

# C++ Real-Time Audio Programming with Bela

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# Course topics

## Programming topics

Working in real time  
Buffers and arrays  
Parameter control  
Classes and objects  
Analog and digital I/O  
Filtering  
Timing in real time  
Circular buffers  
State machines  
MIDI  
Block-based processing  
Threads  
Fixed point arithmetic  
ARM assembly language



## Music/audio topics

Oscillators  
Samples  
Wavetables  
Control voltages  
Gates and triggers  
Filters  
Metronomes and clocks  
Delays and delay-based effects  
Envelopes  
ADSR  
MIDI  
Additive synthesis  
Phase vocoders  
Impulse reverb

Today

# Lecture 11: Circular buffers

## What you'll learn today:

Keeping track of previous audio samples  
Why and how to use circular buffers  
Creating an audio delay

## What you'll make today:

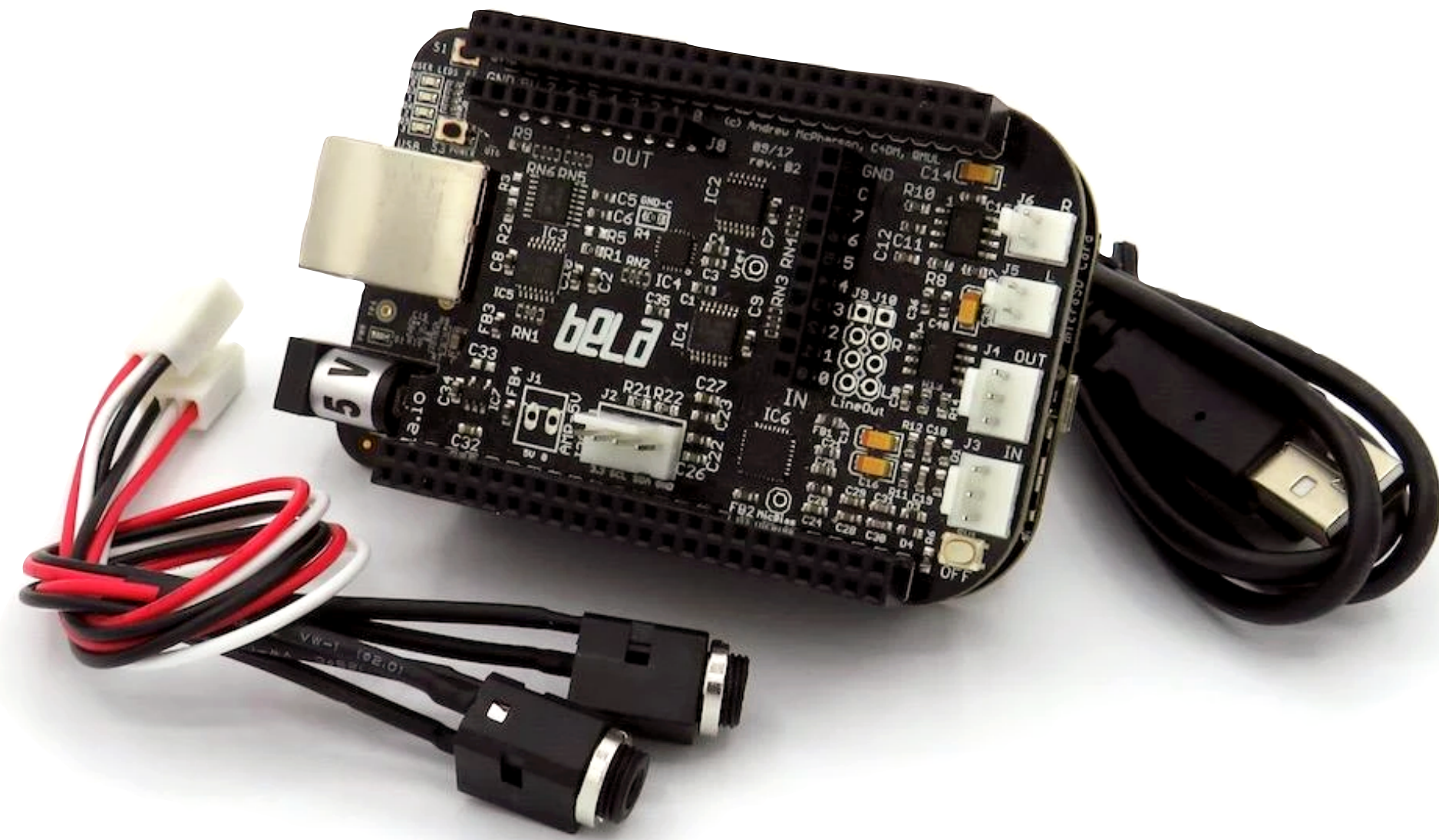
An adjustable delay and an echo effect

## Companion materials:

**[github.com/BelaPlatform/bela-online-course](https://github.com/BelaPlatform/bela-online-course)**



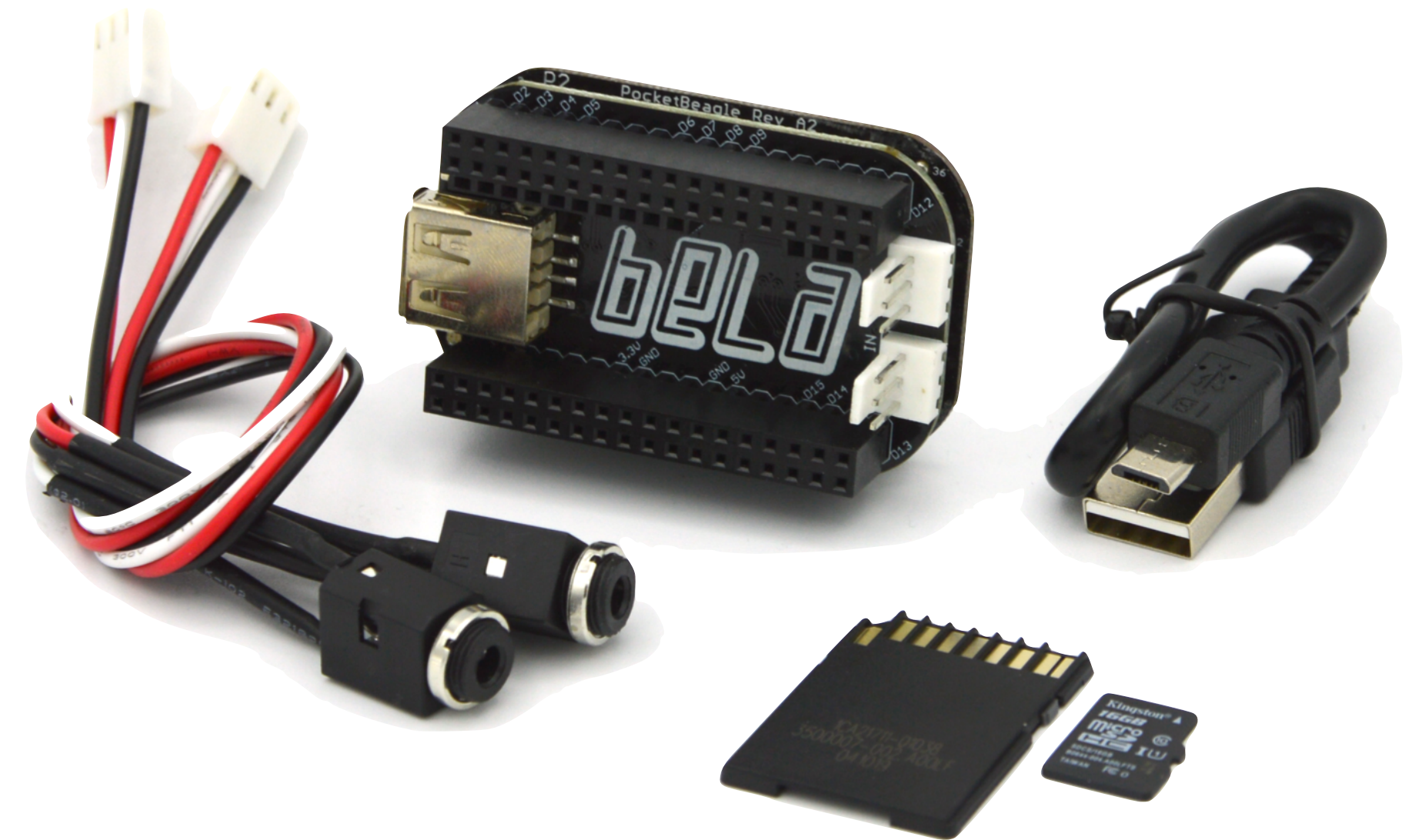
# What you'll need



Bela Starter Kit

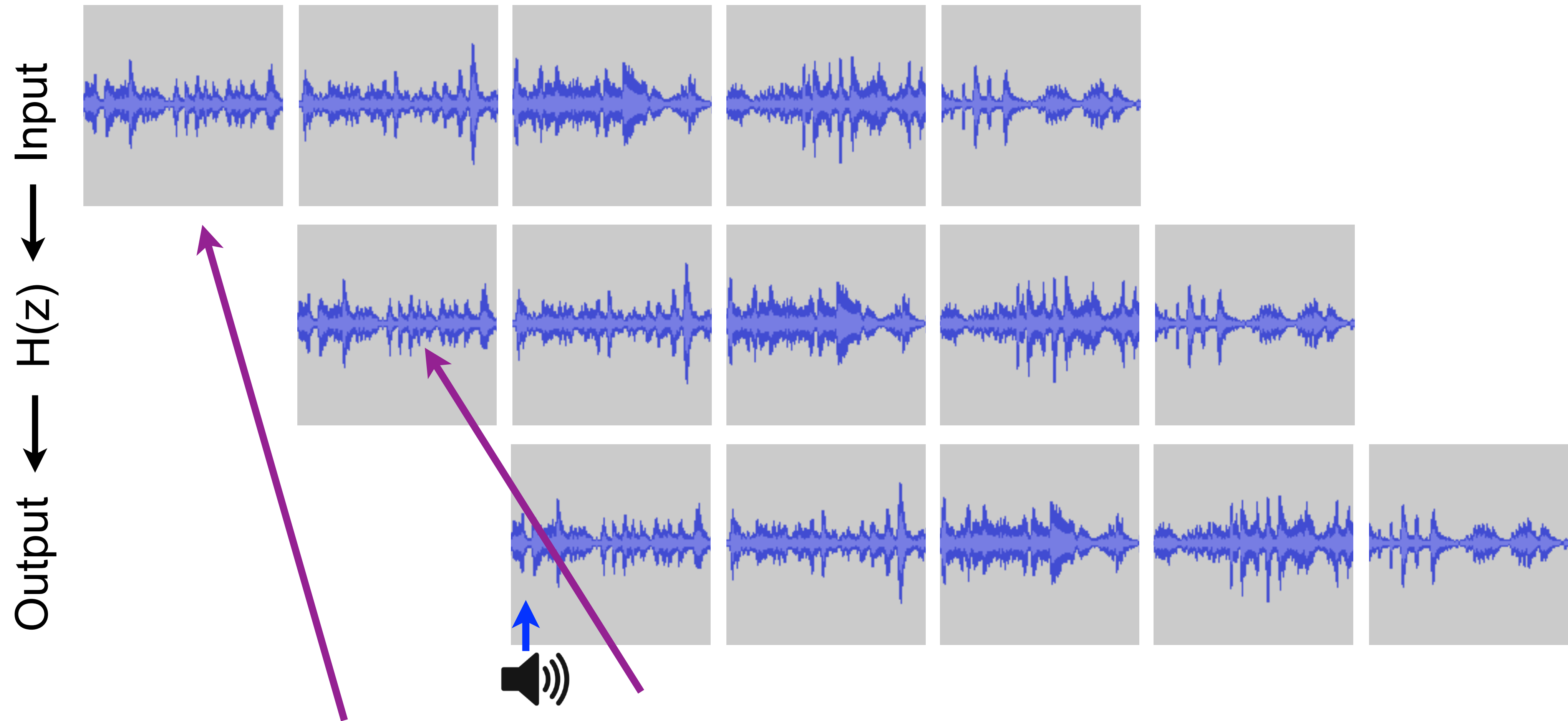
or

[\[shop.bela.io\]](http://shop.bela.io)



