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 17: Block-based processing

What you'll learn today:

Generating and processing windows of a signal
The Fast Fourier Transform (FFT)
Sending data to the Bela GUI

What you'll make today:

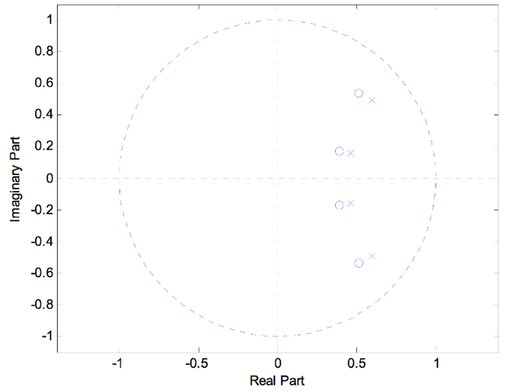
Two simple spectrum analysers with LEDs and GUI

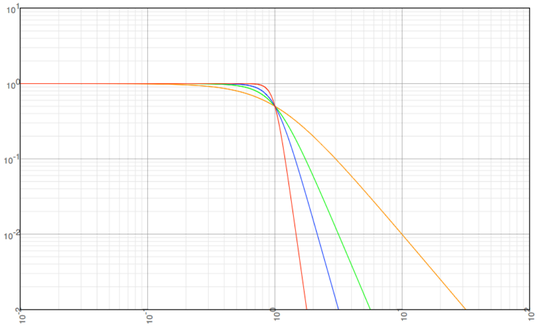
Companion materials:

github.com/BelaPlatform/bela-online-course

Theory and implementation

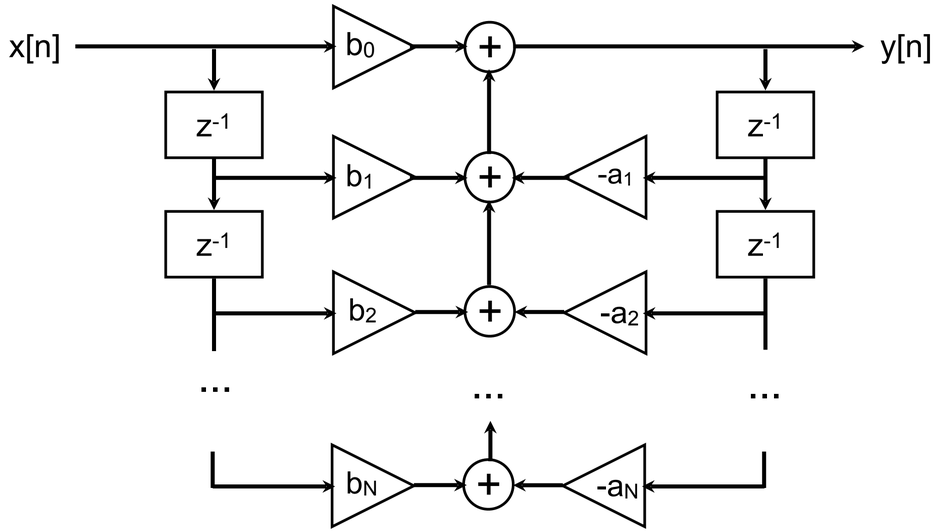
Specifications





Design

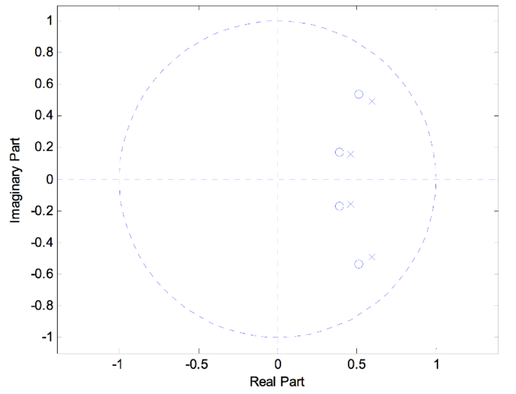
$$y[n] = \sum_{k=1}^{N} a_k y[n-k] + \sum_{k=0}^{M} b_k x[n-k]$$

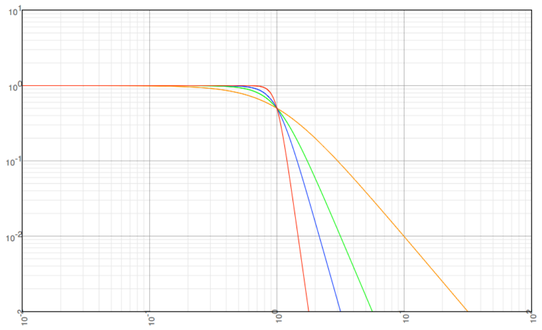


Typical digital signal processing course

Theory and implementation

Specifications





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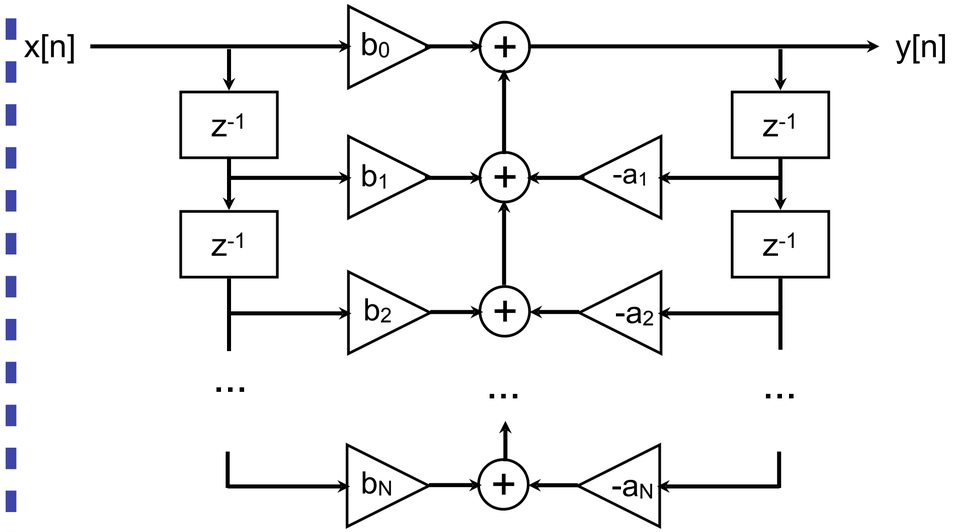
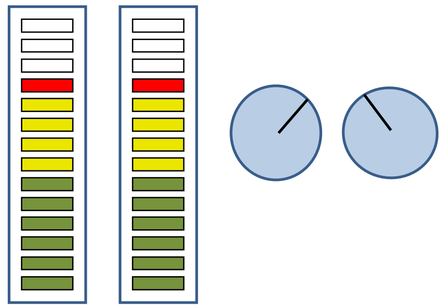


Implementation

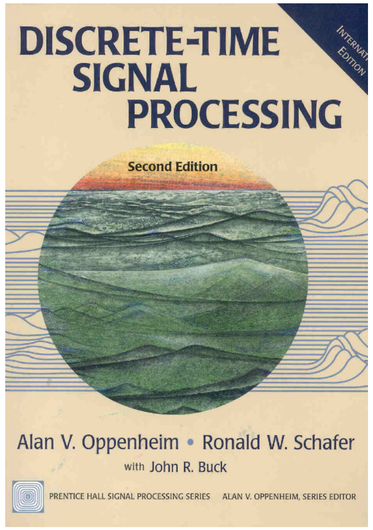
```
void render(BelaContext *context, void *
{
  /* Iterate over each channel */
  for(int channel = 0; channel < numChan
    /* Then iterate over each sample wit
    for(int n = 0; n < numSamples; n++)
      /* Calculate the sample... */
      float sample = gAmplitude *
                     sin(2.0 * M_PI * (f
                                 * gFrequ
```

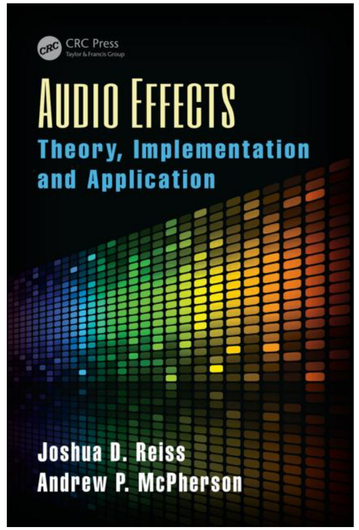


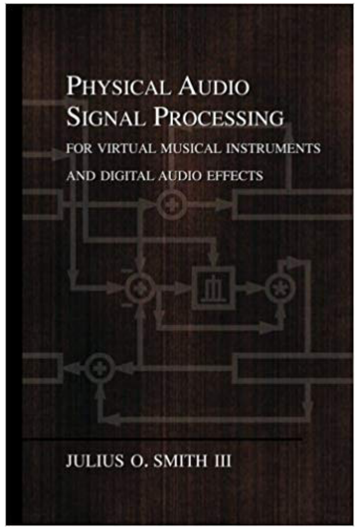
- We will turn signal processing theory into efficient C++ code
- There are plenty of excellent introductions to DSP mathematics

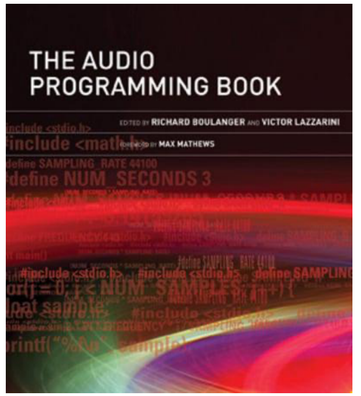
Books and references

- There are lots of good books and online resources about DSP and audio programming. For example:
  - Oppenheim, Schafer and Buck, *Discrete-Time Signal Processing*
  - Julius O. Smith, *Physical Audio Signal Processing*
  - Richard Boulanger and Victor Lazzarini (editors), *The Audio Programming Book*
  - Reiss and McPherson, *Audio Effects: Theory, Implementation and Application*
- See the GitHub companion site for links: **github.com/BelaPlatform/bela-online-course**

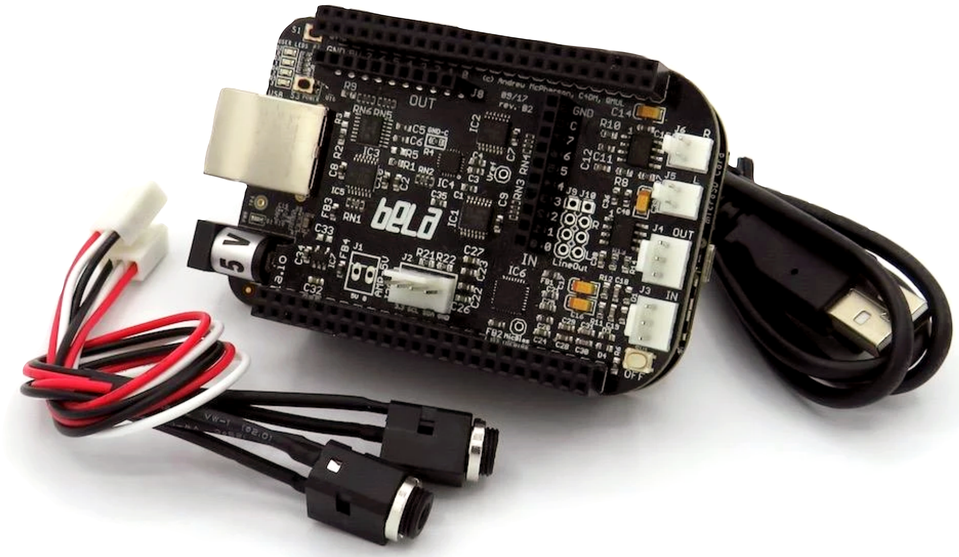






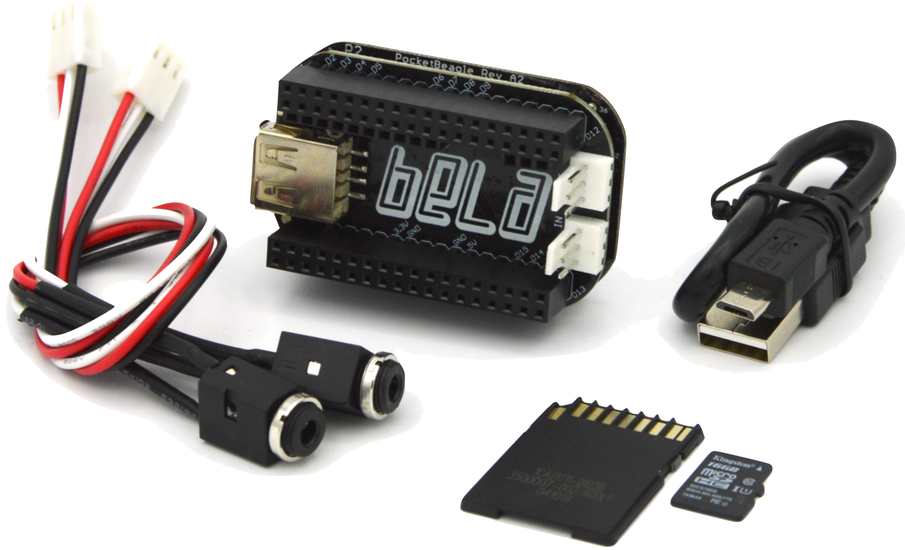


What you'll need



Bela Starter Kit

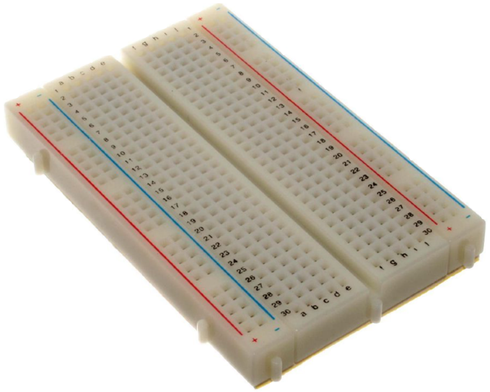
or

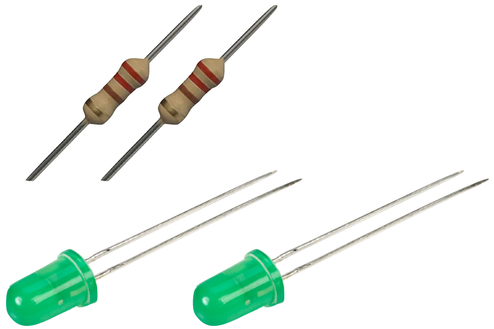


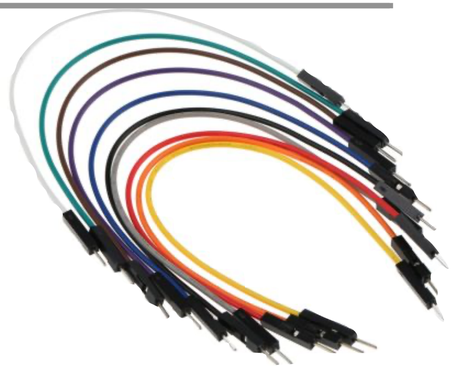
Bela Mini Starter Kit

[shop.bela.io]

