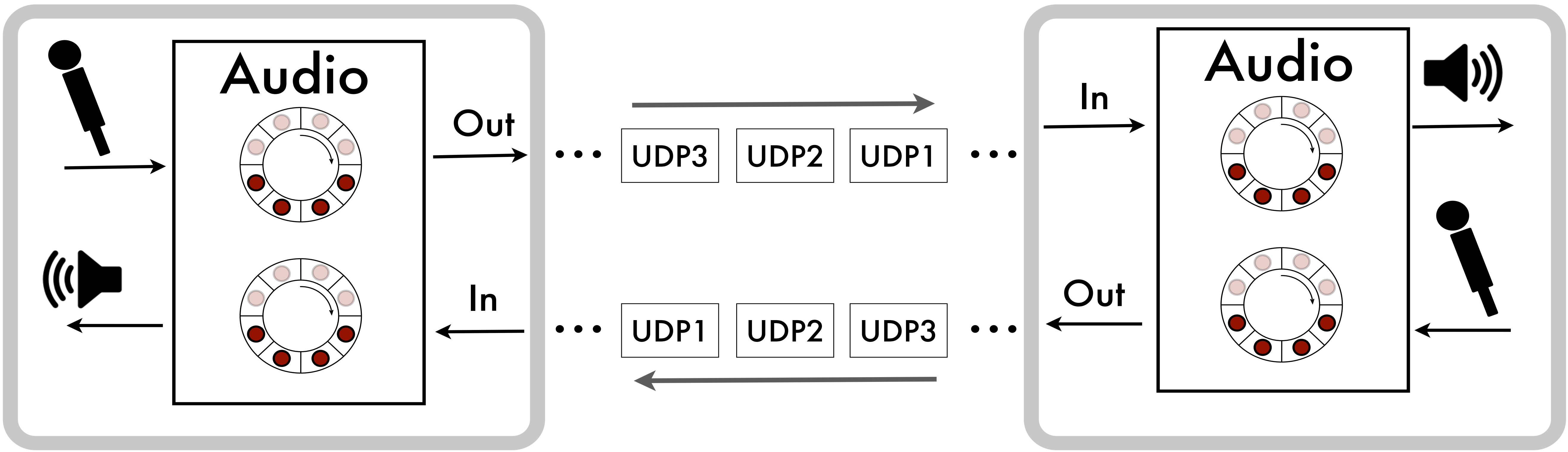
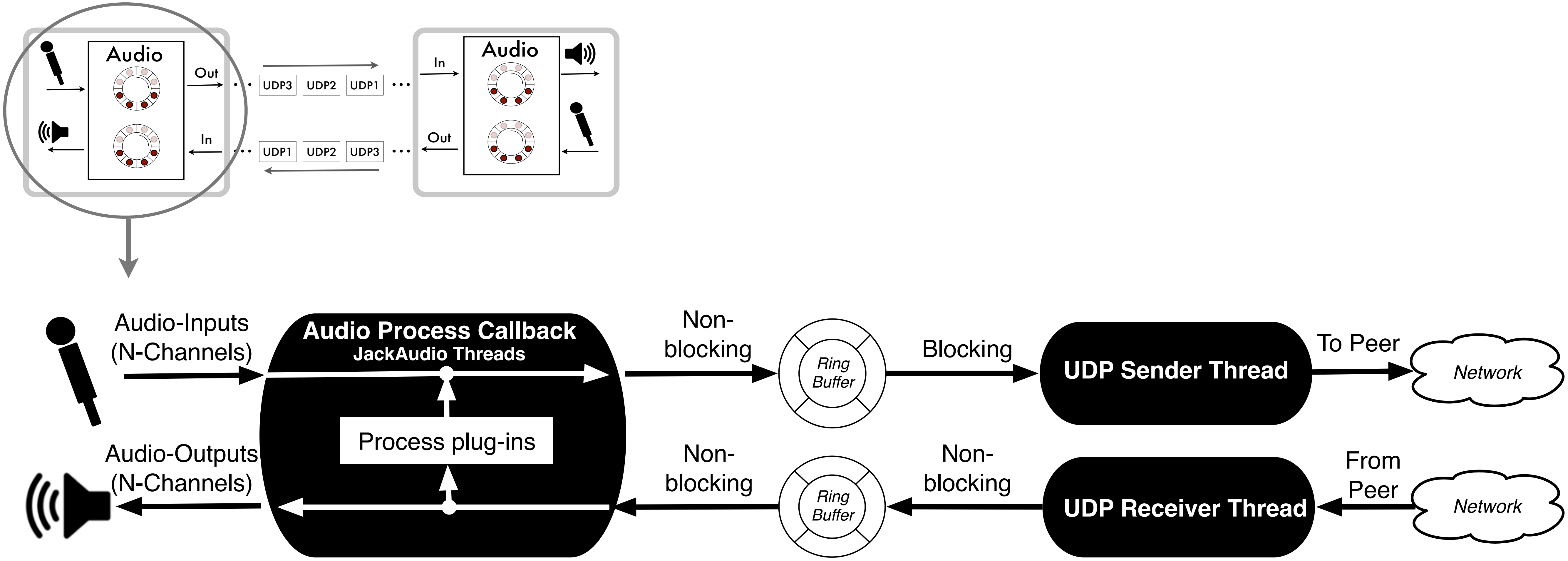
What is Jack?

