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

Today

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

Lecture 18: Phase vocoder, part 1

What you'll learn today:

Segmenting a real-time signal into windows
Working with overlapping windows (overlap-add)
Multi-threaded audio processing

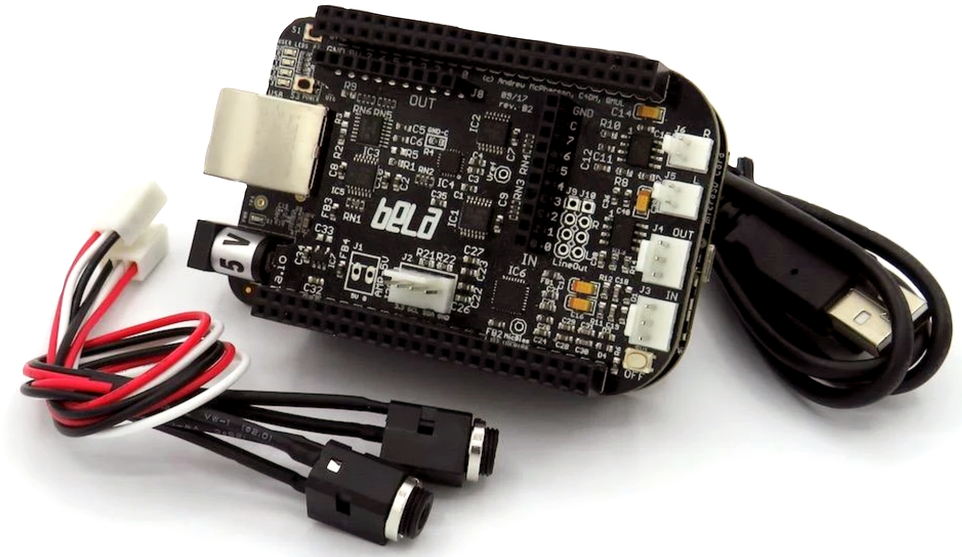
What you'll make today:

A phase vocoder framework with simple effects

Companion materials:

github.com/BelaPlatform/bela-online-course

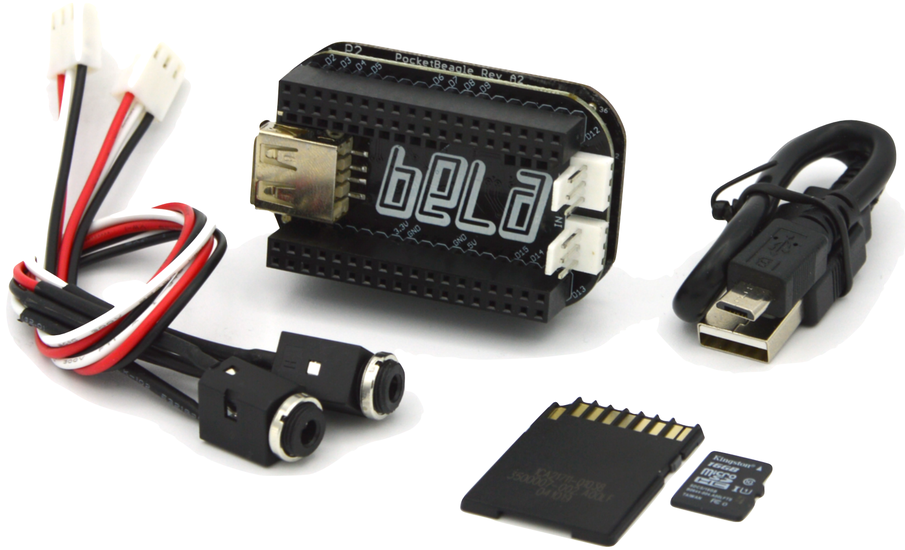
What you'll need



Bela Starter Kit

or

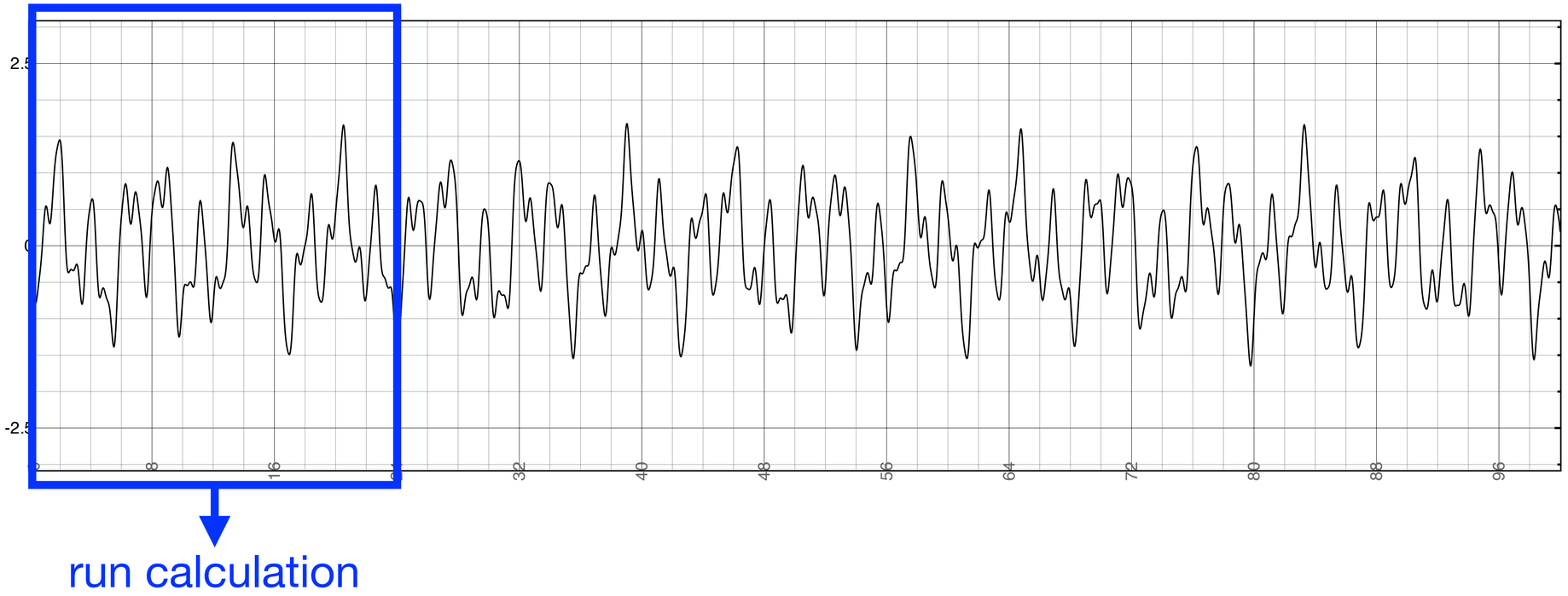
[shop.bela.io]



Bela Mini Starter Kit

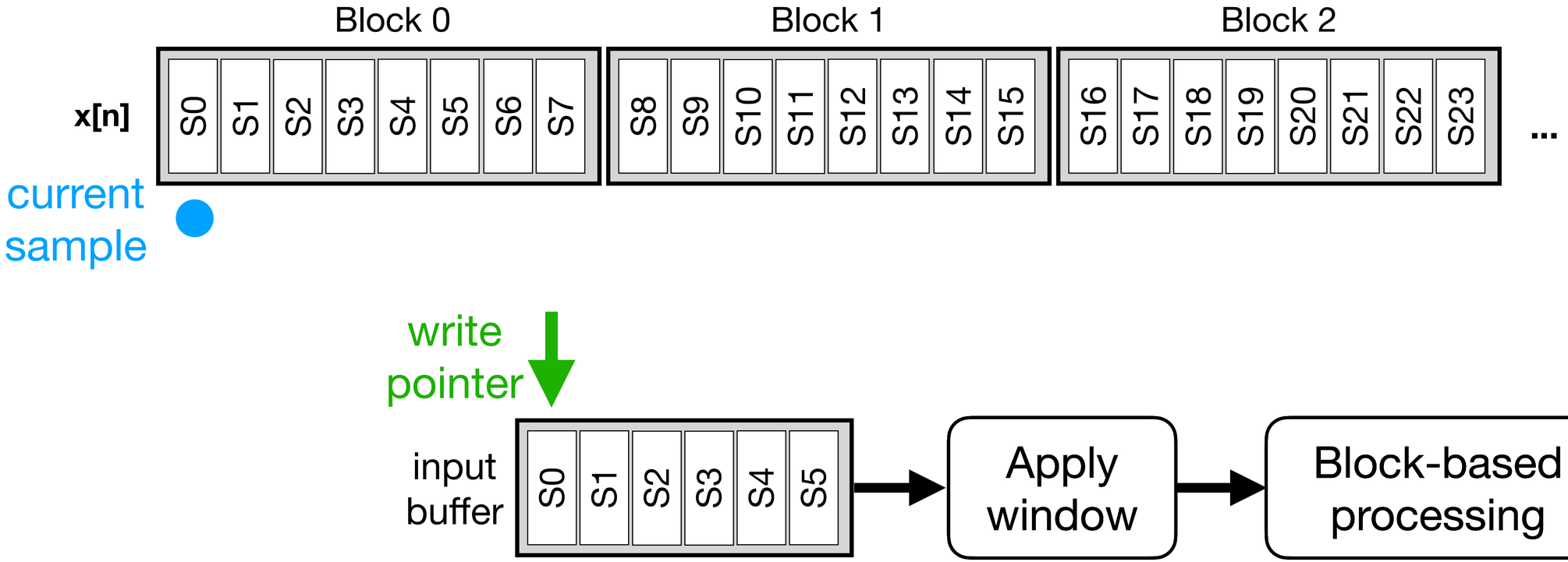
Block-based processing

- 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

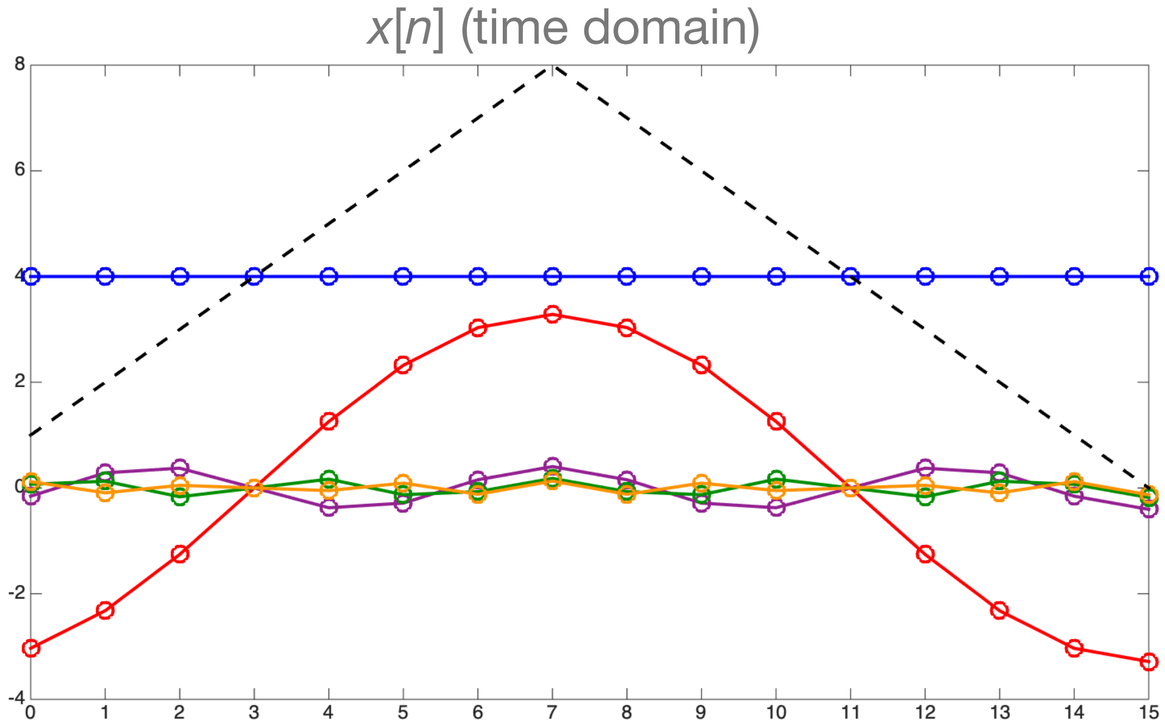
Block-based input



- 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)

