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 20: Phase vocoder, part 3

What you'll learn today:

Modifying signals in the frequency domain
Converting frequency to phase
Analysis and synthesis windows

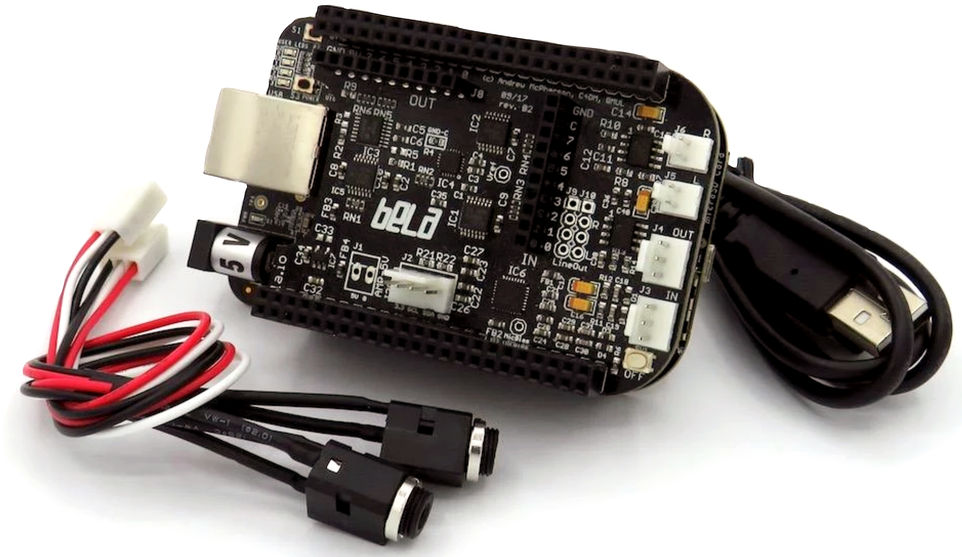
What you'll make today:

Pitch shifting, robotisation and whisperisation effects

Companion materials:

github.com/BelaPlatform/bela-online-course

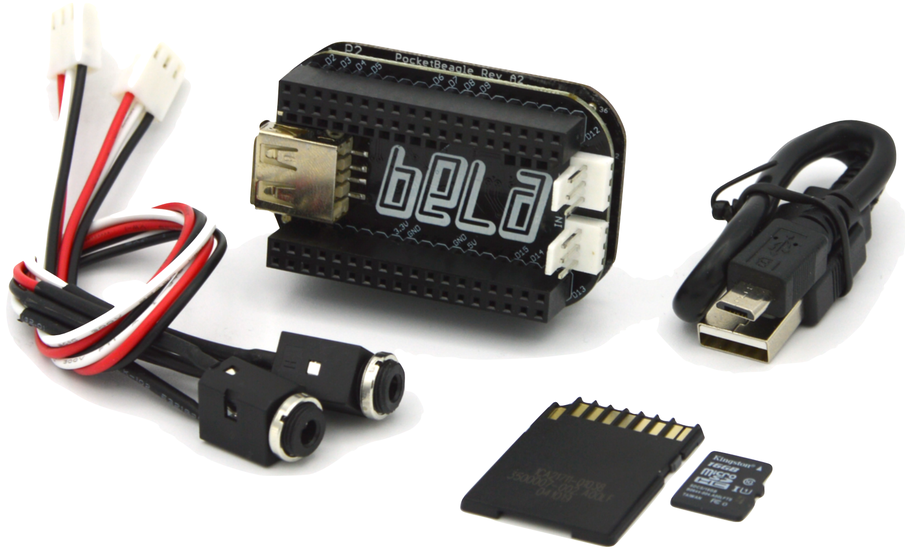
What you'll need



Bela Starter Kit

or

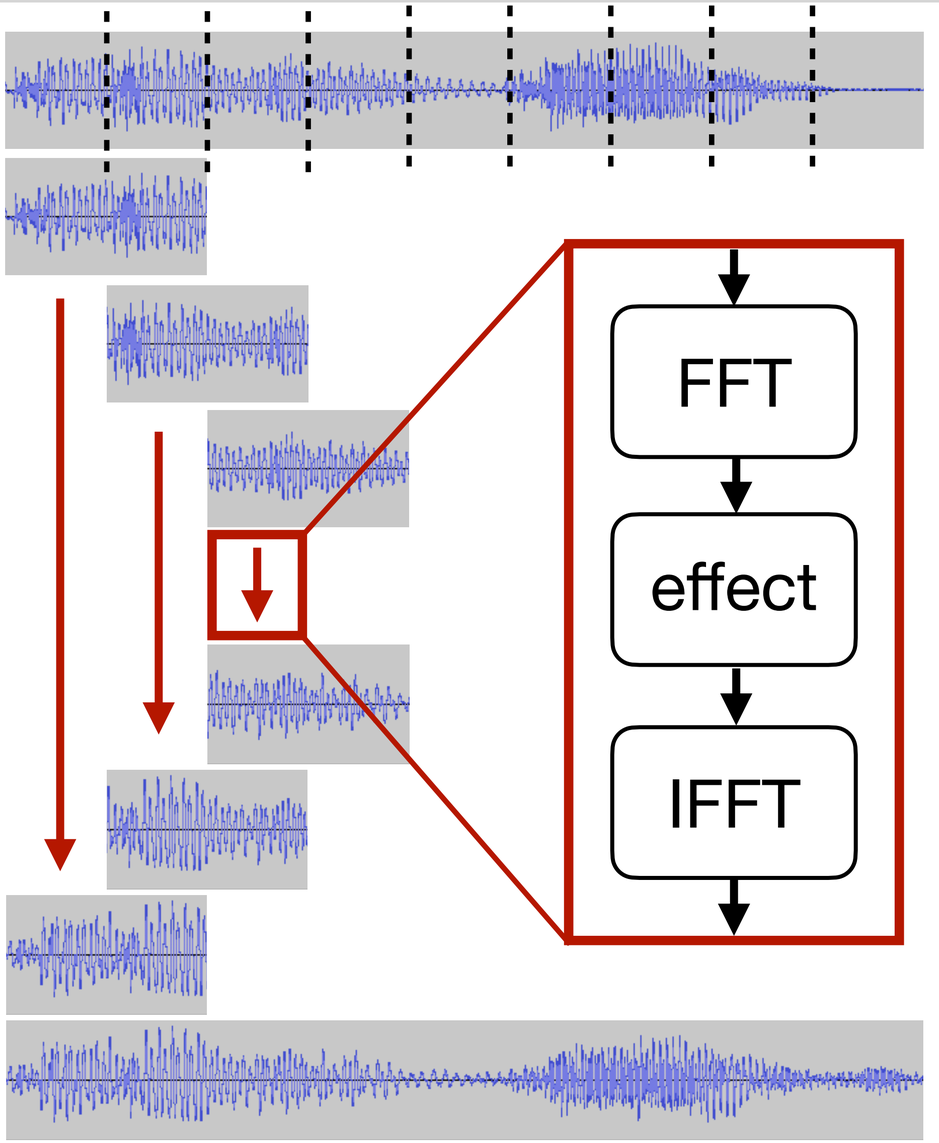
[shop.bela.io]