Demo: *Jack*

Real-Time Audio: Under the Hood



Real-Time Audio: Under the Hood



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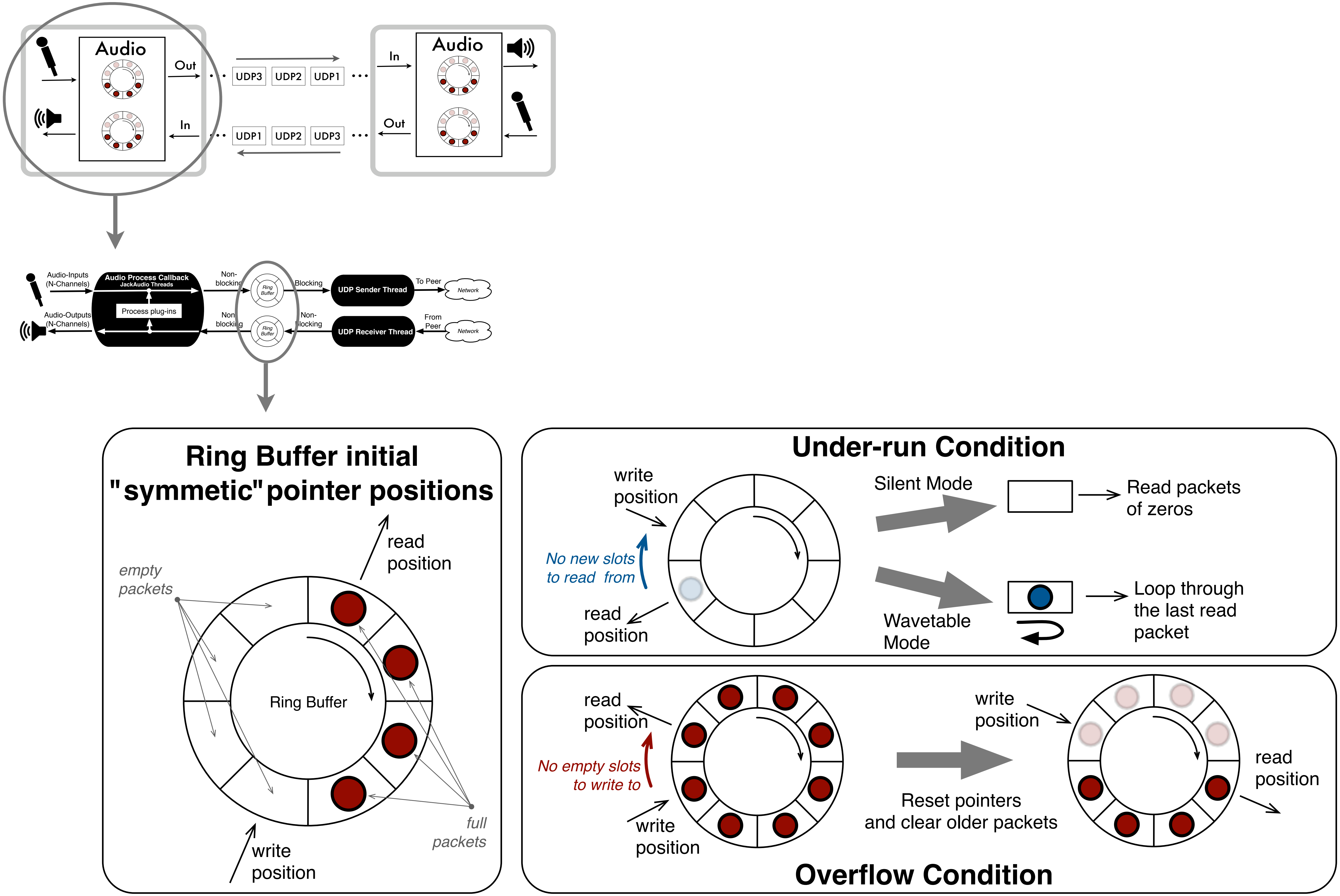