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

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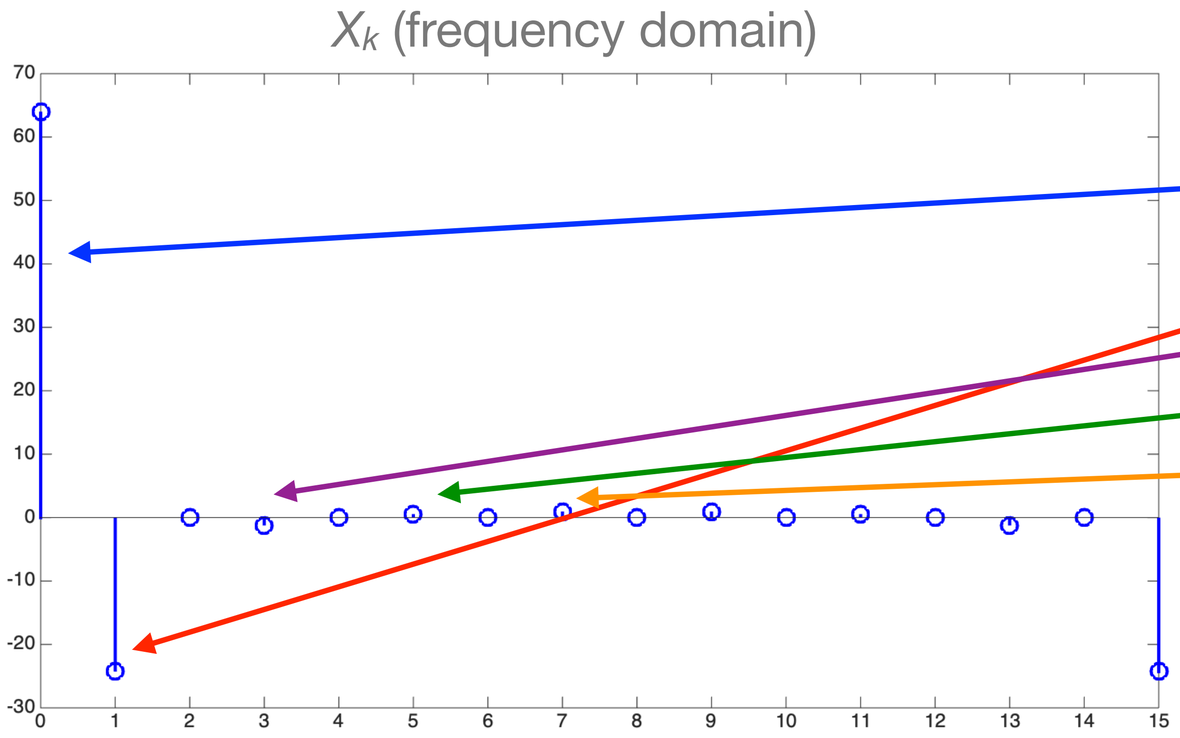

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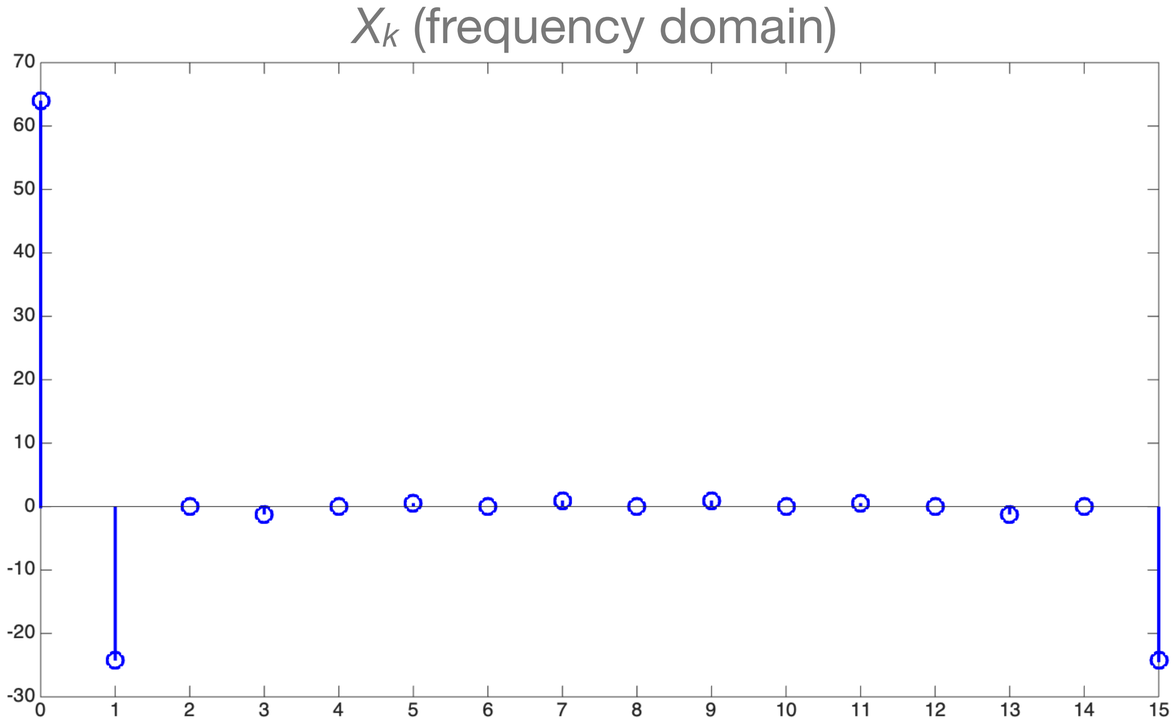
The DFT calculates the amplitudes and phases of each frequency component

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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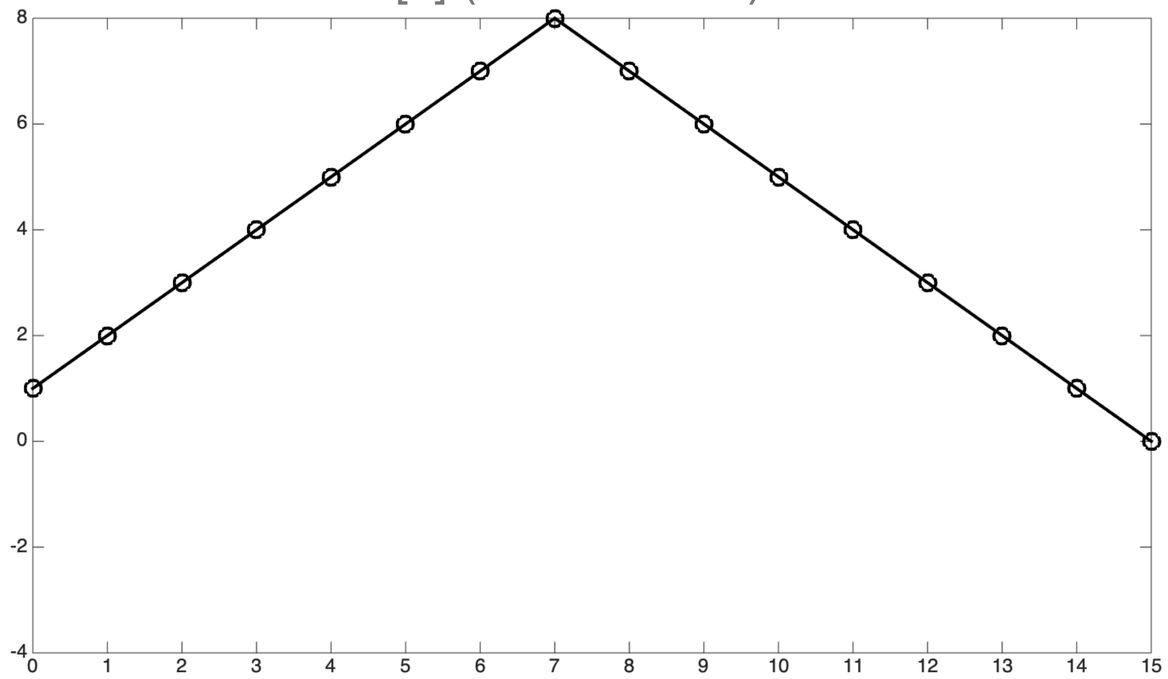
The Discrete Fourier Transform

- 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

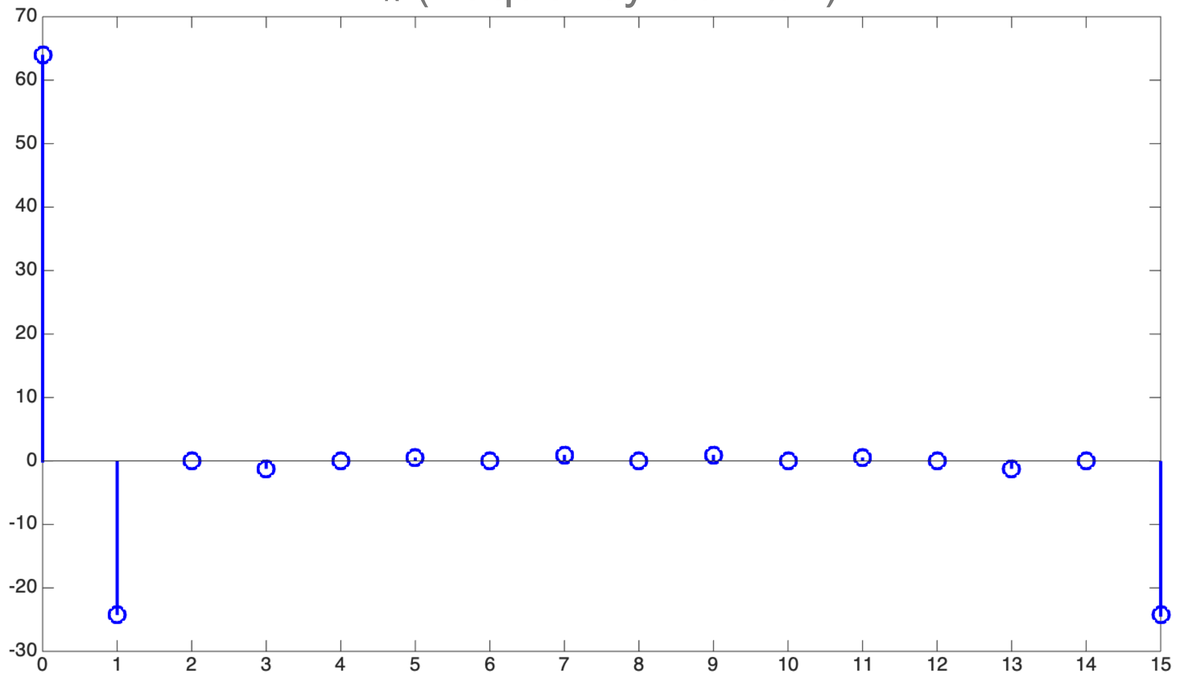
$x[n]$ (time domain)