Bela Mini Starter Kit

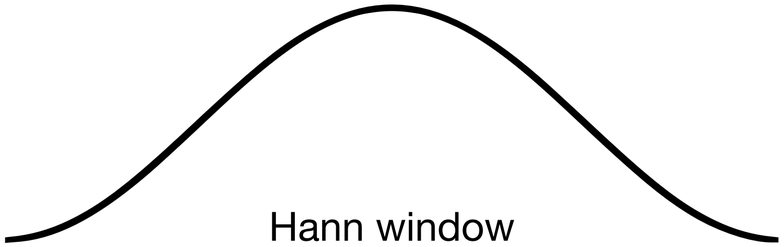
Overlap-Add

- 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**

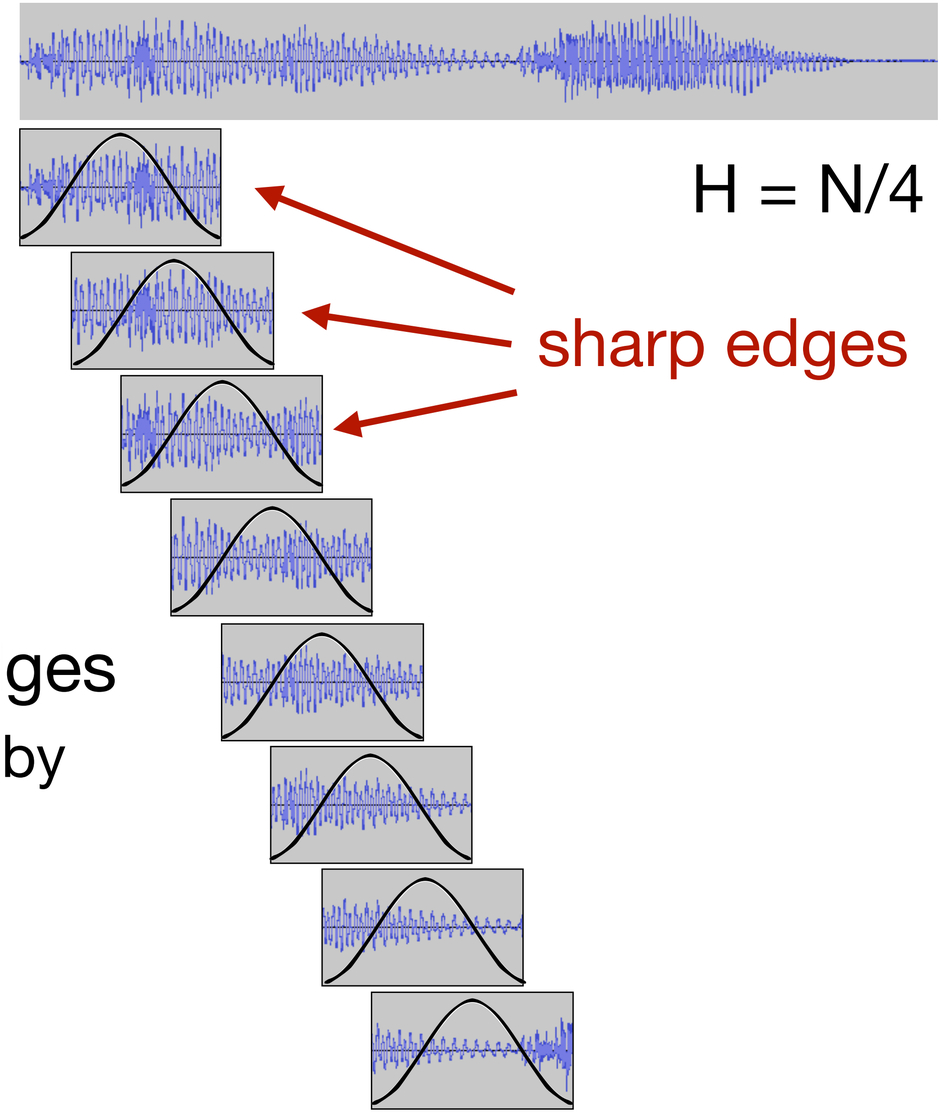


Windowing

- Overlap-Add process segments the signal into finite-length windows
  - These segments can have discontinuities at the edges (assuming 0 outside the window)
  - We might not notice this if we just reconstruct the signal with no frequency-domain processing
  - If we modify the signal, these sharp edges can lead to crackling or buzzing sounds
- We can use a window function to taper the edges
  - Pre-calculated shape that we multiply the segment by before FFT



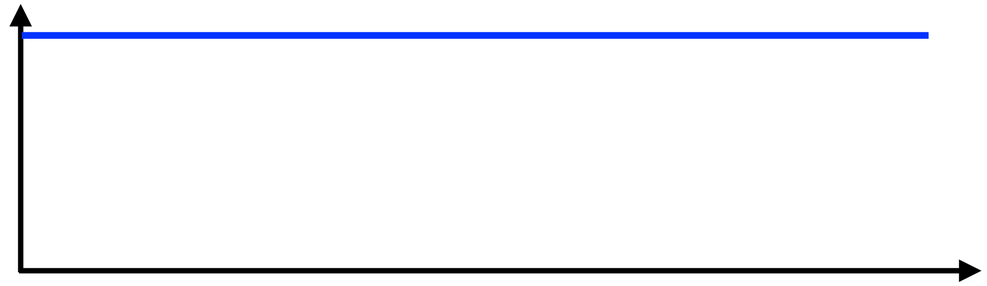
Hann window



Types of window function

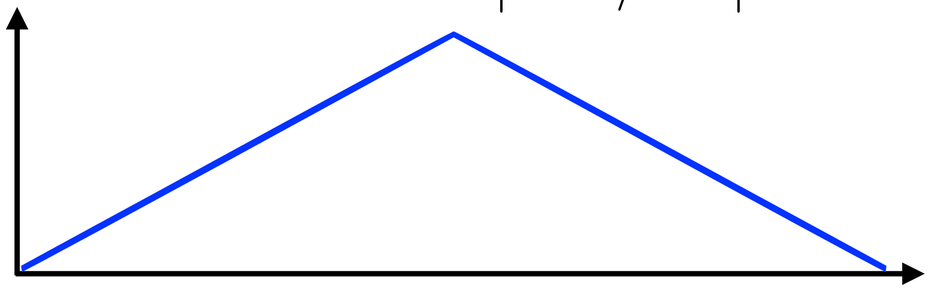
Rectangular

$$w[n] = 1 \quad \text{for } 0 \leq n < N$$



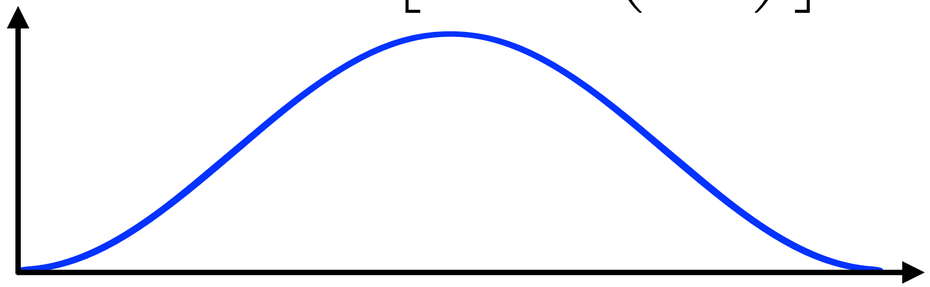
Triangular (Bartlett)