Also needed for this lecture:

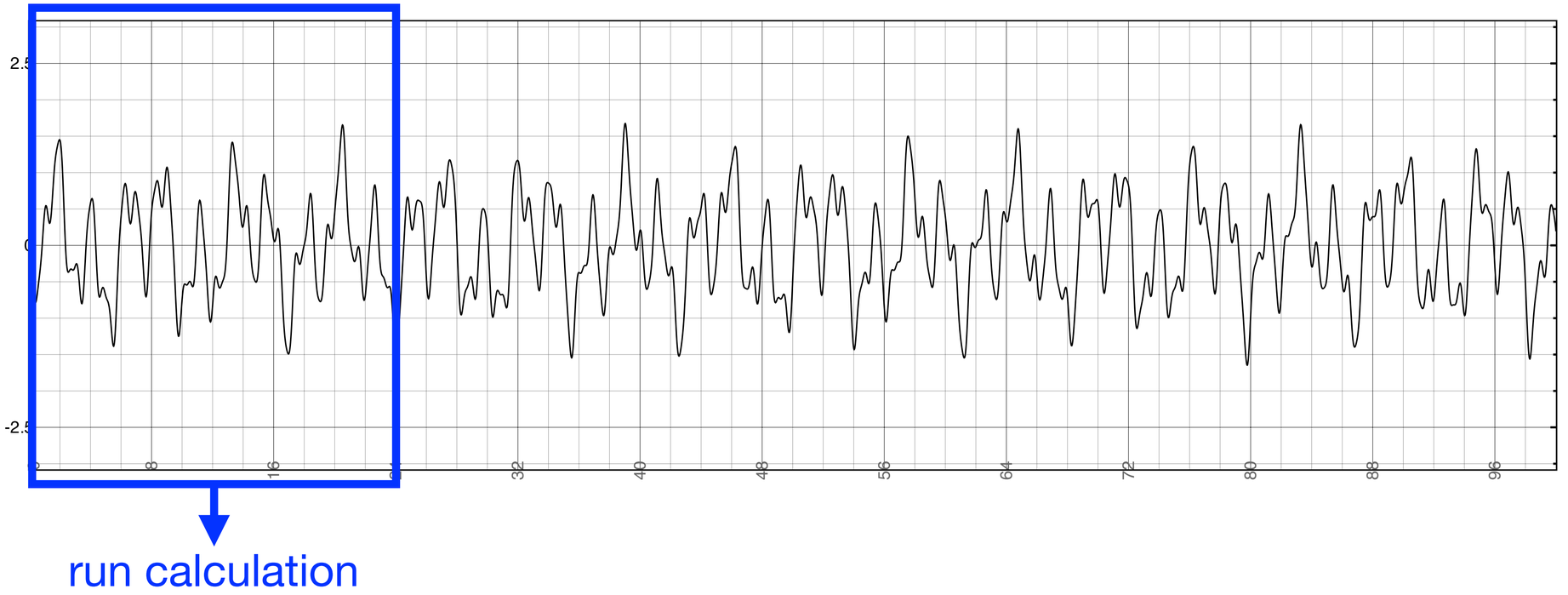






Block-based processing

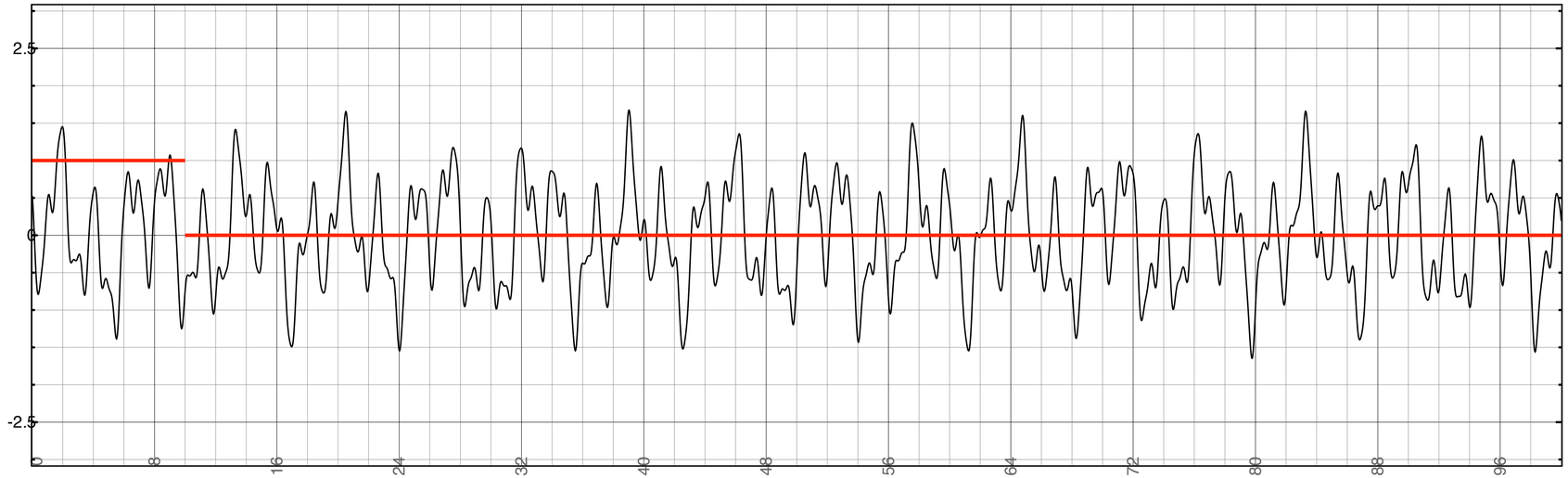
- Sometimes audio calculations need to run on a block of samples simultaneously
  - For example: wait until 512 samples arrive, then do something with all of them at once



- Compare this to filters, where we calculate a new output every sample
  - With block-based processing, we typically run the calculation periodically, not every sample

Block-based processing

- Block-based processing is not the same as non-realtime processing
  - We are not trying to run a calculation on the whole signal all at once
- Instead, we want to work with finite segments of the input signal
- How do we generate a block? Apply a window to the signal
  - Multiply by a function that's nonzero for a fixed length M samples
  - Window might have a shape: rectangular, triangular (Bartlett), Hamming, Blackman, etc.



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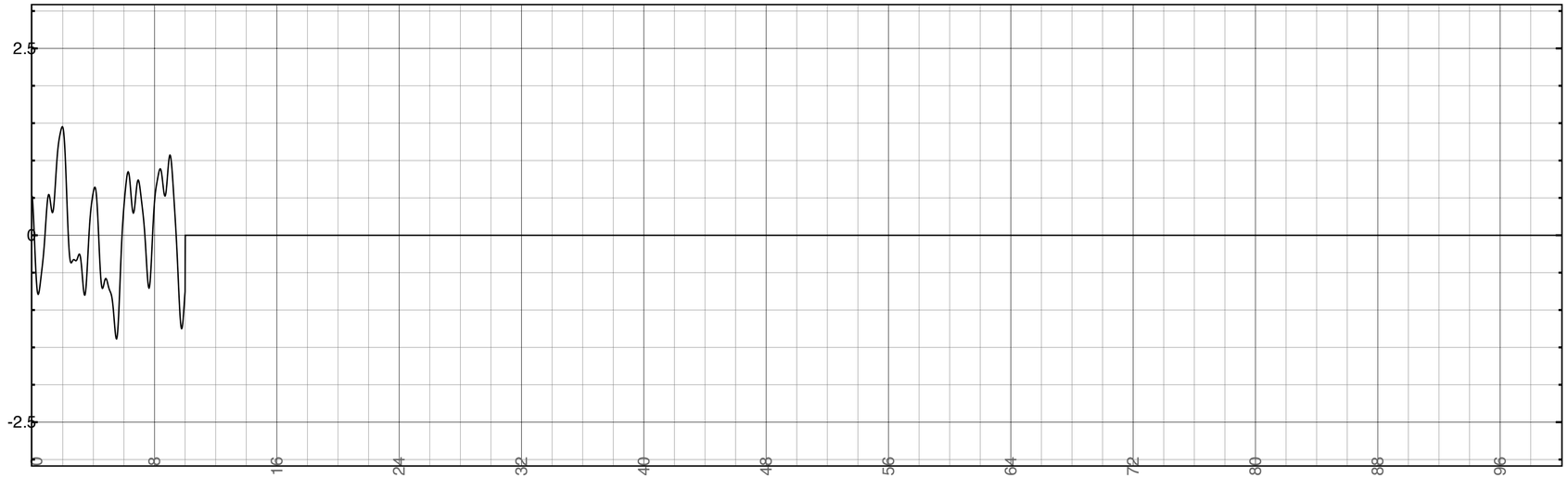

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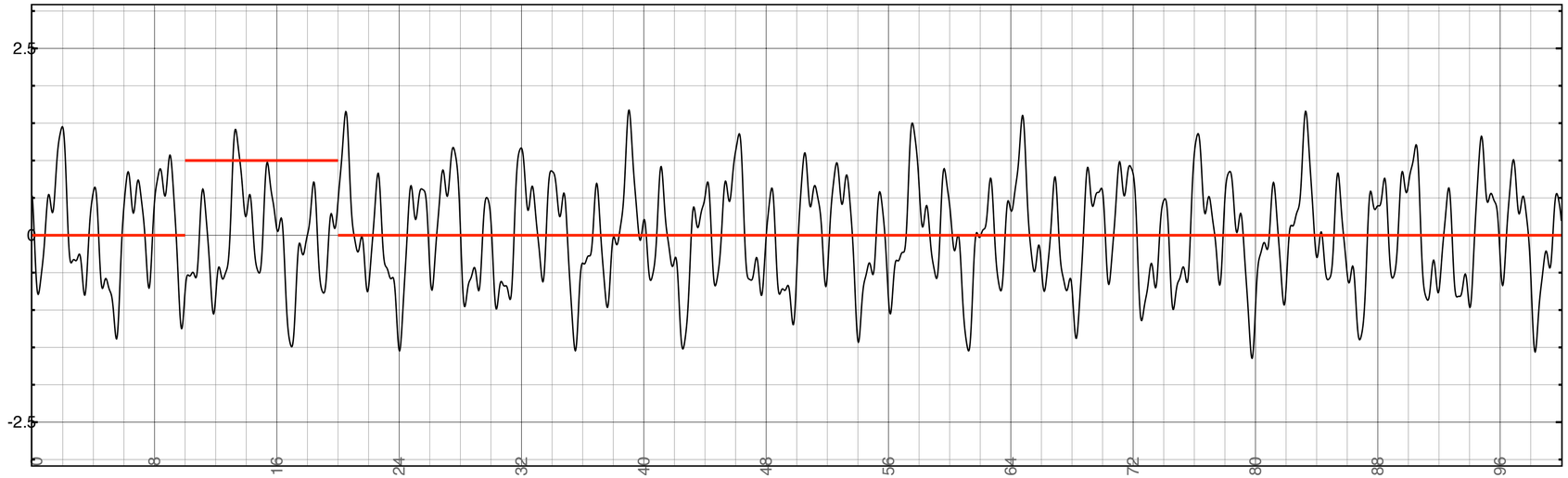

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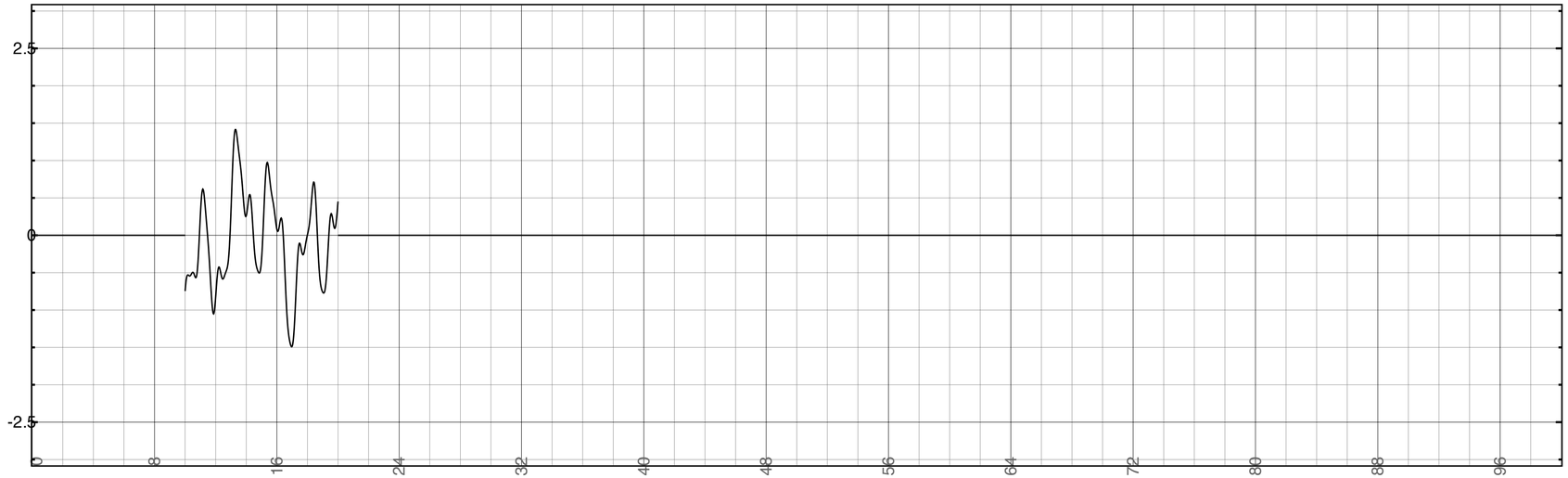