DFT

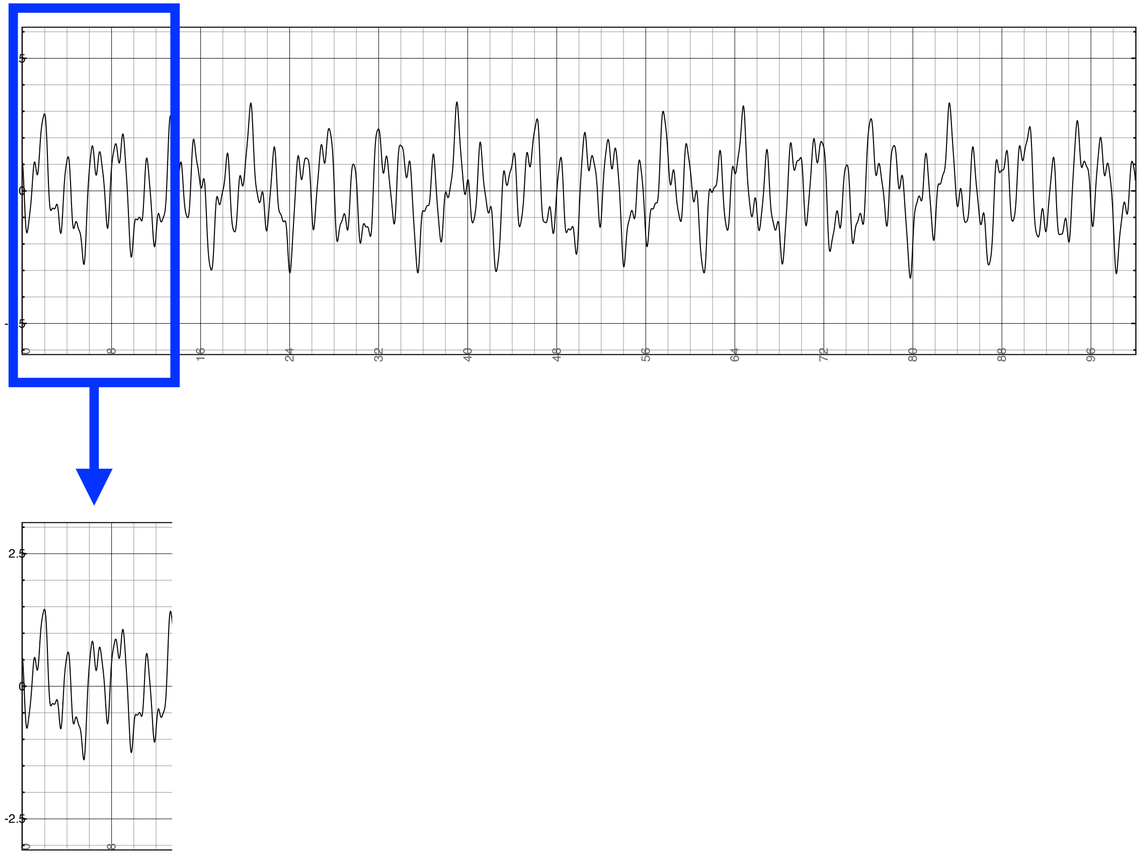
IDFT

X_k (frequency domain)



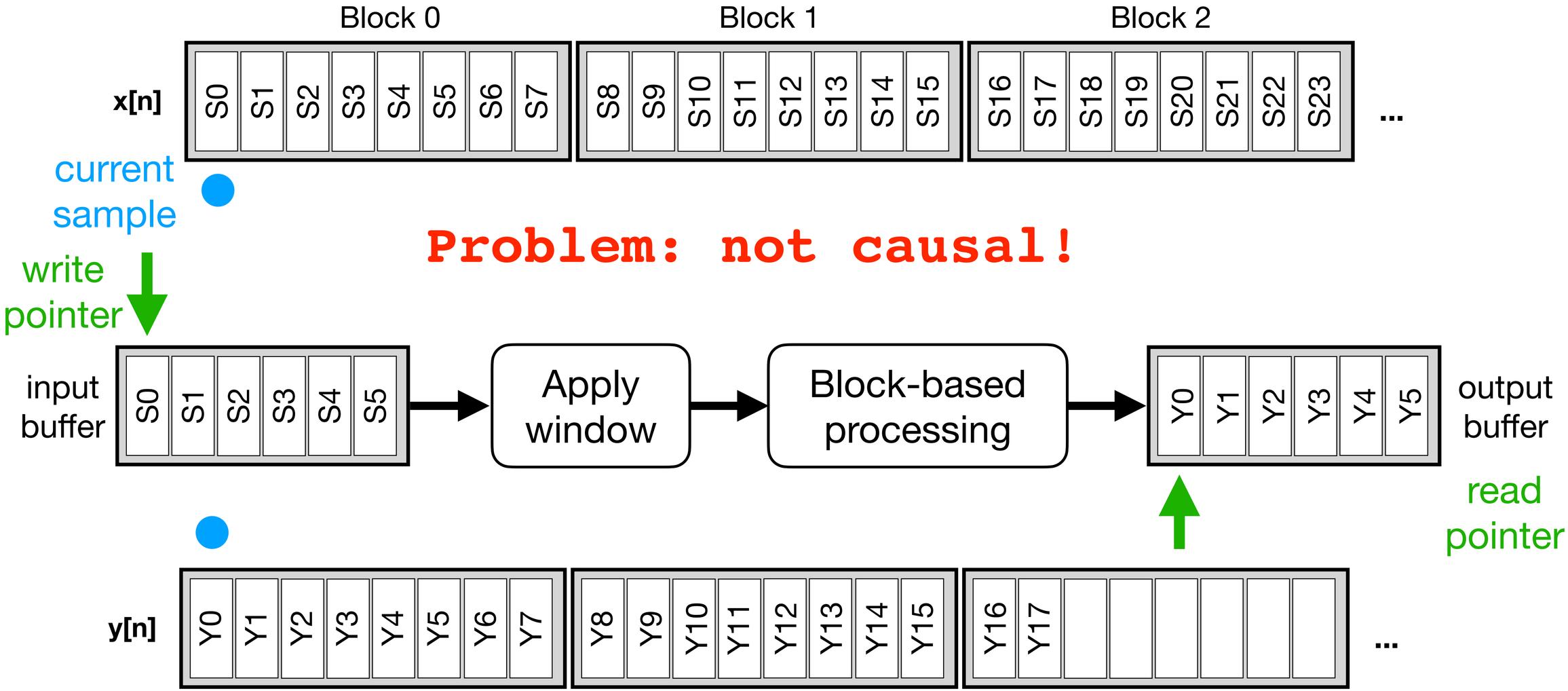
Block-based input and output

- After we divide our signal into windows, we often want to reconstruct it
  - Do some processing on each block
  - Then stitch the processed blocks back together into a new signal
  - In the ideal case, if we don't do any processing on each block, we should be able to reconstruct the signal exactly
- Application: phase vocoder
  - Take an FFT on each block
  - Do processing in the frequency domain using the phase of the bins for a more accurate frequency estimate (more soon!)
  - Take the IFFT on each block



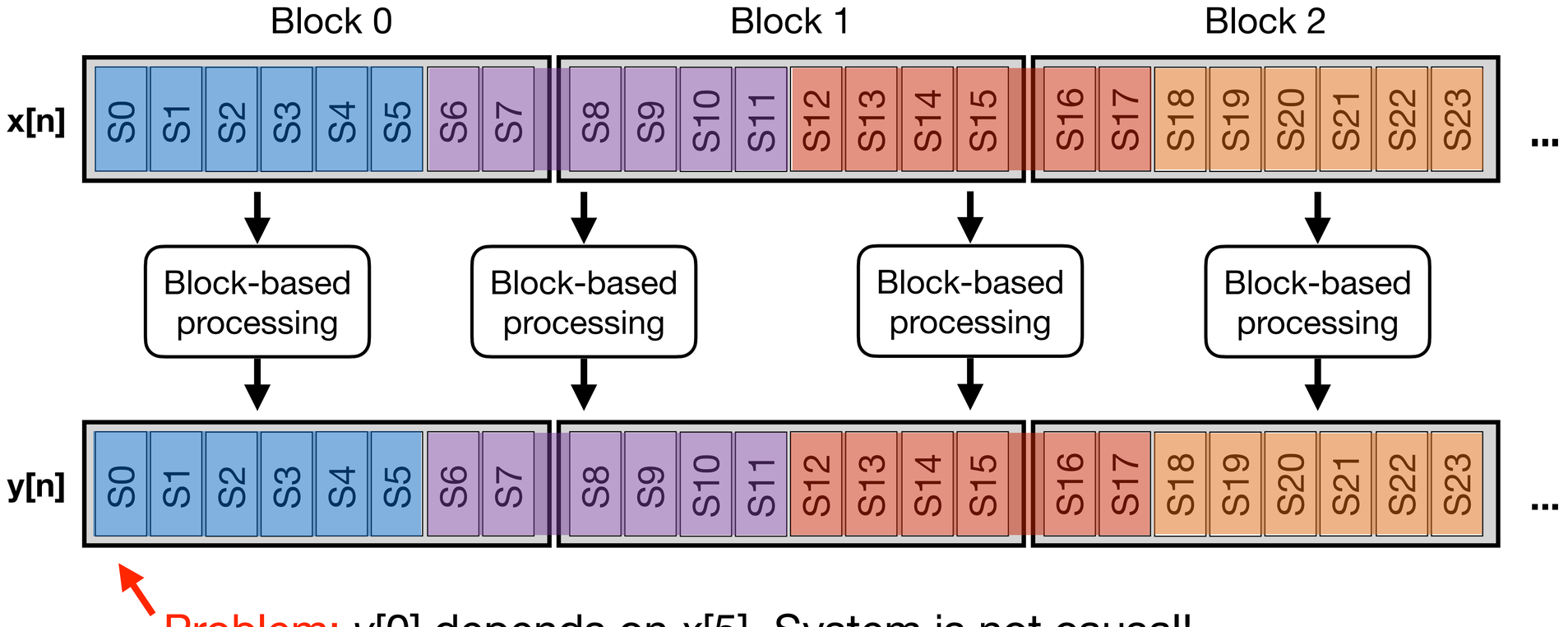
Block-based input and output

- Our ideal block-based processing system might work something like this:



Block-based output: causality

- To have a causal system, an output cannot depend on future inputs



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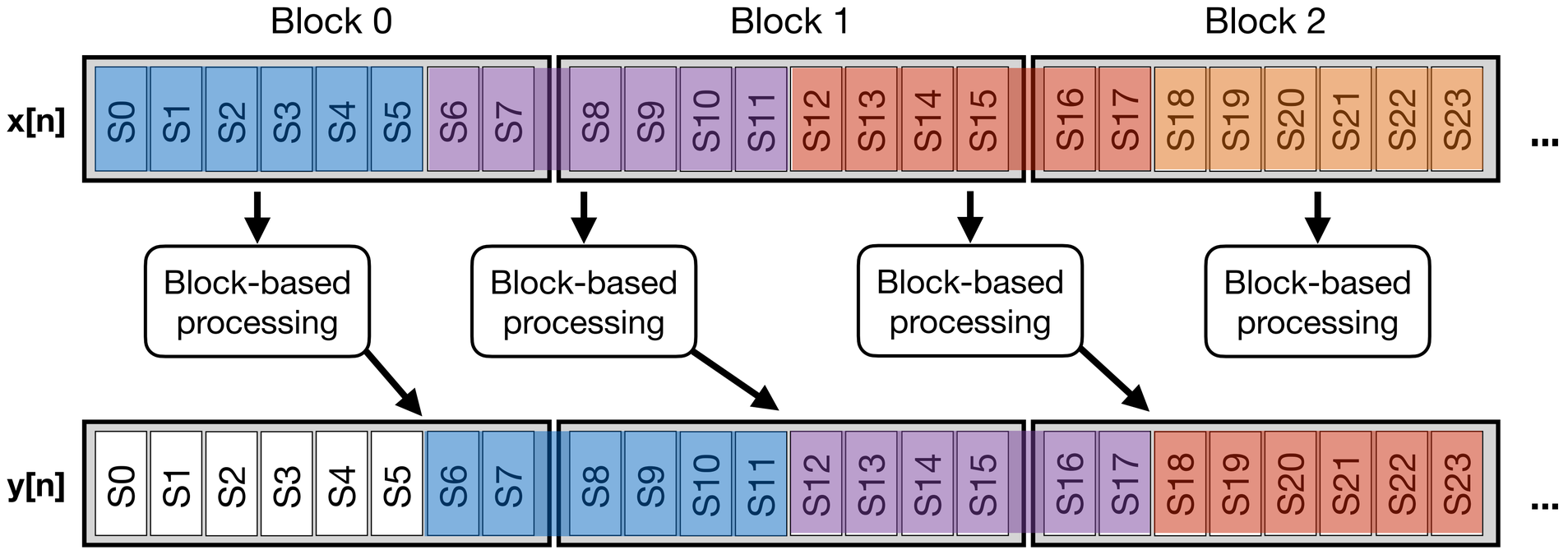
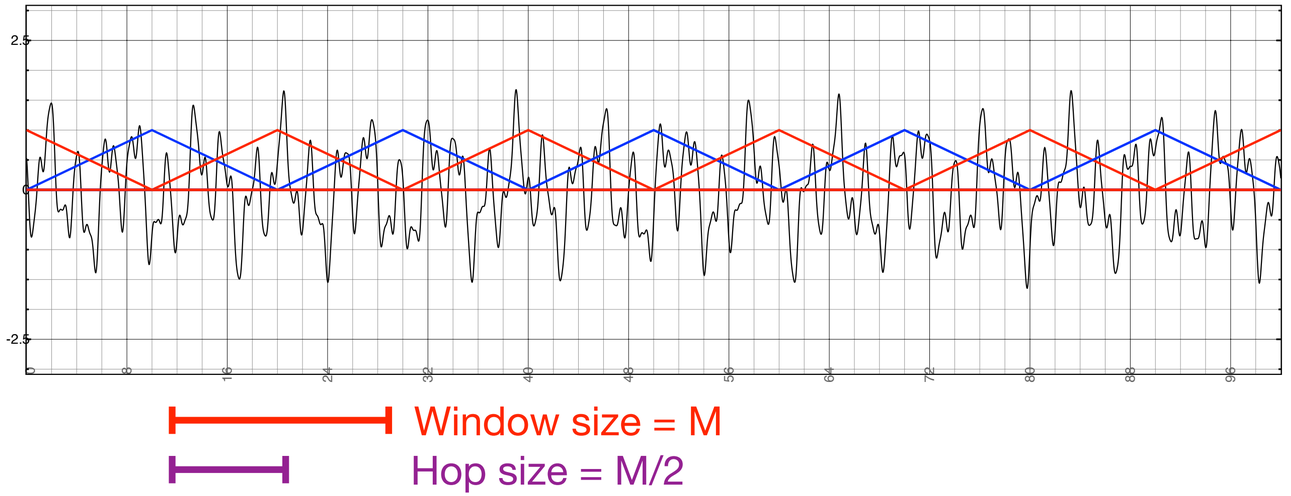


- Partial solution: deliberately add a 1-window latency to the output
  - As we gather input samples for the next FFT calculation, we are copying samples from the last FFT into the output

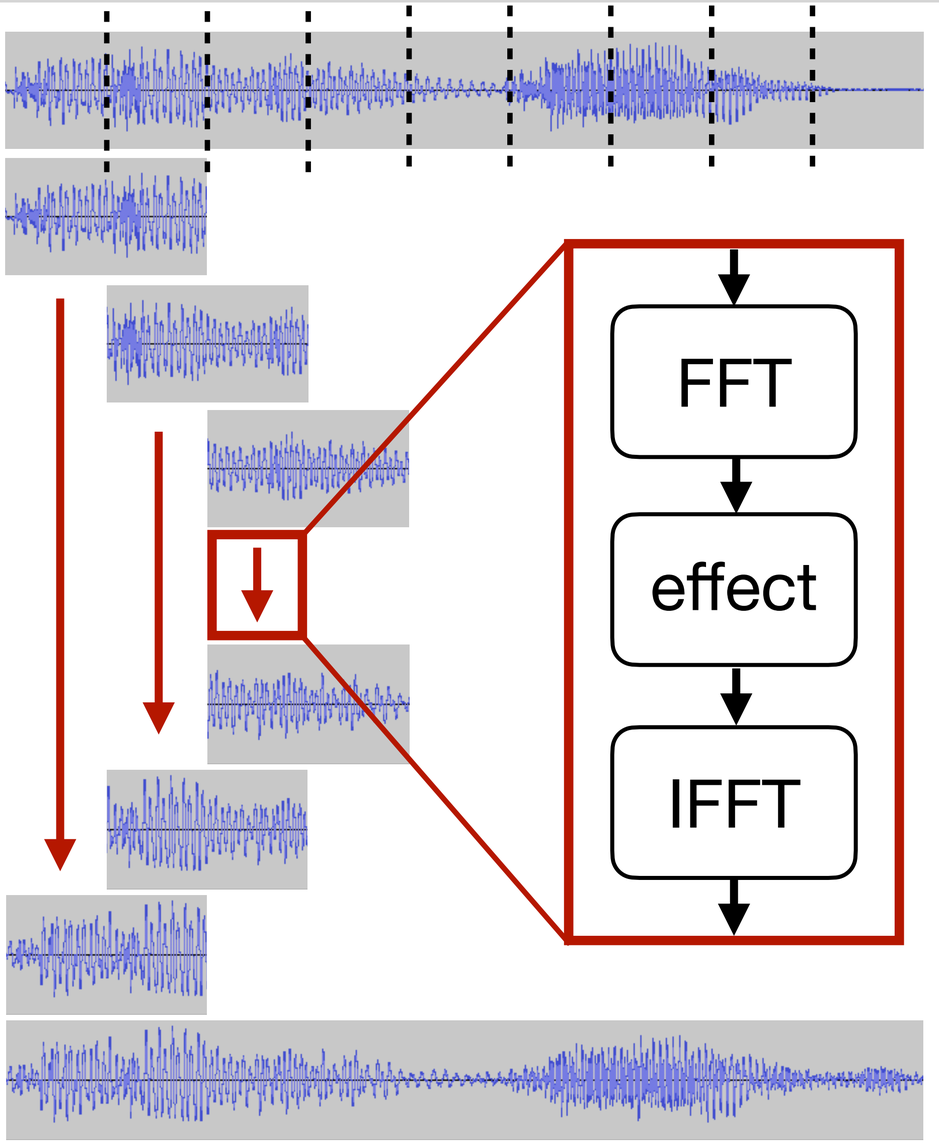
Overlapping blocks

- The previous example performed a block calculation at regular intervals
  - Once a full block of samples was assembled, it was processed (e.g. using an FFT)
  - The larger the window (block) size, the less frequent the calculations
  - In that example, each input sample was part of exactly one block
- It can be useful to have blocks that overlap one another
  - For example, use larger FFTs but calculate them more often
  - The number of samples between the start of consecutive blocks is called the hop size
  - This is often a fraction of the block size (or window size)
    - For example, FFT size 1024, hop size 512
- We might apply a window function before calculating the FFT
  - In this example, a triangular window

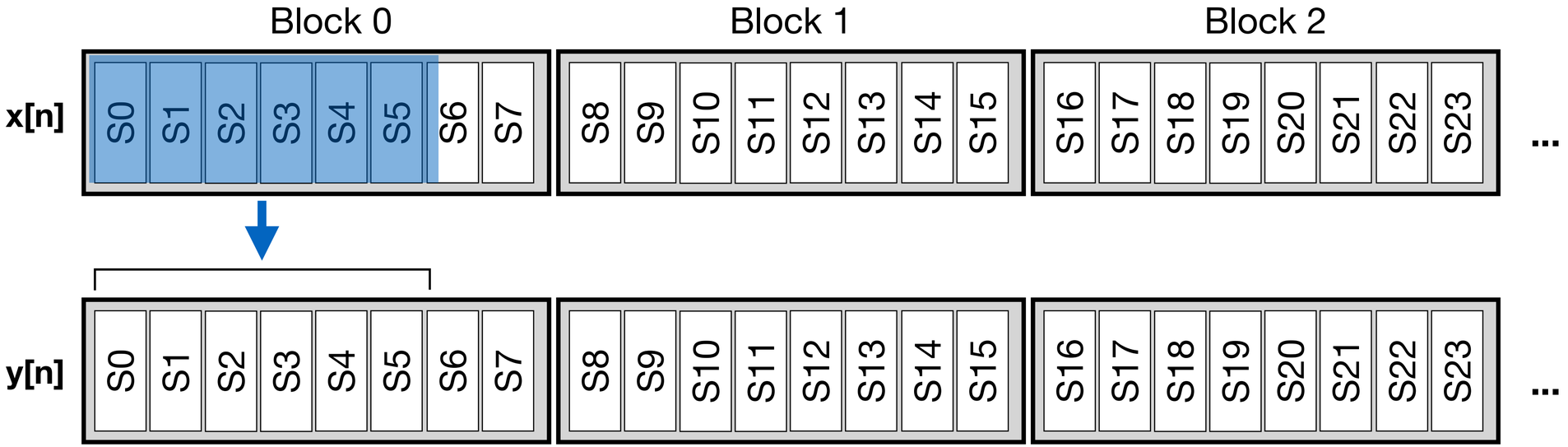


Overlap-Add

- A standard approach for block-based processing with overlapping blocks is called overlap-add:
  1. Isolate a block of length M using a windowing function
  2. Take an FFT of length N ≥ M of the segment
     - If N > M, zero-pad the block (add zeros to end of the window)
  3. Do something interesting to frequency-domain data
  4. Take an IFFT to get a new time domain segment
  5. Add the segment to an output buffer which also contains the preceding segments
     - As we'll see, we can't write the segment directly to the audio output
  6. Advance by the hop size (H) to the next frame and repeat
     - In real time, count samples until H more have arrived
     - Hop size H is less than window size M, hence the **overlap**



Review: overlapping blocks (input side)



- The goal is to work with overlapping blocks in real time
  - 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

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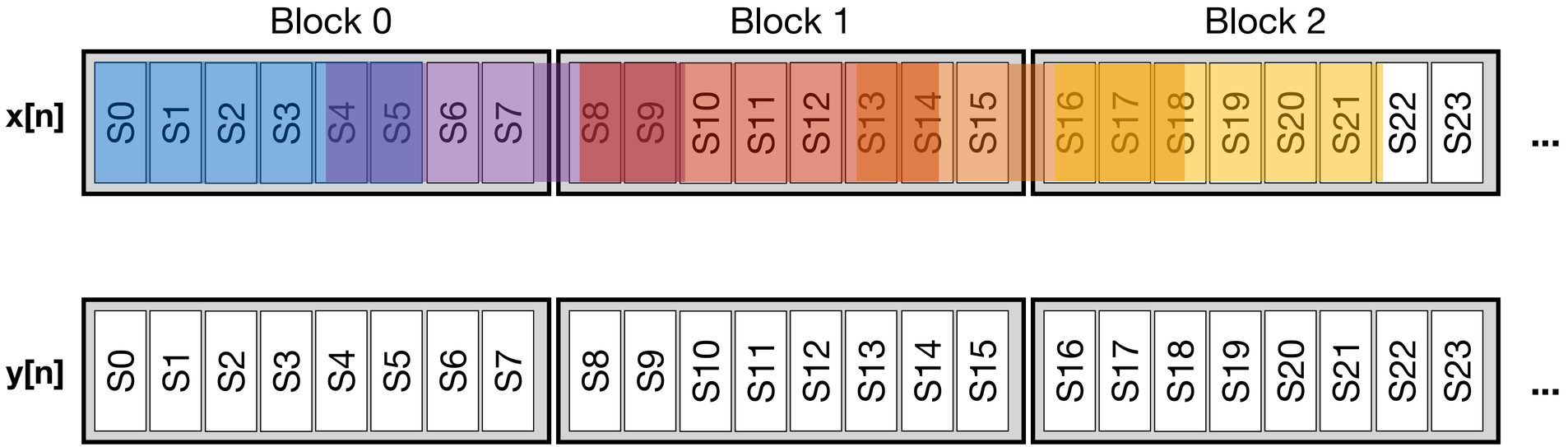

