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Course Title: EC516 DSP

Implementing a polyphase uniform DFT filterbank

Objective:

To study the implementation of a 2 channel DFT filter bank with h[n] = [1,1,1,1] in order to appreciate its computational efficiency over implementing separate filters.

Procedure:

Step 1: Deriving a 2 channel DFT filter bank

We know that h[n] = [1,1,1,1] and since M = 2:

$H_0(z) = E_0(z^2) + z^{-1}E_1(z^2)$

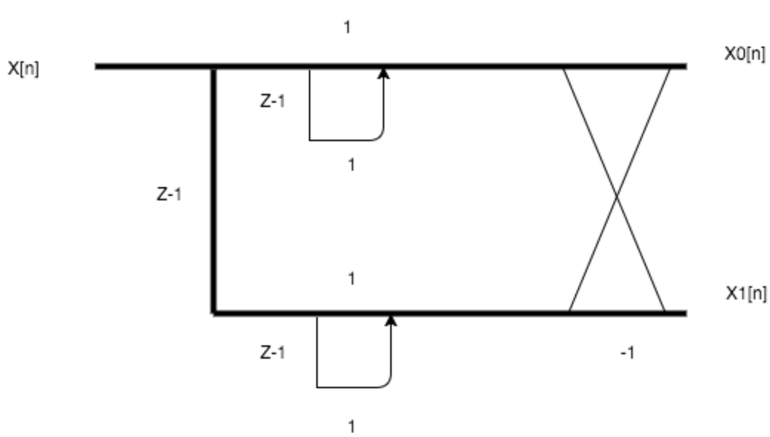
$e_0[n] = h_0[2n]$

$e_1[n] = h_0[2n+1]$

$E_0(z^2) = 1 + z^{-2}$

$E_1(z^2) = 1 + z^{-2}$

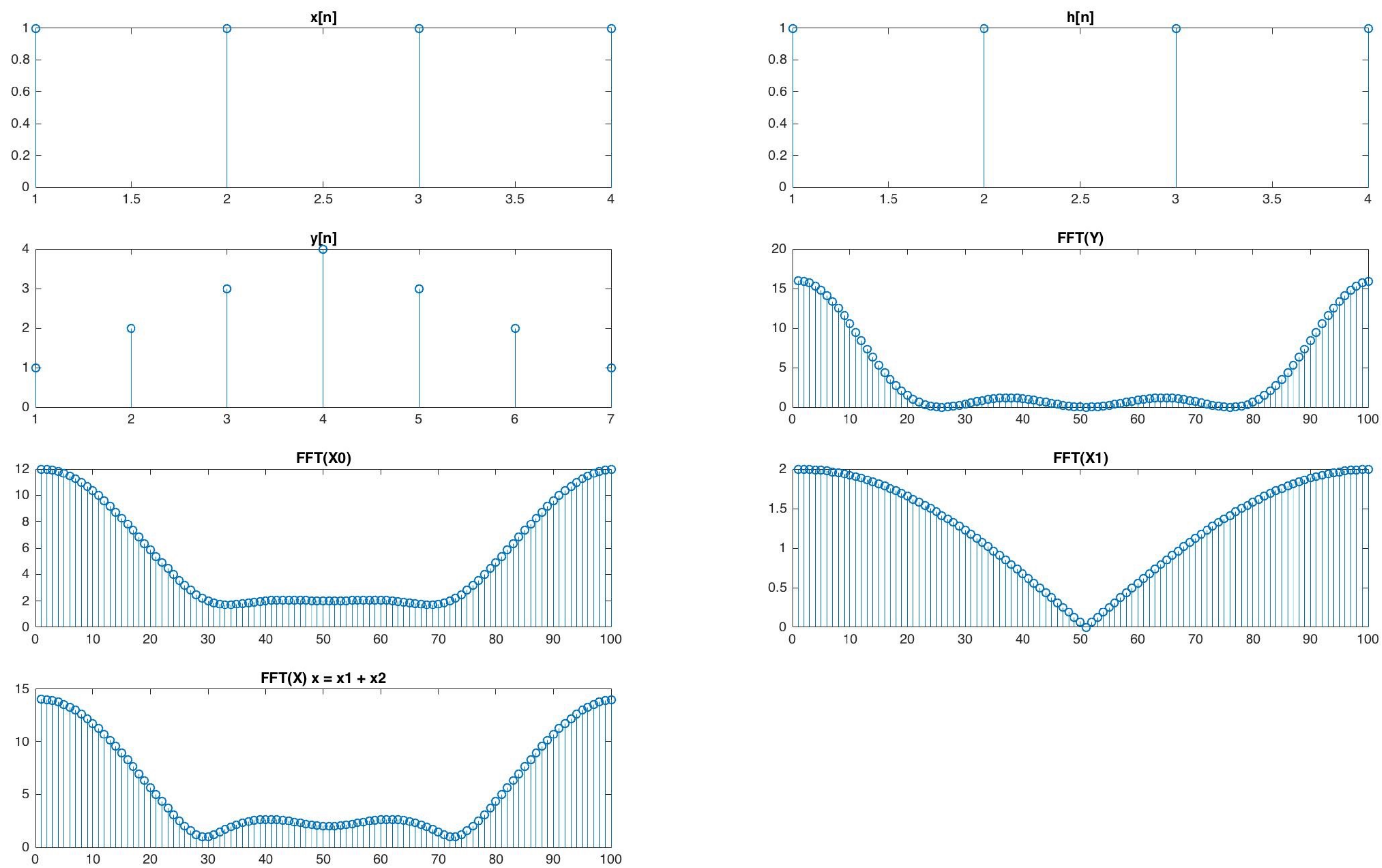
$H_0(z) = 1 + z^{-2} + z^{-1}(1 + z^{-2})$



2 channel uniform DFT filterbank implementation

Step 2: Implementing the filter

I first created the signal h[n] in matlab and x[n] = [1,1,1,1] as the test signal and computed the convolution y after which i implemented the filters E0 and E1. A delay filter z was also created. The outputs at Node 0 and Node 1 were computed (right before the cross in the flow diagram towards X0 and X1 respectively.) With these values X0[n] and X1[n] were found. The following figure shows my results:



As we can see, the result of the convolution between the signal and the filter is a triangular function which appears to be a sinc in the frequency domain. Adding the outputs from the filter bank x = x0 + x1 and computing the DFT of x, we observe that it is nearly a perfect reconstruction of the original signal in the frequency domain.

Conclusions:

The 2 channel polyphase implementation of the DFT filter bank was successful and the output X0 and X1 was computed efficiently by reusing filter elements through this implementation.