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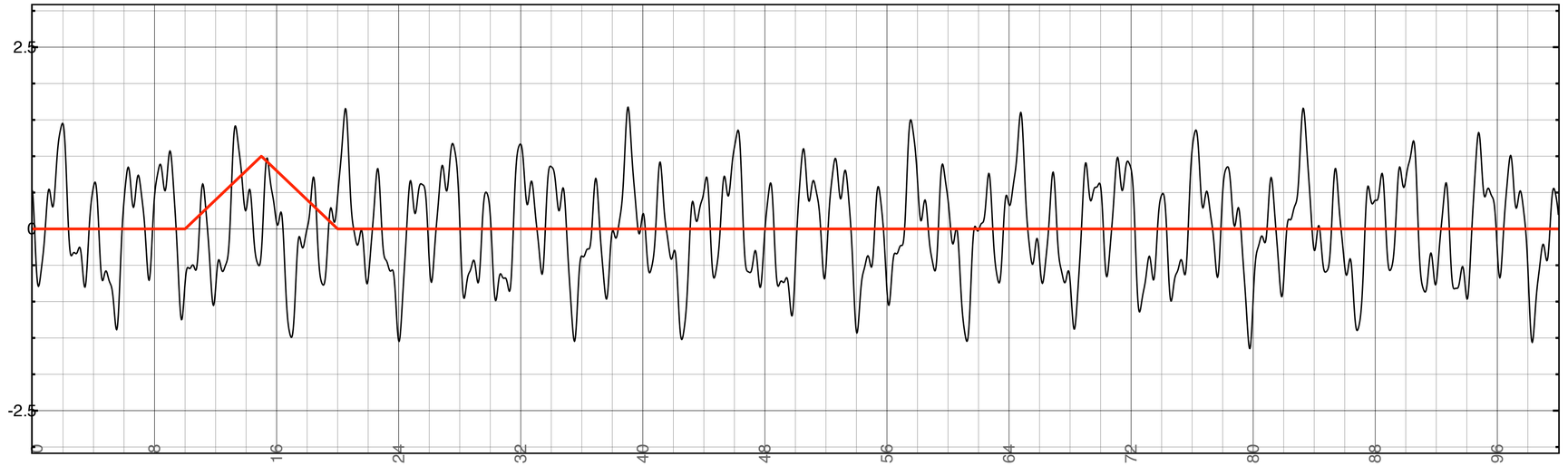

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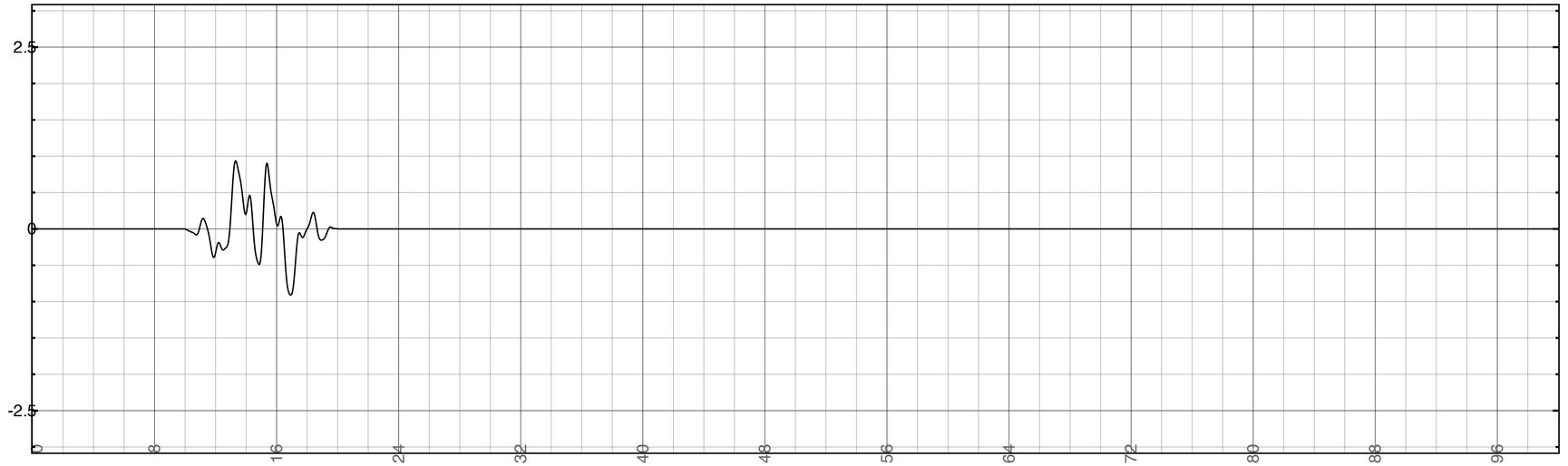
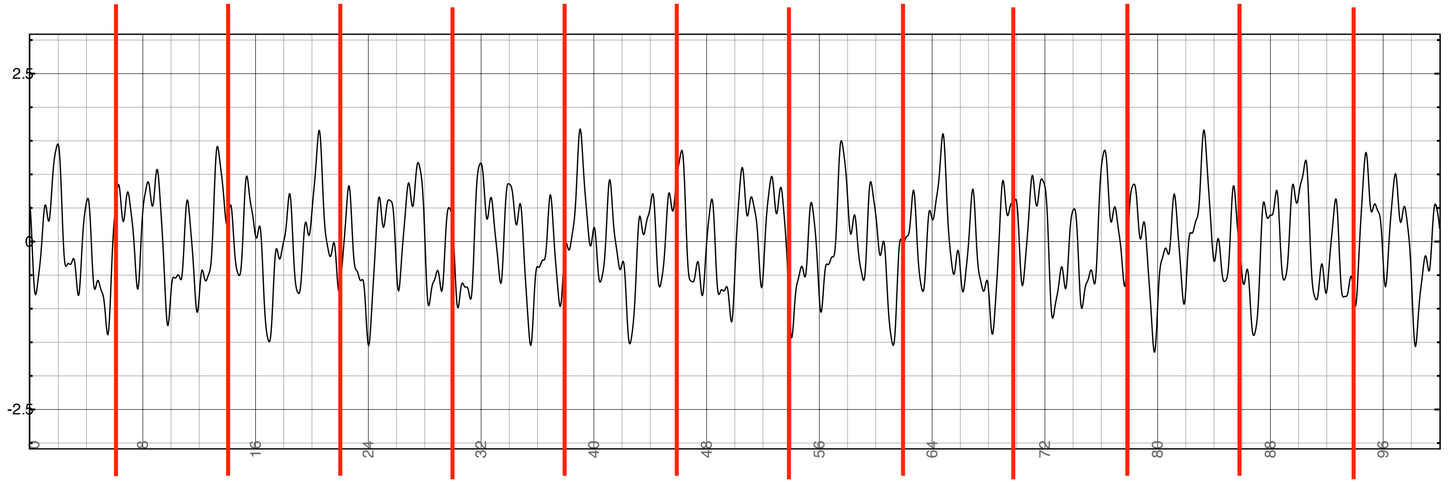


Windowing a signal

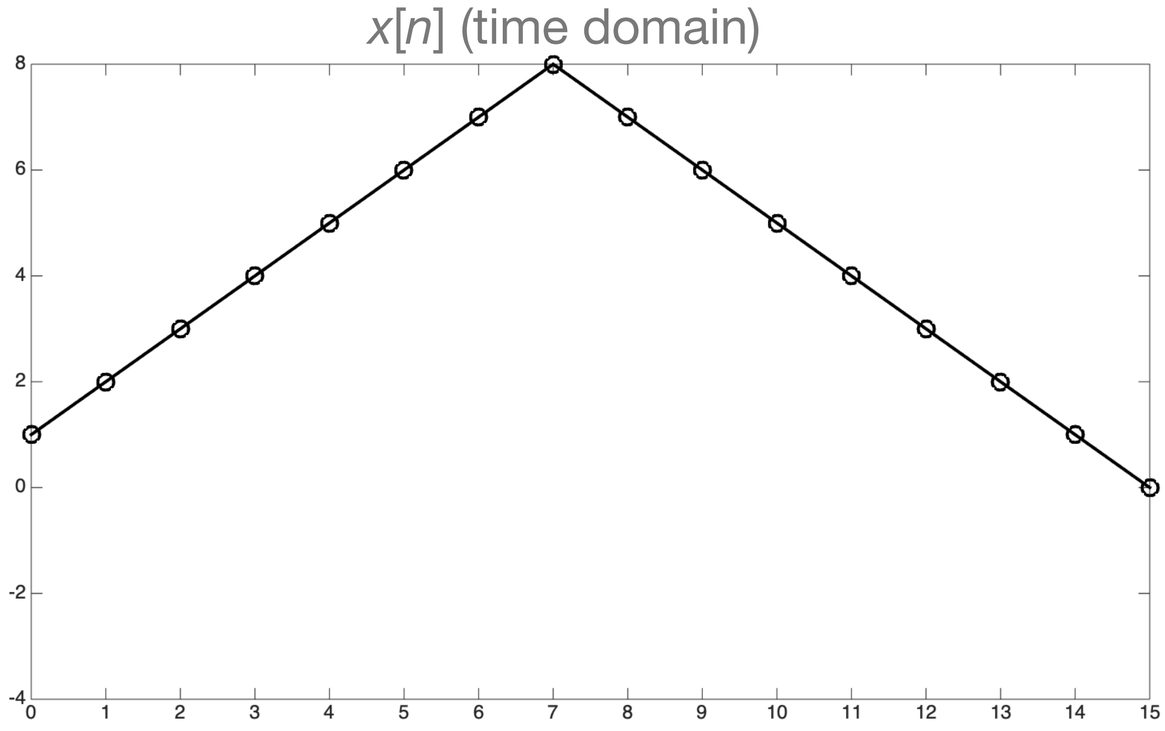
- Any arbitrarily-long signal can be divided into successive windows
  - This segmentation can be done in real time, as the signal comes in



- In the right circumstances, we can use the windows to reconstruct the signal
  - Need window functions to overlap so that they all add to a constant value
  - Constant Overlap-Add (COLA) criterion: we'll come back to this later
- We call this segmentation and reconstruction block-based processing
  - We work on an entire window (or block) of samples at once

The Fast Fourier Transform

- Most common use of block-based processing: the Fast Fourier Transform (FFT)
  - Efficient computational algorithm for calculating the Discrete Fourier Transform (DFT)
  - The DFT is a mathematical formula for calculating the frequency content of a signal
- Any discrete-time signal of N points can be expressed as the sum of N sinusoids:



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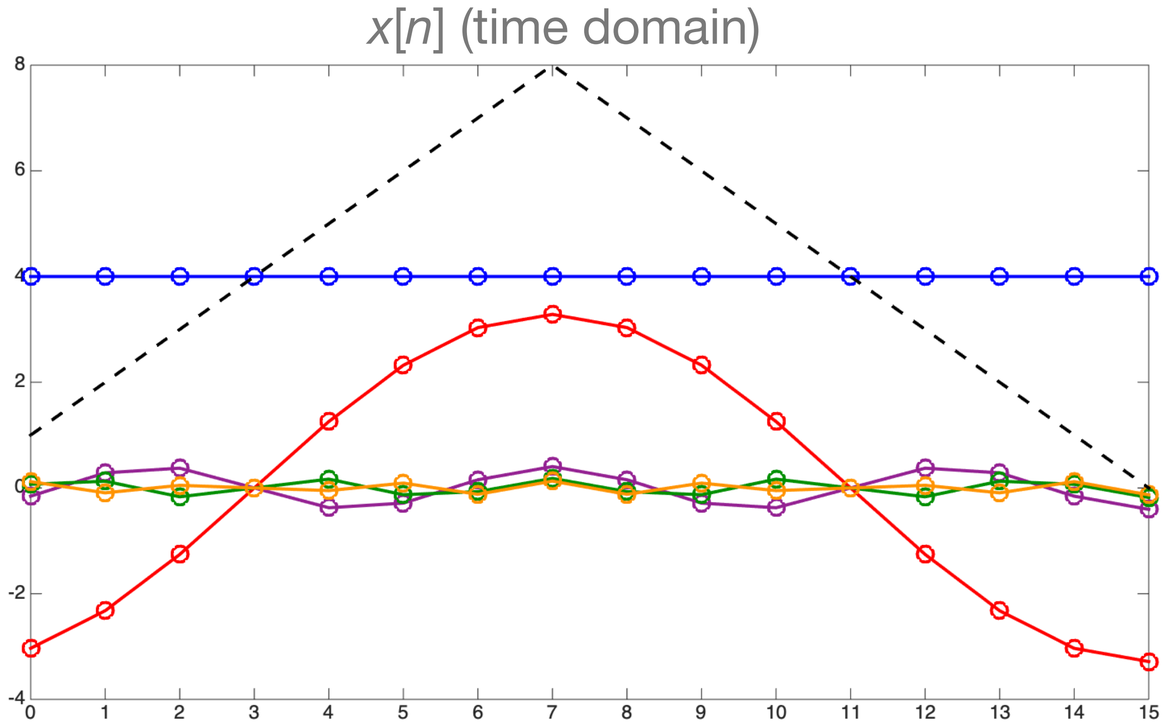

Frequency	Amplitude	Phase
0	4.00	0
π/8	1.64	-.875*π
3π/8	0.20	-.625*π
5π/8	0.09	-.375*π
7π/8	0.65	-.125*π

(other components are 0 or redundant)