Bela Mini Starter Kit

# Review: audio buffering

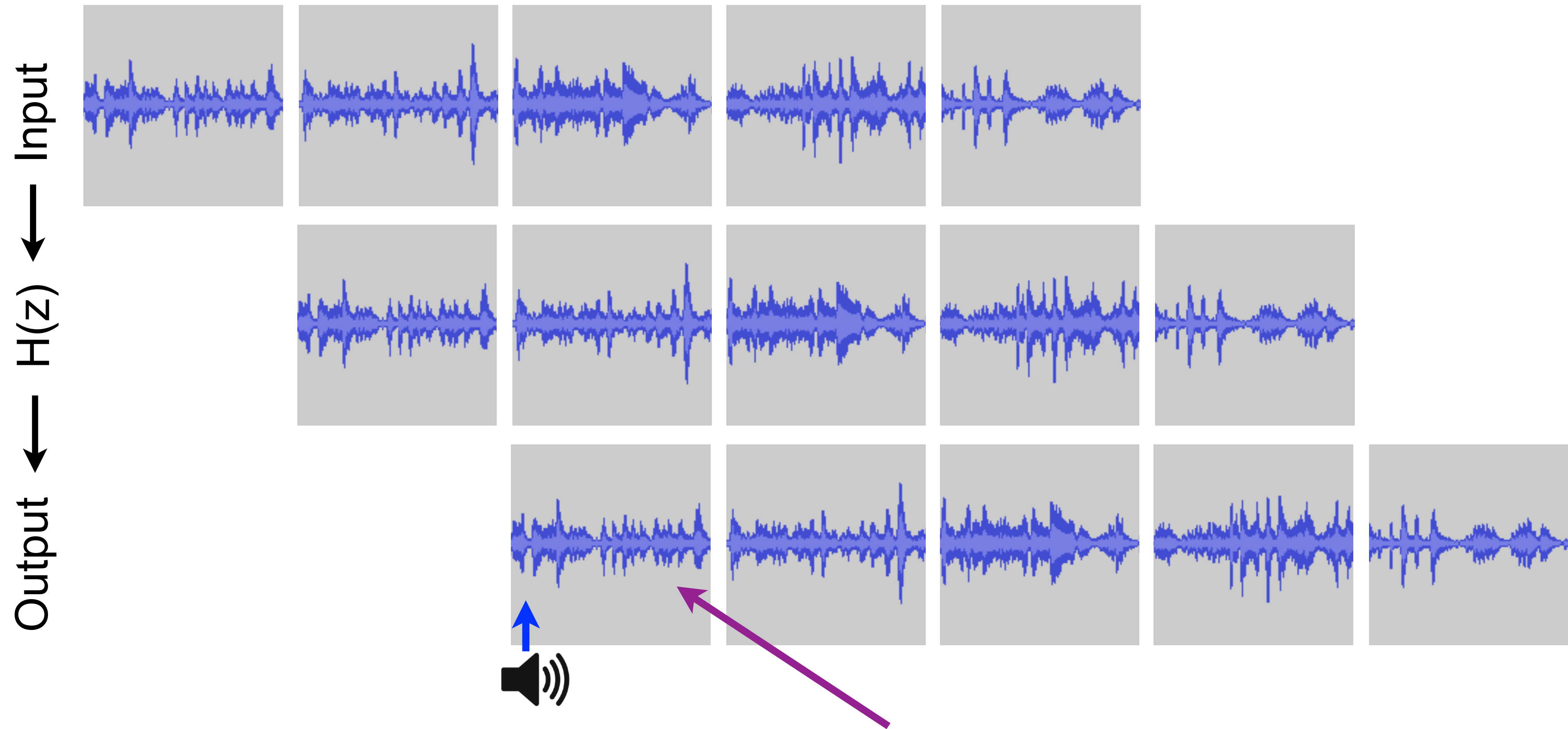


1. First we fill up  
a buffer of samples

2. We process this buffer  
while the next one fills up

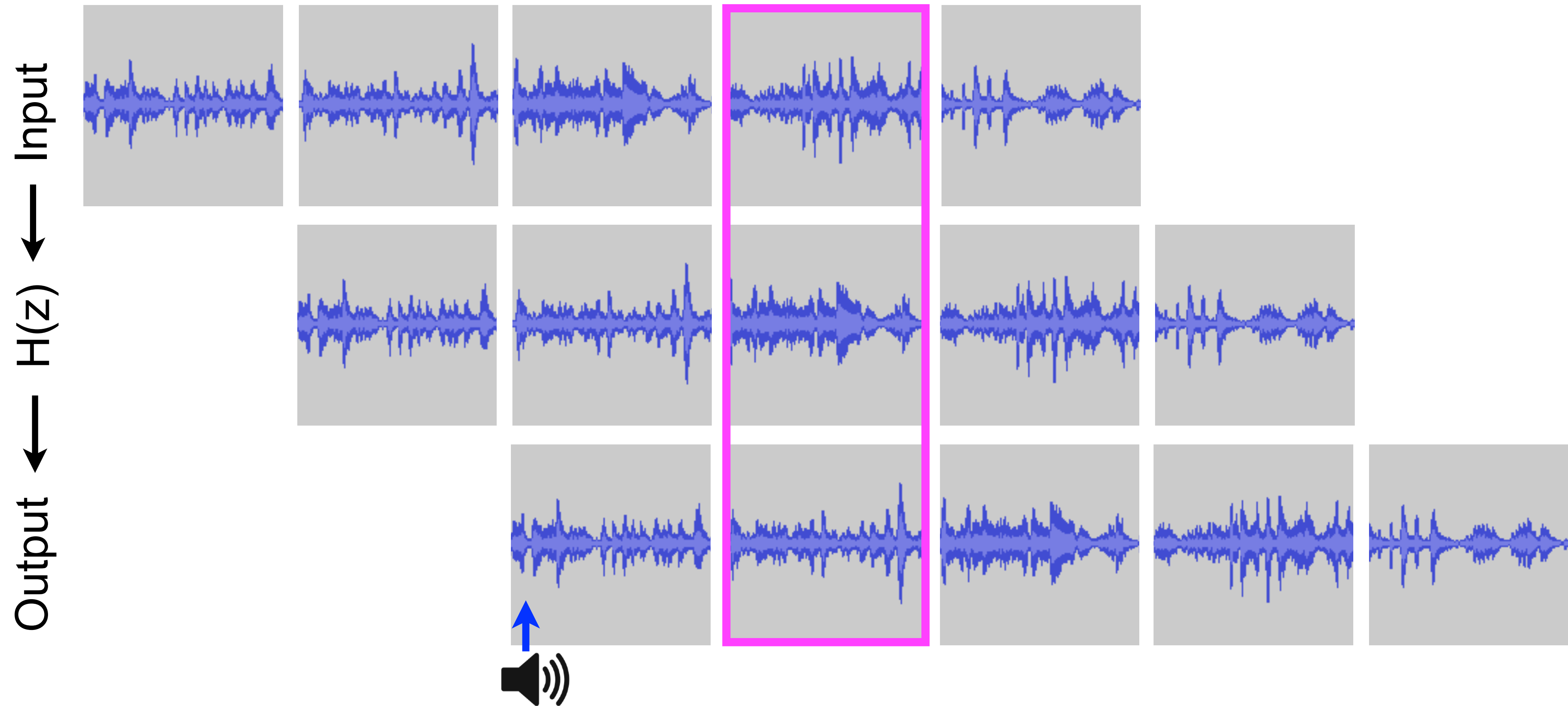


# Review: audio buffering



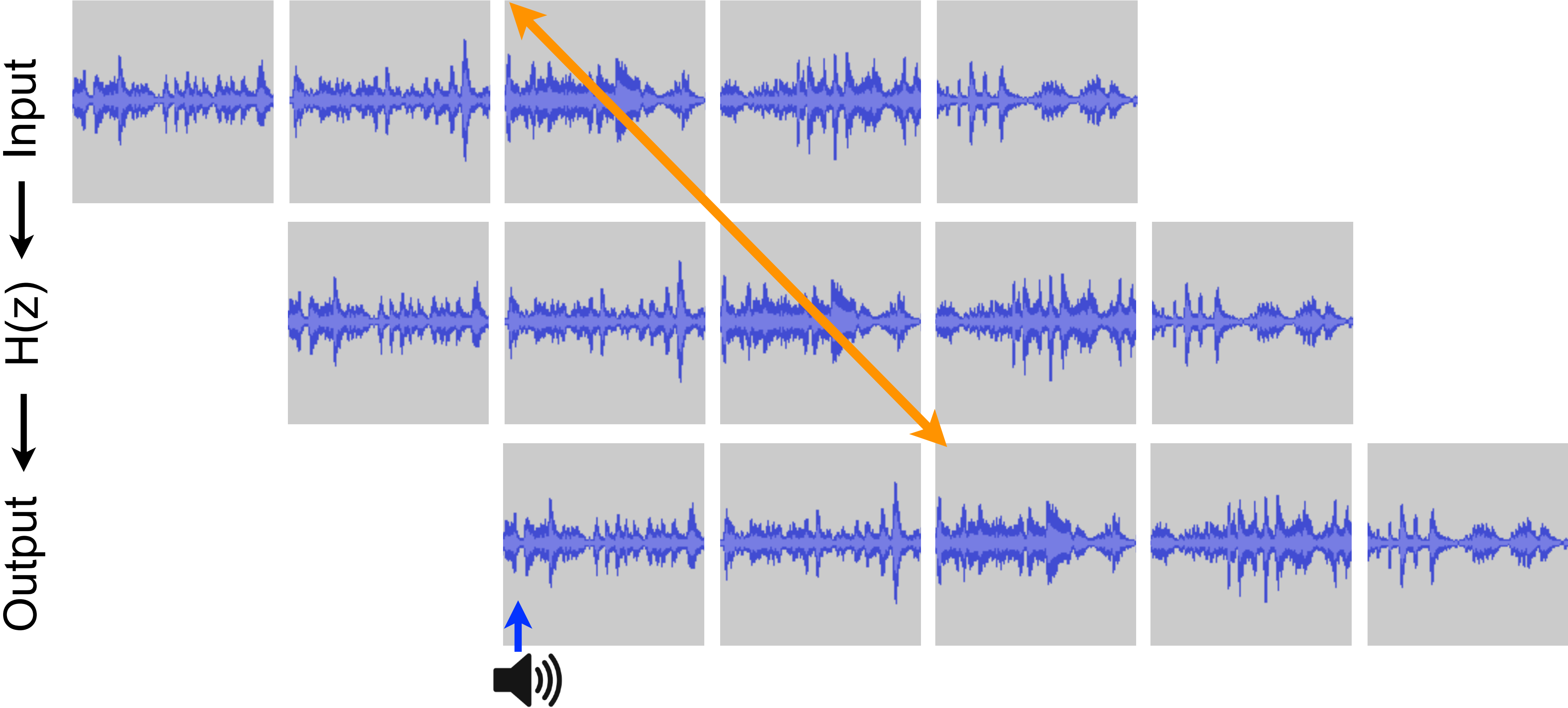
3. Next cycle, we send this buffer to the output

# Review: audio buffering



At any given time, we are reading from ADC,  
