sndio – OpenBSD audio & MIDI framework for music and desktop applications

Alexandre Ratchov
alex@caoua.org

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Outline

Introduction

What is digital audio?

- Sequence of samples at *fixed rate*.
- Played or recorded by the audio interface.
- Full-duplex : n-th sample played while n-th sample recorded.

Consequences

- Clock source (each sample is a clock tick).
- Can be streamed (samples can be buffered).

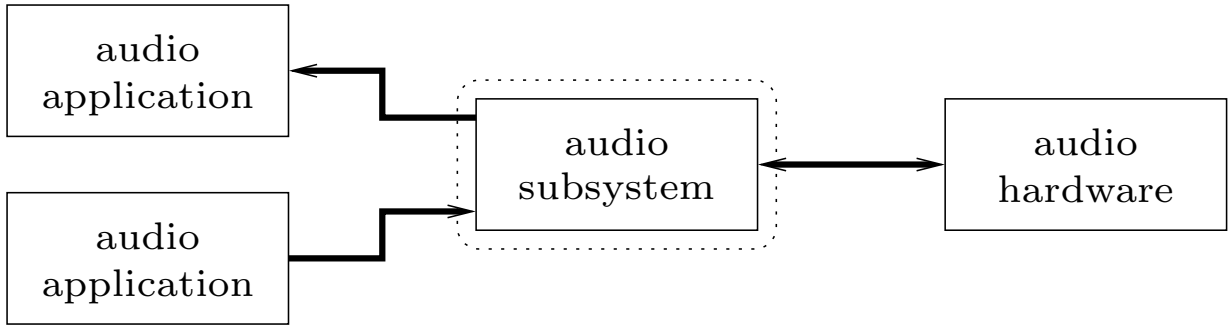
What is MIDI?

- Slow unidirectional serial link.
- Transmits events to control audio (start, stop, volume).
- Standardized in 1985, used by most professional audio equipment.
- Real-time (i.e., events are processed immediately).

Consequences

- Not a clock source, but can carry clock ticks.
- Usable to control audio (start, stop, volume).

Purpose of the audio & MIDI subsystem



- Fill the gap between the software and the hardware.
  ⇒ Format conversions, resampling.
- Allow multiple programs to use the hardware concurrently.
  ⇒ Mixing, splitting hardware in subdevices.
- Allow multiple programs to cooperate.
  ⇒ Synchronization, communication between programs.

The problem to solve

Conversions & resampling

- Application may not support hardware parameters.
- Two programs may not support common parameters – at least one of them requires conversions.

Splitting hardware in subdevices & mixing

Using a 4-channel card as two *independent* stereo cards.

Example

- Headphones (channels 0,1) for telephony.
- Speakers (channels 2,3) for music.

Remark : *mixing two streams is trivial at this stage.*

Synchronization and communication

Unix philosophy

> *"Write programs that do one thing and do it well. Write programs to work together."*
>
> — *Doug McIlroy*

To work together, audio programs must:

- Be synchronized.
- Communicate.

⇒ This is the role of audio and MIDI subsystems.

Fault tolerance and correctness

- Error recovery
  - Effects of transient errors must be transient
    e.g., a load burst may not cause a program to go out of sync.
- Isolation
  - Error in one program should break no other program.
- Correctness
  - Complicated architecture leads to bogus implementation.
  - Complicated APIs are misused and lead to broken programs.
  - Complicated tools are misused and lead to broken configurations.

Design considerations

Performance *vs.* responsiveness

Performance

CPU time consumed – e.g., how many CPU cycles to compress a file.

Responsiveness

Latency in processing events – e.g., how long it takes to read a block from disk.

For any audio subsystem:

- The CPU usage is negligible.
  ⇒ Performance is not a concern.
- Delays causes the sound to stutter.
  ⇒ Responsiveness is of first importance.

User-space *vs.* kernel: extra latency?

What is latency?

The time between a program produces samples and the time they are played.