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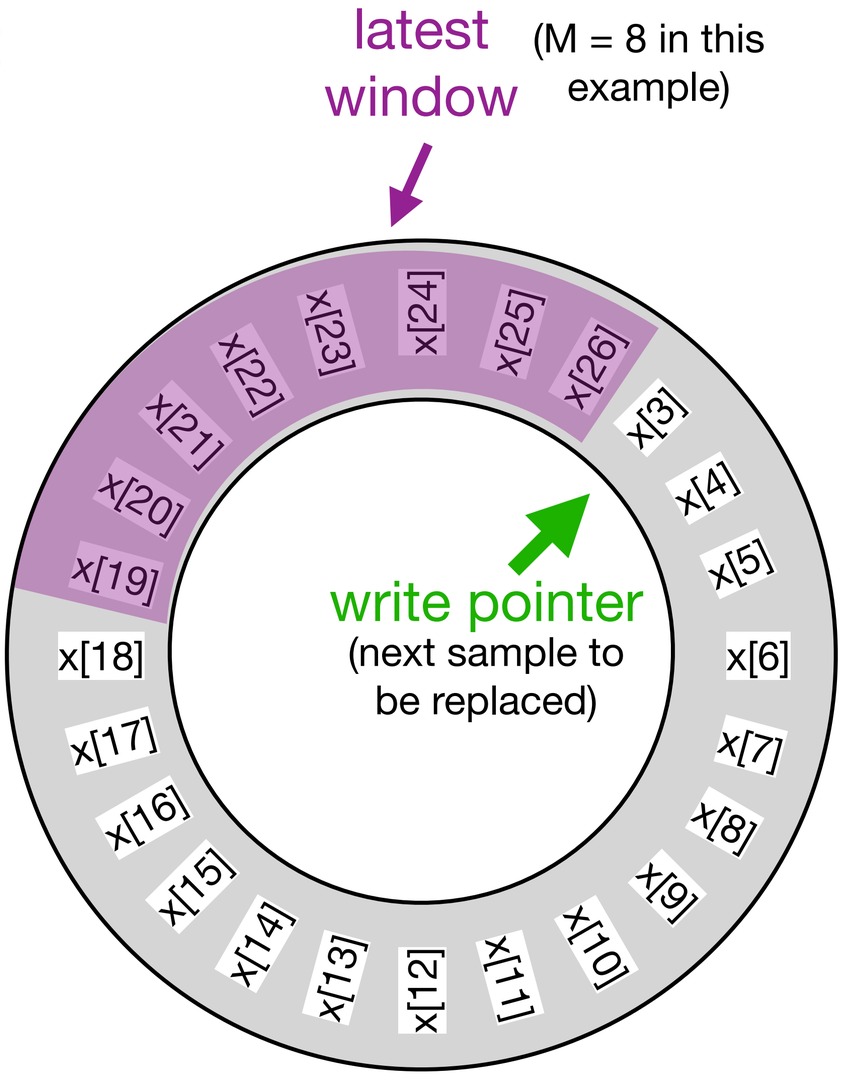
Review: overlap-add with a circular buffer

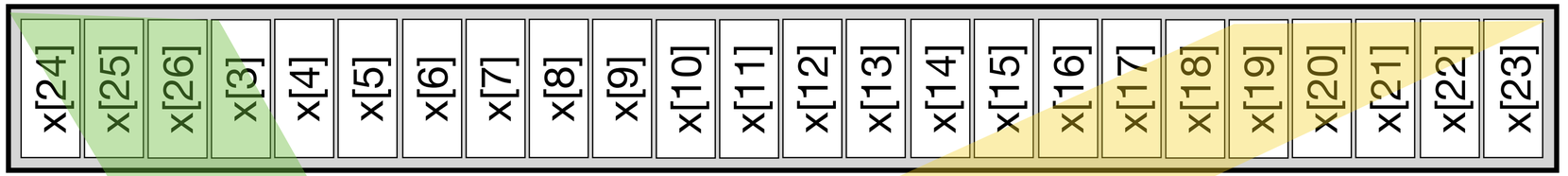
- We use a circular buffer to keep track of the input
- Necessary characteristics:
  - The buffer always holds (at least) the last M input samples (where M is the window size)
  - If using multiple channels, one buffer per channel
- Using the circular buffer:
  - Each iteration of the `for()` loop:
    - 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!

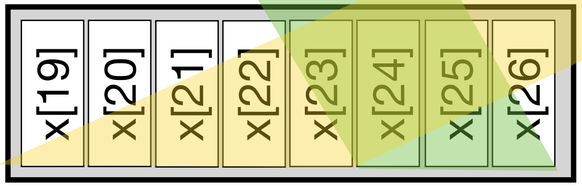


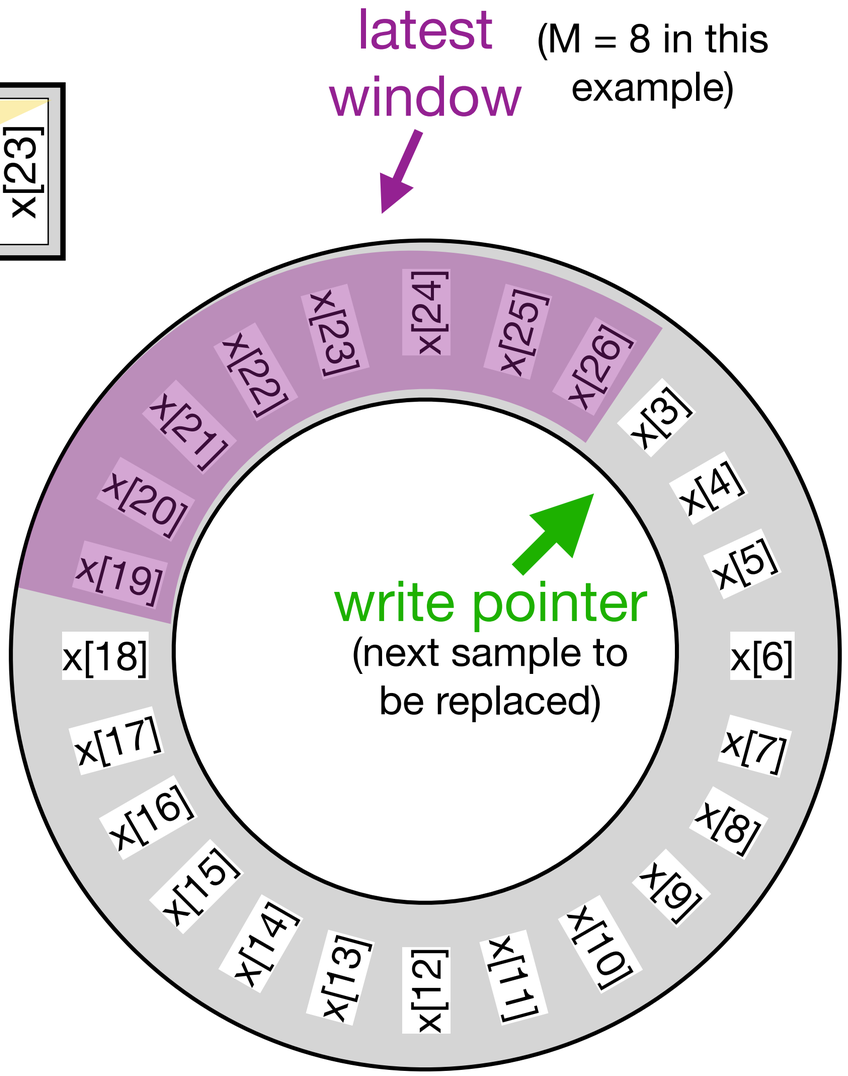
Review: unwrapping the circular buffer

How the circular buffer is actually stored in memory:



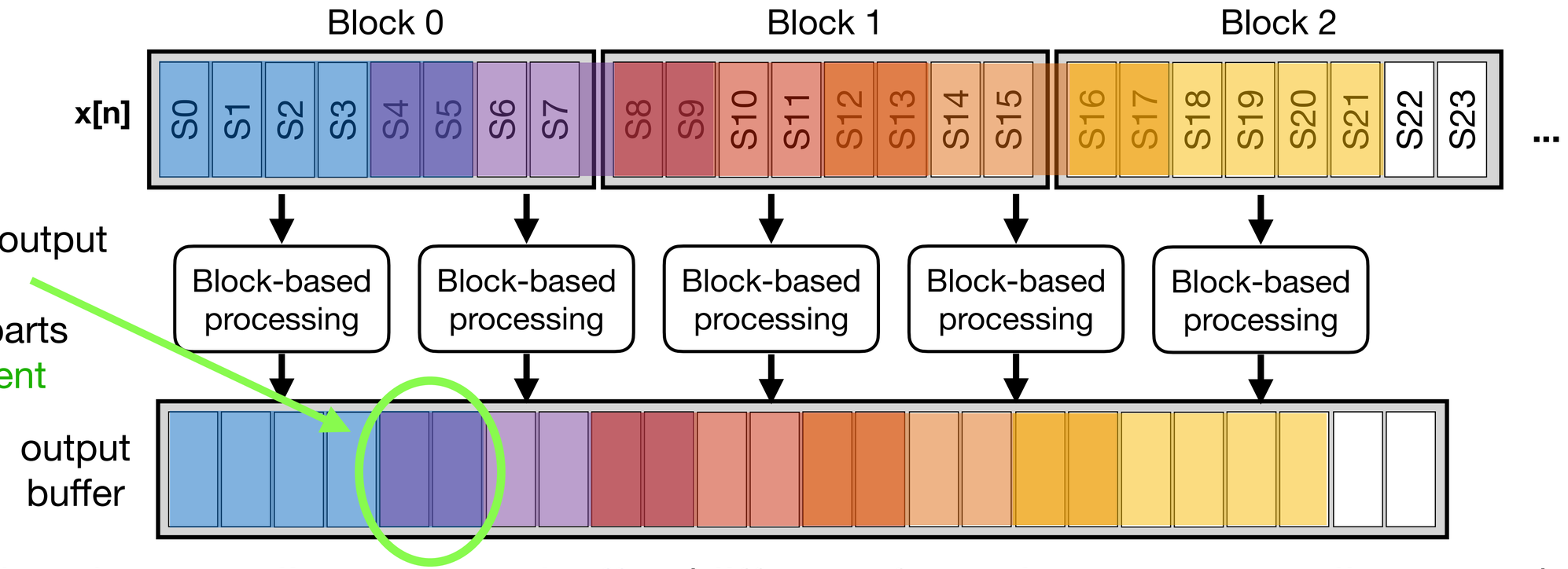
What we need to calculate the FFT (the latest 8 samples):