processing a block, and writing to DAC

# Review: audio buffering



Total latency is 2x buffer length



# Review: a simple filter

- We want to implement this filter:  $y[n] = x[n] - x[n - 1]$

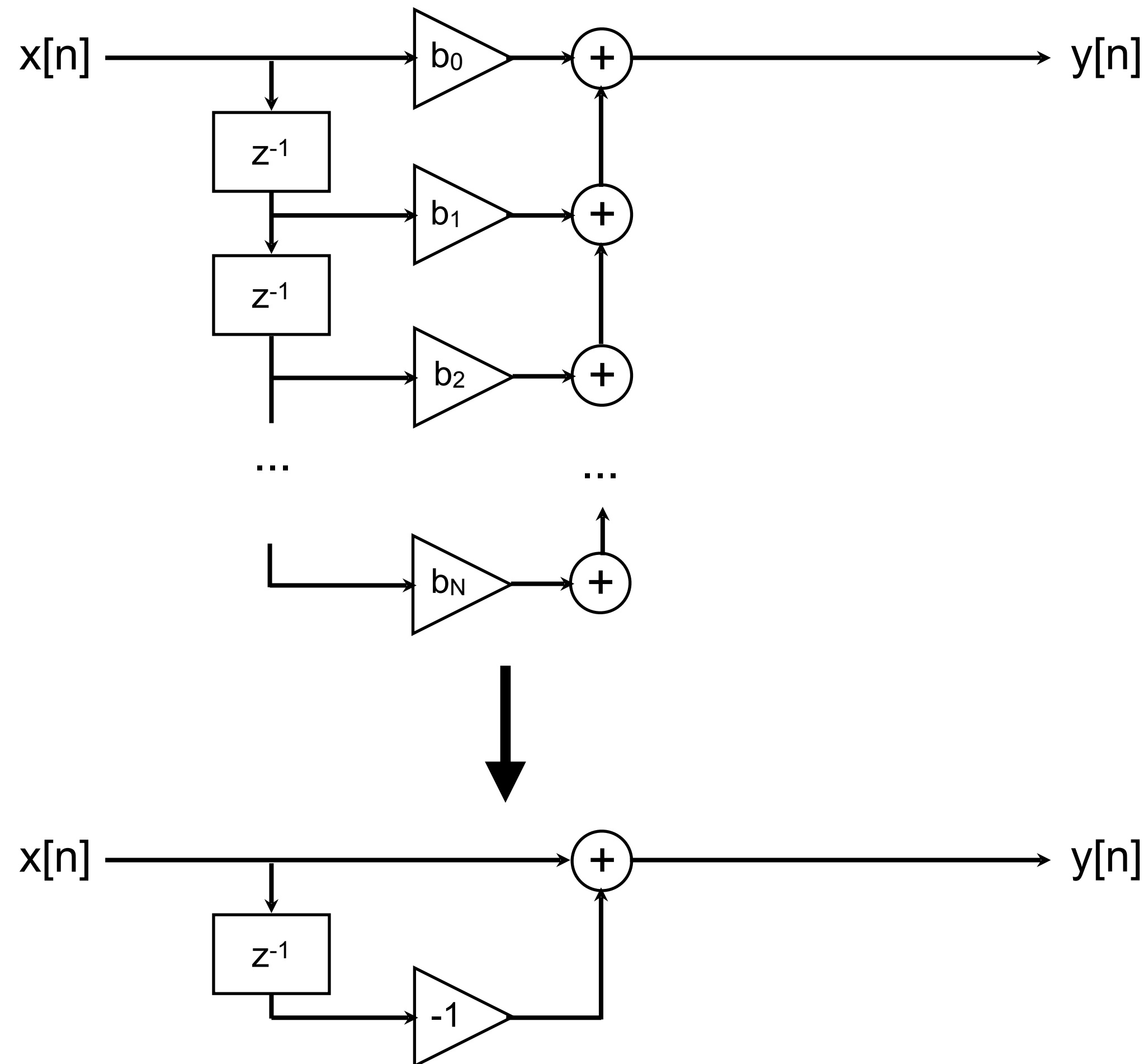
- Block diagram for an FIR filter:

- What are the **coefficients**  $b_n$ ?

- $b_0 = 1$
- $b_1 = -1$

- What information do we need to keep track of?