- Samples are buffered.
- Buffers are consumed at fixed rate (sample rate).

$\Rightarrow$ The latency *is* the buffer usage.

No extra latency

Whether buffers are located in kernel or user space doesn't matter for the latency.

User-space *vs.* kernel: stability?

Why do the sound stutter?

The machine is busy, the program (or the audio subsystem) don't get enough CPU to produce samples to play.

During a load burst:

- User-space components may underrun.
- Kernel components can't underrun.

⇒ During a load burst, the application underruns, so we're toast. No matter whether the audio subsystem has user-space components.

Fixing stuttering

- Write programs that don't block.
- Give enough CPU to programs.

Sample formats choice

- Integers (*a.k.a* fixed point numbers) – yes.
  - Any combination of width, signedness, byte order and alignment.
  - Used by most hardware.
- IEEE floats used only in the $[-1; 1]$ range – no.
  - Equivalent to integers (fixed range).
  - Trivial to convert from/to integers (FPU required).
  - Mostly used to save development costs.
- μ-law & a-law – no.
  - Not usable for audio processing (not linear).
  - Already handled by telephony applications.
- Encrypted/compressed opaque formats – no.
  - Not desirable (computers are to process data).
  - Alternatives exist, *i.e.*, the user is not locked.

Architecture and implementation

Overview

- Audio server
  - Conversions, resampling, mixing, channel mapping.
  - Subdevices – per-subdevice properties.
  - MIDI controlled per-application volume.
  - MIDI exposed clock source for non-audio applications.
  - MIDI controlled synchronization between applications.
- MIDI server – software MIDI thru box
  - “hub” for MIDI data.
  - Software or hardware can be connected to it.
- Library-based programming interface
  - Very simple.
  - Mimics kernel APIs (`read`, `write`, ...).
  - No need to handle synchronization and error recovery.

Audio server architecture - processing chain

The audio server framework:

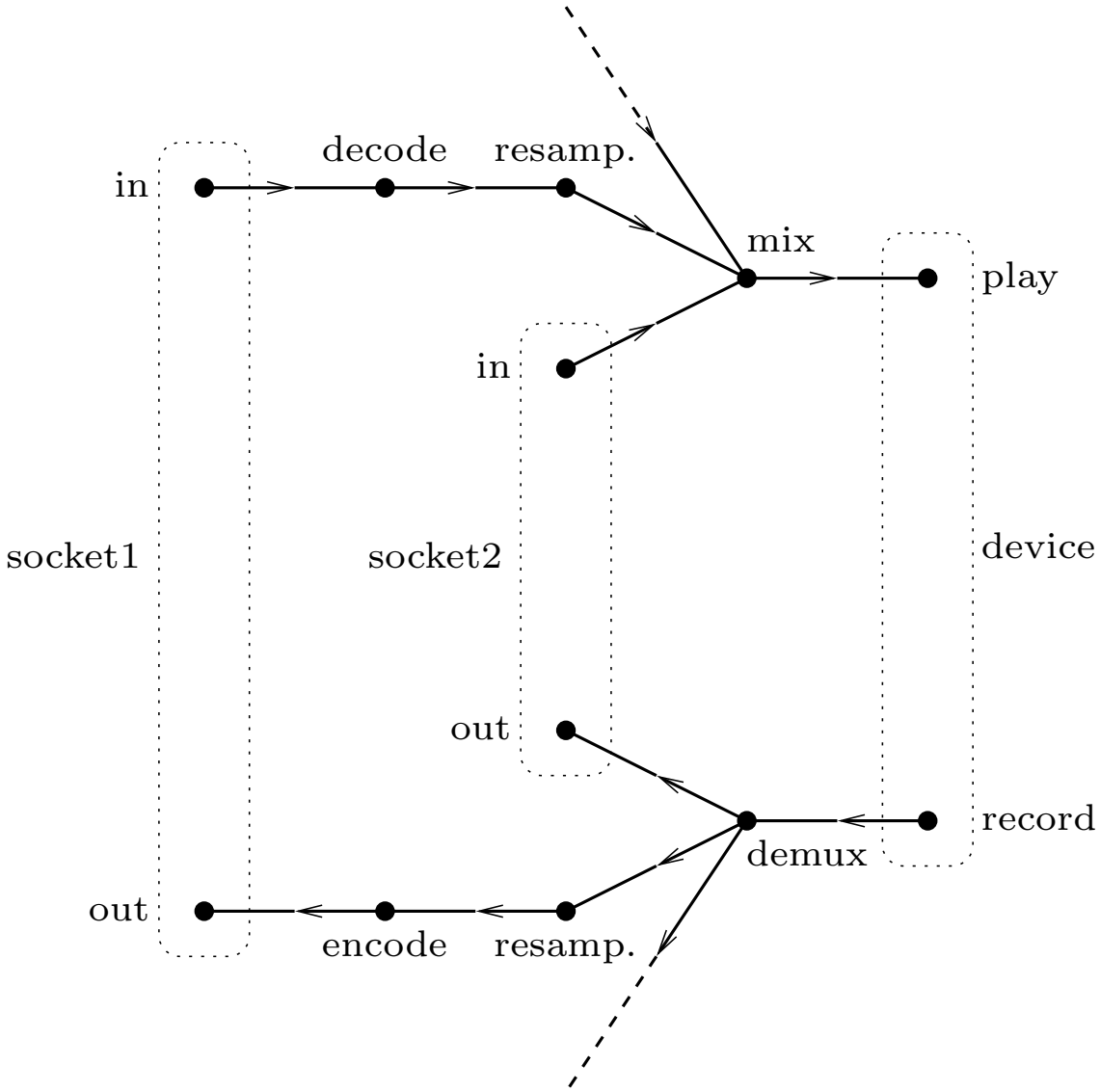
- Elementary data processing units with:
  - conversions, resampling (1 input, 1 output),
  - mixing (N inputs, 1 output),
  - channel extraction (1 input, N outputs),
  - socket I/O, file I/O (either no inputs or no outputs).
- Processing units are interconnected by FIFOs.
- Event-driven framework for non-blocking I/O (no threads).

Server = network of elementary units

Server behavior is determined by:

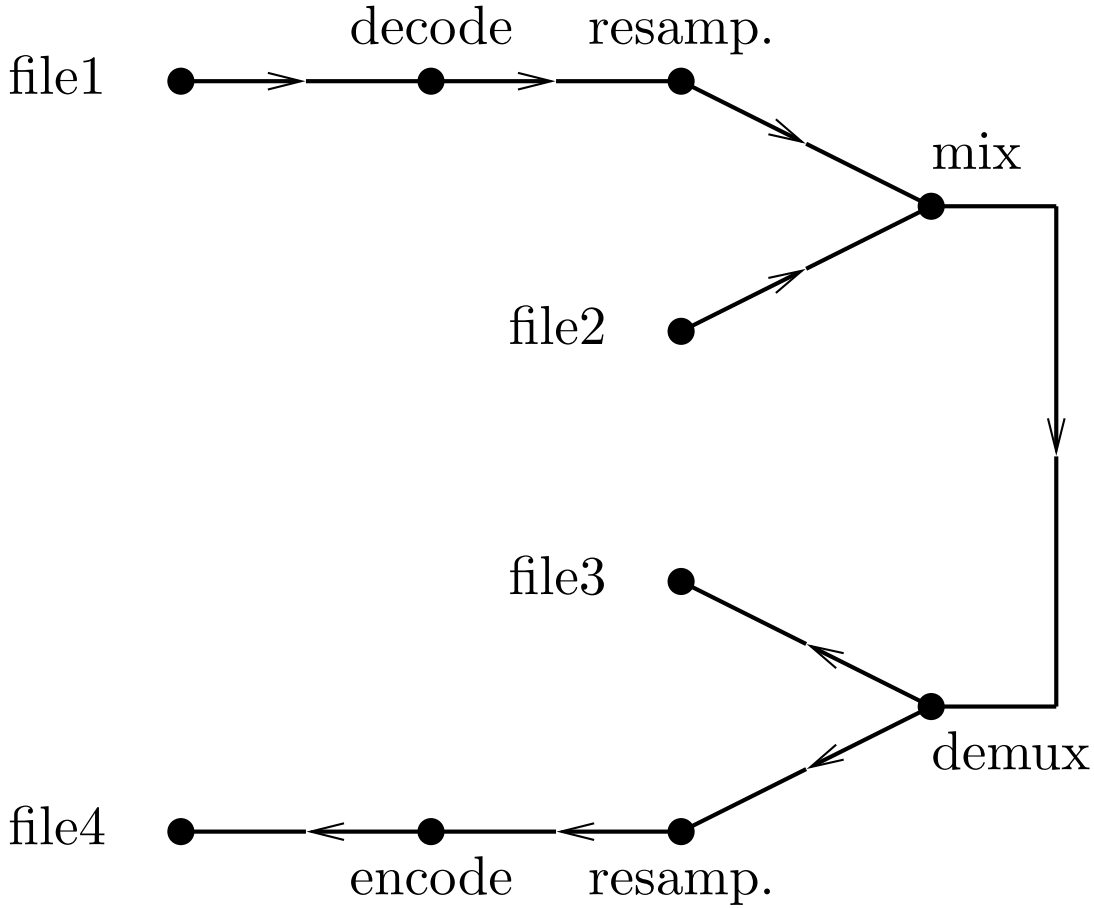
- the choice of processing units,
- the way they are interconnected.

Audio processing chain – server example



Audio processing chain – non-server example

`file1` and `file2` are mixed/merged and the result is stored into `file3` and `file4` (of different formats).



Latency – flow control

Latency

$$\texttt{latency} = \frac{\texttt{buffered}}{\texttt{rate}} = \frac{\texttt{written} - \texttt{played}}{\texttt{rate}}$$

$\Rightarrow$ Control the number of samples written.

Minimum theoretical latency:

- Single application:
  2 blocks (1 for the device + 1 for the application).
- Server with independent applications:
  1 extra block to allow 1 application to underrun without disturbing others.

Current implementation allows 3-block latency (i.e., the minimum).

MIDI controlled per-application volume

Why MIDI control?

- Standard since 1985.
- Very simple to use and impliment.
- Both software and hardware support MIDI.

Current limitations:

- Which MIDI channel corresponds to which application?
  Mapping should be exposed through the standard mixer interface.
- No MIDI control utility in OpenBSD yet
  one must use the bulky MIDI hardware or non-OpenBSD software.

