# TBAG - Execise 03 - Group C

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# Task 3.1

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In the lecture you heard about audio buffer properties. One aspect was the way channels are being represented. What is meant by interleaved and non-interleaved/planar audio buffers? How does SuperCollider Buffers implement channels? Review the documentation from within SuperCollider (Buffer help page).

# Task 3.1 - answer

Audio buffers store the sampled values per channel for all channels. There is interleaved and planar buffer formats.

In **interleaved** format, the channels of data are mixed together in frames (an audio frame is a -set of- sample per channel), where one frame contains one element from each channel. For example, the left and right channels would be stored like this:

LRLRLRLRLRLRLRLRLRLRLRLRLRLRLRLR (for a buffer of 16 frames)

In **planar format**  all of the samples in each channel are stored consecutively:

LLLLLLLLLLLLLLLLRRRRRRRRRRRRRRRR (for a buffer of 16 frames)

# Task 3.1

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In SuperCollider interleaved buffer format is used. Therefore the actual number of available values when requesting or setting values by index using methods such as set, setn, get, getn, etc., is equal to **numFrames * numChannels**. Indices start at 0 and go up to (numFrames * numChannels) - 1. In a two channel buffer for instance, index 0 will be the first value of the first channel, index 1 will be the first value of the second channel, index 2 will be the second value of the first channel, index 3 will be the second value of the second channel and so on.

# Task 3.2

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Read provided document on audio buffers & latency.

What is meant by latency?

What causes latency and how do you calculate it?

# Task 3.2 - answer

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**Latency** is a period of delay (usually measured in milliseconds) between when an audio signal enters and when it emerges from a system.

Audio latency in a digital audio system is caused by delays processing the audio data as it travels from the outside world to the computer's processor and back out again. Each analog-to-digital and digital-to-analog conversion adds latency on the order of milliseconds to the system.

# Task 3.2 - answer

In computer based audio systems a certain amount of latency, known as audio buffering, is necessary to ensure that playback, recording and processing results in an error-free audio stream without dropouts or glitches. The input buffer must fill up before the digitized audio data is sent along the audio stream to output. Buffer size influences latency. For example, if buffer size is 256 samples and sampling rate is 44.1 kHz, then it will take (256/44,100) seconds which is 0.0058 seconds or 5.8ms to fill the buffer.

Thus, total latency including the time for ADC, DAC, and buffer-filling is on the order of milliseconds.

# Task 3.3

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Take the example implementation of filling a Buffer with sine waves and play around with the sample rate argument. In which way does the sound change if you provide a sample rate that is (a) lower resp. (b) higher than that of the server?

Read the documentation of Buffer.loadCollection and allocate, fill, and play a Buffer with a 2-second stereo sine wave at 624 Hz with Amplitude 0.7 (left) and 625 Hz with Amplitude 0.58 (right). What happens at the loop point?

# Task 3.3

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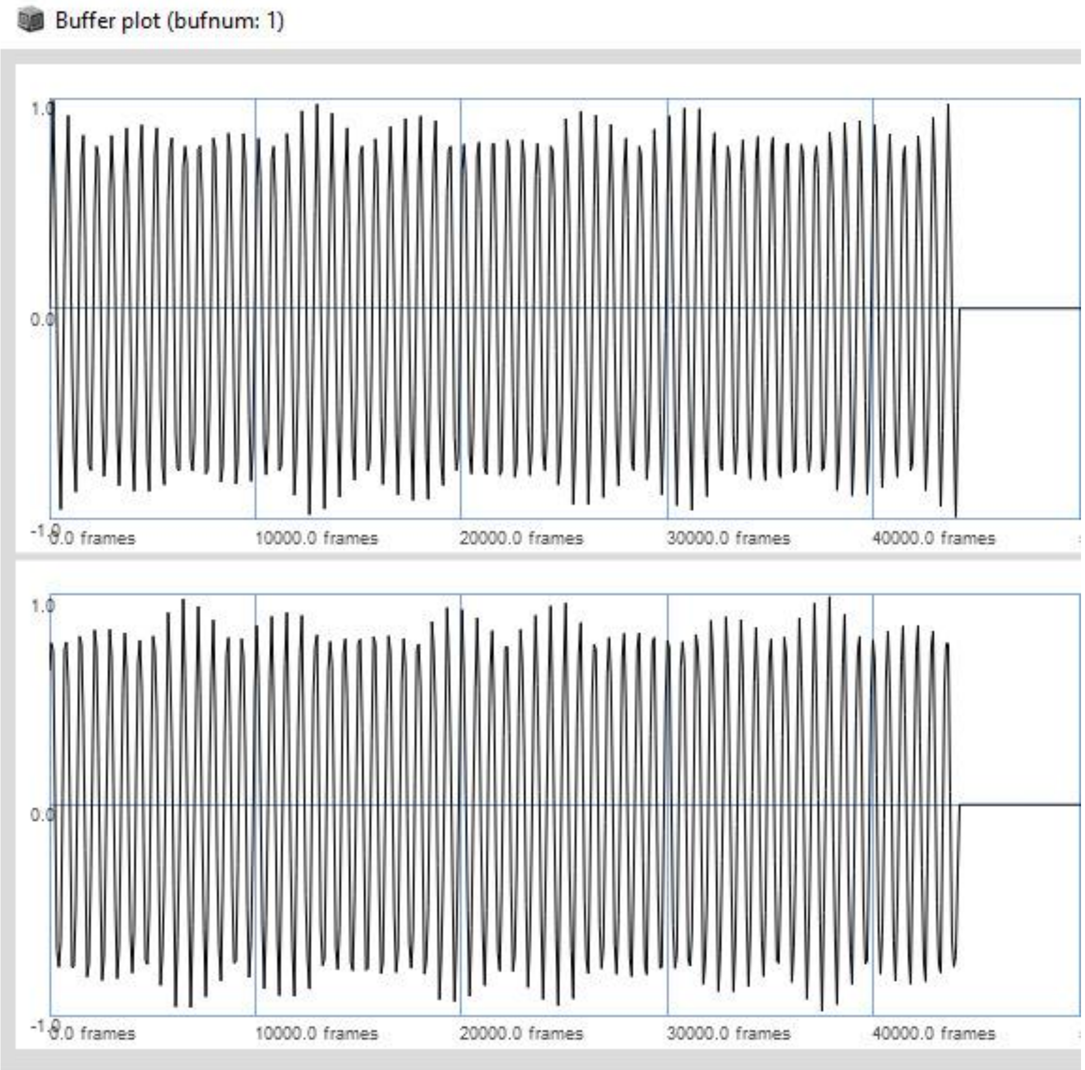
Lower sample rate



# Task 3.3

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Higher sample rate