$$w[n] = 1 - \left| \frac{n - N/2}{N/2} \right|$$



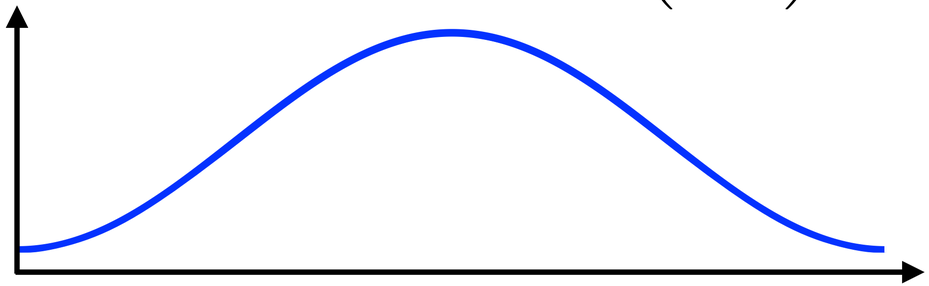
Hann

$$w[n] = 0.5 \left[1 - \cos\left(\frac{2\pi n}{N} \right) \right]$$



Hamming

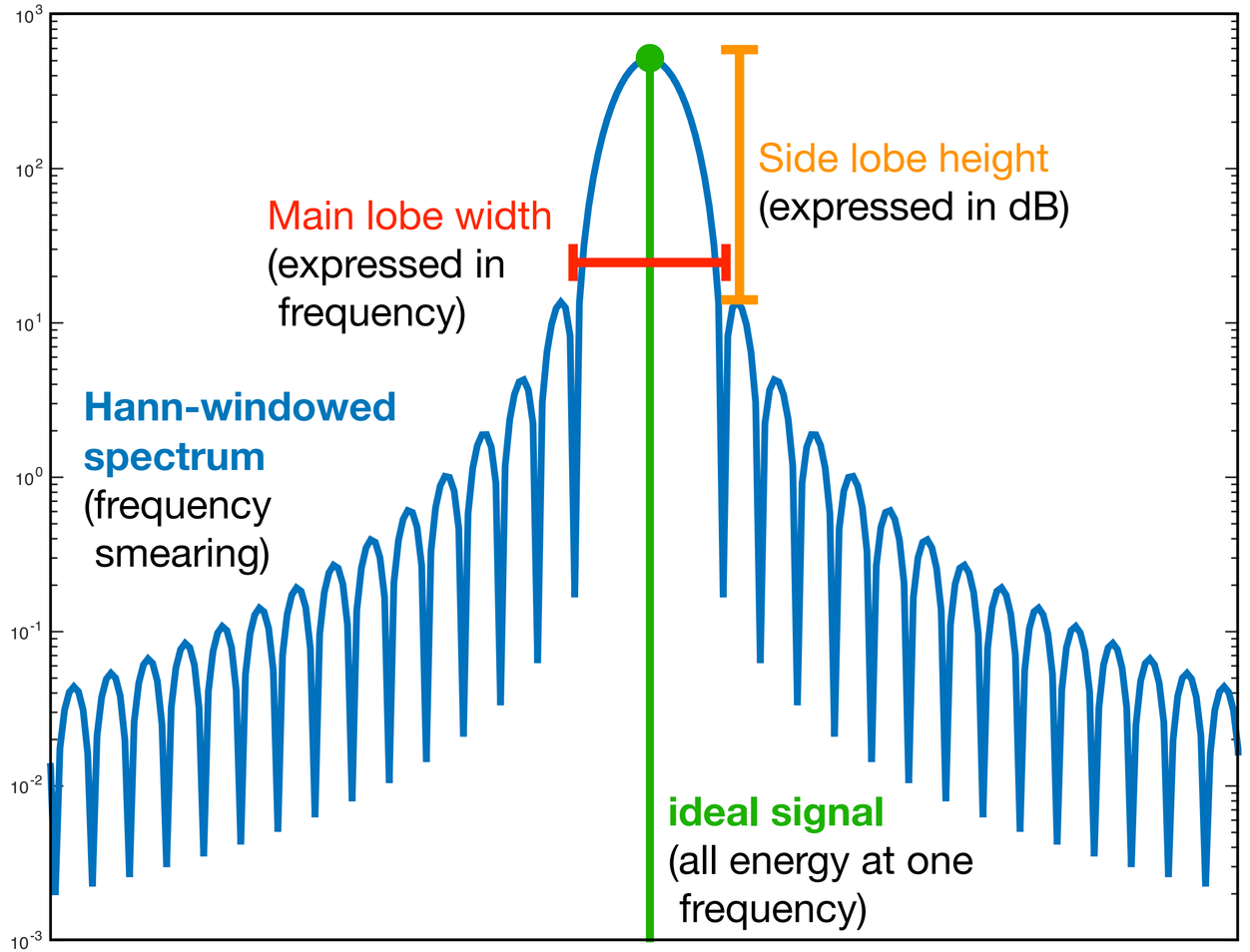
$$w[n] = \frac{25}{46} - \frac{21}{46} \cos\left(\frac{2\pi n}{N} \right)$$



And more types: Blackman, Kaiser, Flat Top, ...

Effects of windowing

- Windowing lets us isolate a particular time segment of our signal
- The cost is smearing of energy in the frequency domain
  - Energy appears at frequencies where it doesn't exist in the original signal
- Each frequency component of the signal appears as a series of lobes
  - Main lobe: a broad peak centred around the actual frequency
  - Side lobes: secondary peaks at other frequencies
- We typically want to minimise:
  - Main lobe width
  - Side lobe height

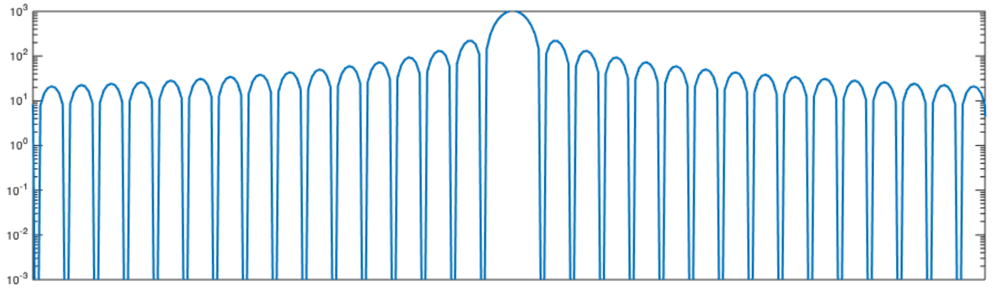


Window spectra

Rectangular

Main lobe width: 4π/N

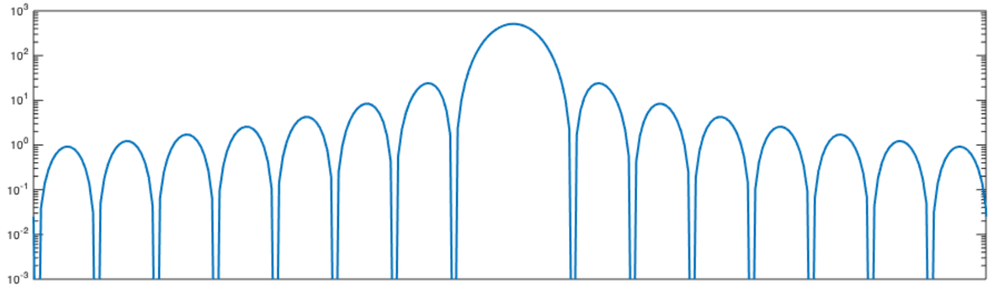
Side lobe height: -13dB



Triangular (Bartlett)