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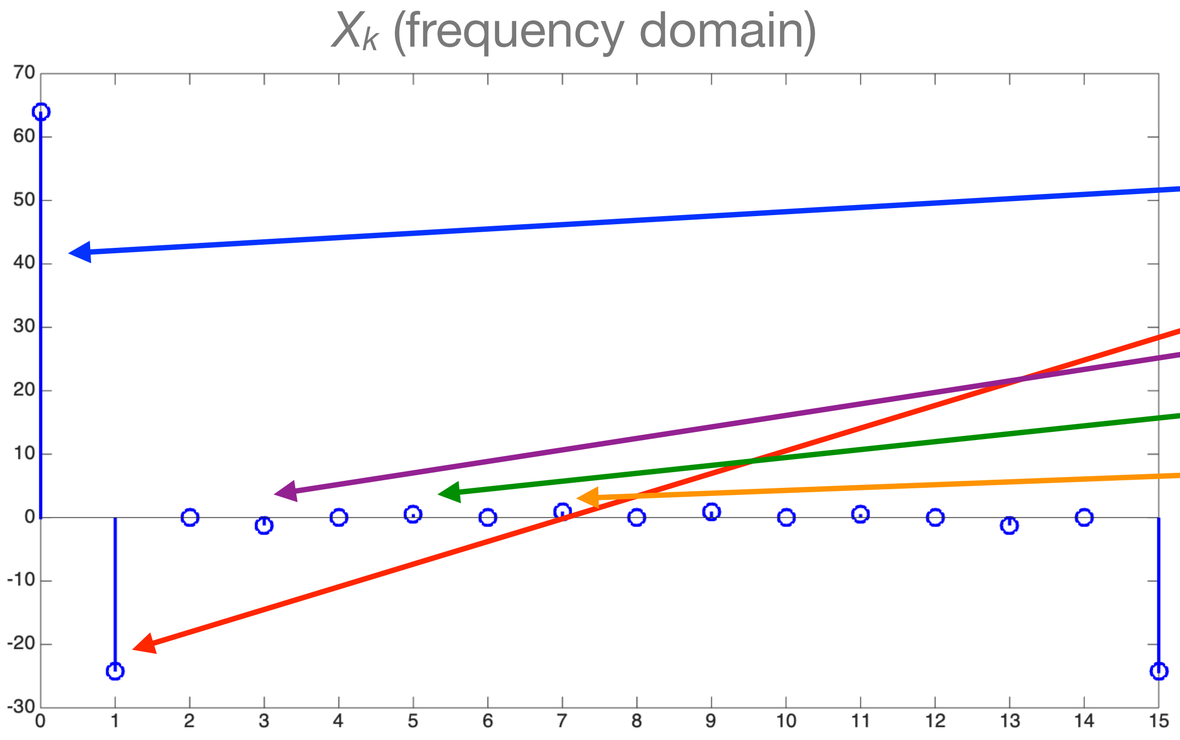

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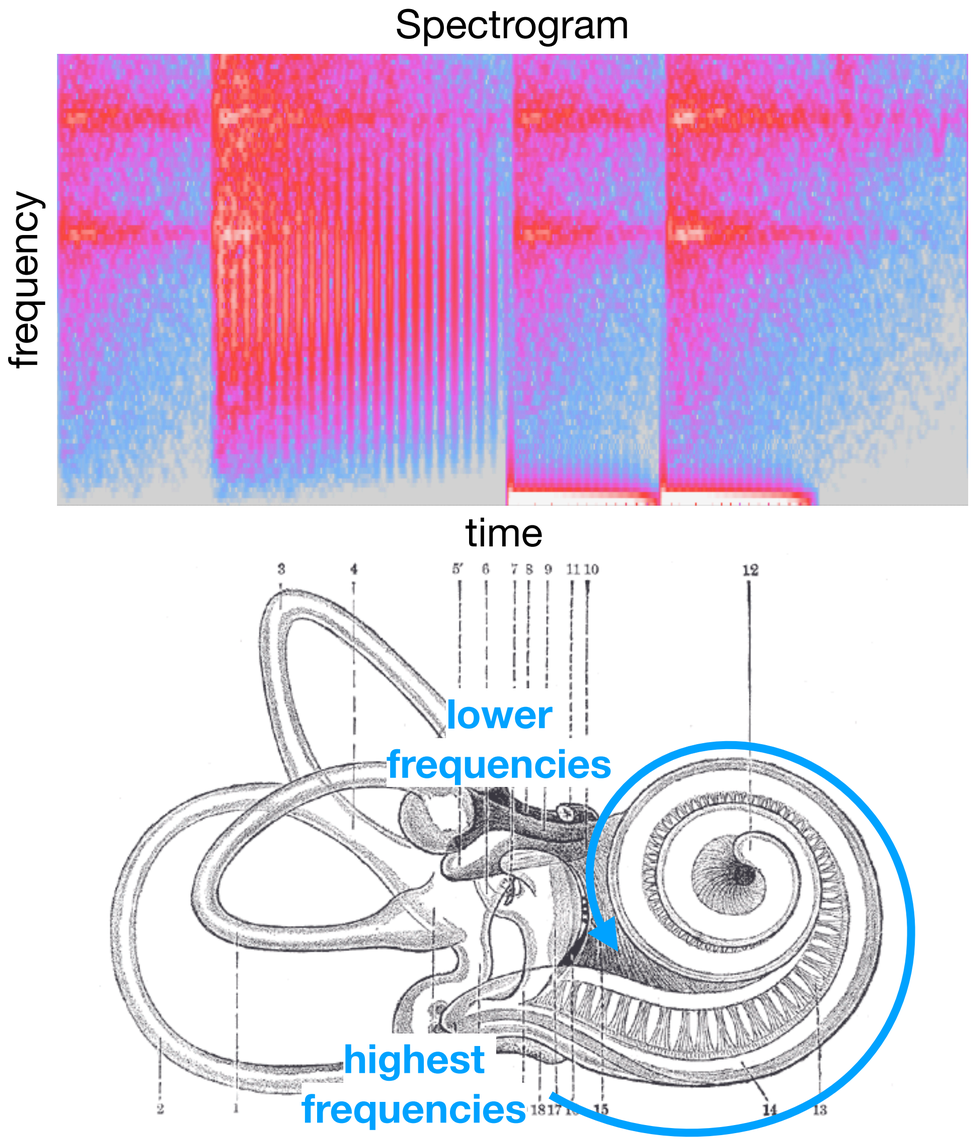
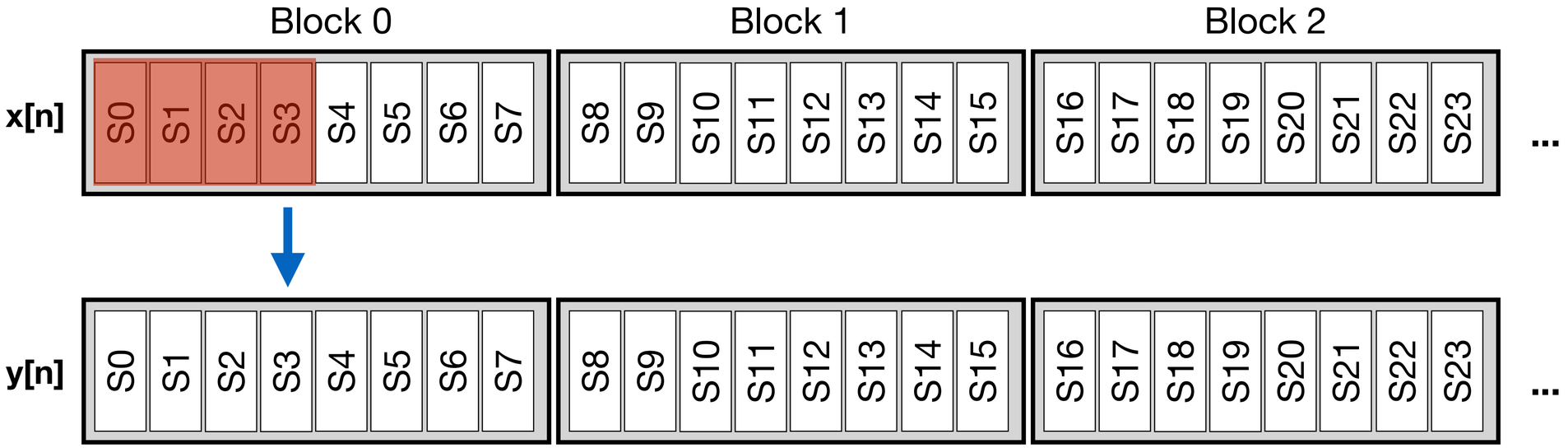
The DFT calculates the amplitudes and phases of each frequency component

Why do we care about the FFT?

- The FFT has transformed digital signal processing, including for audio
- Application 1: audio analysis
  - Calculate the spectrum of a signal (magnitude vs. frequency)
  - Or the spectrogram (how the spectrum changes over time)
  - Can extract higher-level audio features from the FFT
    - This is the basis of the field of music informatics
- Application 2: sonic transformation and resynthesis
  - Many useful audio effects are done in the frequency domain
    - Pitch shifting, denoising, cross-synthesis, ...
  - These are known as phase vocoder effects
- Interesting aside: the human cochlea (inner ear) acts like a kind of biological Fourier transform
  - Each location is sensitive to a different frequency



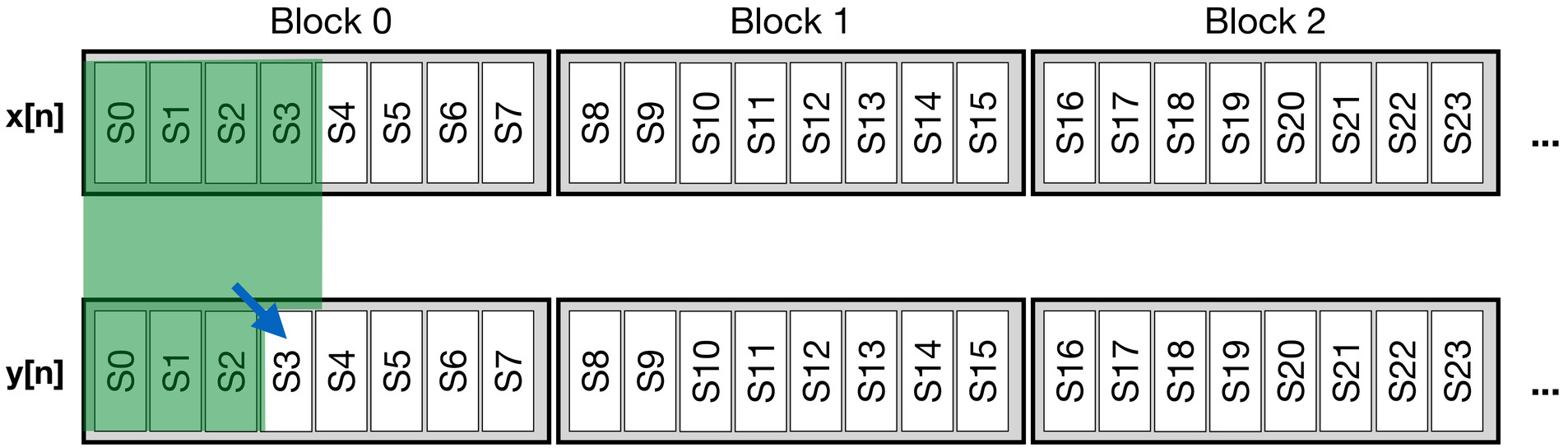
Filters vs. block-based processing



FIR Filter: new calculation each sample

- Important: Window size is not necessarily the system audio buffer size!
  - Might need multiple calls to `render()` to fill a single window
  - Or window might fill up in the middle of `render()`

Filters vs. block-based processing



FIR Filter: new calculation each sample

IIR Filter: new calculation each sample

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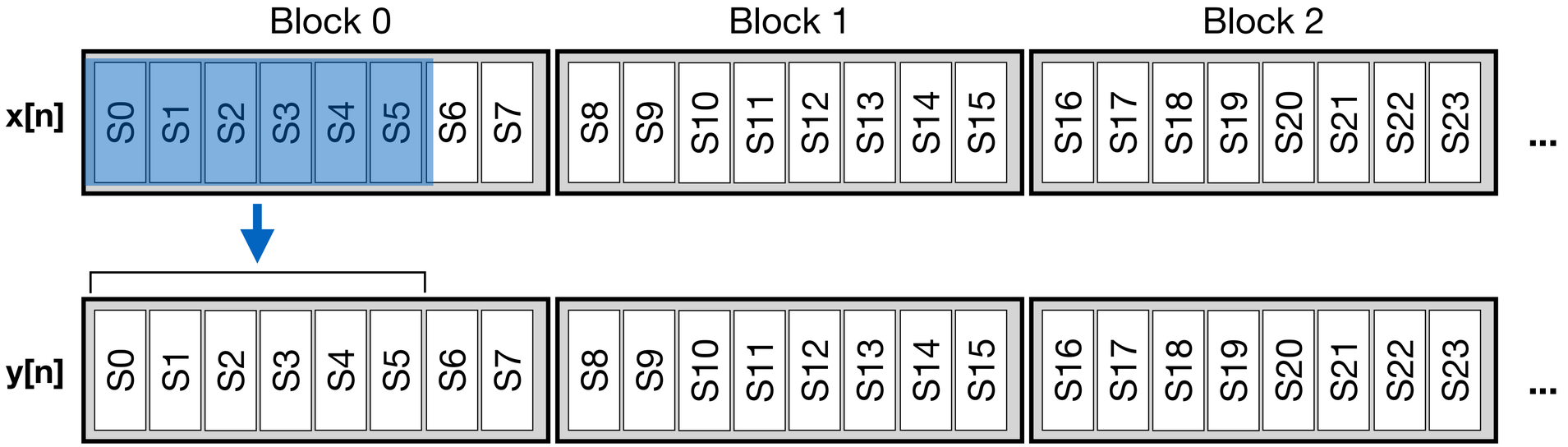
Filters vs. block-based processing



FIR Filter: new calculation each sample

IIR Filter: new calculation each sample

DFT: calculation once per block*

**note: size of DFT may be different than interval between calculations*

- Important: Window size is not necessarily the system audio buffer size!
  - Might need multiple calls to `render()` to fill a single window
  - Or window might fill up in the middle of `render()`