- Previous value of  $x[n]$
- Use a global variable






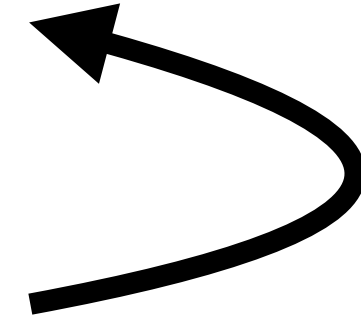
# Saving a previous input

Calculating the filter  $y[n] = x[n] - x[n - 1]$

`float gLastSample = 0;`  global variable to hold  $x[n - 1]$

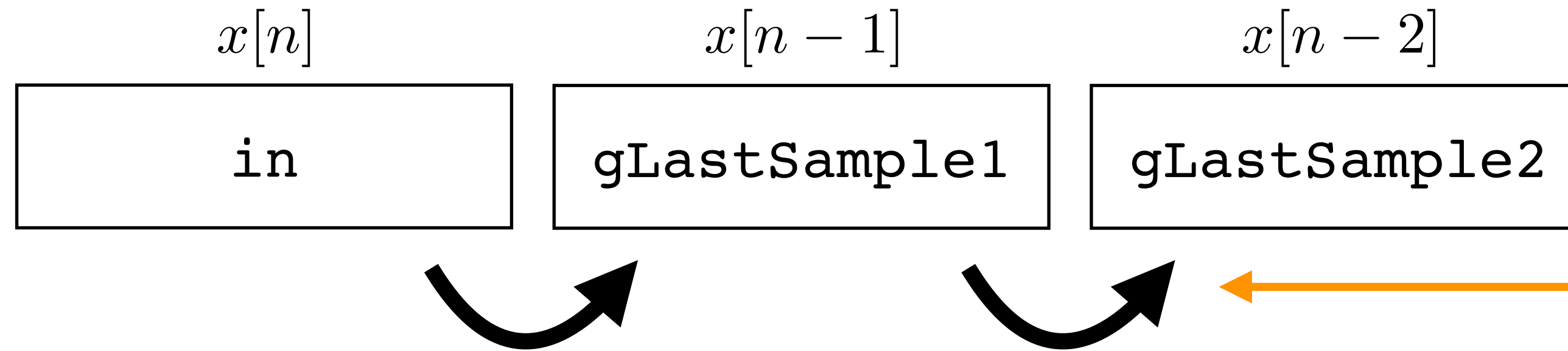
---

```
for(unsigned int n = 0; n < context->audioFrames; n++) {  
    // ...let's say "in" holds our input, calculated somehow:  
    float in =   
  
    // Here we implement a first-order FIR filter:  
    // y[n] = x[n] - x[n-1]  
    float out = in - gLastSample;  so that the next time  
    gLastSample = in;  set gLastSample to  $x[n]$ ...  
}
```



# Saving 2 previous inputs

- Say we want to calculate  $y[n] = x[n] + x[n - 1] + x[n - 2]$
- How do we save 2 previous inputs?
  - Need two global variables (we'll call them gLastSample1 and gLastSample2)
  - Update them at the end of the loop, like before:



**Important!** This assignment has to happen first. (Why?)

```
// Here we implement a second-order FIR filter:  
//  $y[n] = x[n] + x[n-1] + x[n-2]$   
float out = in + gLastSample1 + gLastSample2;  
gLastSample2 = gLastSample1;  
gLastSample1 = in;
```

- Do we still need gLastSample1 here?  $y[n] = x[n] + x[n - 2]$

# Saving many previous inputs


- Let's extend the previous concept to saving the last 100 samples:

```
float gLastSample1;  
float gLastSample2;  
float gLastSample3;  
float gLastSample4;  
float gLastSample5;  
float gLastSample6;  
float gLastSample7;  
float gLastSample8;  
float gLastSample9;  
float gLastSample10;  
float gLastSample11;  
float gLastSample12;  
float gLastSample13;  
float gLastSample14;  
float gLastSample15;  
float gLastSample16;  
float gLastSample17;  
float gLastSample18;  
float gLastSample19;  
float gLastSample20;  
float gLastSample21;  
float gLastSample22;  
float gLastSample23;  
float gLastSample24;  
float gLastSample25;  
float gLastSample26;  
float gLastSample27;  
float gLastSample28;  
float gLastSample29;  
float gLastSample30;  
float gLastSample31;  
float gLastSample32;
```



# Saving many previous inputs

- Let's extend the previous concept to saving the last 100 samples:
- **Better plan:** use an **array** for the previous samples

`float gLastSamples[100] = {0};`  Simple way to initialise all array elements to 0

- ▶ Let's define the indices like this:  $\text{gLastSamples}[k] \longleftrightarrow x[n - 1 - k]$ 
  - So for example,  $\text{gLastSamples}[0]$  corresponds to  $x[n - 1]$
- ▶ Where would we find  $x[n - 100]$ ?  $\text{gLastSamples}[99]$
- ▶ What does  $\text{gLastSamples}[37]$  hold?  $x[n - 38]$
- ▶ What does  $\text{gLastSamples}[100]$  hold?
  - **Nothing!** Array has only 100 elements, so valid indices are 0 to 99
- **Task:** write some pseudocode to save the last 100 samples
  - ▶ Implement the equation  $y[n] = x[n - 100]$

# Saving many previous inputs

```
float gLastSamples[100] = {0};
```

```
for(int n = 0; n < context->audioFrames; n++) {  
    float in =   
    float out = gLastSamples[99];
```

$$y[n] = x[n - 100] \longrightarrow$$

```
// Move every sample back one element in the array  
// Notice: have to start from back
```

```
for(int i = 99; i > 0; i--)  
    gLastSamples[i] = gLastSamples[i - 1];
```

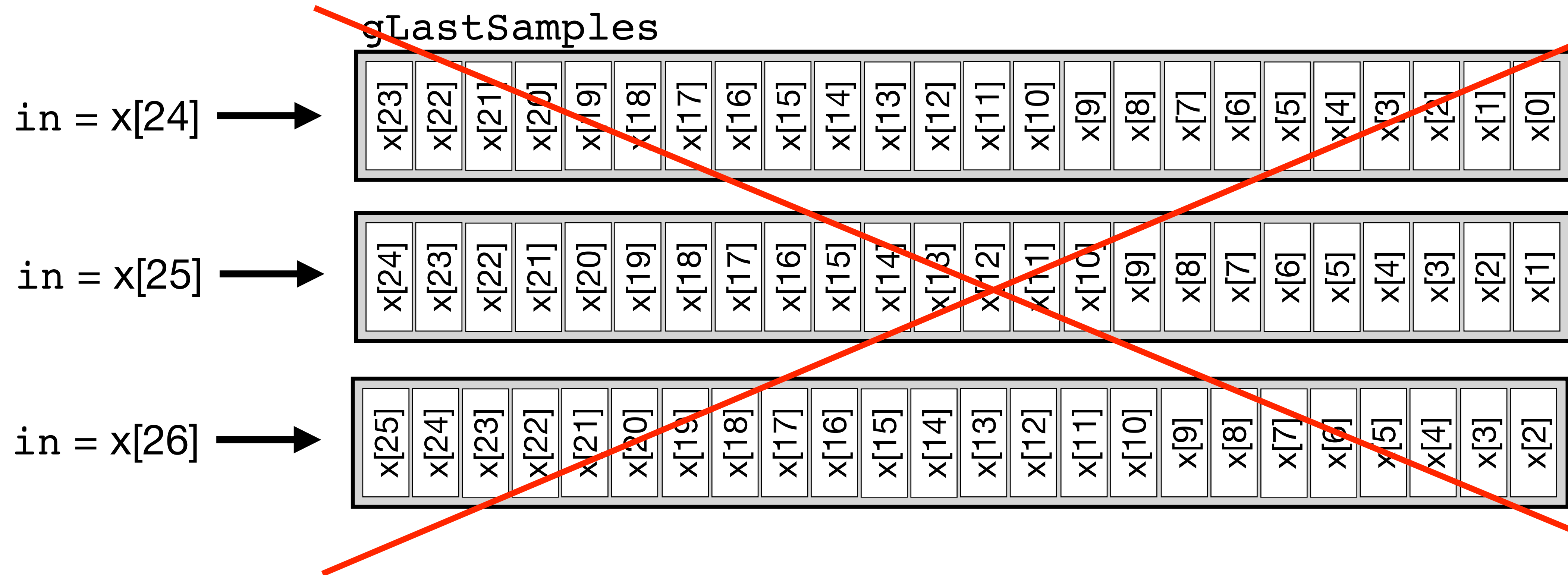
```
// First element in the array is the most recent input sample  
gLastSamples[0] = in;
```

```
}
```

- Notice use of internal `for ( )` loop
  - Also notice its direction: **decreasing**
  - Why `i > 0` and not `i >= 0` ?
    - Can't access `gLastSamples[-1]`
- What is a drawback of this whole approach? **Inefficiency!**

# Moving samples

- Moving memory around is wasteful!
  - In this illustration, we are saving the last 24 samples:

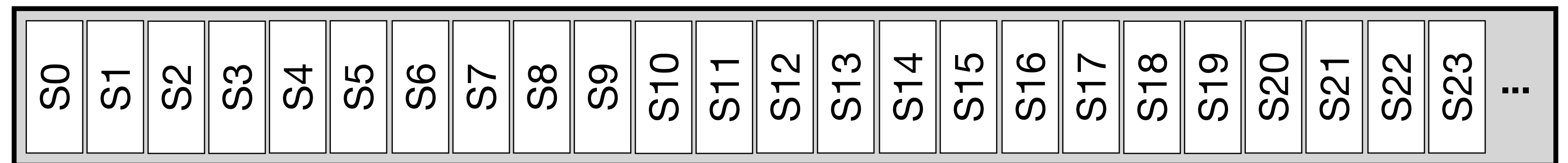
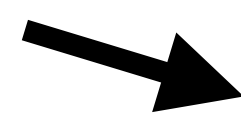