Main lobe width: 8π/N

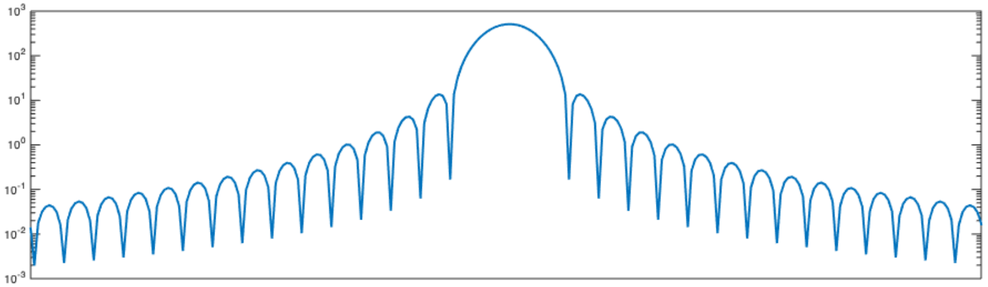
Side lobe height: -26.5dB



Hann

Main lobe width: 8π/N

Side lobe height: -31.5dB

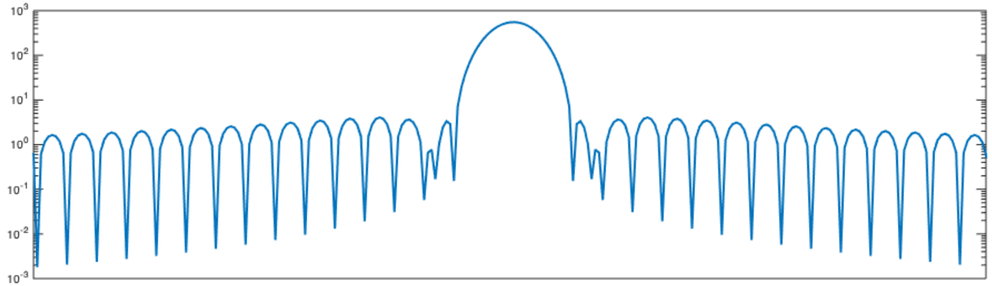


→ When in doubt: use a Hann window

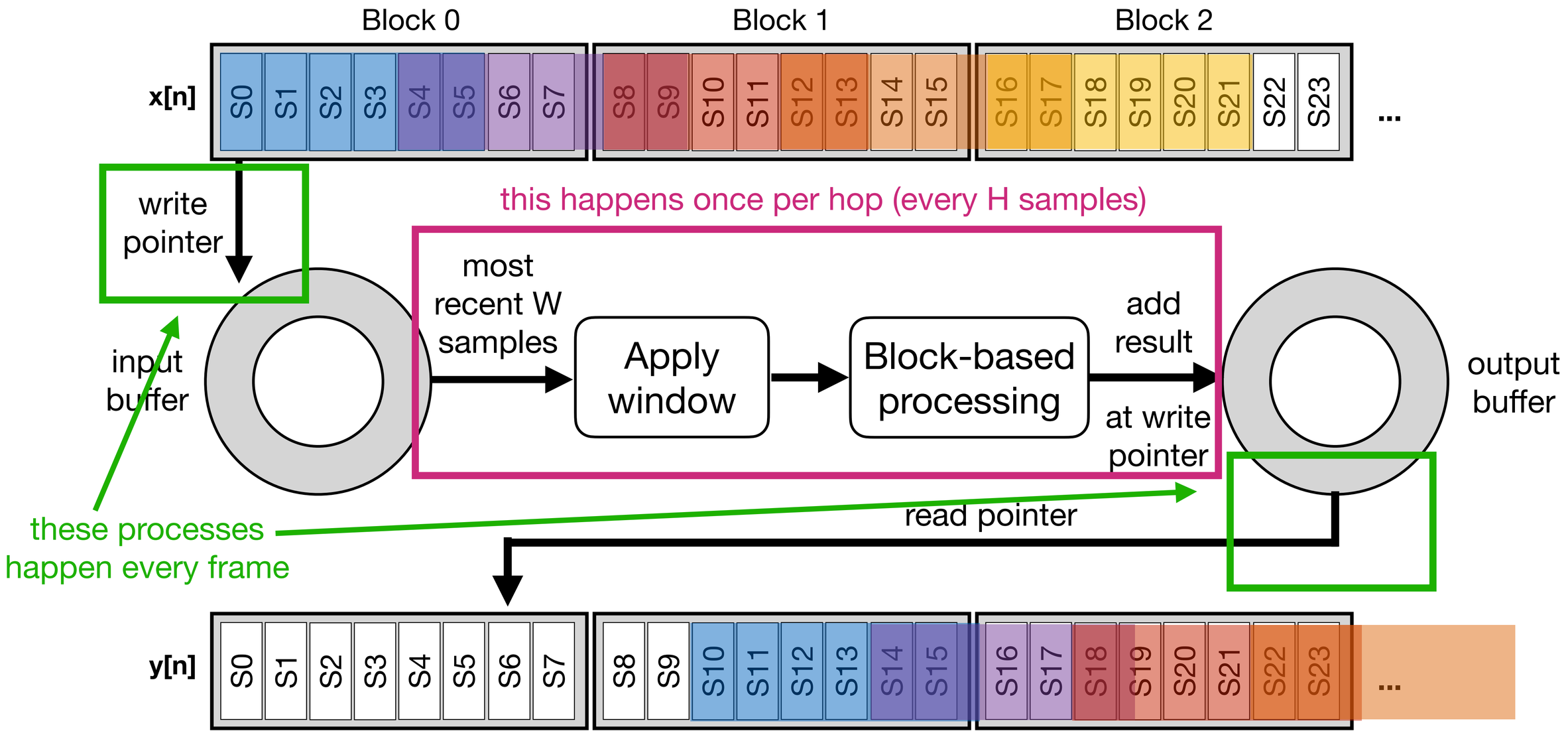
Hamming

Main lobe width: 8π/N

Side lobe height: -42.8dB

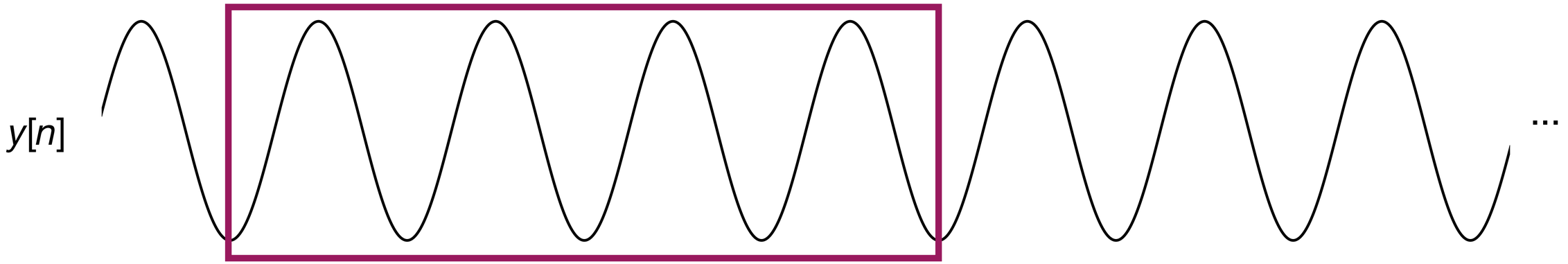


The full signal chain



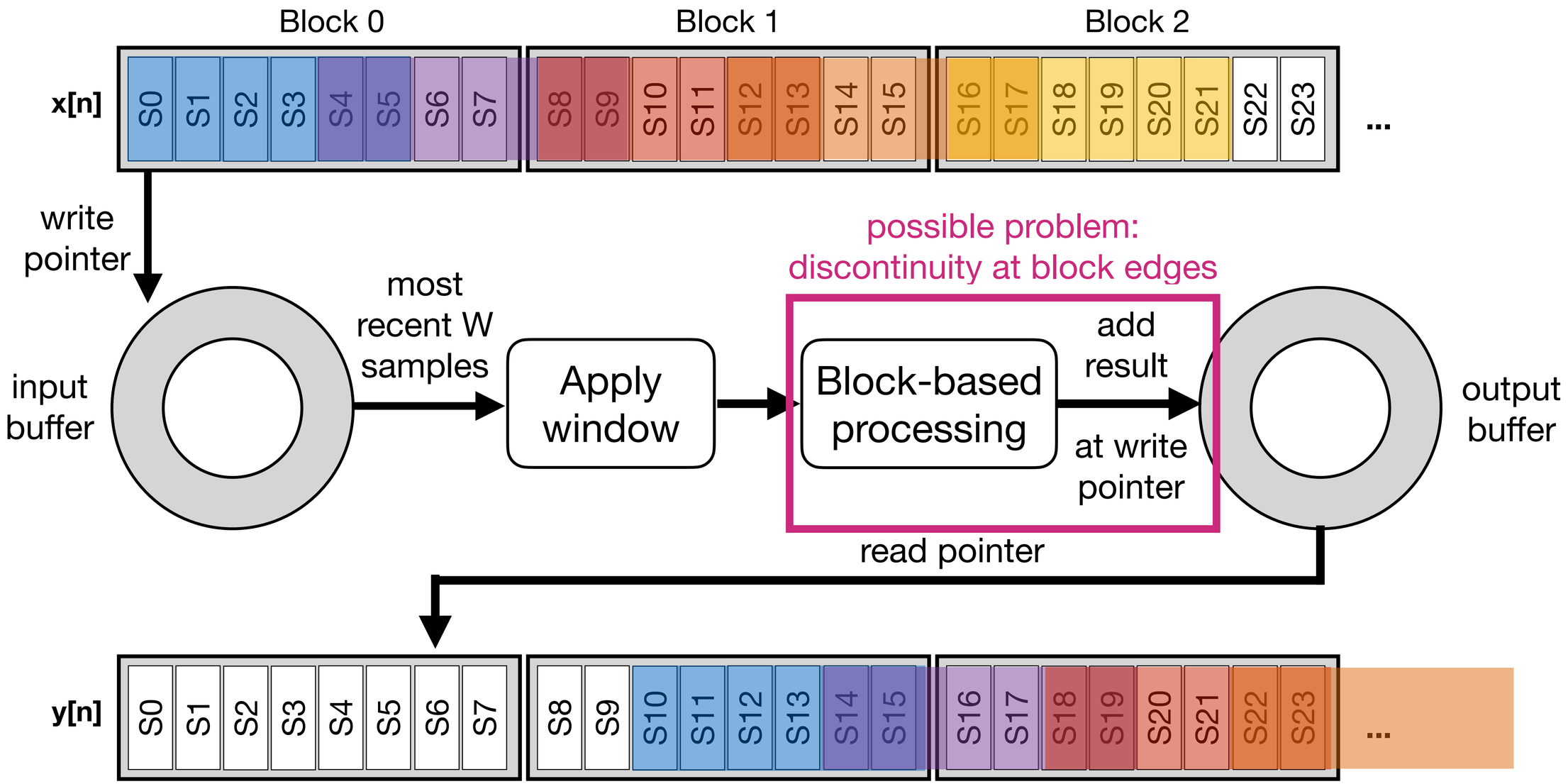
Robotisation

- A simple phase vocoder effect to generate robot-like voices