Real-time windowing for the FFT

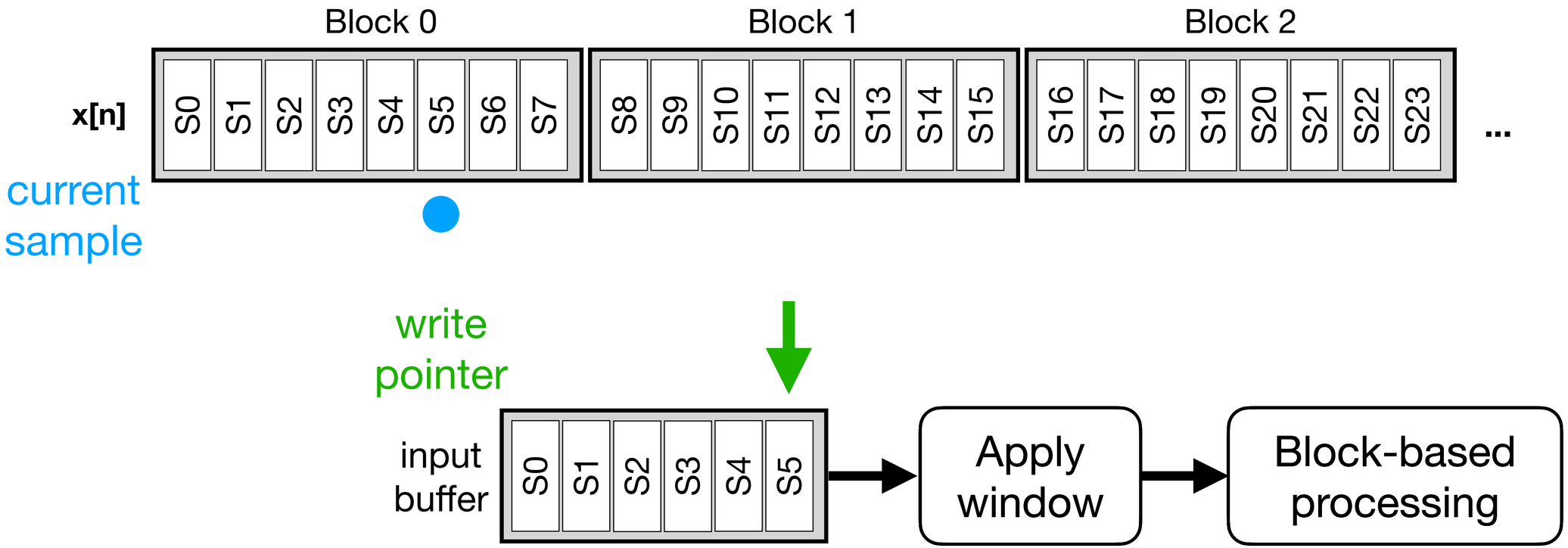
- We need a full window (i.e. block) of samples at once to pass to the FFT
- Suppose that we have a window size of N samples
- To handle FFT processing in real time:
  1. In our initialisation code, create a buffer (array) of type `float`, size N
     - If we have multiple audio channels, we'll need one buffer per channel
  2. In `render()`, as we process samples, store them in buffer and keep count
     - We need a global variable to keep track of how many samples are in the buffer
  3. When the buffer is full:
     - Multiply the contents of the buffer by the window function (if window is not rectangular)
     - Perform an FFT on the result
     - Handle any necessary processing in the frequency domain
  4. After the processing is complete, reset the counter to 0 to start filling the next window

Block-based input

- Again: window size is not necessarily the system buffer size
  - Therefore, we cannot just pass the raw audio buffer inside `BelaContext` to the FFT!
- Instead, we need:
  - Separate buffer to hold incoming samples
  - Counter or write pointer to keep track of how full the buffer is
  - Even though we're processing in discrete blocks, all the code still needs to go inside the `for()` loop in `render()`

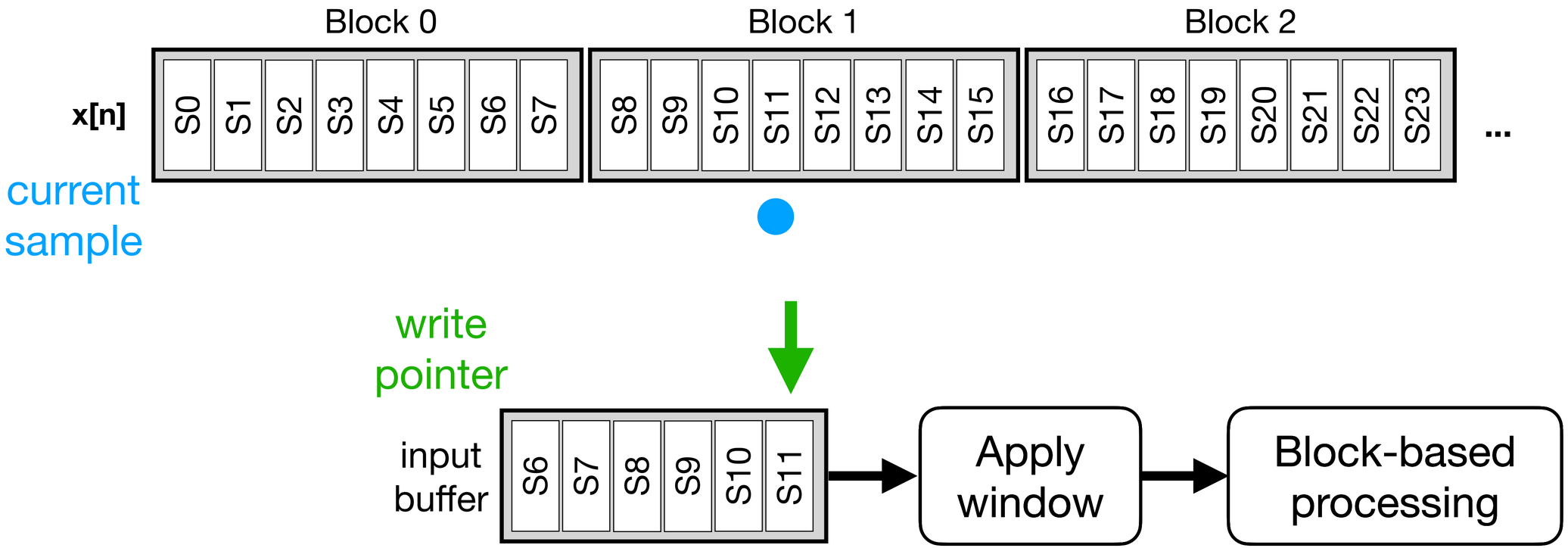
```
float gWindowBuffer[BUFFER_SIZE]; // Buffer for 1 window of samples
int gWindowBufferPointer = 0;     // Tells us how many samples in buffer

void render(BelaContext *context, void *userData) {
    // Important: nothing goes here; everything in the loop!

    for(int n = 0; n < context->audioFrames; n++) {
        // 1. Copy audio input into gWindowBuffer
        // 2. Increment gWindowBufferPointer
        // 3. Check if pointer has reached end of gWindowBuffer
        // 4. If yes, time to do the FFT and any other processing then reset pointer to 0
    }
}
```

Block-based input



- Start with a buffer of fixed size
  - Fill it up sample by sample
  - When it's full, apply the window function and pass it to the FFT (or other block based process)

Block-based input



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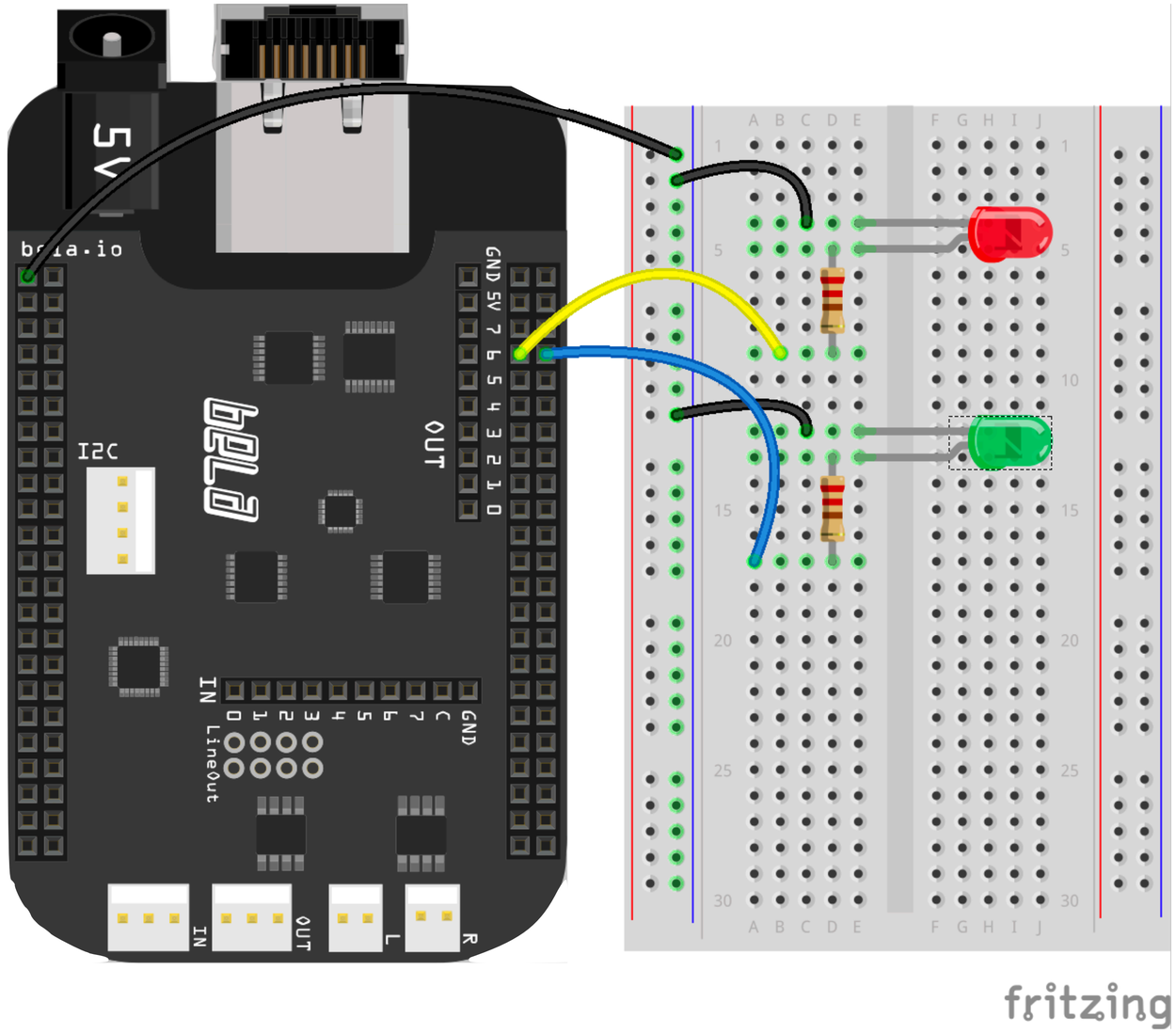
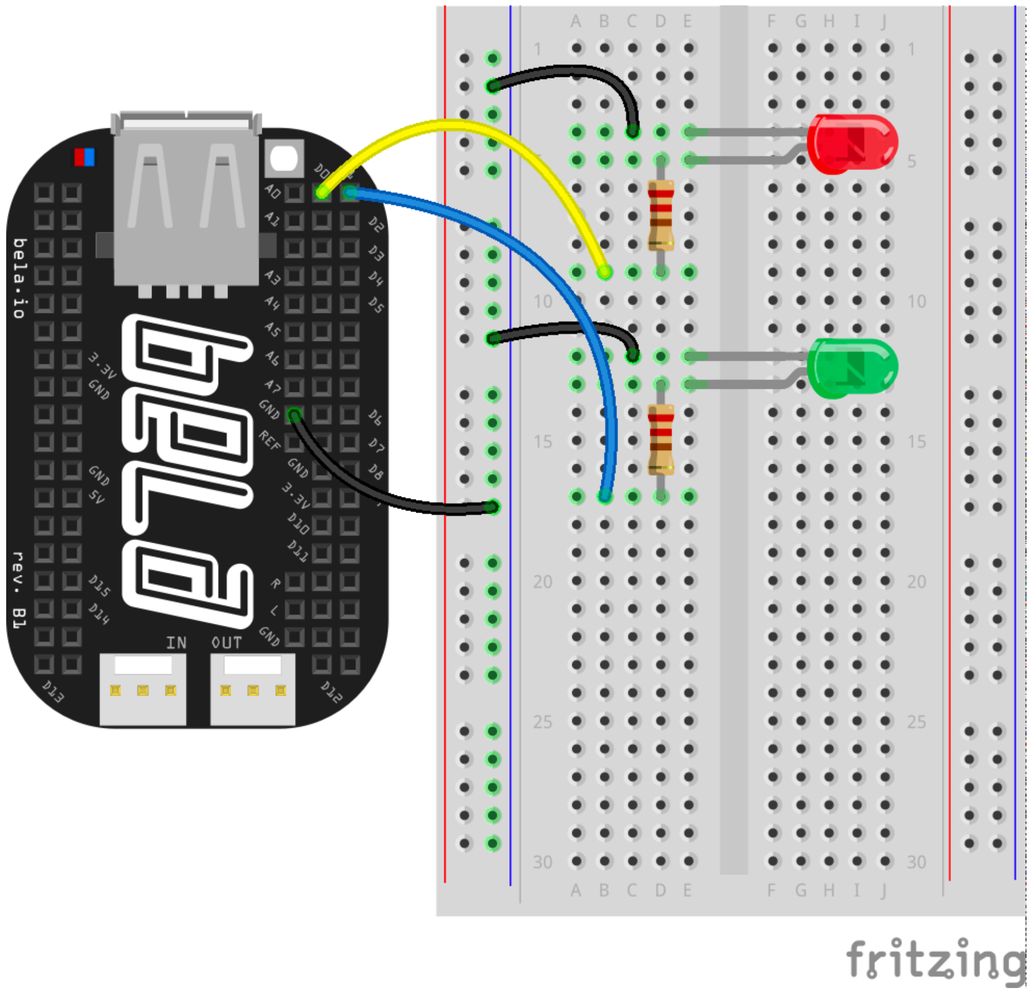
Block-based processing task

- Task: using fft-led project
  - Attach LEDs to digital outputs 0 and 1
  - Implement code to segment the incoming signal into windows
    - Rectangular windows are fine for now (no window function needed)



Block-based processing task

- Task: using fft-led project
  - Attach LEDs to digital outputs 0 and 1
  - Implement code to segment the incoming signal into windows
    - Rectangular windows are fine for now (no window function needed)



Block-based processing task

- Task: using fft-led project
  - Attach LEDs to digital outputs 0 and 1
  - Implement code to segment the incoming signal into windows
    - Rectangular windows are fine for now (no window function needed)
- Steps:
  1. Declare a buffer of size `gFftSize` as a global variable
     - Could use either an array or a `vector` of type `float`
  2. Keep a counter (also global) to record how many samples have been put in the buffer
     - Can also use this as a write pointer into the buffer
  3. Each iteration of the `for()` loop in `render()`, store the current sample in the buffer
     - When the counter reaches `gFftSize`, call the function `process_fft()`
  4. When finished with `process_fft()`, reset the counter and start again

The Bela GUI library

- Bela projects can have a GUI that renders in the browser
  - The GUI is typically written in **p5.js** (but other frameworks can be used)
  - Data can be sent from the Bela program to the browser via a WebSocket connection
  - Can also send data the other way, to control a Bela program from the GUI