- **Bad idea:** don't move 23 samples to add 1

# Circular buffering

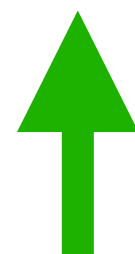
- Instead, leave the old samples in place when we add new ones
  - This is called a **circular buffer**: a memory buffer that acts like a loop
    - Write each new sample in the next location, from beginning to end
    - When we get to the end, go back to the beginning again
    - Keep track of the **write pointer**, which tells us which slot we write to next
- 
- Before getting into those details, let's review **reading** from a buffer...
    - The **read pointer** was a global variable that kept track of which sample we were reading

the buffer doesn't  
change as we play it



but the read pointer moves

**read pointer**

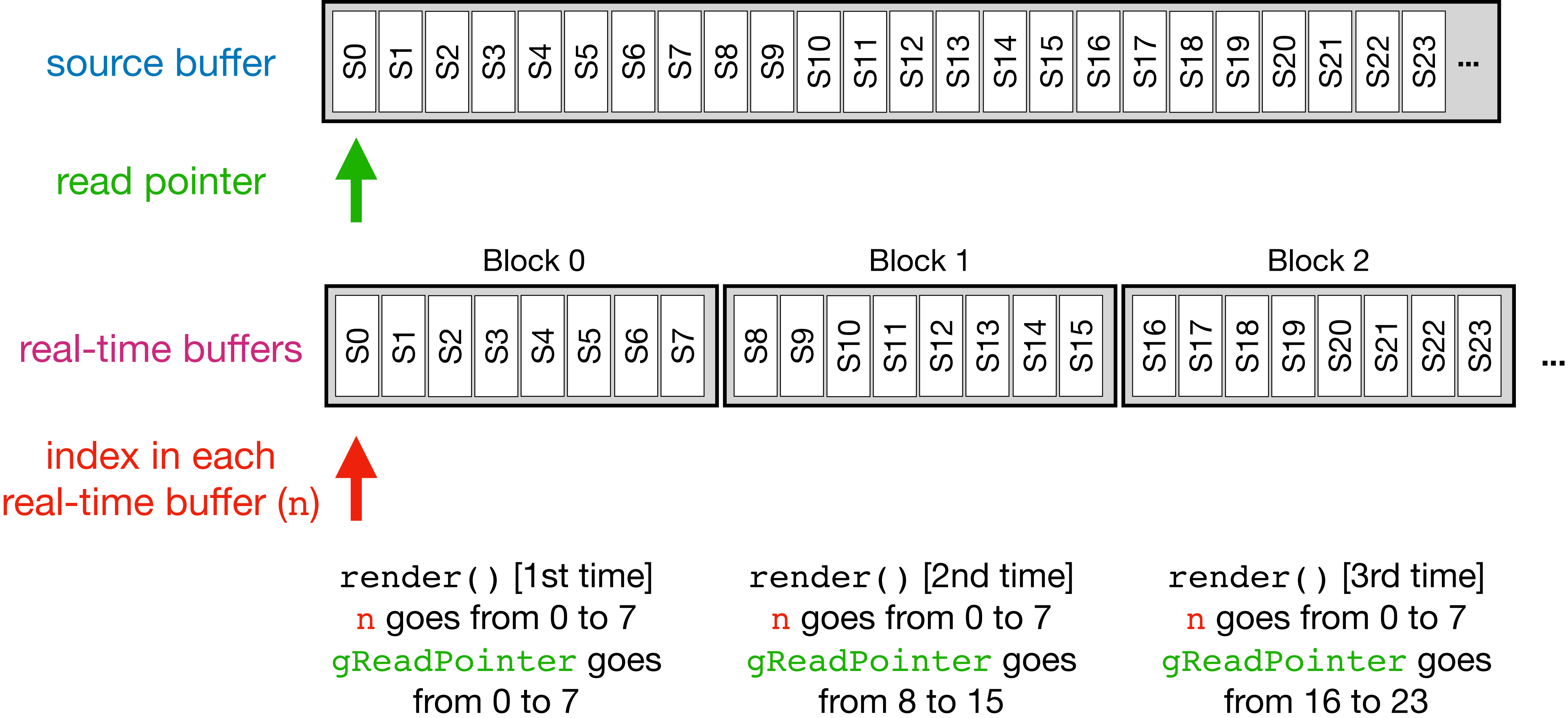




# Review: indexing

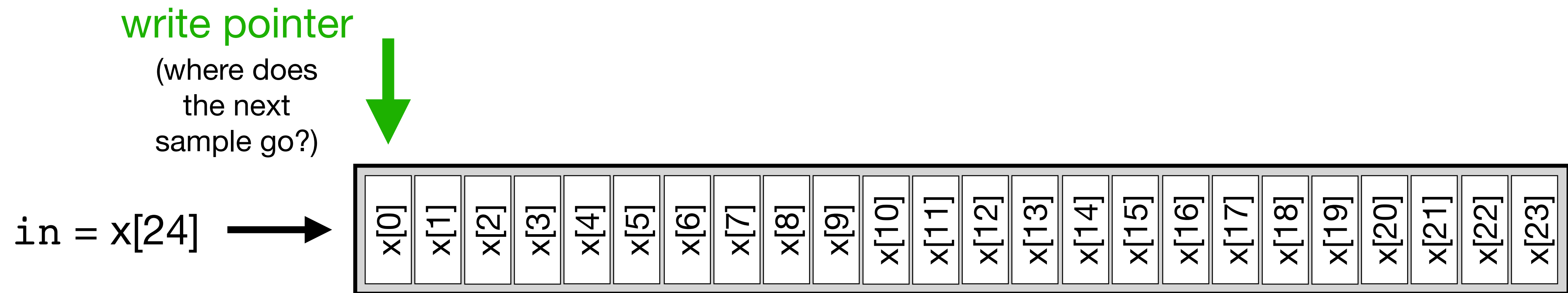
- One of the hardest parts of working with buffers can be keeping track of what each index means
- In this case, we've got two different kinds of **buffers** to think about:
  1. The **recorded sound** (let's call it the **source buffer**)
    - Only one buffer whose contents don't change
    - Length: **number of samples in the source sound** (possibly long)
  2. The buffer for each **real-time audio block**
    - A new buffer each time `render()` is called, accessed via `audioWrite()`
    - Length: **block size of the real-time system** (e.g. 16)
- Therefore, we need to keep track of two **indexes**:
  1. Where are we **playing** in the source buffer? (**read pointer** or **play head**)
  2. Where are we **writing** in the output buffer? (starts over from 0 each block)

# Review: indexing



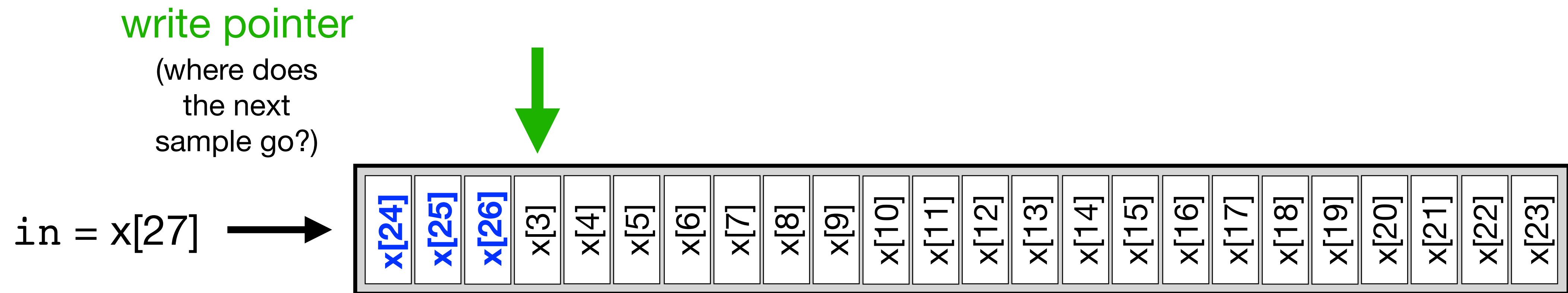
# Circular buffering

- Back to writing to our circular buffer:
- A **circular buffer** is a memory buffer (**array**) that acts like a loop
  - Write each new sample in the next location, from beginning to end
  - When we get to the end, go back to the beginning again
  - Keep track of the **write pointer**, which tells us which slot we write to next
- The buffer always ends up holding the N most recent samples
  - We just need to keep track of which sample is held where