- At each hop, set the phase of every bin to 0
  - But preserve the magnitude of the bin, preserving the shape of the spectrum
- What does zeroing the phase do?
  - No phase change between hops
  - That means that hops always advance by an integer number of cycles

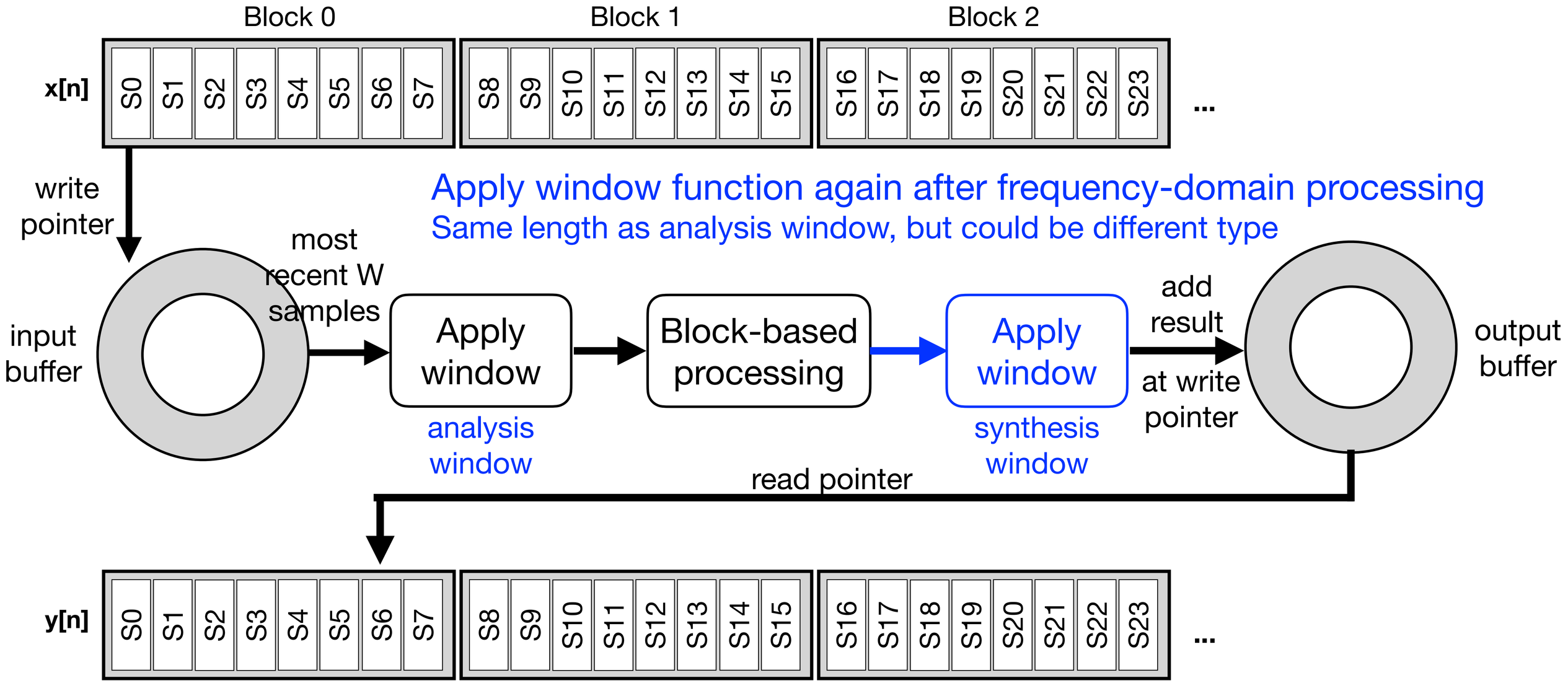


  - Thus the period of every bin must be an integer division of the hop size
    and the frequency of each bin must be a multiple of the the hop frequency f_s/H
  - So the robotisation effect imposes a constant pitch determined by the hop size