The simple GUI from the `fft-led` project

- How to communicate with a p5.js GUI from your Bela C++ code:
  1. Include the GUI library: `#include <libraries/Gui/Gui.h>`
  2. Make an object of type `Gui`: `Gui gLedGui;`
  3. In `setup()`, initialise the `Gui` object: `gLedGui.setup(context->projectName);`
  4. Use the `Gui::sendBuffer()` method to send data to the GUI

The Bela GUI library: communication

- Sending data from C++ on Bela to JavaScript in the browser:

render.cpp

```
// Update the Bela GUI at a 100Hz frame rate
if(++gGuiUpdateCounter >= context->audioSampleRate/100) {
    gGuiUpdateCounter = 0;

    // Send low and high values in two slots of the buffer
    gGuiBuffer[0] = gLedLoOutput;
    gGuiBuffer[1] = gLedHiOutput;
    if(gLedGui.isConnected()) {
        gLedGui.sendBuffer(0, gGuiBuffer);
    }
}
```

limit the rate that we send to the GUI by counting samples

store all the data in a single buffer (here, array of `float`)

check if GUI running

first argument is buffer number (allows multiple pieces of data)

second argument can be any type, array or vector

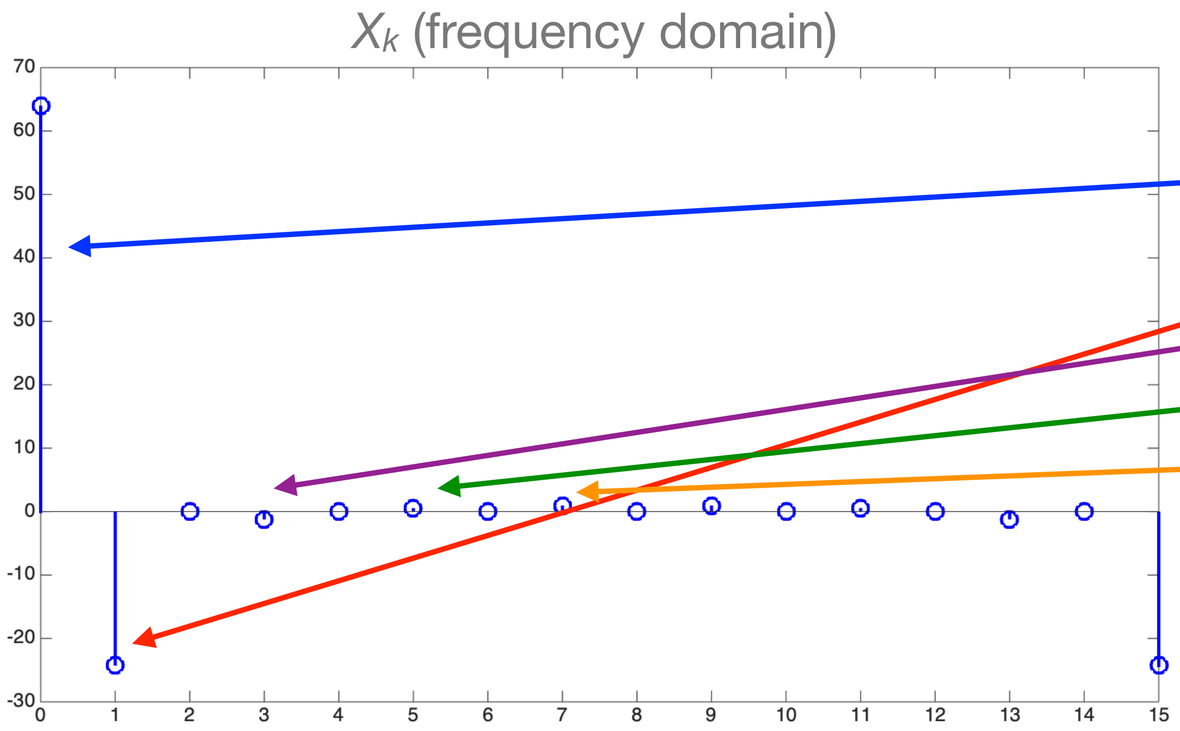
sketch.js

```
// Retrieve the data being sent from render.cpp
let ledLow = Bela.data.buffers[0][0];
let ledHigh = Bela.data.buffers[0][1];
```

this holds the contents of `gGuiBuffer` on the C++ side (because we sent it as buffer number 0)

The Fast Fourier Transform

- Most common use of block-based processing: the Fast Fourier Transform (FFT)
  - Efficient computational algorithm for calculating the Discrete Fourier Transform (DFT)
  - The DFT is a mathematical formula for calculating the frequency content of a signal
- Any discrete-time signal of N points can be expressed as the sum of N sinusoids:



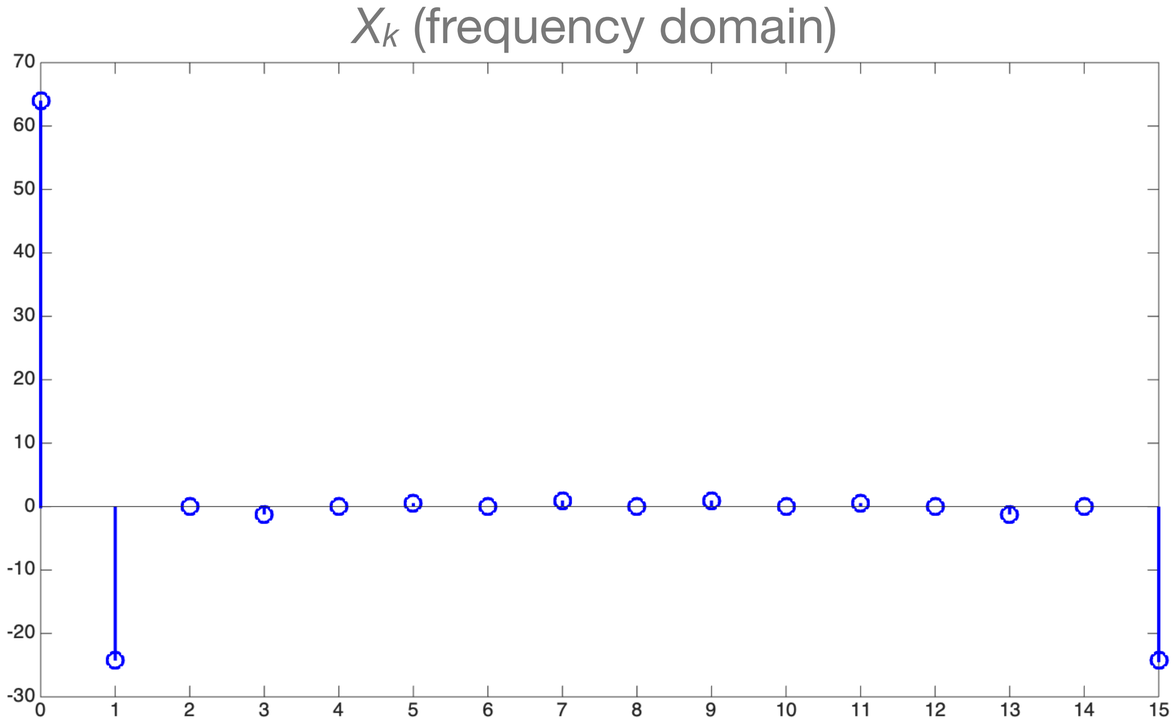
Frequency	Amplitude	Phase
0	4.00	0
π/8	1.64	-.875*π
3π/8	0.20	-.625*π
5π/8	0.09	-.375*π
7π/8	0.65	-.125*π

(other components are 0 or redundant)

The DFT calculates the amplitudes and phases of each frequency component

The Discrete Fourier Transform

- The DFT calculates the amplitudes and phases of each sinusoidal component
  - The frequencies are linearly spaced between 0 and 2π: $\frac{2\pi k}{N}$ for $0 \leq k < N$
  - Each amplitude/phase measurement is called a bin
  - Collectively, the N bins are called the frequency domain representation of the signal



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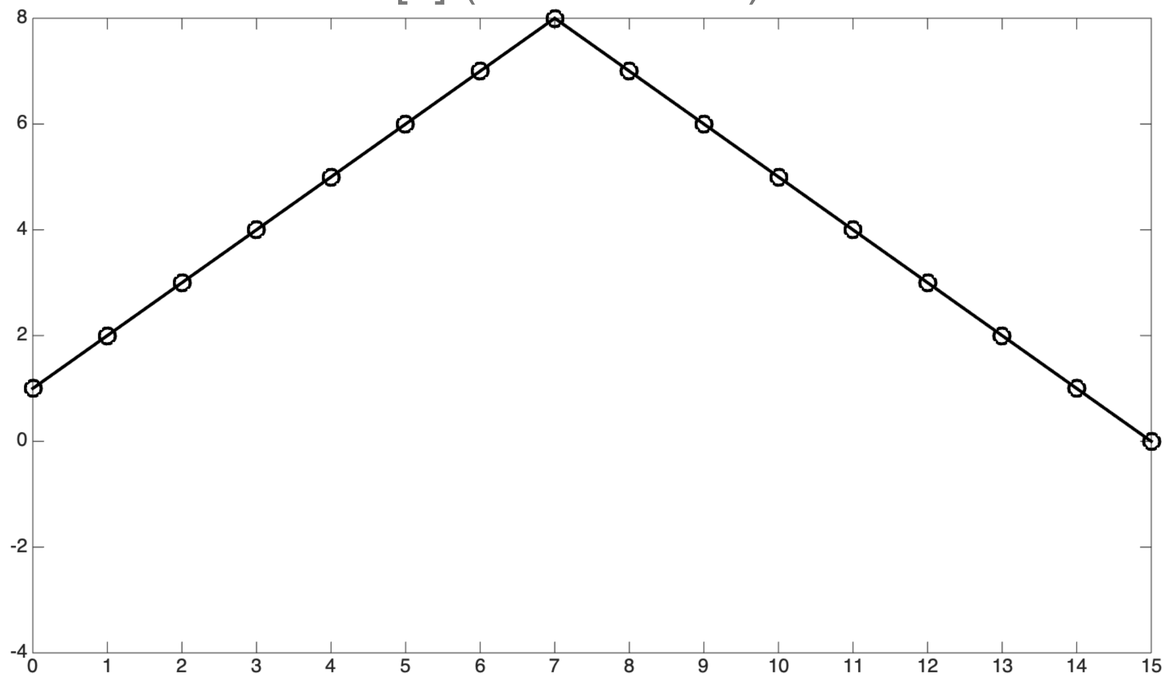
(other components are 0 or redundant)

The Discrete Fourier Transform

- Key point: N time samples ↔ N frequency samples
  - Two different mathematical perspectives on the same signal
  - The Inverse Discrete Fourier Transform (IDFT) converts from frequency domain to time domain
  - The DFT+IDFT is an exact reconstruction as long as signal length $M \leq$ DFT length N

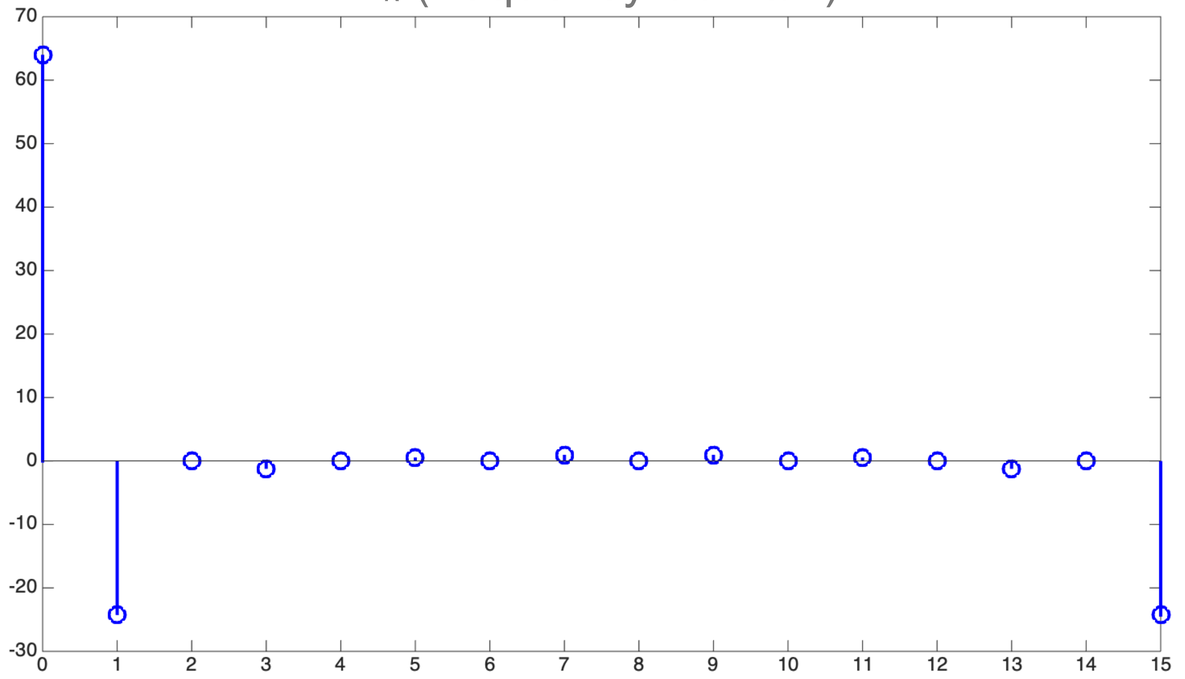
$x[n]$ (time domain)



DFT

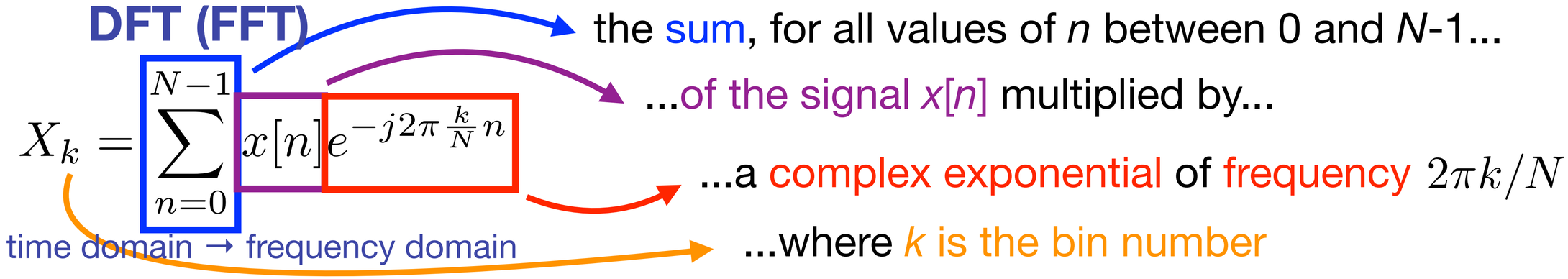
IDFT

X_k (frequency domain)