# Circular buffering

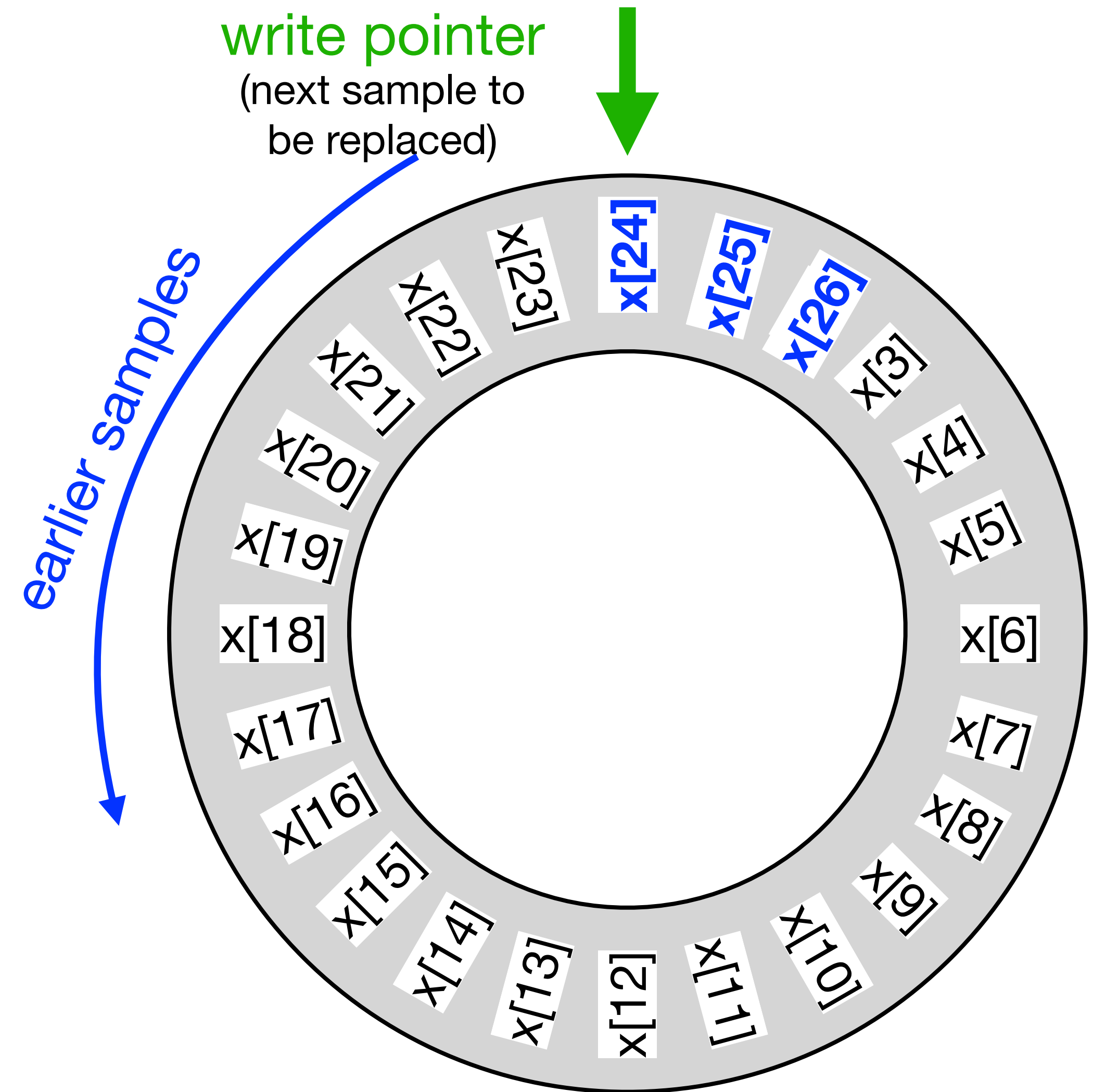
- Back to writing to our circular buffer:
- A **circular buffer** is a memory buffer (**array**) that acts like a loop
  - Write each new sample in the next location, from beginning to end
  - When we get to the end, go back to the beginning again
  - Keep track of the **write pointer**, which tells us which slot we write to next
- The buffer always ends up holding the N most recent samples
  - We just need to keep track of which sample is held where





# Circular buffering

- Another equivalent view:
- The **write pointer** tells us where to find the **front** of the buffer
  - Points just past the most recent sample
    - i.e. it's the **oldest** sample in the buffer until it's replaced
  - To find earlier samples, look backward from the write pointer
- At any given time:
  - The buffer holds the **N most recent** samples
  - **Each individual sample never moves** until it is eventually replaced



# Write pointer

- The circular buffer has two components:
  1. A **region in memory** (array) to store the samples
  2. A **write pointer** to keep track of where we are
- Remember, there is **no functional beginning or end** to a circular buffer!
- In code, we need to declare two (global) variables:

```
std::vector<float> gDelayBuffer;  
unsigned int gWritePointer = 0;
```

- ▶ When we have a new sample, **store it** at the write pointer, then **increment** the pointer

```
gDelayBuffer[gWritePointer] = in;  
gWritePointer++;
```

- ▶ What else do we need to do?

- **Keep the write pointer in range**

```
if(gWritePointer >= gDelayBuffer.size())  
    gWritePointer = 0;
```

# Circular buffer task

- Using the **circular-buffer** code example from the companion materials
- **Task:** implement a **0.5-second delay** on **only the left channel**
  - The right channel should have no delay, so the difference can be clearly heard
- You will need to:
  - **Declare variables** for the buffer (use `std::vector<float>`) and the write pointer
  - **Allocate the buffer** to hold 0.5 seconds (see the `std::vector::resize()` method)
  - **Initialise the write pointer** in a sensible place (e.g. at 0)
  - **Read samples** out of the buffer which are 0.5 seconds old
  - **Store samples** in the buffer as they come in, and **move the write pointer**
- **Hint:** the write pointer always points to the **oldest sample in the buffer**
  - If you set your buffer size correctly, you only need a single pointer!

# Circular buffer code

```
std::vector<float> gDelayBuffer;
unsigned int gWritePointer = 0;

bool setup(BelaContext *context, void *userData)
{
    // [...]
    // Allocate the circular buffer to 0.5 seconds
    gDelayBuffer.resize(0.5 * context->audioSampleRate);
    return true;
}

void render(BelaContext *context, void *userData)
{
    for(unsigned int n = 0; n < context->audioFrames; n++) {
        float in = 

        // Read the output from the write pointer (oldest sample)
        float out = gDelayBuffer[gWritePointer];

        // Overwrite the buffer at the write pointer, the increment and wrap pointer
        gDelayBuffer[gWritePointer] = in;
        gWritePointer++;
        if(gWritePointer >= gDelayBuffer.size())
            gWritePointer = 0;

        // Write the input and output to different channels
        audioWrite(context, n, 0, in);
        audioWrite(context, n, 1, out);
    }
}
```



# Indexing in a circular buffer

- The circular buffer isn't literally a circle in the computer's memory
  - If we fall off the end, we need to wrap the index around to the beginning
- Suppose our buffer is `gDelayBuffer` and is 100 samples long
  - What index is **2 samples older** than `gDelayBuffer[51]`?
    - `gDelayBuffer[49]`
  - What index is **2 samples older** than `gDelayBuffer[1]`?
    - `gDelayBuffer[99]` **(why?)**
  - What index is **5 samples older** than `gDelayBuffer[0]`?
    - `gDelayBuffer[95]`
  - How do I find **100 samples older** than `gDelayBuffer[10]`?
    - **Can't!** If a given sample is in the buffer, then there are only 99 more stored there.
- What is the generic way of doing this?
  - **Modulo arithmetic**

# Modulo arithmetic

- $x \% y$  (“x mod y”) gives the **remainder** after  $x$  is divided by  $y$ 
  - If  $x > 0$  and  $y > 0$ , then the range of  $x \% y$  is **0 to  $y-1$** 
    - For example:  $5 \% 2 = 1$
- Modulo arithmetic completes the “**circle**” in the circular buffer
  - It lets us always stay in the right range of array indices
  - It wraps around when we give it an index off the end of the buffer
- How do we use modulo arithmetic to implement a circular buffer?
  - **What is the value of  $y$**  in the expression above?
    - **The buffer size**
  - `gDelayBuffer[(n + 2) % 100];`  $\longrightarrow$  2 samples forward (later) in buffer
  - `gDelayBuffer[(n - 2) % 100];`  $\longrightarrow$  **2 samples backward (earlier) in buffer?**

# Modulo arithmetic

- $x \% y$  (“x mod y”) gives the **remainder** after  $x$  is divided by  $y$ 
  - If  $x > 0$  and  $y > 0$ , then the range of  $x \% y$  is **0 to  $y-1$** 
    - For example:  $5 \% 2 = 1$
  - But if  $x < 0$ , result will be negative:  **$-(y-1)$  to 0**
    - For example:  $-5 \% 2 = -1$
    - **This is clearly not what we want!**
    - **Even worse, it is language-dependent.** This is not, for example, not how Python implements modulo.
- What is the solution to keep the indices in range?
  - Always **add** one or more **multiples of the buffer size**  

```
std::vector<float> gDelayBuffer;  
float twoSamplesBeforeN =  
    gDelayBuffer[(n - 2 + gDelayBuffer.size()) % gDelayBuffer.size()];
```
  - Here, even if  $n < 2$ , the modulo will be positive

# Circular buffer task 2

- **Task: without changing buffer size**, change delay to **0.1 seconds**
  - Now you can no longer read the oldest sample in the buffer
  - You will need to use **modulo arithmetic** on the write pointer to look backward by 0.1 seconds
  - How many samples is 0.1 seconds at 44.1kHz sample rate?