Review of signal chain

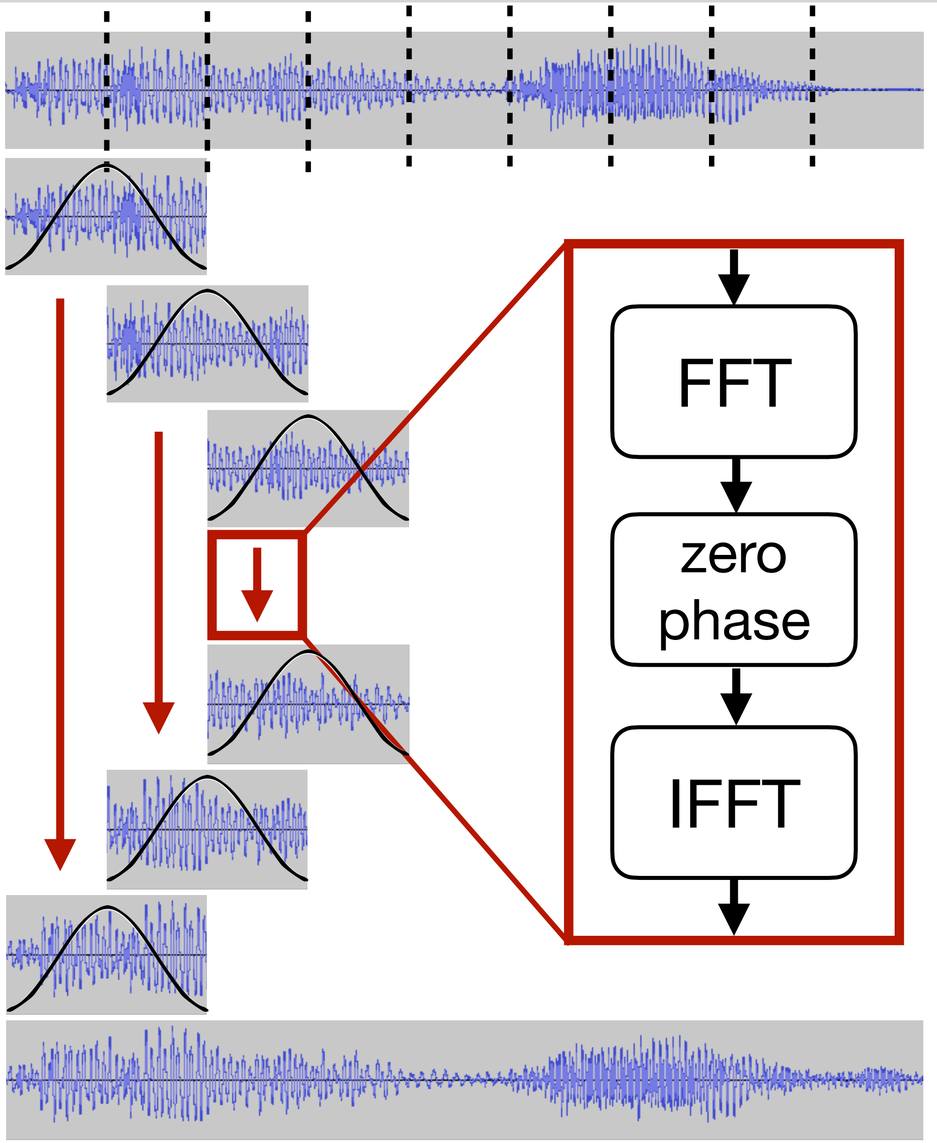


Analysis and synthesis windows



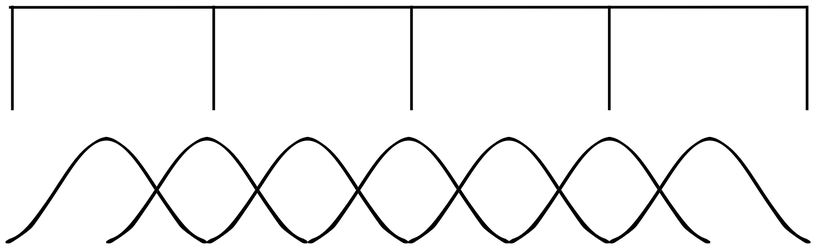
Robotisation

- For the cleanest effect, we need to apply both analysis and synthesis windows
  - Analysis window smoothes the edges of each block coming into the effect
  - However, changing the phase will scramble the shape of the block, so a synthesis window is also needed to smooth the edges again
- Task: in project fft-robotisation
  - Implement a Hann window for analysis and synthesis
  - You can calculate the window once in `setup()` and use the same array for analysis and synthesis
  - Multiply the block by the window in `process_fft()`
  - How does the window change the sound?
  - Next, make the hop size adjustable with the GUI



The COLA criterion

- Often, we want to exactly reconstruct a signal we windowed and transformed to frequency domain
  - i.e. when we do no processing in the frequency domain, we want to be able to get the same result back when done
- Exact reconstruction works for only certain combinations of window function and hop size
- Constant Overlap-Add (COLA) criterion:
  - Means that windows of length *W*, added together with a given hop size *H*, sum to a constant
  - For rectangular windows: *H* = *W*, *W*/2, *W*/3, W/4, ...
    - i.e. any integer division of the window size
  - For Hann, Hamming, Barlett: *H* = *W*/2, *W*/3, W/4, ...
    - i.e. divisions at most half the window size
  - Using Hann windows for analysis and synthesis: *H* = *W*/3, W/4, ...
    - Applying the window twice means it is effectively squared



Whisperisation

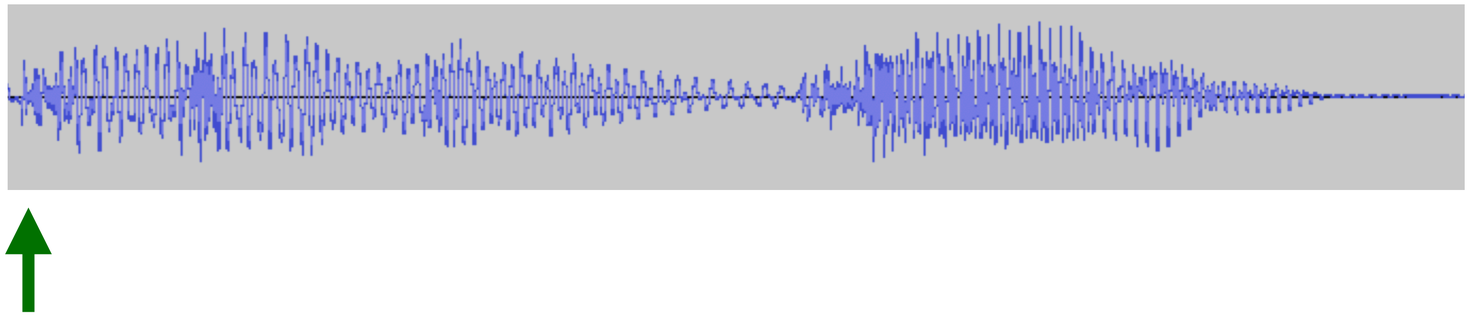
- A simple phase vocoder effect to obscure the pitch of a sound
  - On speech, it can produce a whisper-like effect
- At each hop, randomise the phase of each bin
  - This means the difference in phase between hops is also randomised
  - Therefore the frequency of the bins is also randomised (within a certain range)
- How much frequency variation can whisperisation introduce?
  - Phase difference between hops must be in the range ±π
  - How much difference is that? From earlier: $\omega[n] = \frac{\text{wrap}(\phi_r[n])}{H} + \omega_k$
  - So the frequency of each bin could differ by up to ±π/*H*
  - Smaller hop sizes allow more frequency variation
- As with robotisation, analysis and synthesis windows improve the effect
  - You might also find smaller FFT size produces a more convincing effect

Whisperisation task

- Task: in fft-robotisation code, change the code to whisperise the sound
  - In `process_fft()`, randomise the phase of each bin
    - Hint: to generate a random float between 0 and 1, use this expression: `(float)random() / (float)RAND_MAX`
  - Once you have the magnitude and phase, calculate real and imaginary components
    - We saw how to do this in Lecture 19
    - Use `cosf_neon()` and `sinf_neon()` for efficiency
  - Try different settings for window size and hop size