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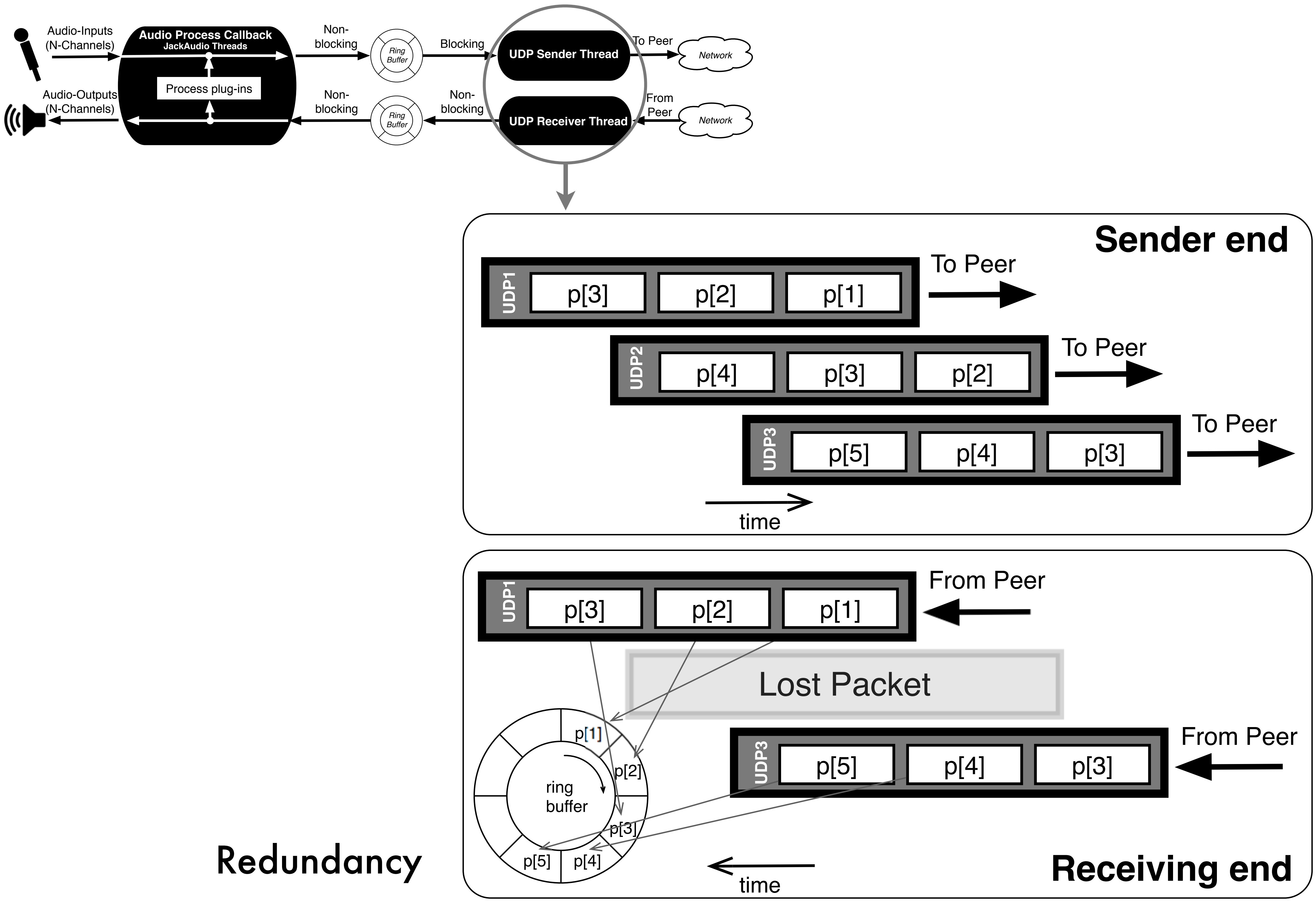
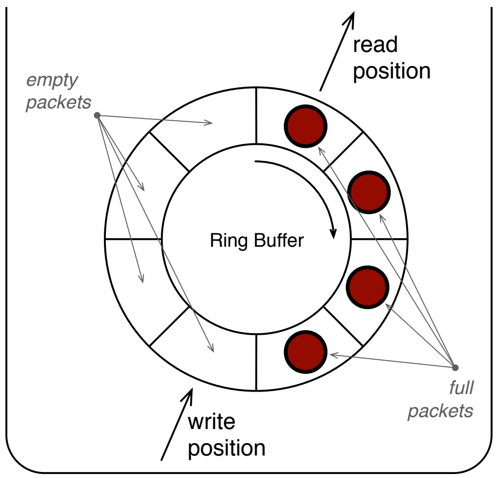
Real-Time Audio: Under the Hood



Redundancy

Parameters that matter



Queue Length (latency/jitter tradeoff)