# Circular buffer code

```
std::vector<float> gDelayBuffer;
unsigned int gWritePointer = 0;
unsigned int gOffset = 0;  ← offset between pointers in samples

bool setup(BelaContext *context, void *userData)
{
    // Allocate the circular buffer to 0.5 seconds
    gDelayBuffer.resize(0.5 * context->audioSampleRate);
    // Calculate the offset based on the sample rate
    gOffset = 0.1 * context->audioSampleRate;  ← offset calculated here
    return true;
}

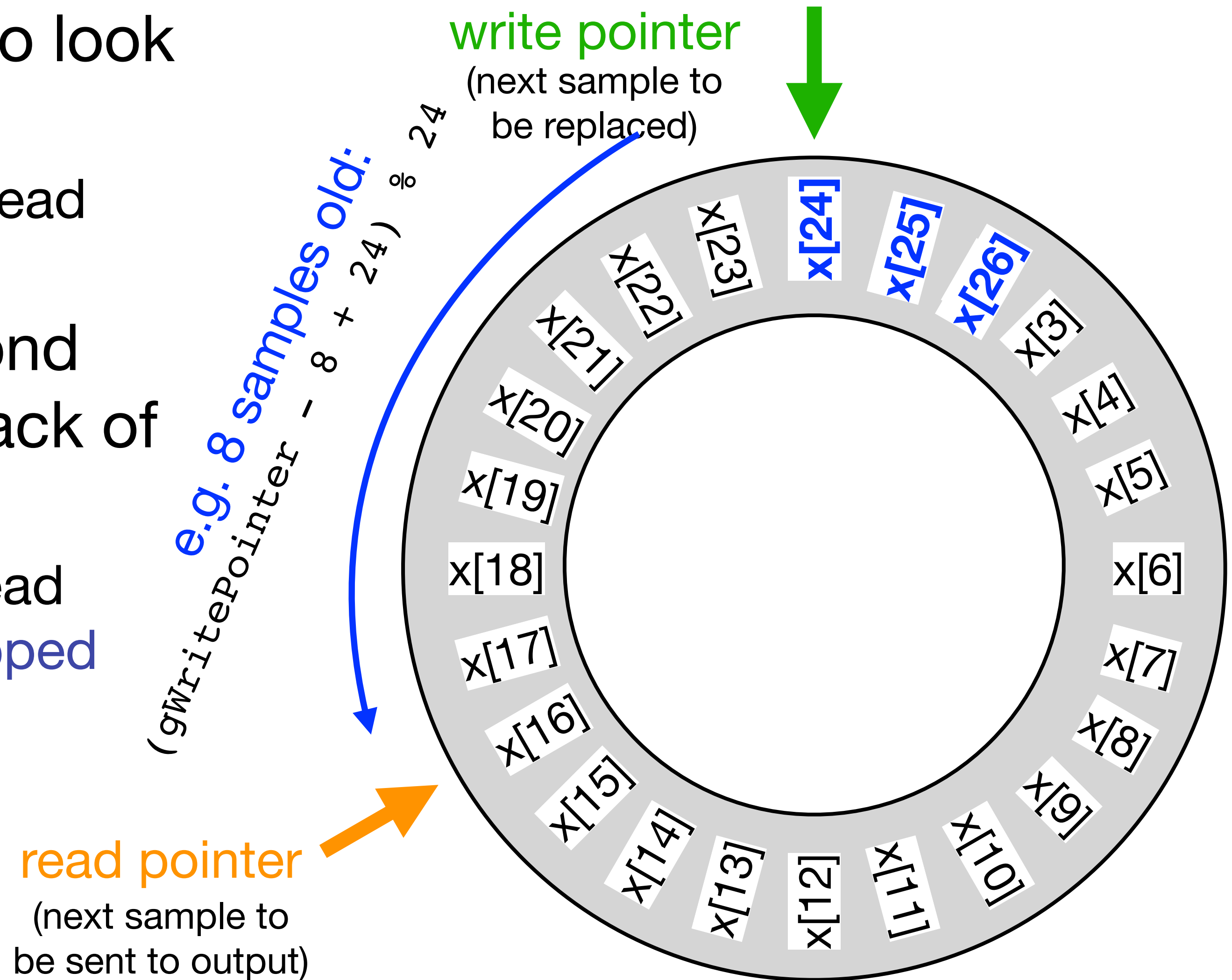
void render(BelaContext *context, void *userData)
{
    for(unsigned int n = 0; n < context->audioFrames; n++) {
        float in =  

        // Read the output from the write pointer (oldest sample)
        float out = gDelayBuffer[(gWritePointer - gOffset + gDelayBuffer.size()) % gDelayBuffer.size()];  ← modulo calculation

        // Overwrite the buffer at the write pointer, then increment and wrap pointer
        gDelayBuffer[gWritePointer] = in;
        gWritePointer++;
        if(gWritePointer >= gDelayBuffer.size())
            gWritePointer = 0;
        // [...]
    }
}
```

# Write and read pointers

- We can use **modulo arithmetic** to look backwards in the circular buffer
  - We could do this every sample to read samples out at a particular delay
- Alternatively, we can use a second pointer (**read pointer**) to keep track of where we are reading
  - Each sample, both the write and read pointers are **incremented and wrapped**
  - The **distance between the pointers** determines the delay



# Write and read pointers

- The circular buffer now has **three** components:
  1. A **region in memory** (array) to store the samples
  2. A **write pointer** to keep track of where we are writing new samples
  3. A **read pointer** to keep track of where we are reading old samples
- Remember, there is (still) **no functional beginning or end** to a circular buffer!
- In code, we need to declare three (global) variables:

```
std::vector<float> gDelayBuffer;
```

```
unsigned int gWritePointer = 0;
```

```
unsigned int gReadPointer = 
```

← What index we put here  
determines the delay

- ▶ For each new sample, **store it** at the write pointer, then **increment/wrap** both pointers

```
out = gDelayBuffer[gReadPointer];
```

```
gReadPointer++;
```

```
if(gReadPointer >= gDelayBuffer.size())
```

```
    gReadPointer = 0;
```

```
gDelayBuffer[gWritePointer] = in;
```

```
gWritePointer++;
```

```
if(gWritePointer >= gDelayBuffer.size())
```

```
    gWritePointer = 0;
```

# Circular buffer task 3

- **Task:** Change your code to implement the delay using a **read pointer**
  - Keep the delay at 0.1 seconds
  - The delay is set up by the difference between the read and write pointer locations
  - You should not need modulo indexing in `render()` anymore

```
std::vector<float> gDelayBuffer;  
unsigned int gWritePointer = 0;  
unsigned int gReadPointer =
```

← What index we put here  
determines the delay

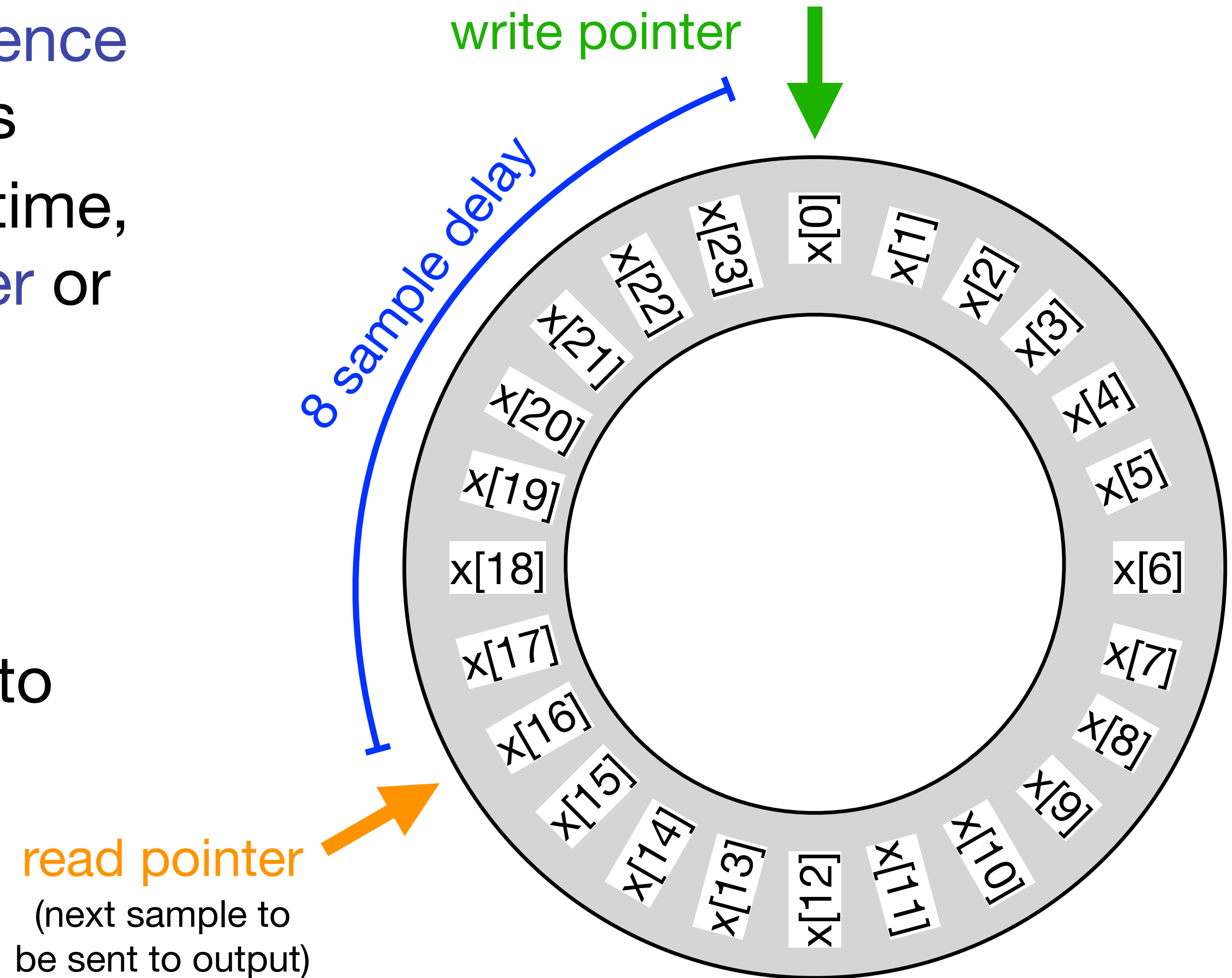
```
out = gDelayBuffer[gReadPointer];  
gReadPointer++;  
if(gReadPointer >= gDelayBuffer.size())  
    gReadPointer = 0;
```

```
gDelayBuffer[gWritePointer] = in;  
gWritePointer++;  
if(gWritePointer >= gDelayBuffer.size())  
    gWritePointer = 0;
```