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 output: overlap

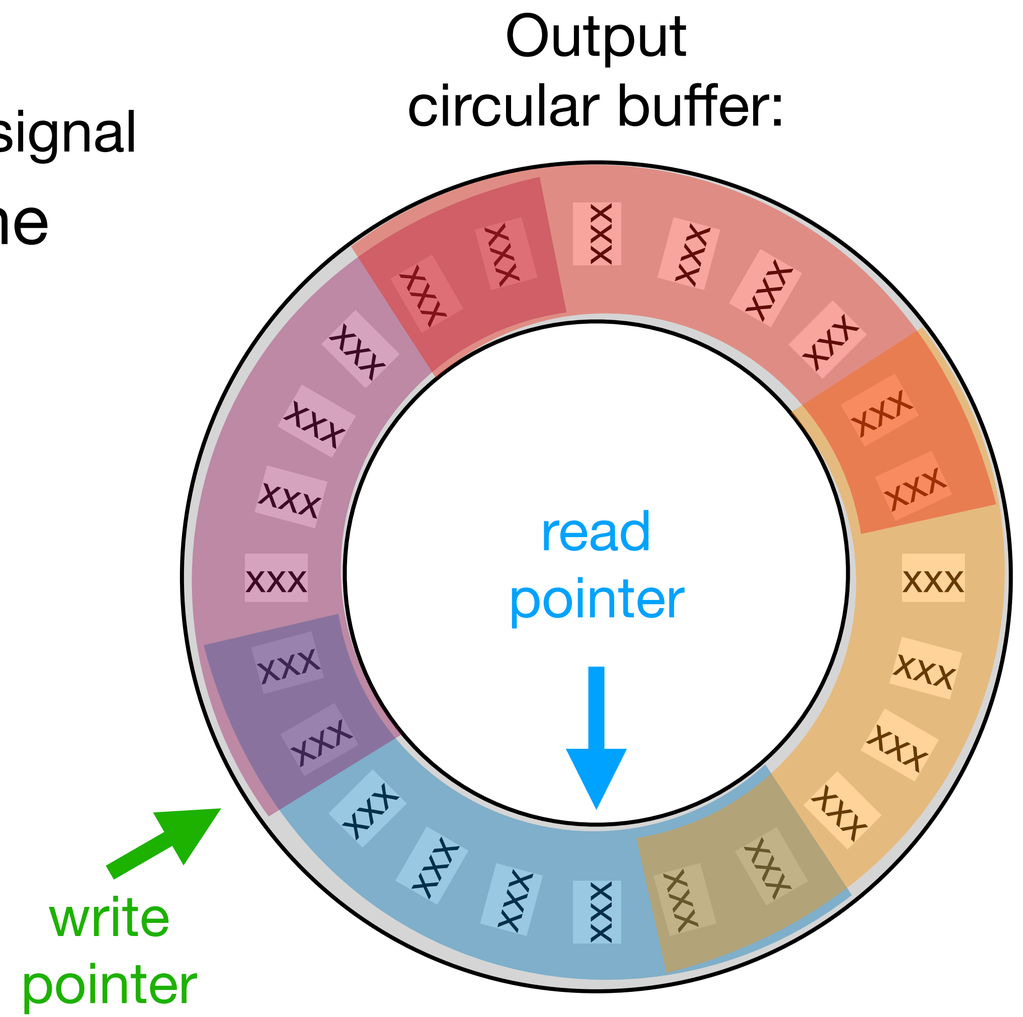


Note: these output samples are the sum of parts of two different blocks

- We need an intermediate output buffer (different from the system audio output) to hold the results of our block-based calculation
  - Each window, the new output should be added into that buffer
  - A sample can be sent from there to the system output as soon as all the blocks it depends on have been calculated (this implies a latency of at least 1 window)

Output circular buffer

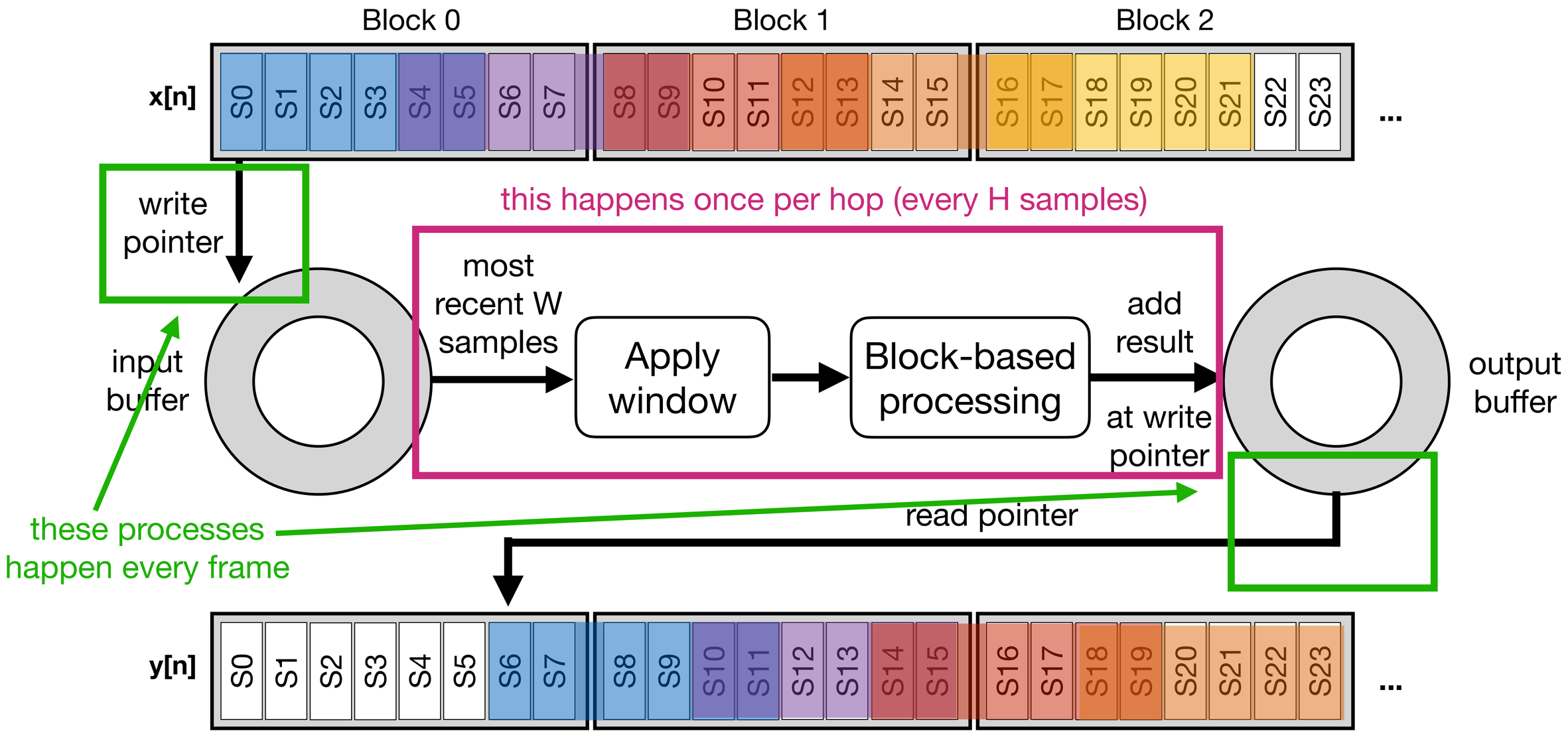
- We will use another circular buffer to gather output from the overlap-add
  - Not the same as the circular buffer of the input signal
- We need to keep track of two pointers in the circular output buffer
  - Write pointer: where does the next block get written once it's calculated?
    - This pointer moves in jumps of the hop size
  - Read pointer: where does `render()` get the next samples for the system output?
    - This pointer moves continuously frame by frame
  - The write pointer should always remain ahead of the read pointer
    - Otherwise we've run out of data!



Output circular buffer

- Necessary characteristics of the output buffer:
  - Should hold at least W + H samples
  - Where W is the window size, H is the hop size
  - One buffer per channel of signal (like before)
  - Needs an array, plus a write pointer (from FFT) and read pointer (to system output)
- Using the output buffer:
  - When you finish the inverse FFT, add the result into the output buffer, starting at the write pointer
    - To reiterate: don't replace the contents, add the new result onto the existing content of the buffer
  - In the main `for()` loop in `render()`, copy samples from the output buffer into the audio output at the read pointer
  - Then (important!) set the sample you just copied to 0
  - Finally, increment the read pointer