UDP3 UDP2 UDP1

UDP3 UDP2 UDP1

UDP3 UDP2 UDP1

Redundancy (bandwidth)

8 / 16 / 24 / 32

Audio Bit Resolution (bandwidth)

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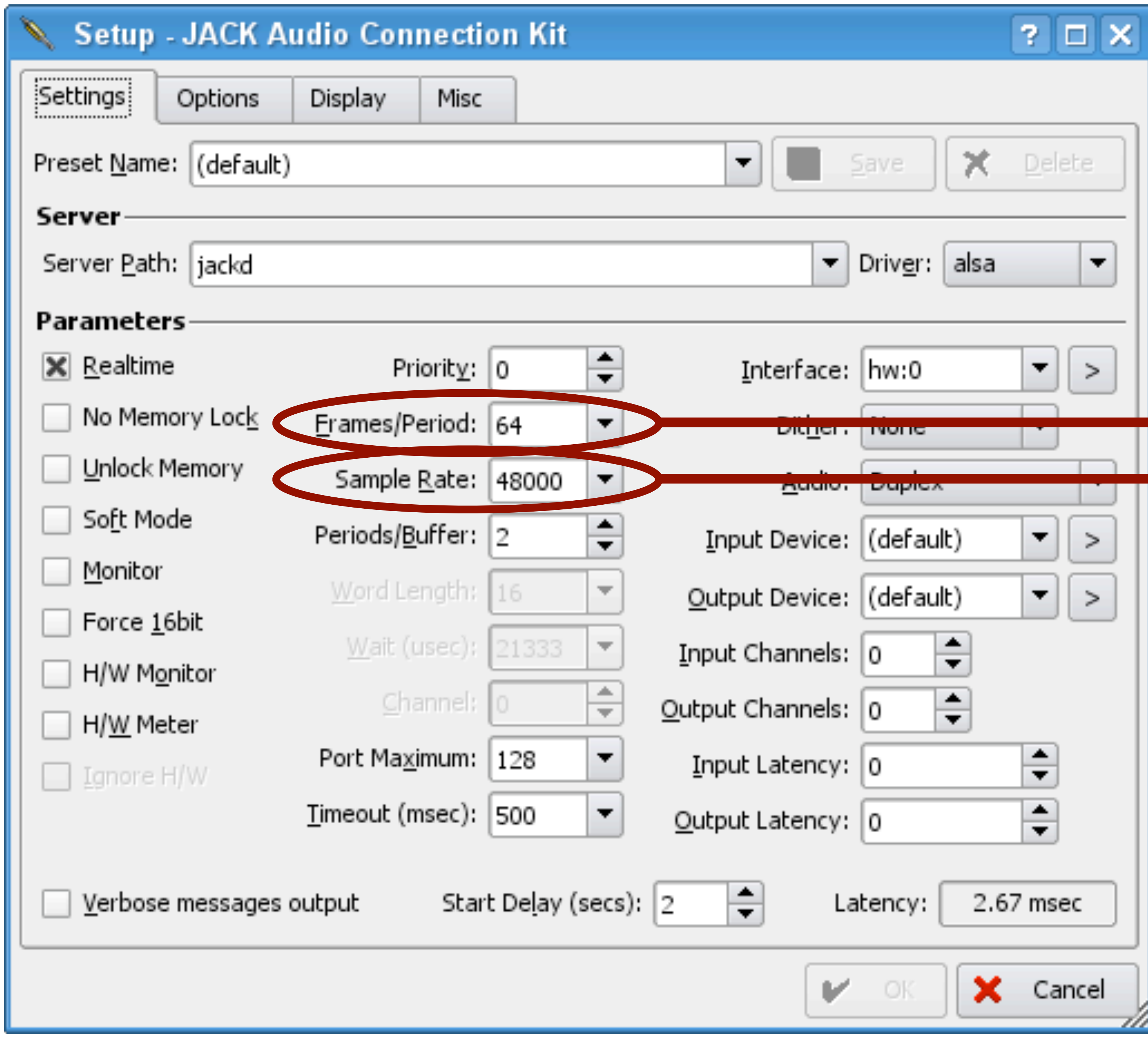
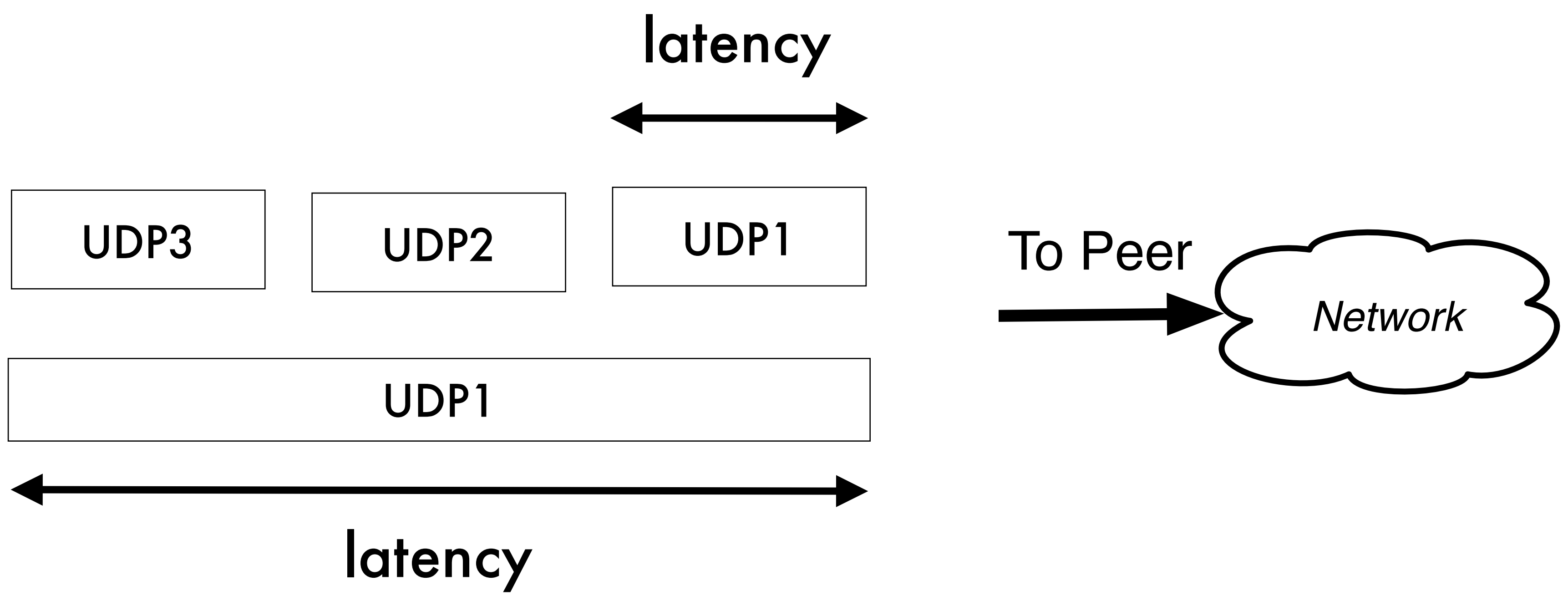
Parameters that matter