Pitch shifting

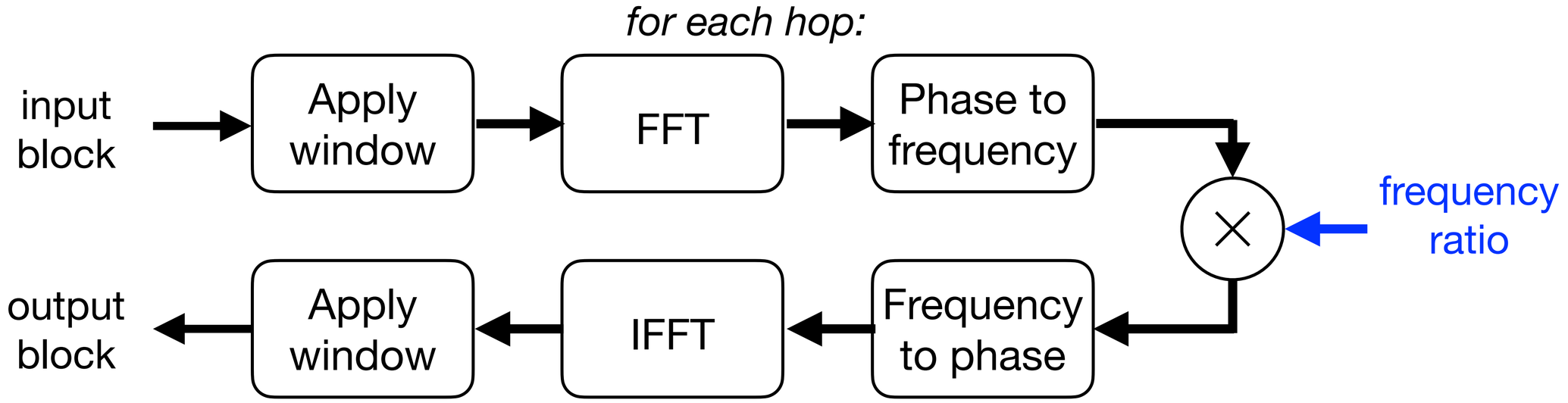
- When playing audio samples, playback speed and pitch are usually coupled
  - This is not a digital phenomenon: it's the same on turntables and audio tapes



- How, then, can we change the pitch of a sound without changing its speed?
- This cannot be done through filtering because the effect is nonlinear
  - Linear systems cannot create energy at frequencies that were not present in the input
  - However, for a pitch shifter, the output signal will have energy at different frequencies than the input signal
- Even though we can't use simple filters, the phase vocoder can help...

Pitch shifting

- We've seen that any signal can be represented as the sum of sinusoids
- The DFT gives us a snapshot of magnitudes and phases
  - From phase, we calculated precise frequencies for each bin
- Pitch shifting in the frequency domain just involves scaling each frequency
  - Multiply each frequency by a constant ratio: >1 raises the pitch, <1 lowers the pitch
  - Then we need to convert frequencies back to magnitudes and phases of each bin



Review: phase to frequency

- First we calculate the phase remainder: $\phi_r[n] = (\phi[n] - \phi[n-H]) - \frac{2\pi k H}{N}$

  measured **phase values** from bin *k* on **two successive hops**

  centre frequency of bin *k* multiplied by the **hop size**

- Then use remainder to calculate the deviation from the bin centre frequency:
  - We should wrap the phase to be between -π and π (i.e. the principal argument)

$$\omega[n] - \omega_k = \frac{\text{wrap}(\phi_r[n])}{H}$$

centre frequency of bin *k* (= 2π*k*/*N*)

- Finally, we can rearrange to get the exact frequency of our sine wave:

$$\omega[n] = \frac{\text{wrap}(\phi_r[n])}{H} + \omega_k$$

  - We could also calculate in terms of (fractional) FFT bins
    - Use the fact that bins are spaced 2π/*N* apart in frequency

fractional bin number of the sine wave

$$b[n] - k = \frac{\text{wrap}(\phi_r[n])N}{2\pi H} \longrightarrow b[n] = \frac{\text{wrap}(\phi_r[n])N}{2\pi H} + k$$

Frequency scaling

- Here we will talk about analysis and synthesis frequencies
  - The analysis frequency $\omega_a[n]$ is the frequency we calculated from our input signal
  - The synthesis frequency is a scaled version of it: $\omega_s[n] = R\omega_a[n]$
    - Where *R* is the pitch shift ratio
  - The notation involving *n* is to remind us that the frequencies change over time
  - Remember, there is one frequency for each bin of the FFT
- Problem: $\omega_s[n]$ might be in a different FFT bin than $\omega_a[n]$
  - Need to find the bin with the closest centre frequency
- Let's look at the analysis in terms of (fractional) bins: $b_a[n] = \frac{\text{wrap}(\phi_r[n])N}{2\pi H} + k$
  - The above was calculated for FFT bin *k*
  - Pitch shifting gives us a new fractional bin index $b_s[n] = Rb_a[n]$
  - Round to the nearest bin to figure out where this frequency goes: $k' = \text{floor}(Rk + 0.5)$
    - For $R > 1$, energy is likely to move into higher bins, and the reverse for $R < 1$

Frequency to phase

- Based on the bin frequency, we calculate a phase difference since last hop
  - Not an absolute phase; we need to remember each bin's phase from last hop
- The process of calculating the bin phase is the reverse of the analysis
  - The frequency shift gave us a bin *k'* and a fractional bin number $b_s[n]$
  - Use it to calculate the phase remainder: $\phi_{rs}[n] = \dfrac{2\pi H(b_s[n] - k')}{N}$
    - How much more or less will the phase advance at the synthesis frequency vs. the bin centre frequency
  - Then consider the expected phase shift based on the bin centre frequency, and add these to the phase of bin *k'* at the last hop

$$\phi_s[n] = \text{wrap}\left(\phi_s[n-H] + \phi_{rs}[n] + \frac{2\pi k' H}{N}\right)$$

  - Wrap the phase to be in the ±π range
- Meanwhile, the magnitude of the bin doesn't change in the pitch shift process

Frequency to phase: code

- We are calculating the phase difference from one hop to the next
  - Therefore, we need a global variable to hold the synthesis phases from the last hop
  - Since we need to hold a value for each FFT bin, this variable should be an array
- Each hop, we add the calculated phase difference to the output phase
  - Magnitude stays the same
  - Then we convert back to real and imaginary values, as before:
  - Real component: $a = M\cos(\phi)$ Imaginary component: $b = M\sin(\phi)$
  - The functions `cosf_neon()` and `sinf_neon()` from the Ne10 library are faster than `sin()` and `cos()`
- Another efficiency trick: exploit the symmetry of the FFT
  - We only need to calculate up to (and including) bin *N*/2
  - Above this, the bins are conjugate symmetric:
    - Same real component; negative imaginary component

Pitch shift task

- Task: in the project fft-pitchshift, implement the pitch shifter
  - The overlap-add process is implemented for you (see Lecture 18 for explanation)
  - Your code goes in `process_fft()`
  - The analysis step (phase to frequency) is already done from Lecture 19
- You should implement the synthesis step:
  1. Multiply each frequency component by the pitch shift value: $b_s[n] = Rb_a[n]$
  2. Check which bin the component belongs in: $k' = \text{floor}(Rk + 0.5)$
  3. Calculate the phase of the new bins:
  4. Convert magnitude and phase to real and imaginary components

$$\phi_{rs}[n] = \frac{2\pi H(b_s[n] - k')}{N}$$

$$\phi_s[n] = \text{wrap}\left(\phi_s[n - H] + \phi_{rs}[n] + \frac{2\pi k' H}{N}\right)$$

- To test:
  - Change pitch shift value in the GUI
  - If you like, try a real-time audio input using `audioRead()`