# Adjusting the delay

- Delay time is given by the **difference** between read and write pointers
- If we want to change the delay time, should we move the **read pointer** or the **write pointer**?
  - **The read pointer.** Why?
  - Don't want a gap in the buffer
- When the delay length should change, use **modulo arithmetic** to recalculate the read pointer

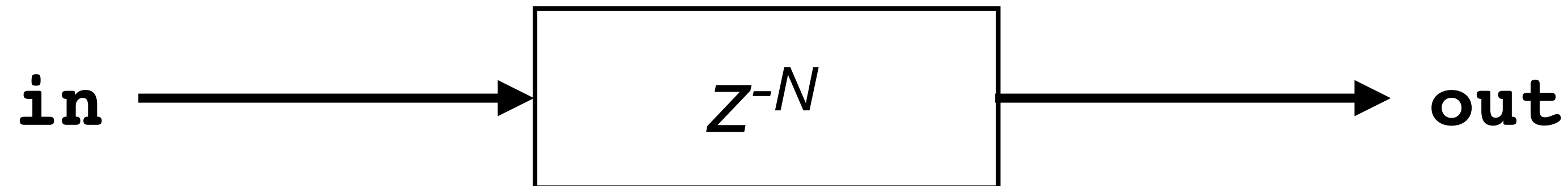


# Circular buffer task 4

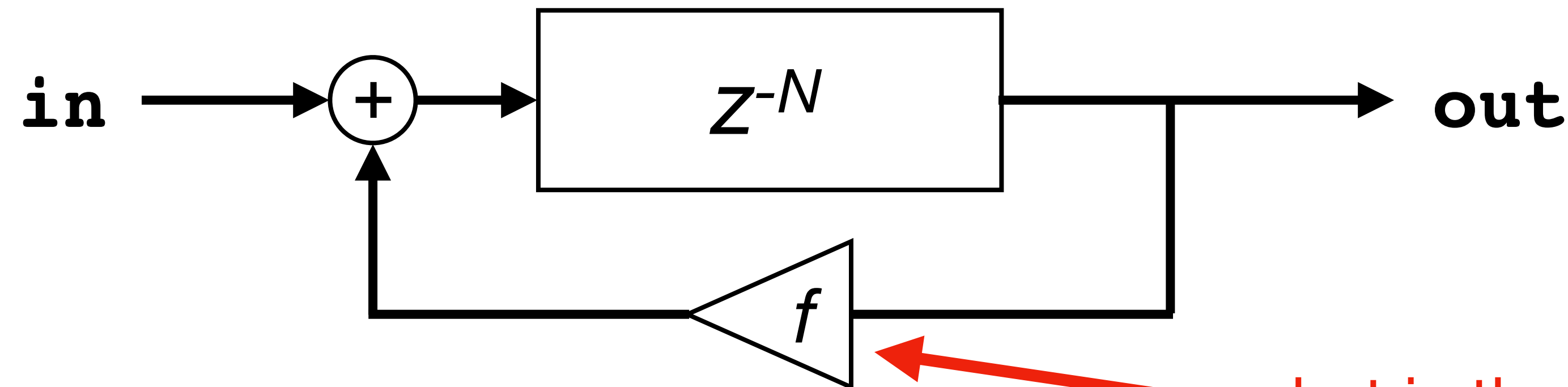
- **Task:** add a **GUI slider** to change the delay length
  - See Lecture 4 for more details on the Bela GUI
  - **Make the delay adjustable between 0 and 0.49 seconds**
  - From time in seconds, calculate how many samples of delay are needed
  - Update the location of the read pointer based on the write pointer location
- **Hint:** make sure your code works with a delay of 0!
  - It might matter whether you read or write to the buffer first

# Echo effect

- With our circular buffer, we have implemented a simple delay:



- We can also add **feedback** (or **regeneration**) from output to input
  - This produces periodic **echoes** of the sound



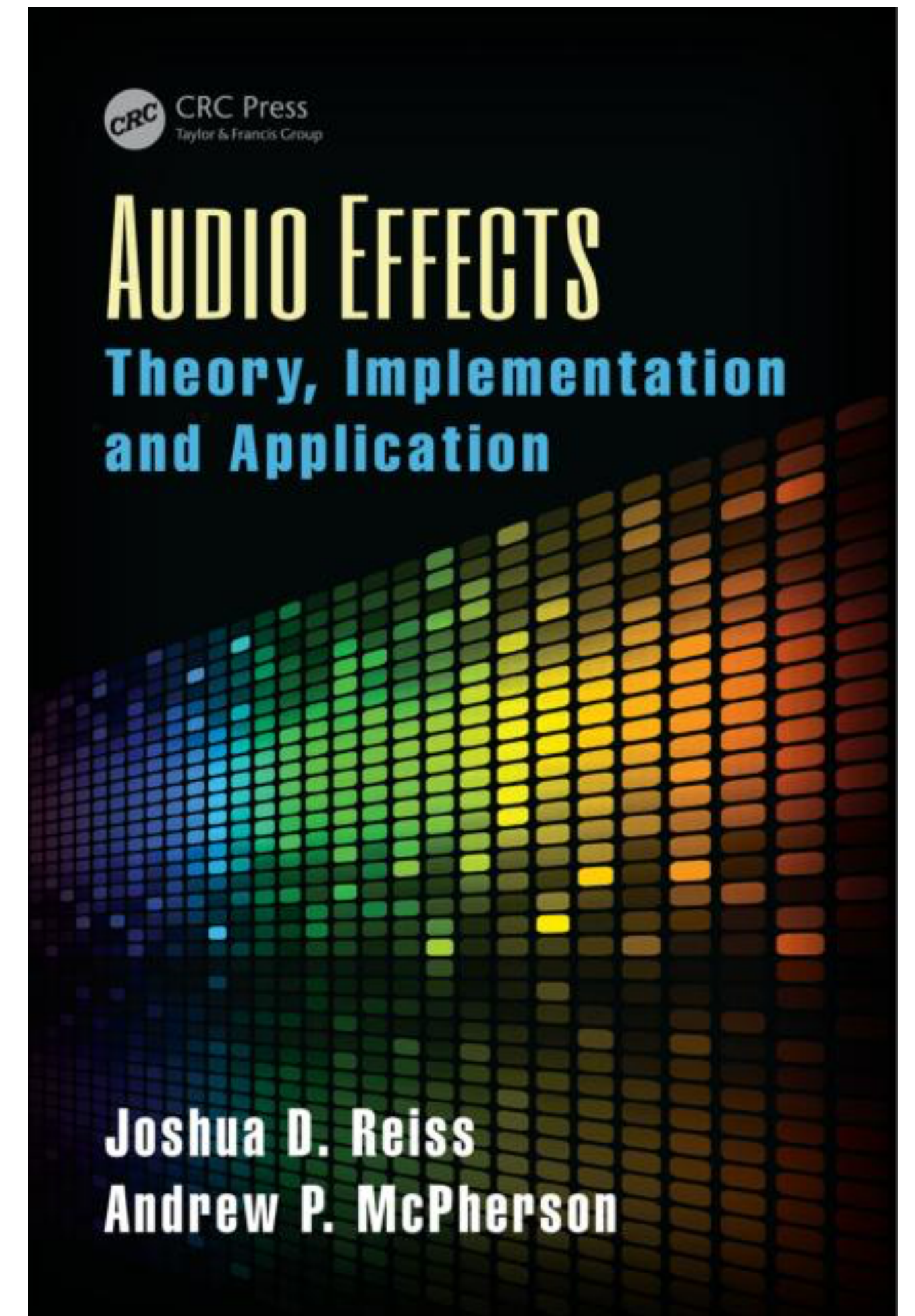
what is the valid range of values for  $f$ ?

- **Task:** add feedback to the **circular-buffer** project to create an **echo**
  - Add a second GUI slider to control the level of feedback



# Other delay-based effects

- Vibrato
  - Created with a variable-speed read pointer
- Chorus
  - A time-varying delayed copy added to original signal
- Flanger
  - Implemented like a chorus, but with lower delay and possible presence of feedback
- All of these require **fractional read pointers** (see Lecture 3)
- They also require **LFOs (low-frequency oscillators)**
- The Audio Effects textbook has more theory and code examples for these effects





# Keep in touch!

Social media:  
**@BelaPlatform**

**forum.bela.io**

**blog.bela.io**

More resources and contact info at:

**learn.bela.io/resources**