## J-DSP Lab 6: Multi rate Signal Processing and QMF Banks

## Introduction

The goal of this exercise is to examine the effects and use of sampling rate conversion and simulate a two-band QMF bank.

## Problem 6-1: Up-Sampling/Down-Sampling

Use a **Sig Gen** block for input. Set the input to a "pulse width" 20 and "gain" 1. Connect it to a **junction** block. One output of the **Junction** block should be connected to a **D-sampling** block that should be set to "rate" 2. The second output of the **Junction** block, along with the output of the **D-sampling** block should now be connected to two **FFT** blocks. Finally connect the **FFT** block outputs to the inputs of a **Plot2** block. **Plot2** should now display the frequency responses of the original and the sampled signals.

Questions:

1. Try to change the sampling rate and note the change in the frequency response

2. Replace the *D***-sampling** block with an *U***-sampling** block of rate 2. Observe the frequency response and repeat 1.

3. For question 2, add an interpolation filter at the output of the up-sampler. Design a low pass Butterworth filter with cut-off frequency  $\pi/M$ , where M is the up-sampling rate. Examine both the time and frequency domain signals before and after the filter.

## Problem 6-2: Two-Band Filter Bank (QMF)

Go to the File menu at the top of the editor and select 'new' to clear the workspace. Use again a signal generator as the input. Connect it through a *Junction* block to two *Filter* blocks. Connect the *Filter* blocks to two *D-sampling* blocks and then to two *U-sampling* blocks. Set the sampling rates to 2. The outputs of the **U-sampling** blocks should now be connected to two more *Filter* blocks. The output of the two filters should be added together using a *Mixer* block. As a last step connect the *Mixer* block output to a *Plot* block. The system should look like this:



To set the filter coefficients, add **Coeff.** blocks to **Filter** blocks and just type the coefficient values given below. When finished, press the [update] button. After all filters have been set, look at the output signal.

Nominal Filter coefficients:

G1(z)	0.5	0.5
G2(z)	0.5	-0.5
F1(z)	1.0	1.0
F2(z)	-1.0	1.0

Connect a *Freq Resp* block to the top output of each filter.

- Plot and observe the frequency response of each filter.
- o Compare the input and output of the filter bank. Are the signals identical?
- What is the overall delay of the filter bank?
- Change G1(z) to 0.7+0.5z<sup>-1</sup>
- Compare the input and output.
- Design another QMF filter with third order FIR filters and repeat the above steps.

Note: After you have entered the coefficient values in the **Coeff.** blocks, some may appear to be zero but they are just truncated due to limited space. After you enter the coefficients, press [update] just ONCE, the second time will pass the truncated values to the filter.

Filter coefficients obtained from: Sanjit K. Mitra and James F. Kaiser, Handbook for Signal Processing. John Wiley & Sons, Inc., New York 1993