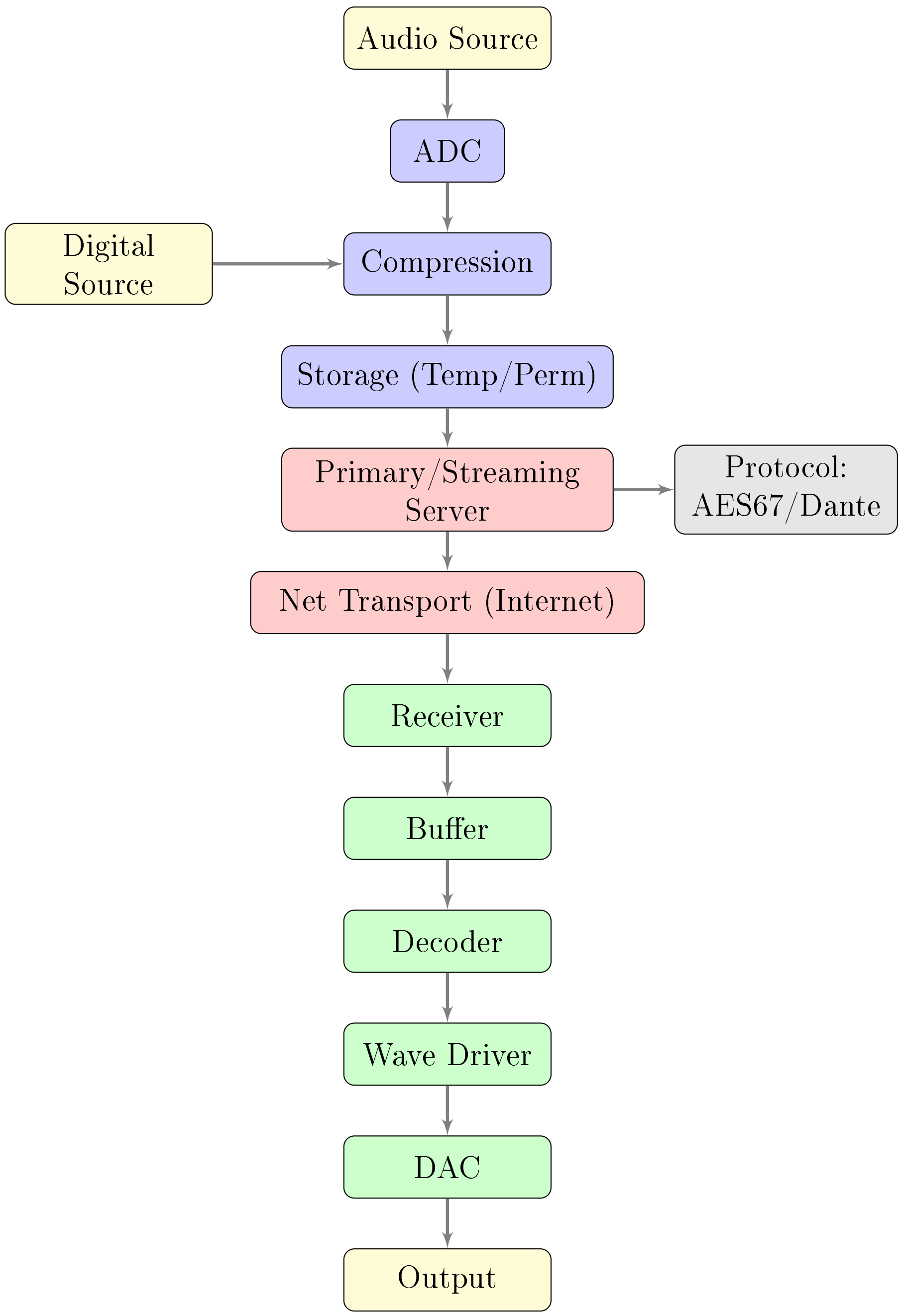
Transmission of High Quality Audio over IP Networks

1 General Concept of Streaming



2 Issues to Be Addressed

- the audio quality reduction is already noticable at 25 ms delay
- secure acoustical network

2.1 Overall Challenges

- low end-to-end (overall) delay in order to have high Quality of Experience (QoE)
- dealing with varying bandwidth (system scalability)
- high quality audio

2.2 Client

- encoder: hardware or software
- encoder optimisation: it may contribute to the total delay if not optimised properly
- error protection

2.3 Transport

2.4 Enduser

- Choosing the audio quality should be an option for either the sender or the receiver, depending on their broadband speed.
  - better quality — higher broadband speed
  - preferably loseless compression
  - perceptual codec(?)
- error concealment: packet loss replaced by the decoder
- scalable format — same media stream should be used on wide range of devices to avoid transcoding (which will increase overall delay)

AES67 is a briding protocol between Ravenna, Dante, etc. . .

3 First Iteration

- stereo in, stereo out
- no hardware involved
- plug-in format (VST and/or AU)
- using the same network, not over the internet
- loseless compression

Consider the following:

- version control
- cross-platform development tools (such as CMake)
- visual feedback (depends on the user's environment)

Useful example: VST3NetSend[1] on GitHub

[1] `https://github.com/vgorloff/VST3NetSend`