The Discrete Fourier Transform

- The mathematical basis of the Discrete Fourier Transform:



- N is the size of the DFT
  - For the FFT, N is almost always a power of 2 (the Cooley-Tukey algorithm)
- Each time sample *x*[*n*] is a real number
- Each frequency bin X_k is a complex number: $a + bj$ where $j = \sqrt{-1}$
  - Magnitude of the bin is given by: $\sqrt{a^2 + b^2}$
    - We usually use magnitude to plot a spectrum or spectrogram of a signal
  - Phase of the bin is given by: $\arctan(b/a)$

The Discrete Fourier Transform

- The mathematical basis of the Discrete Fourier Transform:

DFT (FFT)

$$X_k = \sum_{n=0}^{N-1} x[n] e^{-j2\pi \frac{k}{N} n}$$

time domain → frequency domain

IDFT (IFFT)

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{+j2\pi \frac{k}{N} n}$$

frequency domain → time domain

- N is the size of the DFT
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The Discrete Fourier Transform

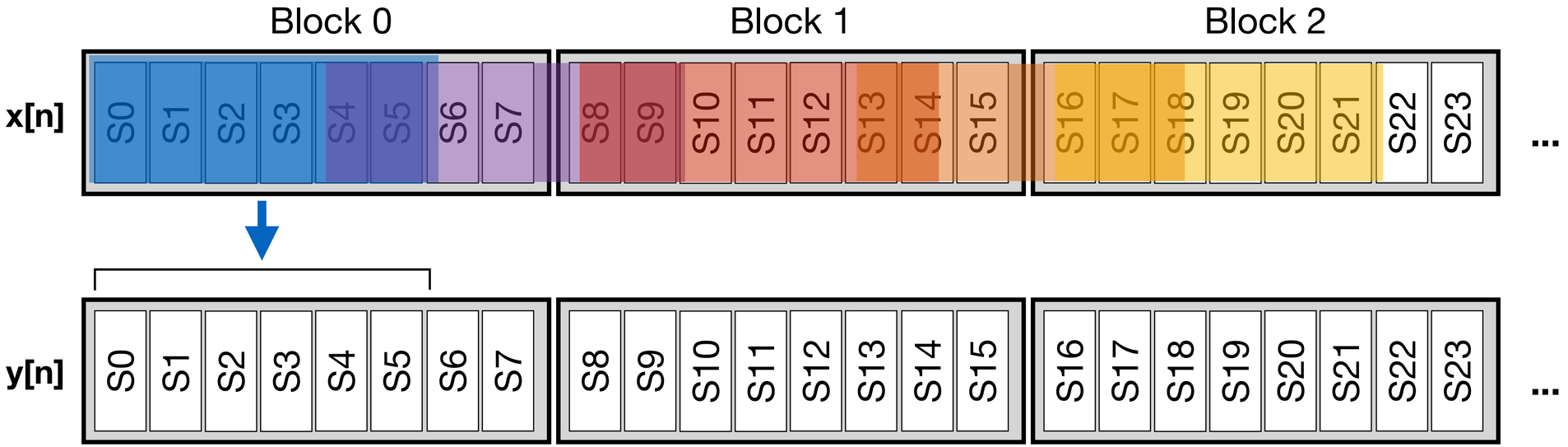
- Frequency bins are equally spaced between 0 and 2π:
  - Where, for sampled analog signals, 2π corresponds to the sample rate

$$X_k = \sum_{n=0}^{N-1} x[n] e^{-j2\pi \frac{k}{N} n} \qquad x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{+j2\pi \frac{k}{N} n}$$

$0 \leq k < N$

- But wait! Doesn't a frequency of π represent the Nyquist rate?
  - Why are there bins with frequency higher than π?
- The higher bins don't contain any unique information in our case:
  - The DFT works generally with complex signals, but audio signals are always real-valued
  - With a real signal, the frequency bins are conjugate symmetric: $X_k = X^*_{N-k}$
- Therefore, we only care about bins $0 \leq k \leq N/2$
  - This is why you'll see audio spectrum plots stop at the Nyquist frequency

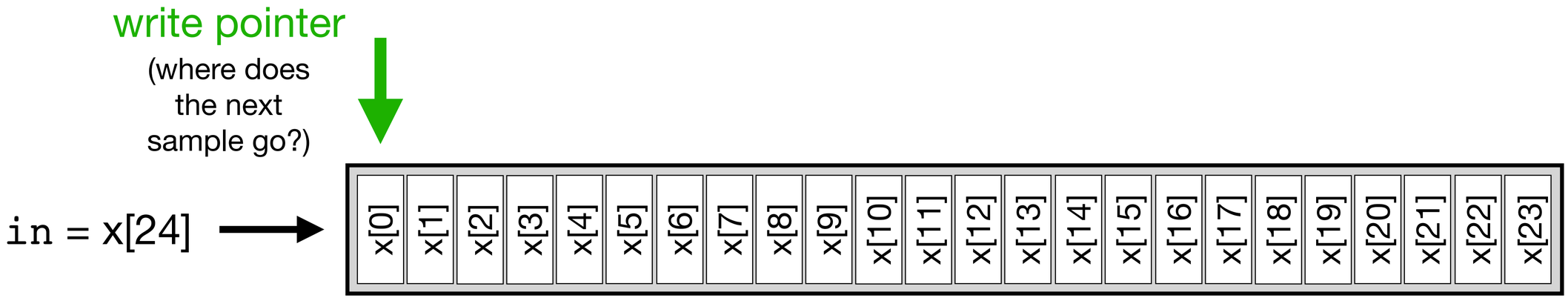
Overlapping blocks in real time



- As we'll see next lecture, we often want to work with overlapping blocks
  - This means block *k* shares some samples with block *k*-1
  - How do we adapt our windowing code to handle this?
- Easiest approach: keep a running history of the input samples
  - i.e. a buffer which always has the last M samples
  - What kind of structure have seen that does this? Circular buffer
  - At each hop, pass M samples from circular buffer to FFT

Review: circular buffering

- 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



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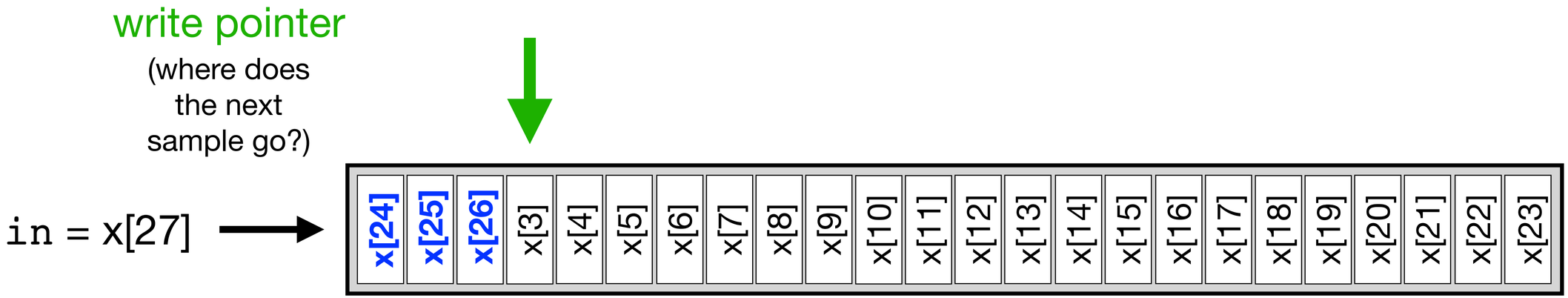
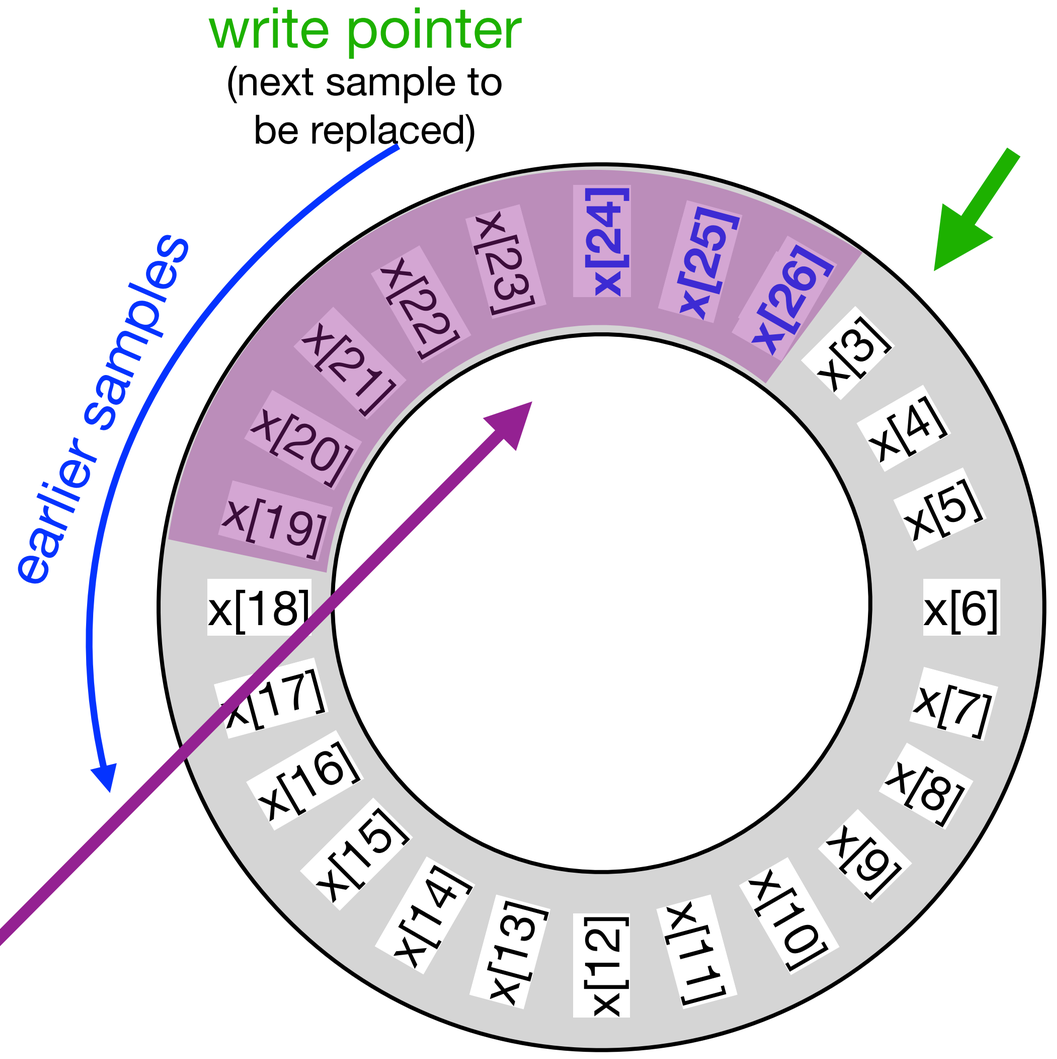


Circular buffering and windowing

- 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
  - As long as our buffer is big enough, we can always get the latest window of the signal



Circular buffer 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:

```
float gLastSamples[gBufferSize] = {0};
int gWritePointer = 0;
```

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

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

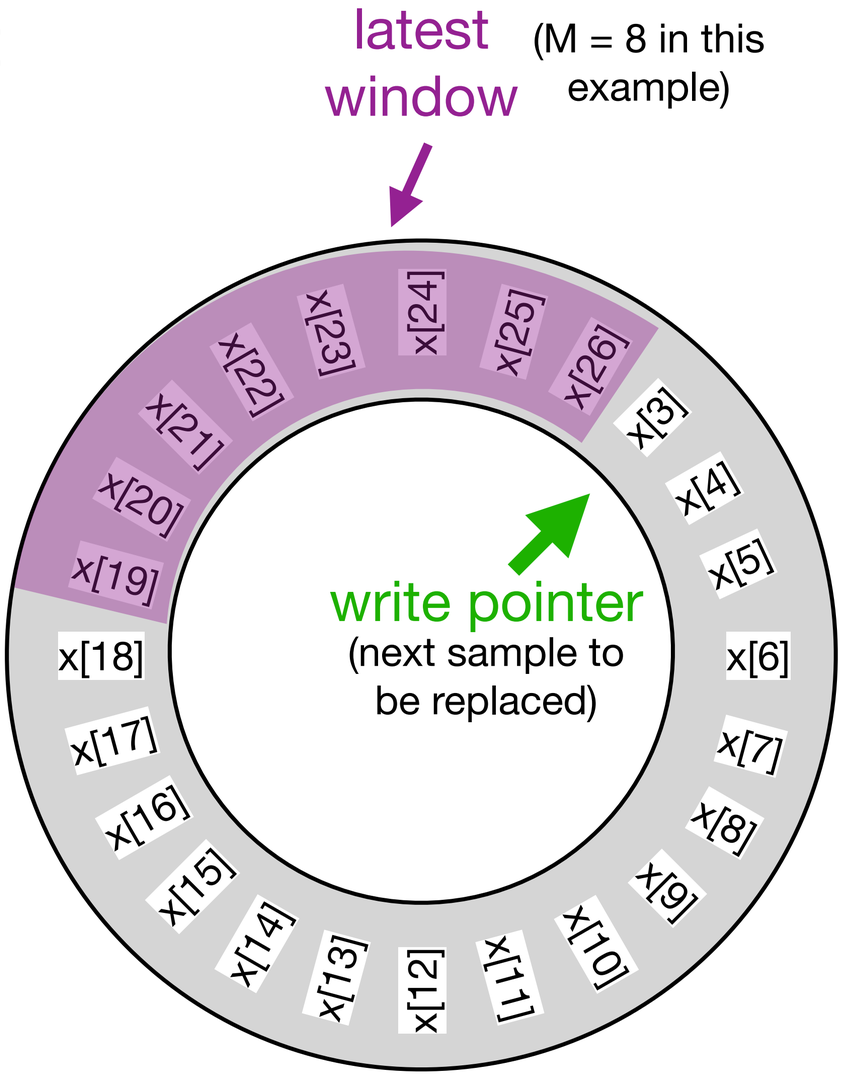
  - What else do we need to do?
    - Keep the write pointer in range

```
if(gWritePointer >= gBufferSize)
    gWritePointer = 0;
```

Overlap-add with a circular buffer

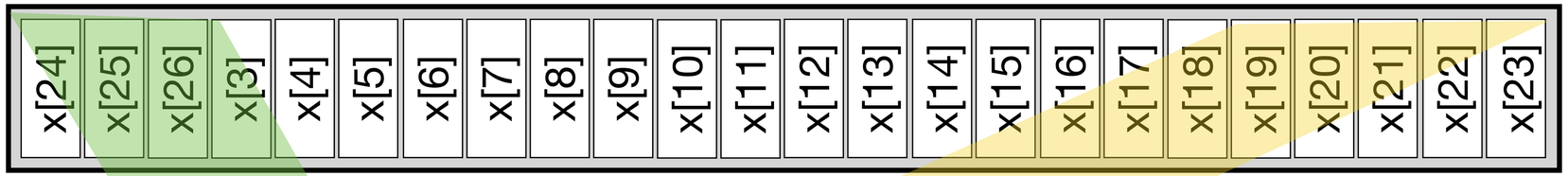
- We use a circular buffer to keep track of the input
- Necessary characteristics:
  - The buffer always holds (at least) the last M input samples (where M is the window size)
  - If using multiple channels, one buffer per channel
- Using the circular buffer:
  - Each iteration of the `for()` loop:
    - store input in buffer
    - increment the write pointer (wrapping as necessary)
    - increment the total count of samples stored
  - When the count reaches the hop size:
    - take one window from buffer, unwrap it, and pass it to FFT
    - in other words: copy it to a new buffer such that the oldest sample appears at index 0 of the new buffer

Important: this is not the same as when the write pointer wraps around!

