towards message based audio systems

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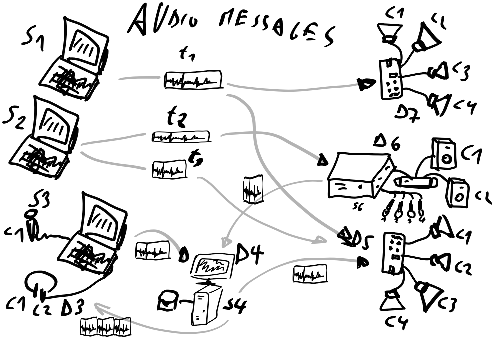
Outline

playing a multi-speaker environment



- idea of and audio network with sources S_n and drains D_n
- Using distributed networked embedded devices
- Playing from different devices
- avoiding a central mixing desk

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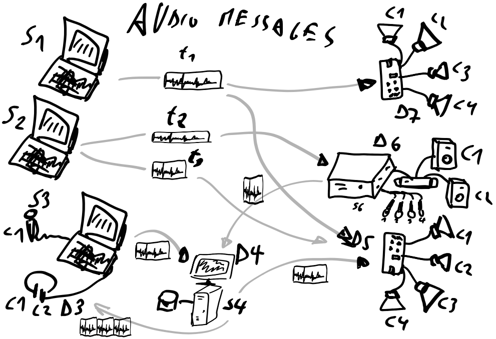

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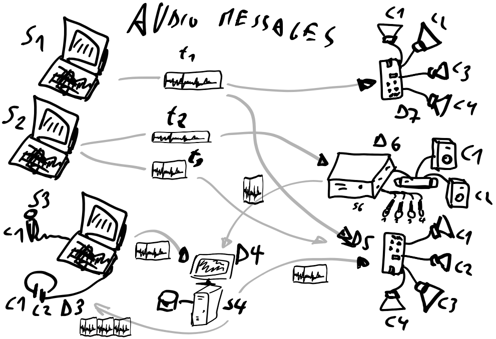

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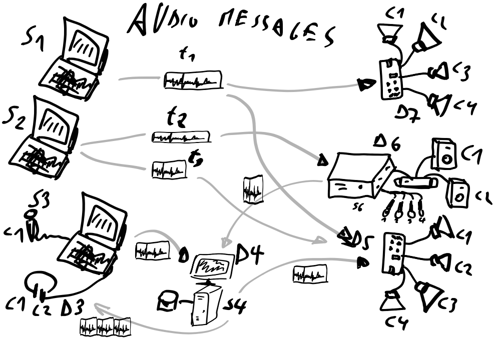

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Features needed:

- audio signal intercommunication between distributed audio systems
- arbitrary ad hoc connections
- various audio formats, sample-rates
- synchronization and lowest latency possible
- audio-data on demand only

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Network

- “Audio over Ethernet“ is a common solution within local networks
- most are Stream based audio transmission, representing the data as a continuous sequence
- audio messages as on-demand packet based streams not available
- -> design and implementation of a new audio transmission protocol
- first implementation in user space (on the application layer)
- the idea of “dynamic audio networks”.

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structure of a audio message

```
AoO message := "#bundle" timestamp
  <format> <channel> [<channel>,...]

format := "/AOO/drain/<d>/format"
  samplerate blocksize overlap mime-type
  [time correction]

channel := "/AOO/drain/<d>/channel/<c>"
  id sequence resolution resampling <data>

d ... number of drain (integer)
c ... channel number (integer)
data ... audio data (blob)
```

AoO Data

sampling rate Different sampling rates of sources are possible, which will be re-sampled in the drain.

blocksize The amount of samples in each package of audio data, which must be greater or equal 1, limited by packet size.

overlapping factor The overlapping factor is 1 (one) by default. Higher values can be used to implement redundancy, to deal with lost packets or needed when sending FFT-frames (in future implementations).

resampling factor is linked to the sampling-rate in order to be able to choose the precision of each channel individually using oversampling or similar.

coding of the audio data using the *Multipurpose Internet Mail Extensions* (MIME) standard[**?**]. In our uncompressed format, the MIME type would be "audio/pcm", whereas "audio/CELP" classifies CELP encoded data.

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