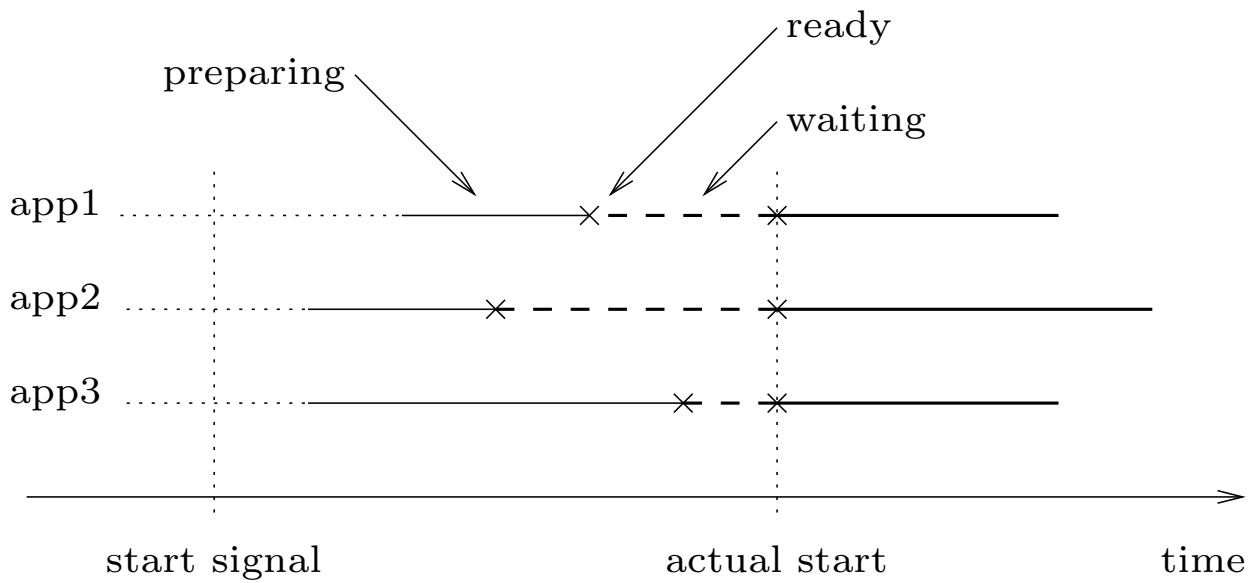
. . . work in progress!

MIDI controlled synchronization

Start process

- MMC “start” message blocks all streams.
- Once all streams are ready, the server starts.

⇒ No need to modify application code.



Sound card clock exposed through MIDI

Why?

Allows MIDI-aware software (or hardware!) to be synchronized to non-MIDI-aware audio programs.

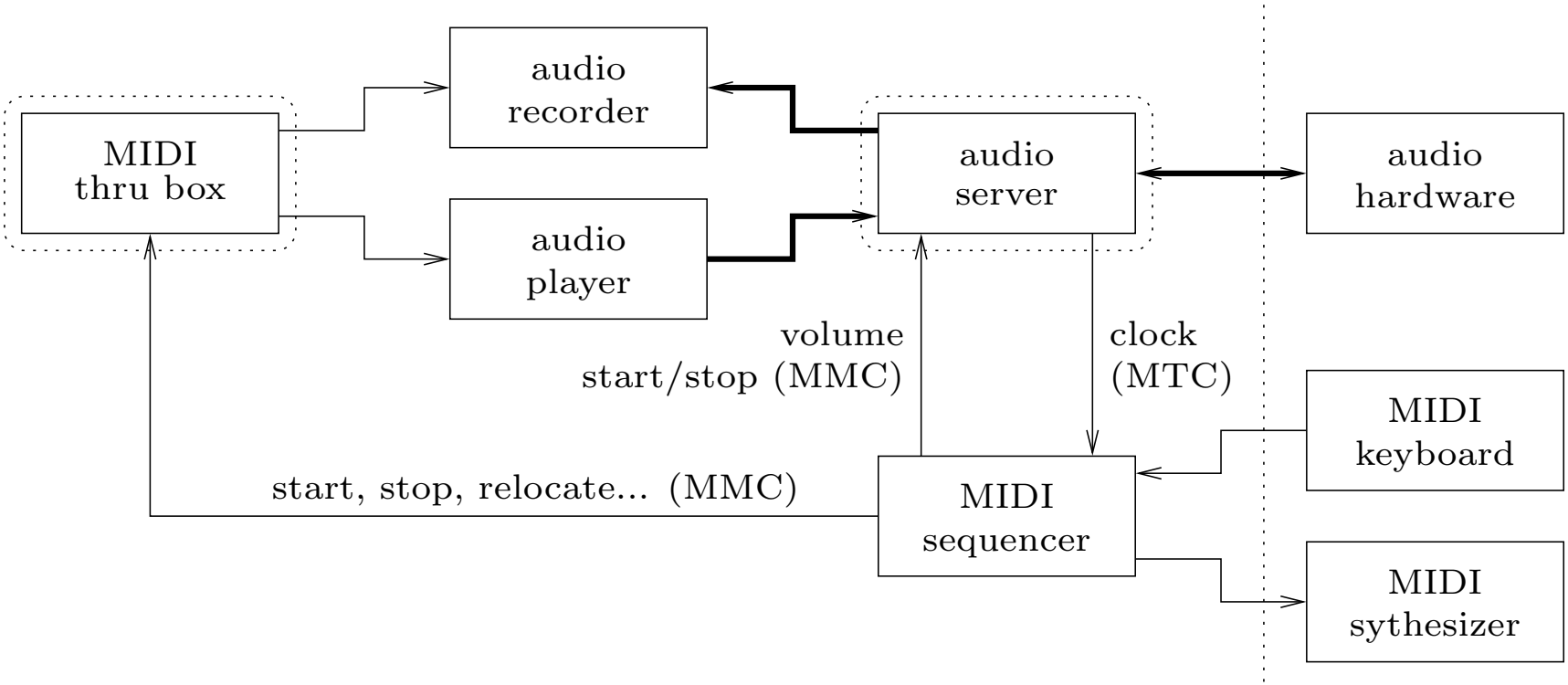
Typical scenario:

