Full Name: ______________________________

EEL 4750 / EEE 5502 (Fall 2021) – HW #11 **Due: 4:00 PM ET, Nov. 22, 2021**

Concept Questions 11

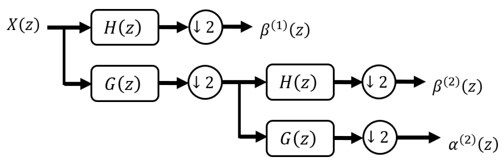
Question #1: Complete the Canvas questions here: `https://ufl.instructure.com/courses/437179/assignments/4812588`

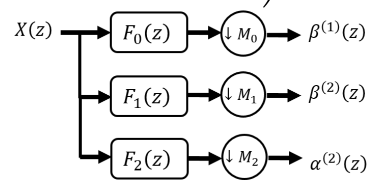
Theory Questions 11

Question #1: Let high pass filter $H(z)$ and low pass filter $G(z)$ be defined by frequency responses:

$$G(\omega) = \sqrt{2}\left(\sum_{k=-\infty}^{\infty} u(\omega + \frac{\pi}{2} - 2\pi k) - u(\omega - \frac{\pi}{2} - 2\pi k)\right)$$

$$H(\omega) = \sqrt{2}\left(\sum_{k=-\infty}^{\infty} u(\omega + \frac{\pi}{2} - \pi - 2\pi k) - u(\omega - \frac{\pi}{2} - \pi - 2\pi k)\right)$$

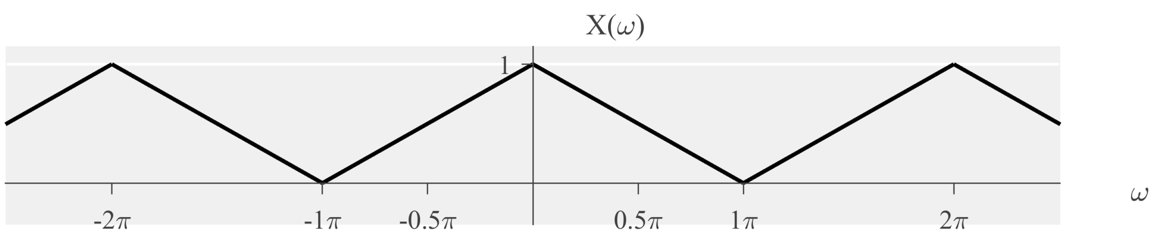




(a) Consider a 3-channel wavelet bank. Use the Noble identities to simplify the analysis wavelet bank and represent it as a filter bank. Determine M_0, M_1, and M_2.

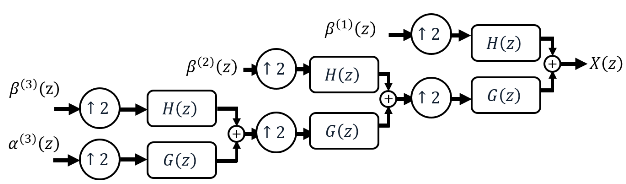
(b) Sketch the resulting $|F_0(\omega)|$, $|F_1(\omega)|$, and $|F_2(\omega)|$.

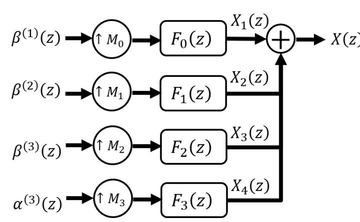
(c) Sketch $|\beta^{(1)}(\omega)|$, $|\beta^{(2)}(\omega)|$, and $|\alpha^{(2)}(\omega)|$ for input frequency-domain signal below



Question #2: Consider the Haar wavelets

$$H(z) = (1/\sqrt{2})[1 - z^{-1}] \qquad , \qquad G(z) = (1/\sqrt{2})[1 + z^{-1}]$$





(a) Use the Noble identities to simplify the synthesis wavelet bank (left) diagram and represent it as a filter bank (right). Determine M_0, M_1, M_2, and M_3.

(b) Sketch the resulting impulse responses, $f_0[n]$, $f_1[n]$, $f_2[n]$, and $f_3[n]$.

(c) Let the input signals be

$$\beta^{(1)}(z) = (1/2)z^{-2} \qquad , \qquad \beta^{(2)}(z) = (1/\sqrt{2})z^{-1}$$
$$\beta^{(3)}(z) = 0 \qquad , \qquad \alpha^{(3)}(z) = 1$$

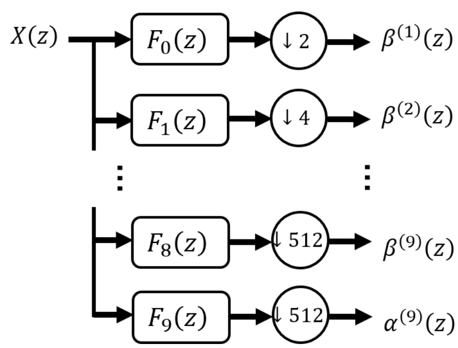
Compute the time-domain output of the inverse discrete wavelet transform $x[n]$

Implementation Questions 11

Question #1: *(Discrete Wavelet Transform)* The Fourier transform uses sinusoids as its basis while wavelet transform (depending on the choice) uses different basis for multi-resolution analysis. This question will create a wavelet basis.

(a) Design a function `f = wb_filters(h,g,M)` that takes two filter impulse responses `h` and `g` for a discrete wavelet tree and generates a matrix of filters `F` for the equivalent filter bank.

The matrix `f` should have dimensions of `(P-1)*2^(M-1)+(P-1)-1` by `M`, where `P` is the length of each filter `h` and `g` (assume they are the same length) and `M` is the number of channels in the filter bank (shown below). You may need to trim zeros from the end of your impulse responses to fit these dimensions.



(b) Use your function in (a) to generate the equivalent $M = 10$ filters for an initial Haar wavelet,

$$h[n] = (\delta[n] - \delta[n-1])/\sqrt{2}$$

$$g[n] = (\delta[n] + \delta[n-1])/\sqrt{2}$$

(c) Define a function `v = wb_analysis(x,f)` that computes the coefficients of the wavelet analysis bank. This function should be similar to `v = fb_analysis(x,h)` from the last assignment (the solution function is provided). A significant difference is that `v` must be a cell array rather than a matrix since each output channel has different output dimensions.[1]

(d) Define a function `y = wb_synthesis(v,fr)` that synthesizes a signal from its wavelet coefficients. This function should be similar to `y = fb_synthesis(v,g)` from the last assignment (the solution function is provided).

(e) Load `zoqfotpik.wav` (audio from Ur-Quan Masters again) to get the signal `x` and compute its wavelet coefficients `v` using `wb_analysis` and the filters defined in (b).

(f) Remove all but one channel by the setting all wavelet coefficients to zero except `v{2}`.

(g) Reconstruct the modified audio using `wb_synthesis`.

[1] Note that this is a very inefficient way to code a discrete wavelet transform, but it is much more conceptually straightforward and easier to code than fast approaches.