The full signal chain

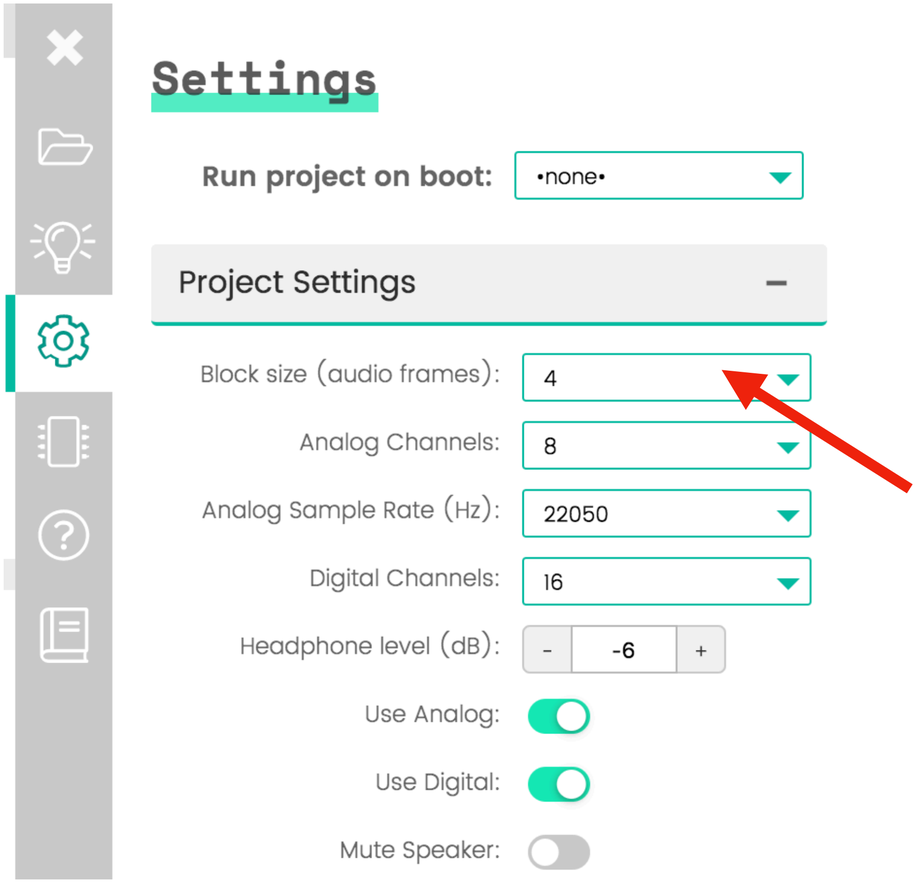


Overlap-add task

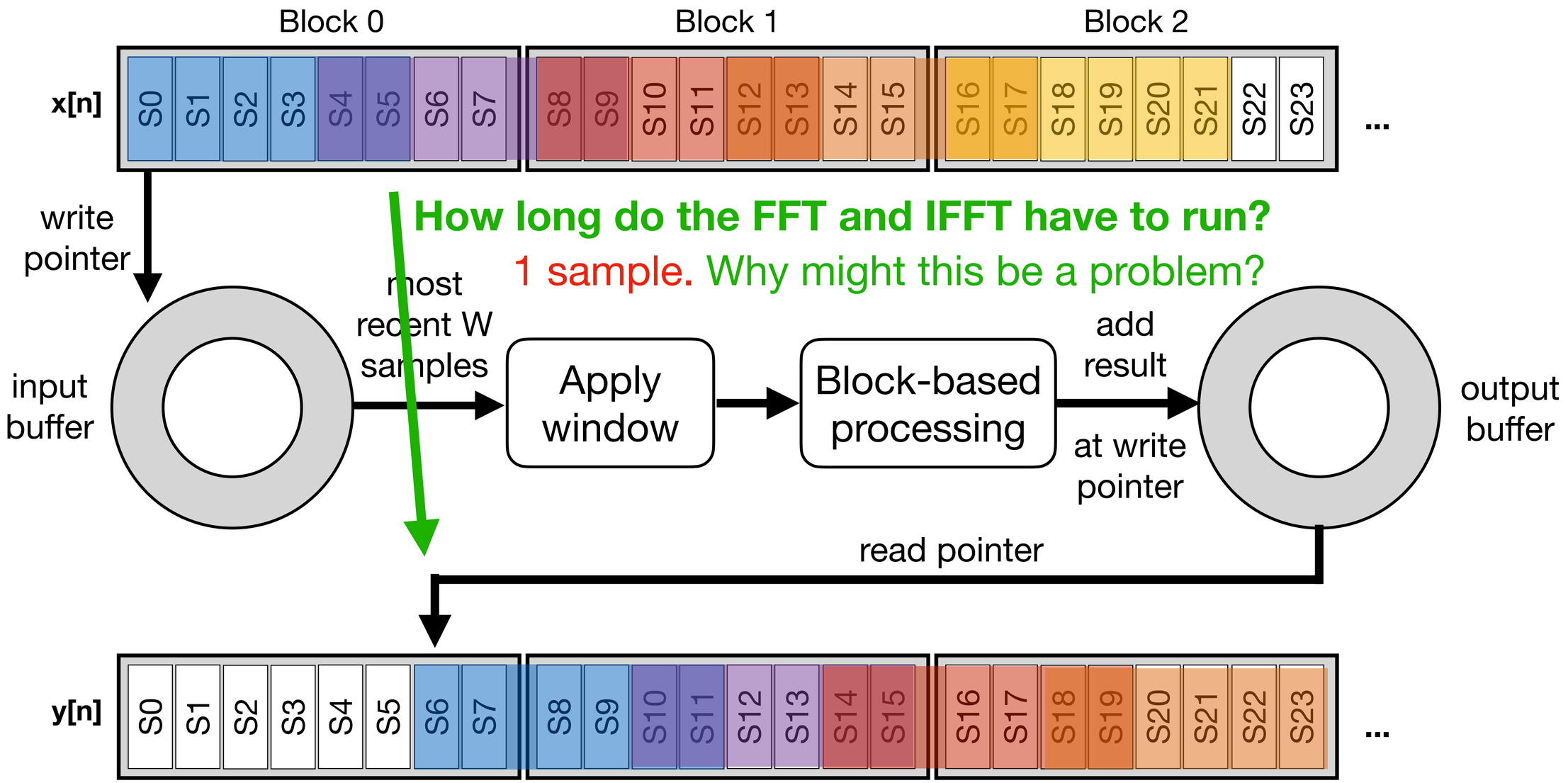
- Task: using fft-overlap-add project, implement the output circular buffer
  - Variables are already declared for you at the top of the file
  - Write pointer is for keeping track of where the next block is written
  - Read pointer keeps track of where to copy samples to the Bela audio output
  - Notice that `process_fft()` takes four arguments:
    - input buffer, input pointer, output buffer, output write pointer
  - Look at bottom of `process_fft()` to see how output is copied to the buffer
- Add your code in `render()`:
  - Copy samples from the output buffer to the audio output (use the variable `out`)
  - After you copy the output, set that sample in the output buffer to 0
  - Increment the output circular buffer pointers in the relevant places
    - Which one happens every frame? Which one advances once per hop?
- Once it works, try uncommenting the robotise code in `process_fft()`

Block-based processing performance

- In the Settings tab of the Bela IDE, try changing the block size to 4 samples
- What happens to the output? Why?

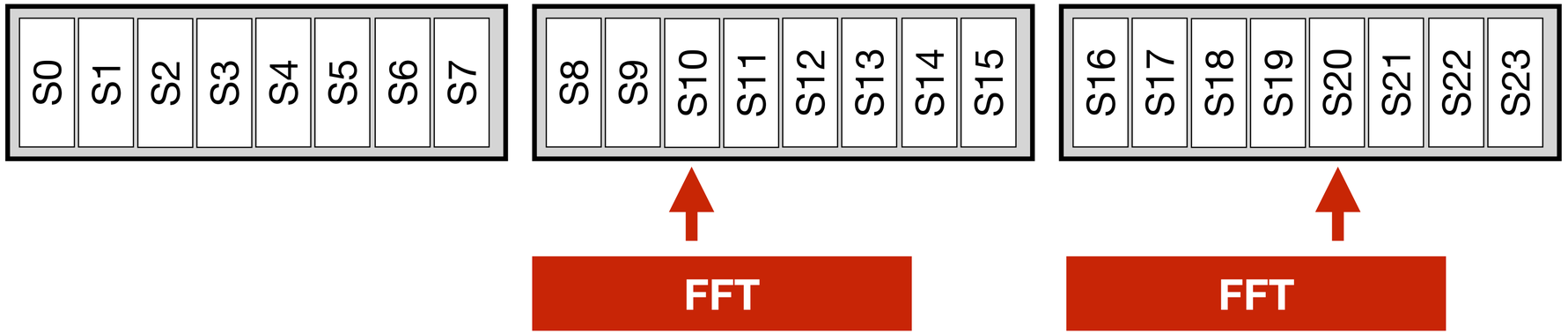


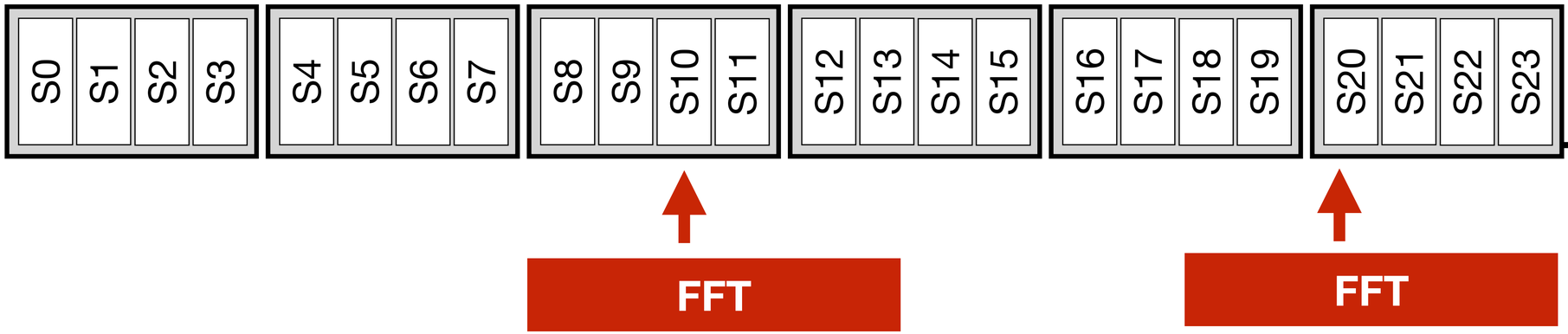
The full signal chain: performance



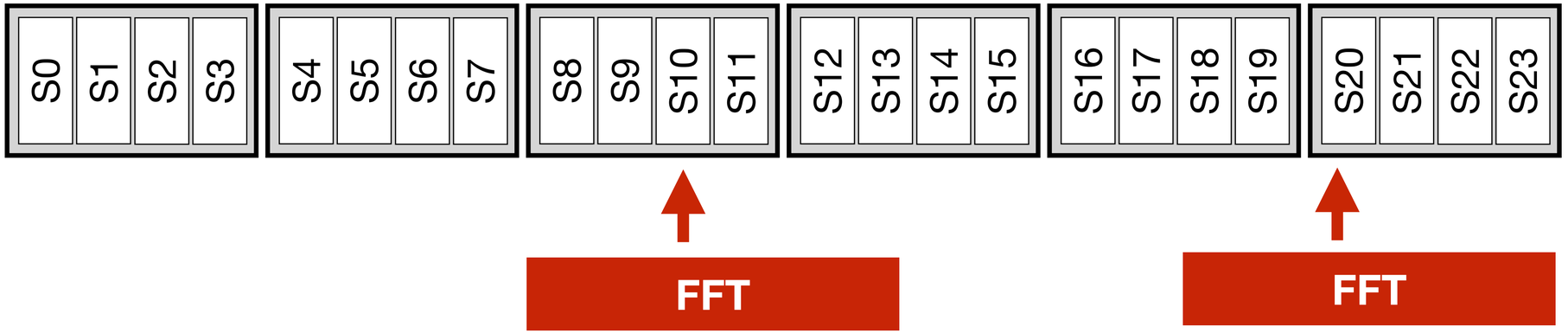
Intermittent CPU load

- We've been assuming the CPU load is roughly constant over time
  - Even if the average load is within limits, the instantaneous load might not be!
- Example: say we do an FFT every 10 samples
  - Let's assume only the FFT takes any time
  - With a buffer size of 8, we're fine:
  - Calculation takes less than one buffer length
  - But with a buffer size of 4, we're not...
  - Calculation takes longer than a call to `render()`



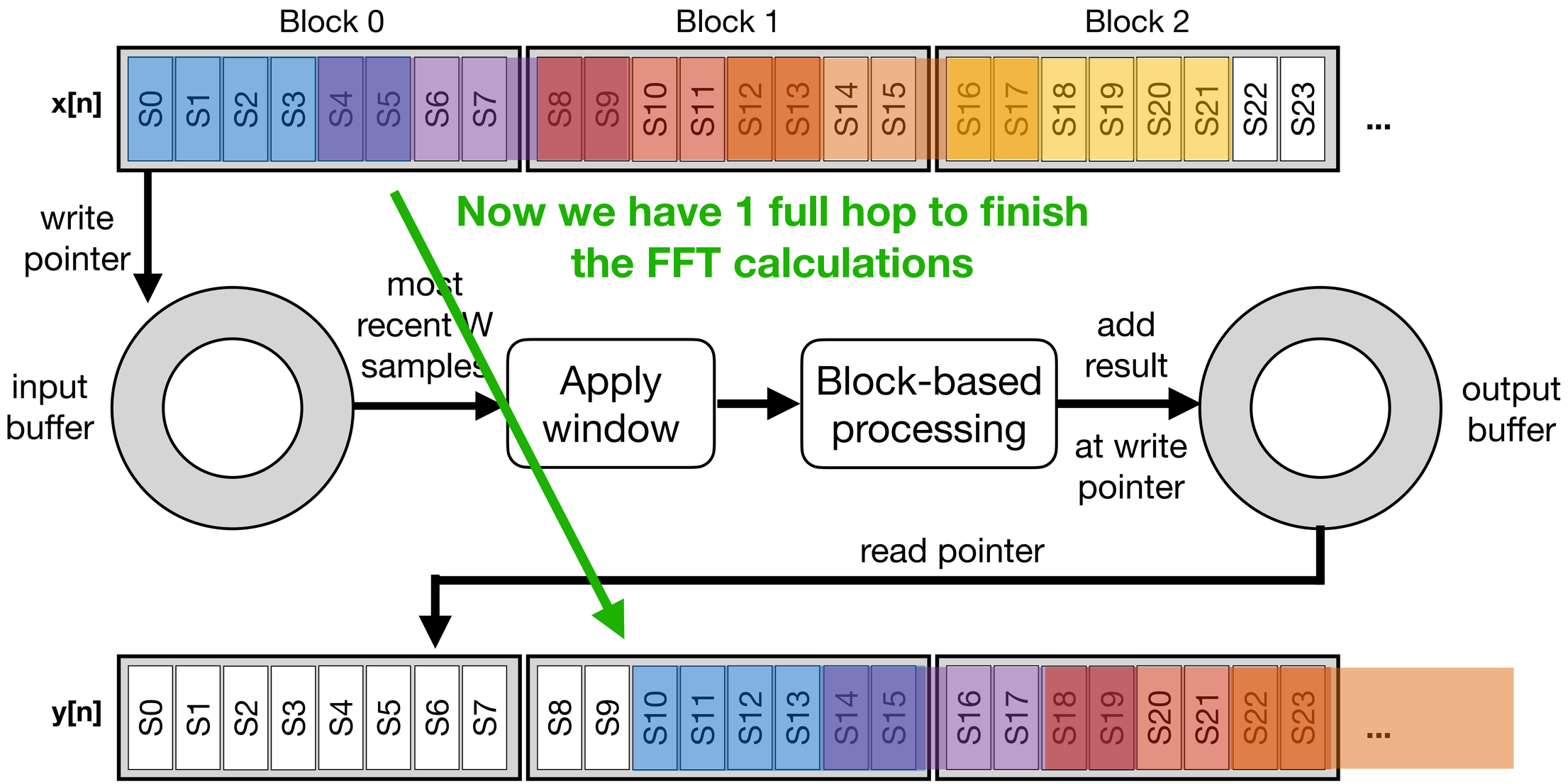


Handling intermittent load



- What can we do about occasional, heavy load?
  - Most calls to `render()` do almost nothing; some calls can't finish on time...
- We should add latency so the result doesn't need to be ready in 1 sample
  - But that still doesn't solve the problem if a calculation spans multiple calls to `render()`
- Solution: move the expensive process to another thread
  - Run the FFT on a thread with lower priority than `render()`
  - The FFT thread will use whatever processing is left over when `render()` is not running