- The user manipulates a MIDI sequencer.
- The sequencer controls audio streams.
- The audio server sends feedback to the sequencer.
- The sequencer stays in sync to audio streams.

⇒ Simple programs work together to achieve a complex task.

Examples

Example: MIDI sequencer, audio player, audio recorder



Without the audio server and the MIDI thru box, the same task would require a monolithic application.

Integration in OpenBSD

- Single binary – to start the server on the default device:

  ```
  $ aucat -l
  ```

- No configuration file, options are on the command line:
  E.g., to create "`spkr`" and "`hp`" subdevices:

  ```
  $ aucat -l -c 0:1 -s spkr -c 2:3 -s hp
  ```

- Audio player, recorder and off-line conversion utility:

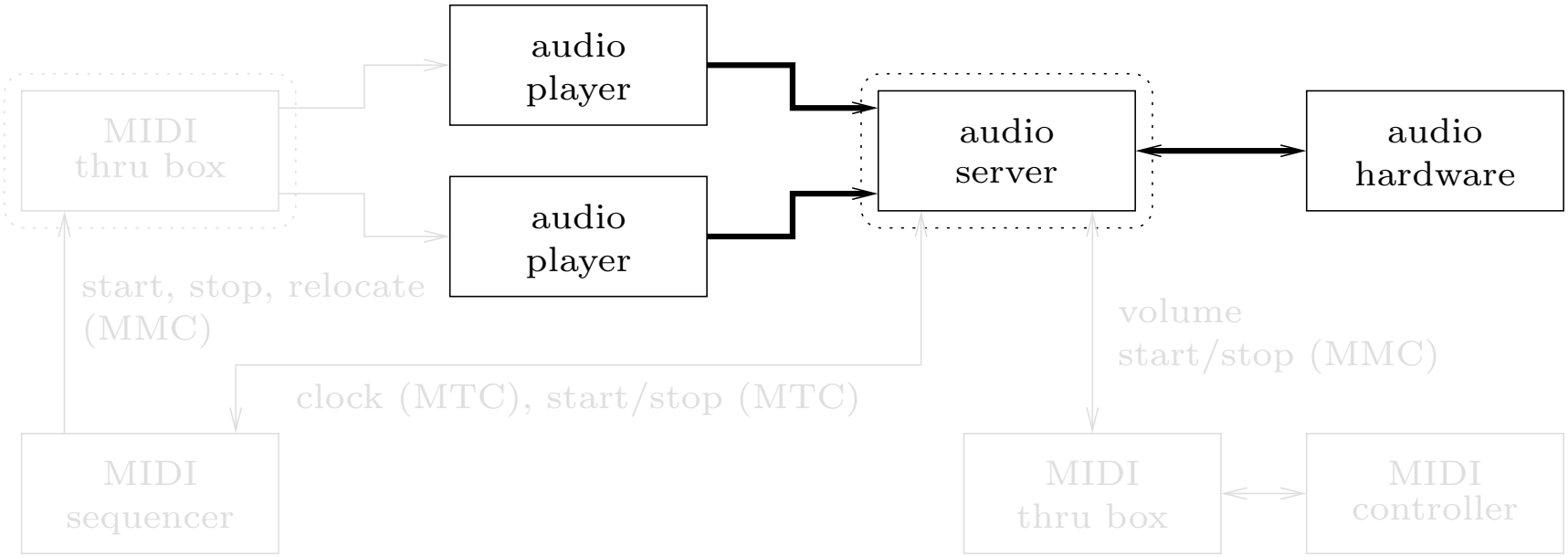
  ```
  $ aucat -i file_to_play.wav
  ```