Packet size (latency)

Sampling Rate (bandwidth)

Packet Size / Latency



The smaller the packet size,
the lower the latency (sampling rate constant)

Sampling Rate / Latency

For the same
packet size

$\boxed{\text{64 samples}} \longrightarrow$ 48kHz: 64/48000 = 1.3 ms

$\boxed{\text{64 samples}} \longrightarrow$ 96kHz: 64/96000 = 0.7 ms

The higher the Sampling Rate,
the lower the delay (packet size constant)

A Simple (default) JackTrip Session

at CCRMA (Server)

```
jacktrip -s
```

server mode

at UK (Client)

```
jacktrip -c [ccrma-IP-number]
```

client mode

More Control over a JackTrip Session

at CCRMA (Server)

```
jacktrip -s  -n 8  -q 4  -r 2
```

server mode

num chans

Queue

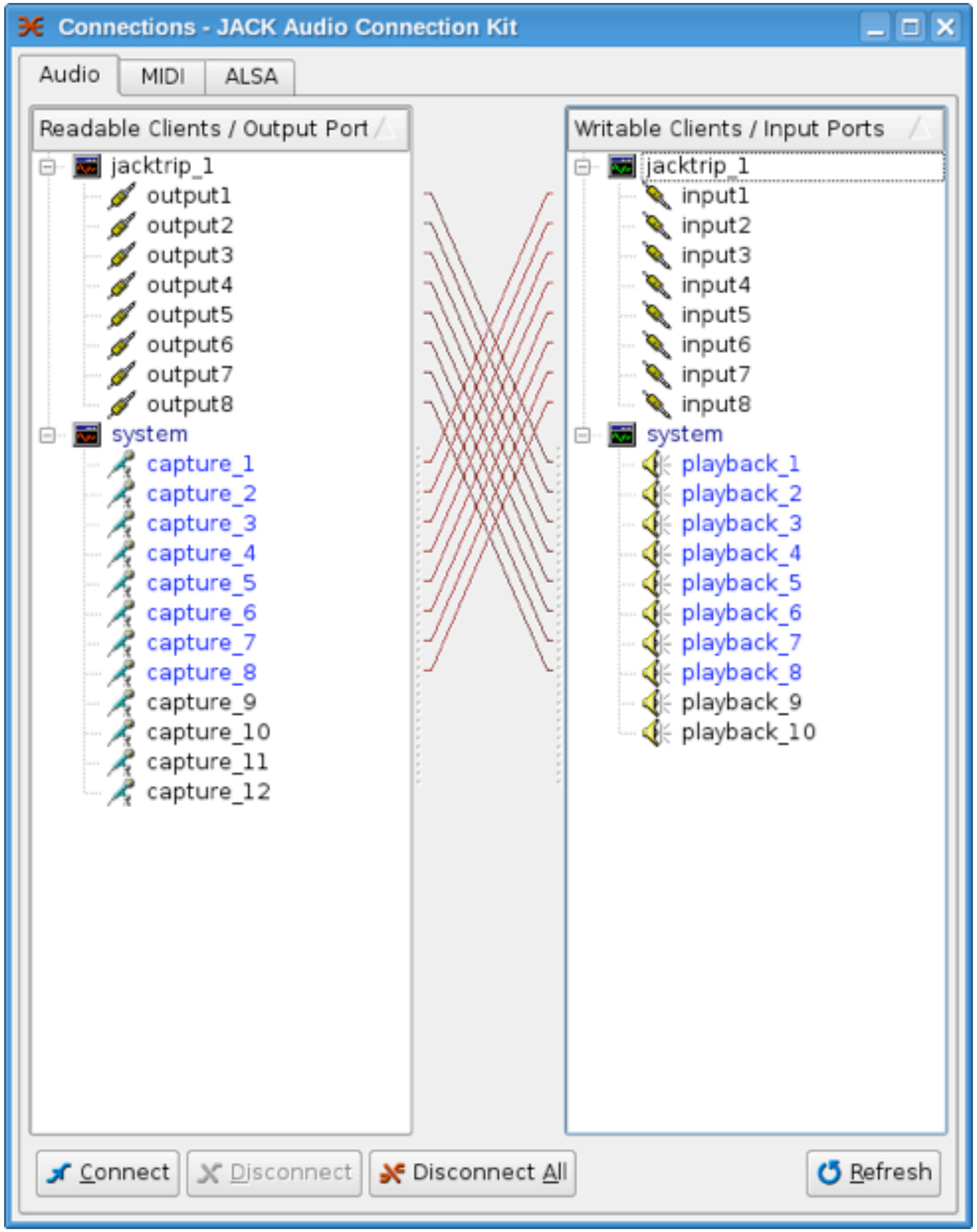
redundancy

at UK (Client)

```
jacktrip -c [ccrma-IP-number] -n 8  -q 4  -r 2
```

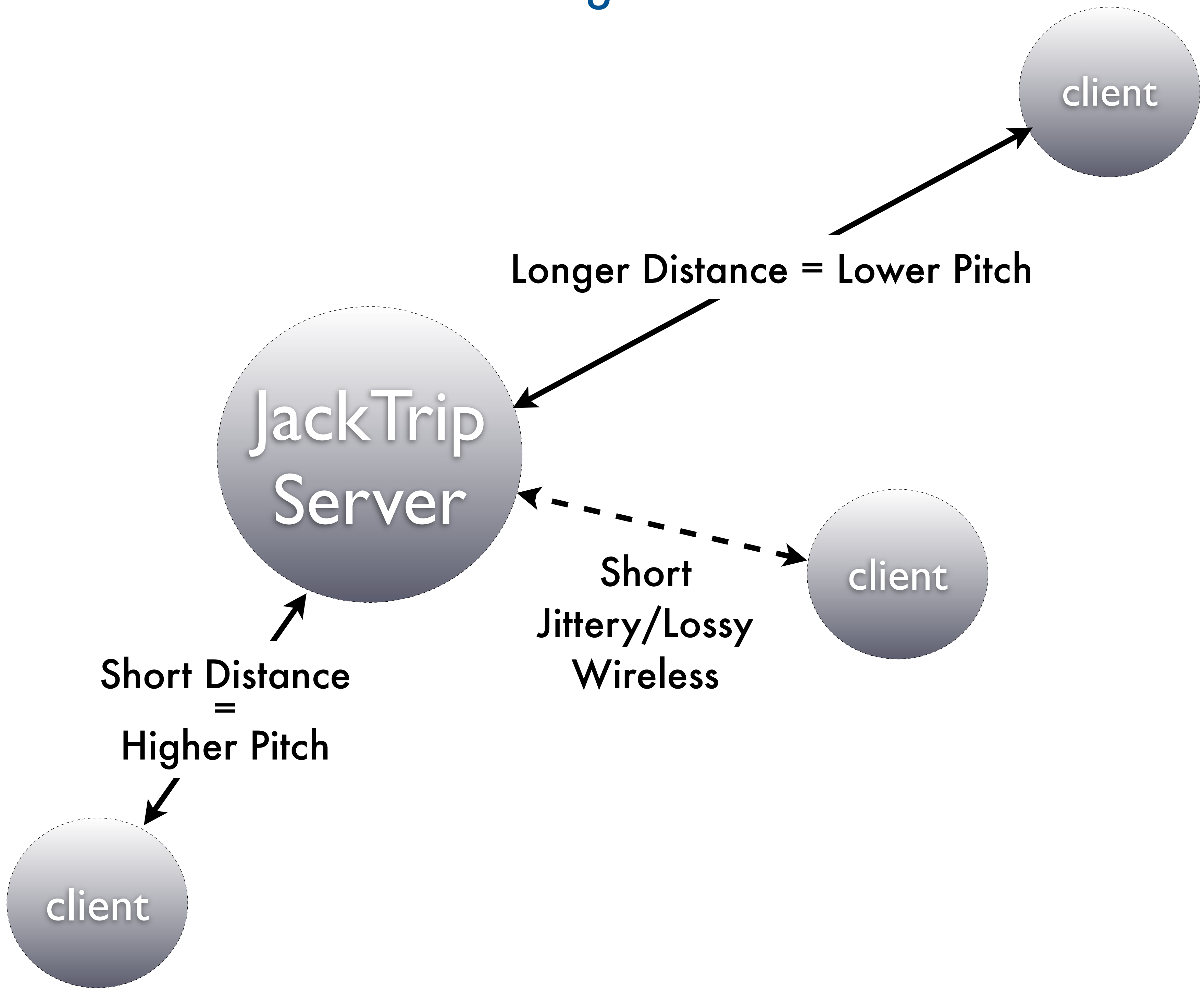
client mode

A JackTrip Session (demo)

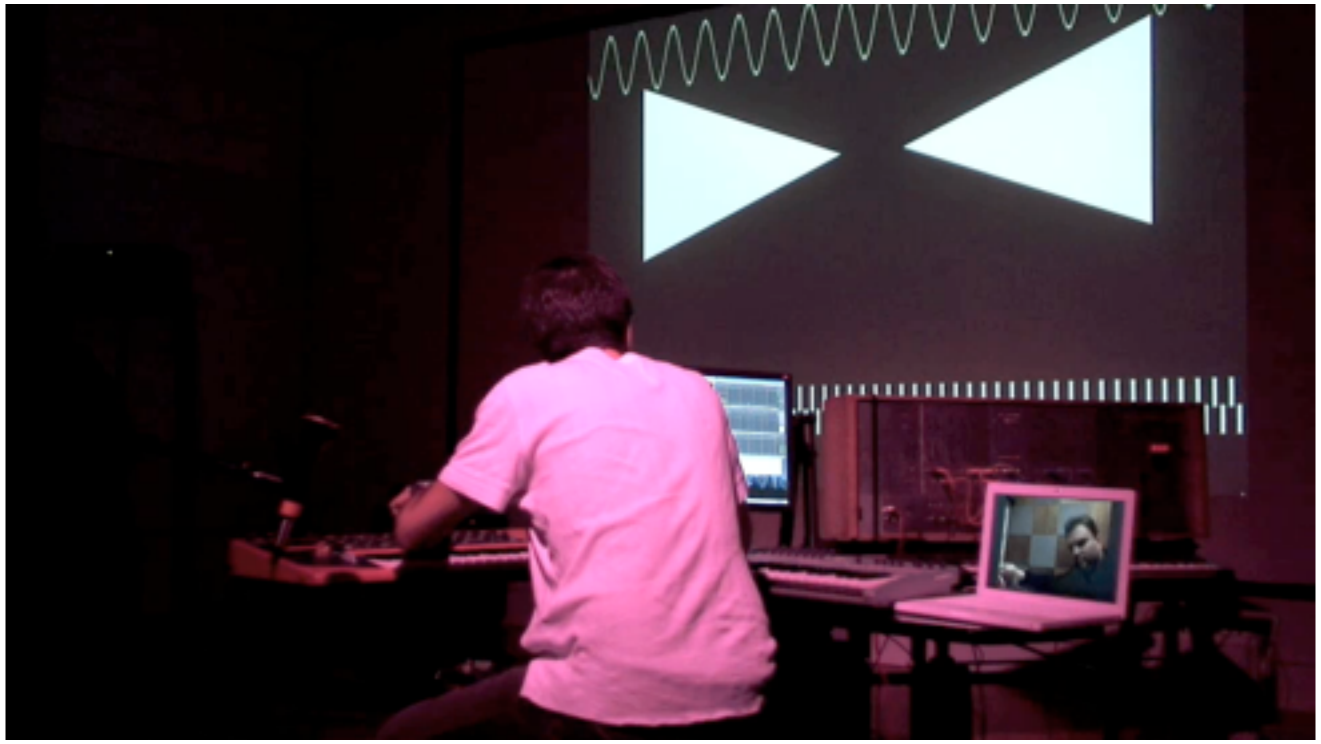


QoS Network "Audible Distances"