The full signal chain: added latency

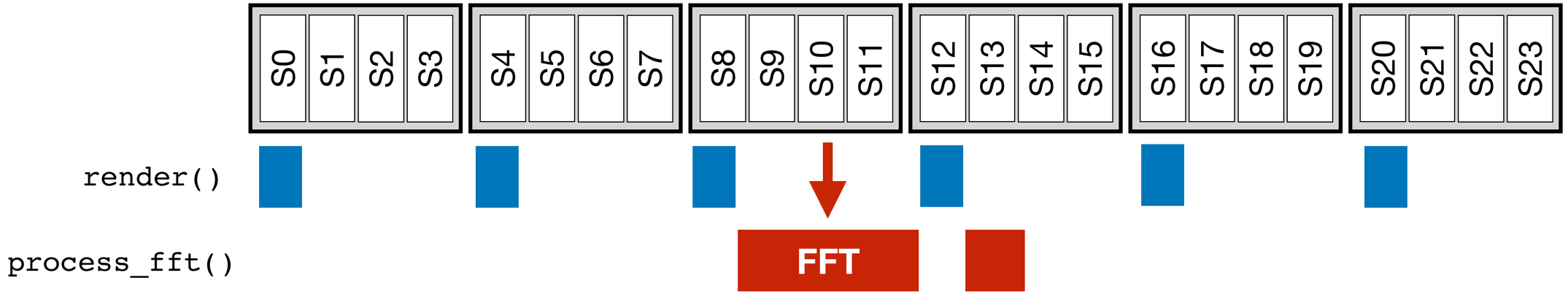


Multi-threaded processing

- Bela / Xenomai maintain a strict priority system for real-time threads
  - The thread with the highest priority always runs first, until it's finished
  - Then execution passes to the thread with the next highest priority
- Regular Linux threads (secondary mode) has a lower priority
  - These threads run when there are no Xenomai real-time threads available to run
  - The Linux scheduler decides which Linux threads will run at what time
    - This is a more complex question that we will not address in this class
- Upshot for block-based processing: use two threads
  - `render()`: high priority, runs every callback, finishes quickly
  - `process_fft()`: lower priority, runs occasionally, takes longer to finish
    - This will run whenever `render()` is not running
    - Still higher priority than the rest of Linux

Multi-threaded processing

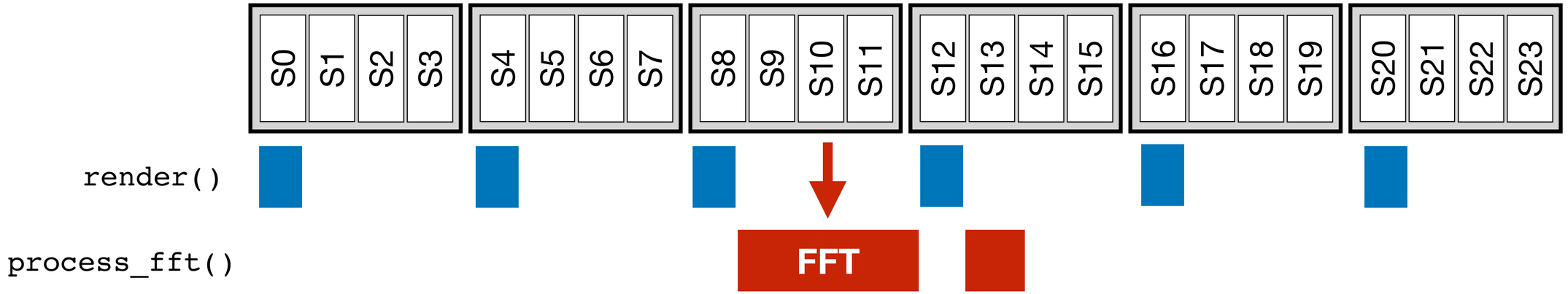
- With 2 threads, `render()` always runs first
  - The rest of the time is spent on the FFT, which could span multiple callbacks



- FFT thread should sleep and wait for a signal to run
  - `render()` should send the signal when a new hop has elapsed
  - `render()` also needs to tell the FFT thread where to find the data (locations of the buffers and pointers it needs)
  - FFT thread should run calculation, store the output in the buffer, then return

Multi-threaded processing

- With 2 threads, `render()` always runs first
  - The rest of the time is spent on the FFT, which could span multiple callbacks

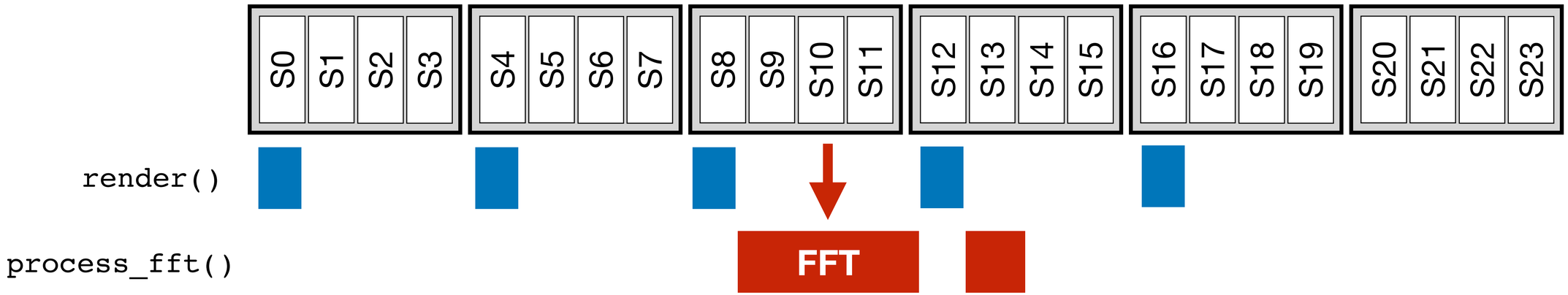


- `process_fft()` is allowed to be interrupted by `render()`
  - That means we need to be careful not to let `render()` modify any variables `process_fft()` was using
  - Main concern here is the input buffer pointer, which will keep changing before `process_fft()` is even allowed to run. Need to cache this value (save a copy that `render()` won't change).
  - Reverse isn't true: `render()` won't be interrupted because it has higher priority

Bela AuxiliaryTask API

- Bela provides a way of making Xenomai real-time threads (tasks)
- Each task requires four pieces of code:
  1. An `AuxiliaryTask` object
  2. A function to call which should be of type: `void function(void *argument);`
     - In practice you don't usually pass anything in the argument
  3. A call to `Bela_createAuxiliaryTask()` inside of `setup()`
     - This will pass in the object, specify which function should be called, and its real-time priority
     - `render()` has a priority of 95; valid levels are 0 to 99
  4. A call to `Bela_scheduleAuxiliaryTask()` whenever you want the task to run
     - Usually, but not always, inside of `render()`
     - Remember that the function won't actually be called until higher priority tasks finish
- You will generally use global variables to communicate between the tasks
  - For example, can't directly pass the `BelaContext` structure to an AuxiliaryTask

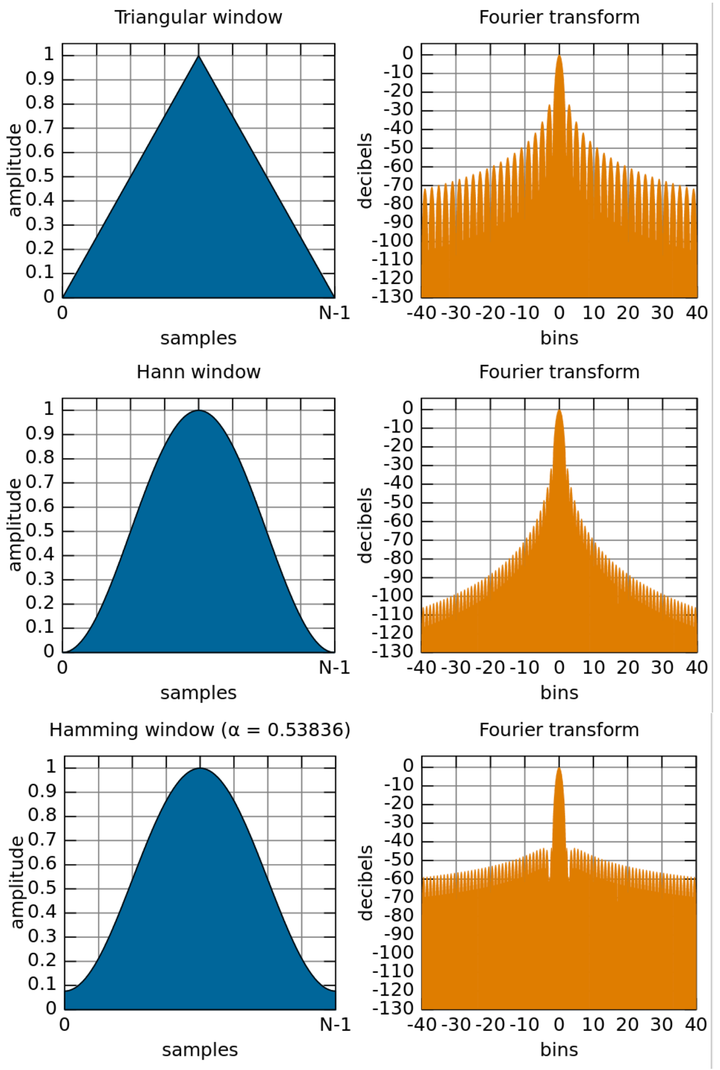
Threading task



- Task: using the fft-overlap-add-threads project
  - Convert the code to run the FFT in a second thread
  - Use the Bela `AuxiliaryTask` API: see `gFftTask` at top of the file
  - The initialisation is done for you in `setup()` - see `Bela_createAuxiliaryTask()`
  - You need to call `Bela_scheduleAuxiliaryTask()` each hop
  - Also finish implementing `process_fft_background()`

Next lecture: Phase vocoder, part 2

- Next time, we will look more at what we can do in the frequency domain
- Window functions: why and how
  - Constant Overlap-Add (COLA) criterion
  - Analysis and synthesis windows
- Reconstructing exact frequency from FFT bin phase
- Phase vocoder effects:
  - Robotisation (phase zeroing)
  - Whisperisation (phase randomisation)
  - Pitch shifting



[images: wikipedia]

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