- Devices can be either hardware (character devices) or software subdevices (server connections) new naming scheme is required:

  `<type>:<unit>[.subdevice]`

  E.g., "`aucat:0.hp`", "`rmidi:5`", "`midithru:0`", ...

Demo: simple tools working together



- Playing two files simultaneously.
- Controlling volume through MIDI.
- Synchronizing a MIDI sequencer to record automation.

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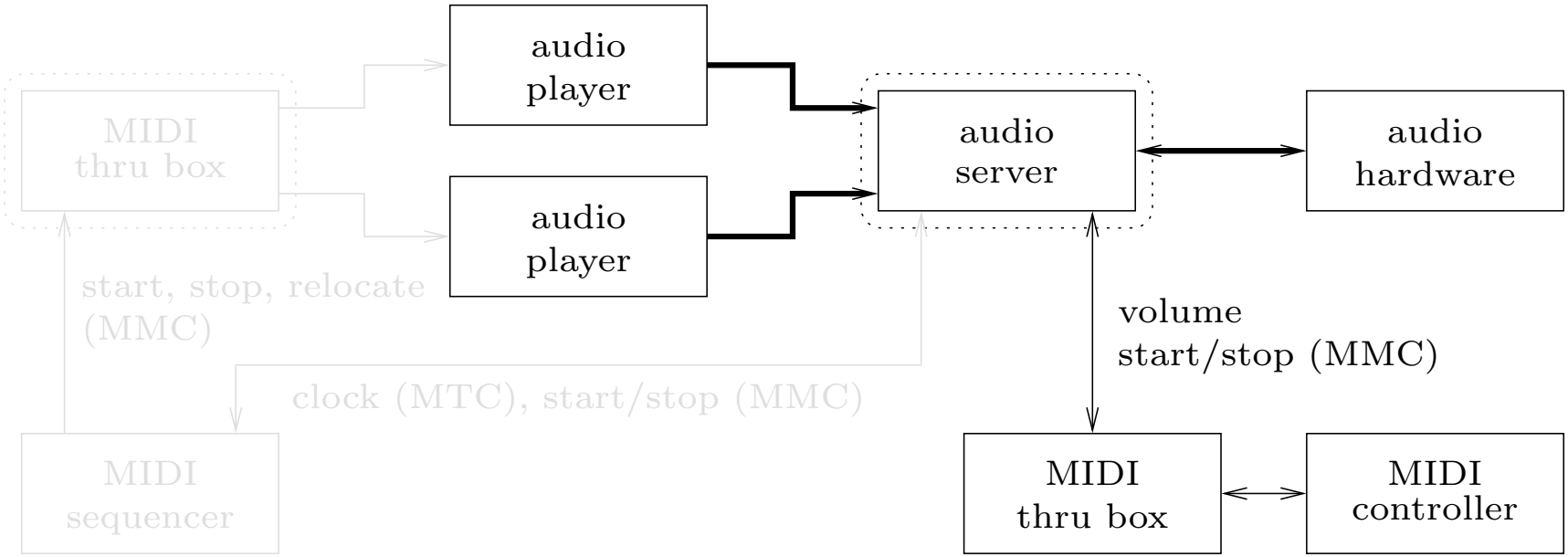