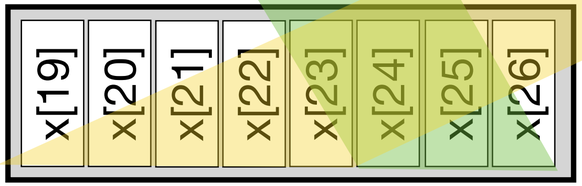


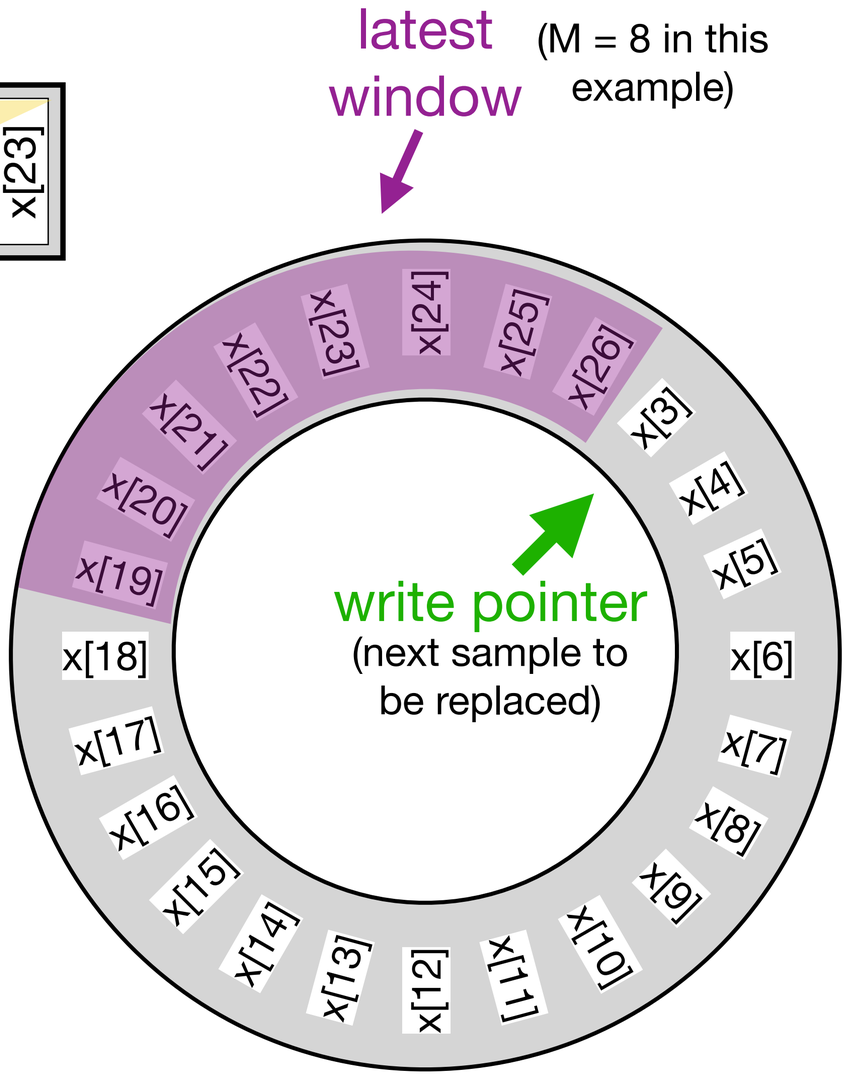
Unwrapping the circular buffer

How the circular buffer is actually stored in memory:



What we need to calculate the FFT (the latest 8 samples):





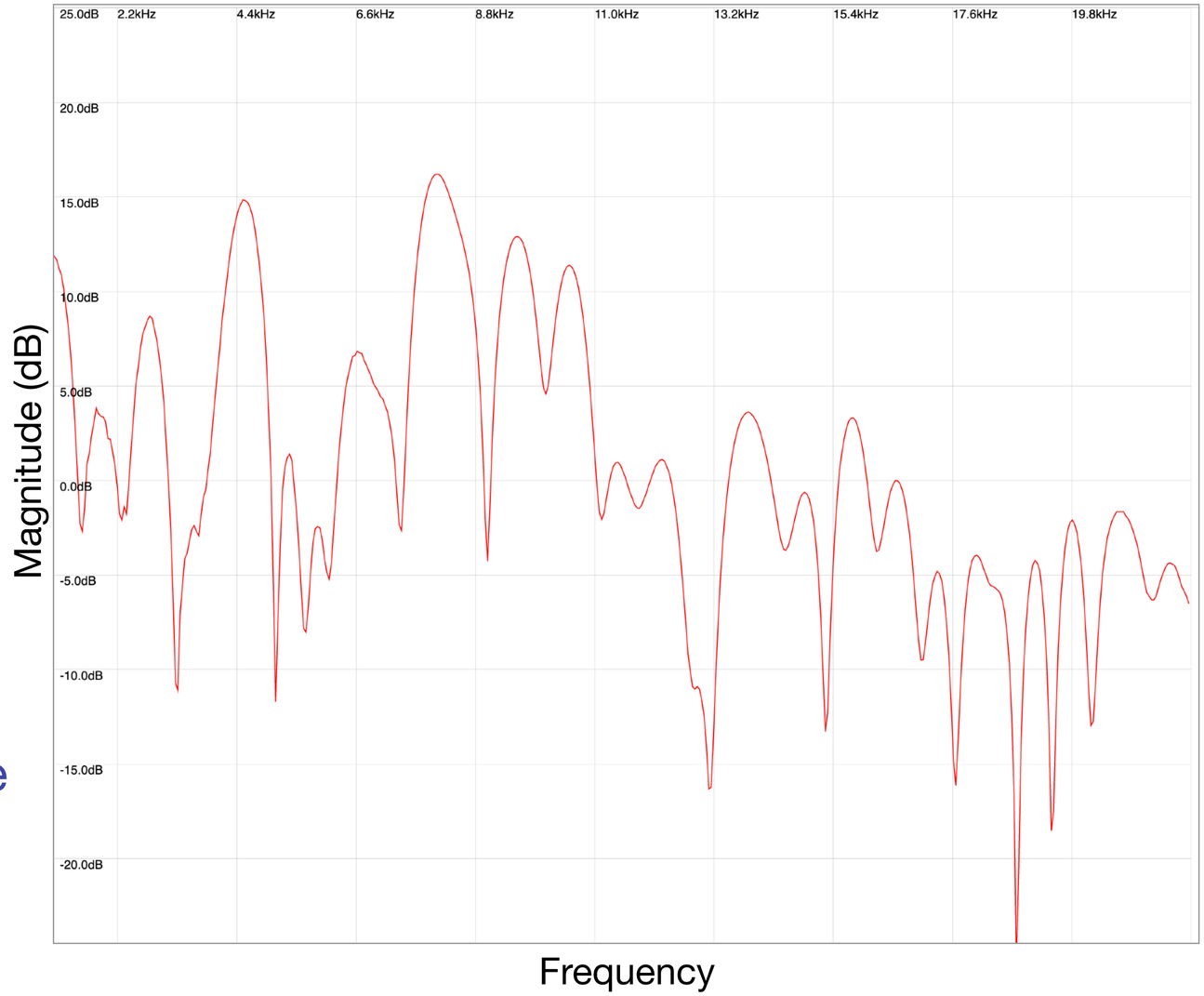
In other words: having assembled the samples into a circular buffer, on each hop it is helpful to copy them into a linear (standard) buffer to pass to the FFT library

Block-based processing: input summary

- Block-based processes are calculated at regular intervals, not every frame
  - Unlike filters, where the output is updated every frame
  - The block-based calculation typically involves a recent window of the signal
- Two relevant quantities:
  - Window size (block size): how long of a segment of the signal is used in the calculation?
  - Hop size: at what interval is the calculation performed?
    - The hop size will typically be no larger than the window size, and is commonly a fraction of window size
  - Neither of these quantities is the same as the real-time audio system buffer size!
    - Your block-based code should work identically regardless of audio buffer size
- We can use a circular buffer to keep track of a recent window of samples
  - Needs the usual array and write pointer
- We also keep a counter of how many frames have elapsed
  - Different from the write pointer of the circular buffer
  - When the counter reaches the hop size, we perform a new calculation

Plotting the audio spectrum

- The bins of the DFT can be used to plot the audio spectrum
  - Frequency on the X axis, magnitude on the Y axis (remember, bins are complex)
  - The Y axis is usually logarithmic (decibel scale)
  - Sometimes the X axis is logarithmic too
- We only plot the first half of the bins
  - The higher bins are the mirror image of the lower ones (conjugate symmetry)
  - For an N-sample DFT, bin N/2 corresponds to a frequency of π, which is the Nyquist rate
- The spectrum is typically a local snapshot, based on a window of signal

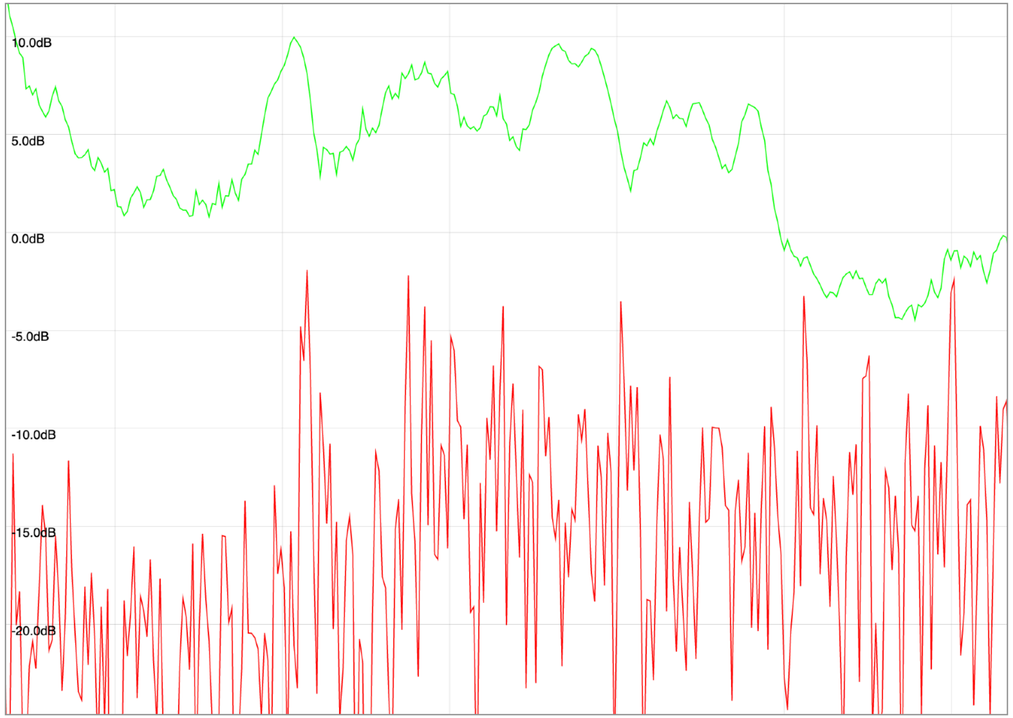


Circular buffer task

- Task: using fft-circular-buffer project (copying in drumloop.wav)
  - Implement the circular buffer in `render()`
  - See variables `gInputBuffer` and `gInputBufferPointer`
    - You still need both of these, and they work similarly to the last example, except...
    - `gInputBufferPointer` should not reset to 0 on each hop
      (it should wrap around whenever it hits the end of the buffer)
  - You need to use the additional global variable `gHopCounter`
    - Use this to keep track of how many samples have gone by,
      rather than using `gInputBufferPointer` to do this
    - When `gHopCounter` reaches the hop size, call `process_fft()` and reset the counter to 0
  - Implement the circular buffer in `render()`
  - Unwrap the buffer in `process_fft()`
    - See the code at the top of the function where the data is copied into the FFT data structure
    - Use modulo indexing here to copy the right samples

Peak detection

- Task 2: add peak detection to the plot
  - We'll calculate a second spectrum which stores the most recent peak value for each bin
  - Need to declare a persistent (global or static) array
    - This will remember the peak value for each frequency bin
    - Each hop, iterate through the bins:
      if the current value is greater than the peak,
      then update the peak
    - Multiply the peak by a decay factor (try 0.98) so the peaks decay over time
  - Send the peak spectrum to the GUI with buffer ID 1



- Task 3: also add smoothing to the peak spectrum
  - Calculate a local average of the nearest 11 bins (from bin *k*-5 to *k*+5)
  - Be careful not to run off the end of the array!

Keep in touch!

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forum.bela.io

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More resources and contact info at:

learn.bela.io/resources