T-61.3010 Digital Signal Processing and Filtering

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The problems marked with [**Pxx**] are from the course exercise material (Spring 2009), where **Pxx** refers to the problem.

In the end of this session **you should know**: (a) how DFT-4 is computed with an efficient FFT algorithm, (b) what upsampling and downsampling mean both in time and frequency domain.

1. [**P74**] Using radix-2 DIT FFT algorithm with modified butterfly computational module compute discrete Fourier transform for the sequence $x[n] = \{\underline{2}, 3, 5, -1\}$ (*Mitra 2Ed Sec. 8.3.2, p. 538 / 3Ed Sec. 11.3.2, p. 610*). The equation pair on rth level (*Mitra 2Ed Eq. 8.42a, 8.42c, p. 543 / 3Ed Eq. 11.45a, 11.45c, p. 614*)

$$\begin{aligned}\Psi_{r+1}[\alpha] &= \Psi_r[\alpha] + W_N^l \Psi_r[\beta] \\ \Psi_{r+1}[\beta] &= \Psi_r[\alpha] - W_N^l \Psi_r[\beta]\end{aligned}$$

2. [**P84**] Consider the multirate system shown in Figure 1 where $H_0(z)$, $H_1(z)$, and $H_2(z)$ are ideal lowpass, bandpass, and highpass filters, respectively, with frequency responses shown in Figure 2(a)-(c). Sketch the Fourier transforms of the outputs $y_0[n]$, $y_1[n]$, and $y_2[n]$ if the Fourier transform of the input is as shown in Figure 2(d).

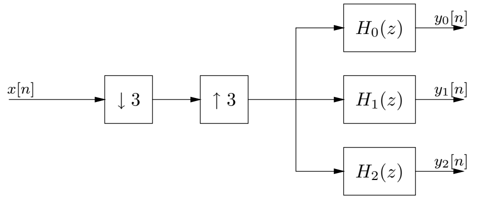

Figure 1: Multirate system of Problem 2.

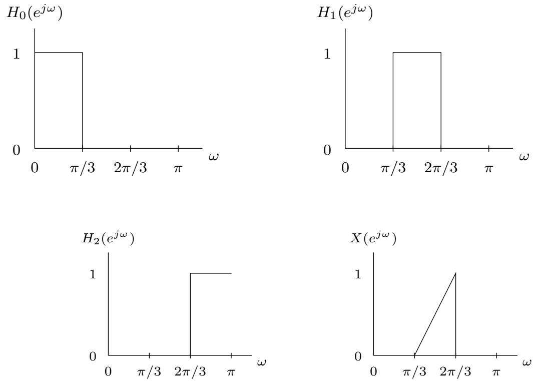

Figure 2: (a)-(c) Ideal filters $H_0(z)$, $H_1(z)$, $H_2(z)$, (d) Fourier transform of the input of Problem 2.