Pitch shift limitations

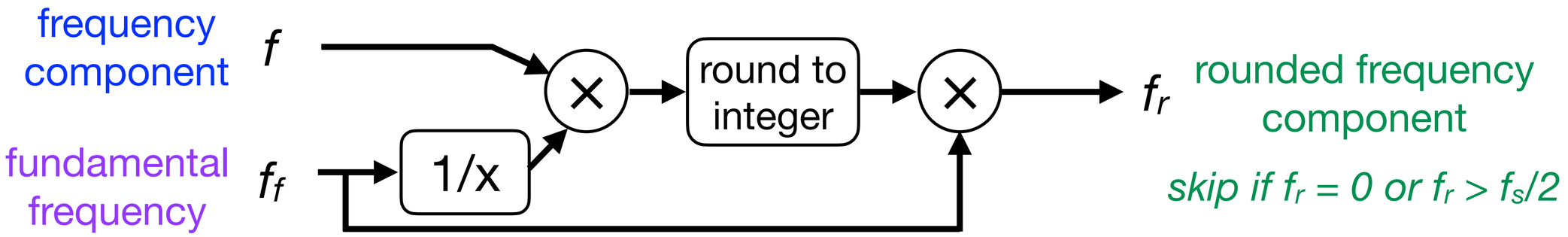
- Limitations of the pitch shift algorithm
  - When shifting pitch downward, components which started out in different bins might end up in the same bin (since frequencies get closer together when $R < 1$)
  - Artefacts from windowing (e.g. the main lobe and side lobes) will also be shifted
  - If the original signal has frequency components spaced more closely than the FFT bins, they may not get resolved and shifted correctly
  - No global preservation of phase which distorts transient events
    - "Phasiness" is a classic problem with phase vocoder effects
  - Latency: the longer the FFT, the better the pitch resolution but the longer the latency
- There is a big market for high-quality pitch shifting effects
  - Antares Auto-Tune, introduced in 1997, has become ubiquitous in pop music
  - Many different pitch shifting guitar pedals on the market
  - These effects often involve many careful optimisations for sound quality and efficiency

Revisiting robotisation

- The robotisation effect worked by setting all phases to 0 each hop
- The fundamental frequency of the effect was f_s/H
  - f_s is the sample rate, H is the hop size
  - For example, $f_s = 44100$, $H = 256$ yields a base frequency of 172Hz
- Limitation 1: H must be an integer
  - This means the frequency can't be finely tuned
- Limitation 2: H typically doesn't meet the COLA criterion
  - It's not an integer division of the FFT size
  - Non-COLA isn't a big deal since we don't need exact reconstruction, but it is a limitation
- Limitation 3: can't do other frequency transformations at the same time
  - Other phase vocoder effects will assume a more regular hop size
  - It's computationally expensive to run two simultaneous overlap-add FFT processes

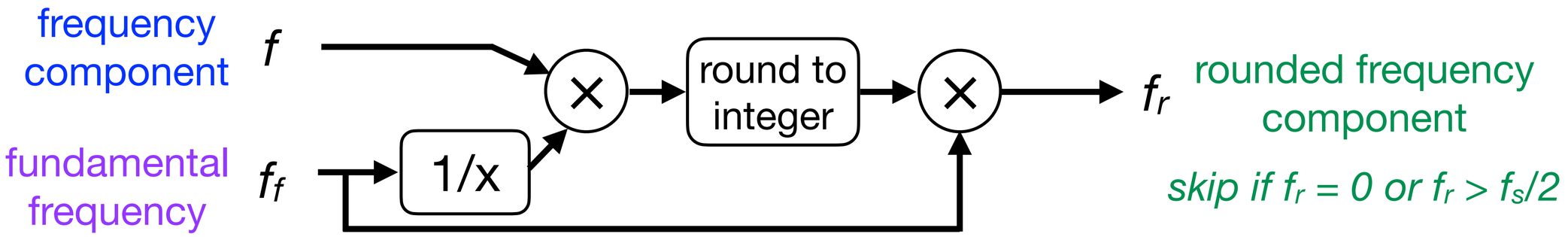
Revisiting robotisation

- How else could we achieve the robotisation effect?
  - Manipulate the frequency components with the phase vocoder
- The pitch shifter used a magnitude-frequency representation of the signal
  - The hard part was calculating that representation (and getting back again)
  - The pitch shift itself was simple: multiply by a constant
- Let's use the same magnitude-frequency representation for robotisation
  - Round each frequency to the nearest integer multiple of the desired fundamental
  - Keep the magnitudes of each frequency component constant



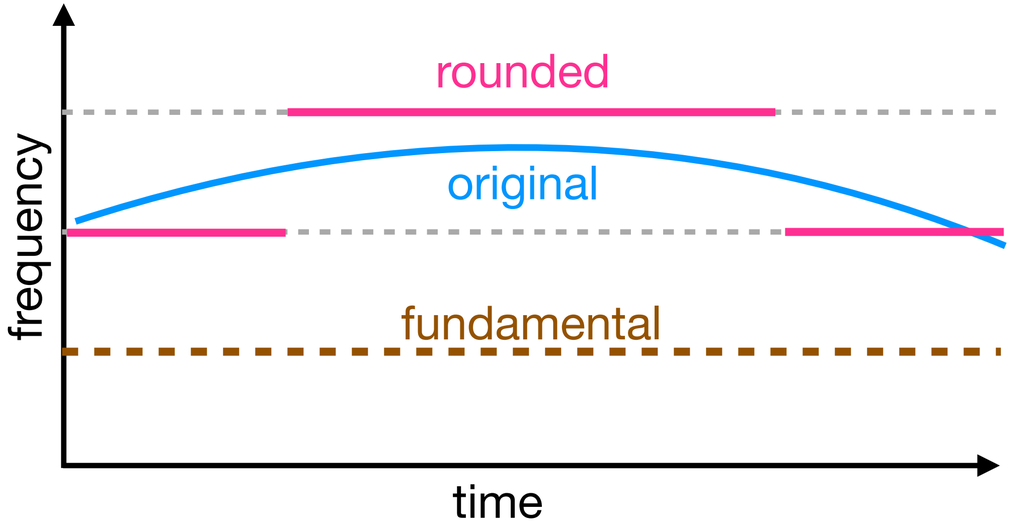
Revisiting robotisation: task

- Our previous projects expressed frequency in fractional bin numbers
- Here, it may be easier to think about actual (discrete time) frequency
  - In discrete time, frequency ranges from 0 to 2π (corresponding to the sample rate)
  - We need to rescale our target fundamental frequency by multiplying by $2\pi/f_s$
- Task: in project fft-robotisation-v2
  - Round the frequency of each bin to a multiple of the target fundamental
  - Remember that the rounded frequency might be in a different FFT bin
  - Keep magnitudes of each frequency component the same



Improving the robotisation effect

- You might have heard some artefacts
  - Frequency components glide smoothly up and down in the original signal
  - Now, they jump from one harmonic to another
  - The result is a trill-like effect



- Alternative approach: interpolation
  - For each frequency component, generate energy at harmonics above and below
  - Scale the magnitude of the upper and lower harmonics depending on which harmonic is closer to the original frequency
  - This is a form of linear interpolation (see Lecture 3 for more)
- Task: in fft-robotisation-v2, implement linear interpolation
  - Create two synthesis frequencies for each analysis frequency
  - Will often need to mix multiple components together in one FFT bin

Phase vocoder: more ideas

- Ways to improve the quality of the effects:
  - When mixing components into an FFT bin, use RMS amplitude rather than linear addition
  - When mixing components into an FFT bin, find the weighted average of the frequencies
  - Implement peak tracking to reduce some of the effects of window lobes
- Other effects using the phase vocoder:
  - Combine a frequency detector with a pitch shift to implement auto-tuning
  - Cross-synthesis: combine frequencies from one signal with magnitudes from another
  - Noise reduction: suppress frequency components below a threshold or matching a known profile
- As the effects become more complex, CPU becomes a limitation
  - Try larger hop sizes (which may also require larger FFTs)
  - Pre-calculate frequently used constants (e.g. bin frequencies)
  - Look for other optimisations to reduce the number of multiplies per hop
  - The Bela NEON library (`math_neon`) contains optimised versions of common functions

Keep in touch!

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More resources and contact info at:

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