Plucking the Net

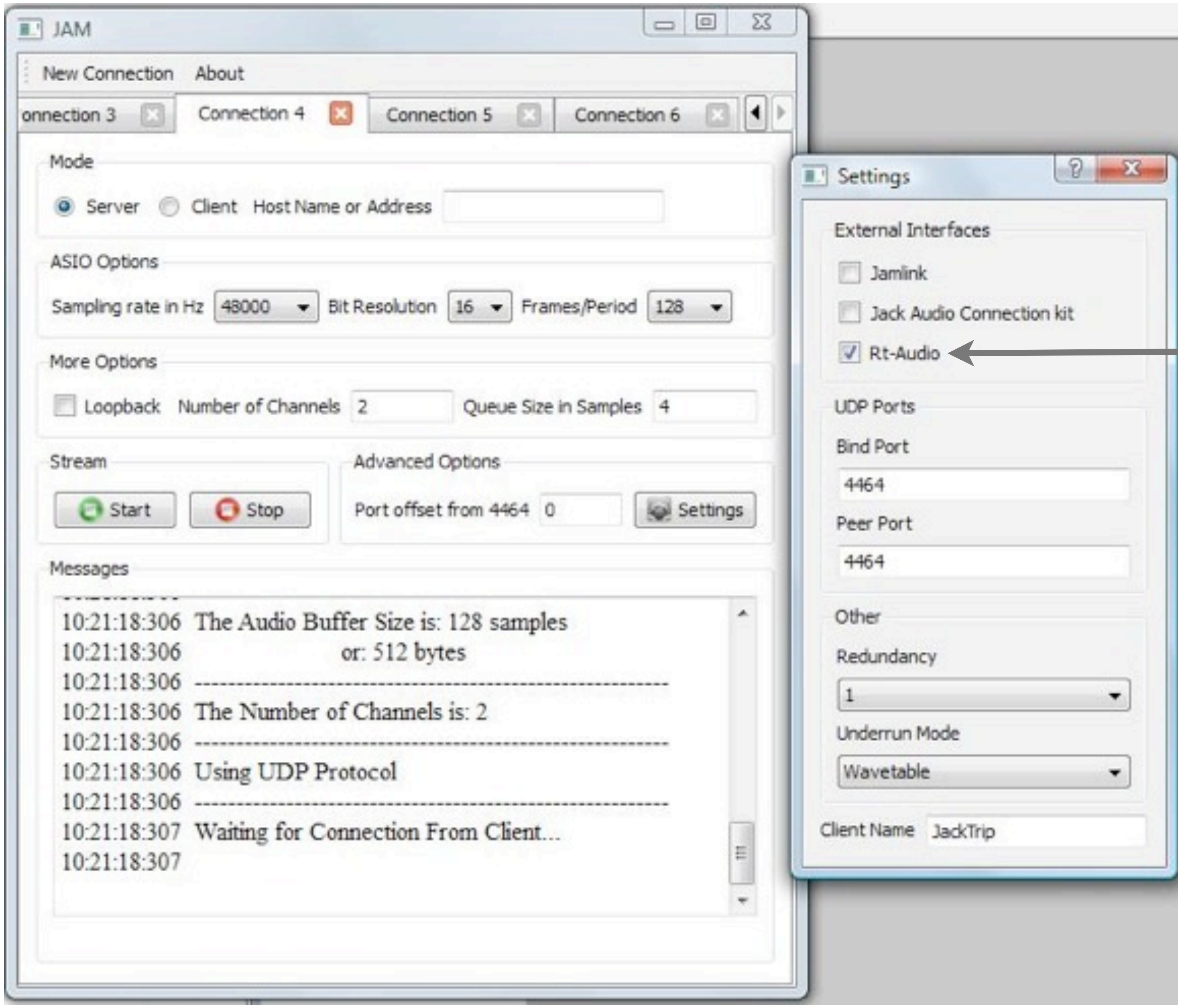


Net vs. Net Collective



Chris Chafe | Chopper

What’s next for JackTrip



Garry Scavone’s
RtAudio

Windows XP, Vista Port | *Elie Noune*

JamLink

http://www.musicianlink.com/

More Information

SOUNDWire
Sound Waves on the Internet from Real-time Echoes

http://ccrma.stanford.edu/groups/soundwire/

JackTrip at Google Code:
http://code.google.com/p/jacktrip/

JackTrip Mailing List:
http://groups.google.com/group/jacktrip-users