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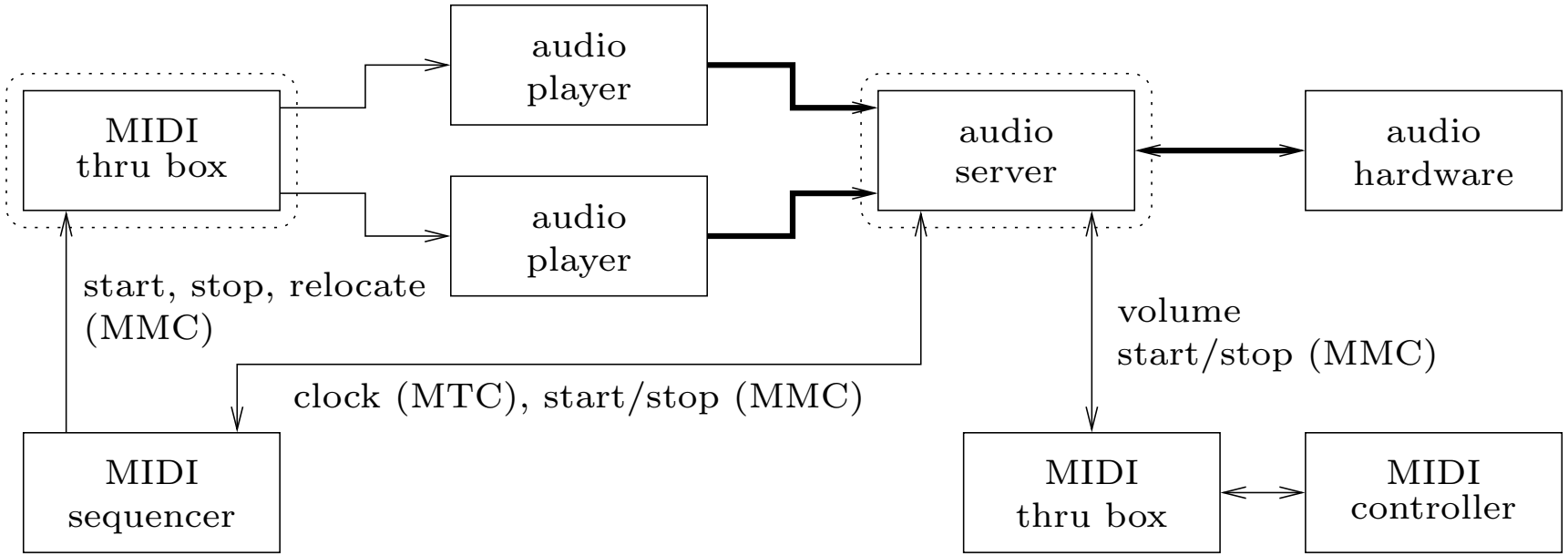

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Future work

- Less is more — keep it simple!
- Port more code to `sndio`, improve quality of existing code.
- MIDI mixer — use a single mixer framework.
- Record played streams (e.g., record from softsynths).
- Avoid useless data copying — use shared memory.

Conclusion

- Very simple framework (hopefully!).
  - User-space implementation.
  - Single binary, no configuration file.
- Stable and reliable by design.
  - Fast and structured non-blocking framework.
  - Strict latency control (theoretical minimum reached).
  - Synchronization maintained after underruns.
- Problems of desktop application addressed:
  - Conversions, resampling, mixing, channel mapping, volume control.
  - Subdevices – per-subdevice properties.
- Problems of music applications addressed:
  - MIDI controlled synchronization between applications.
  - Software MIDI ports allowing applications to communicate.