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EE433 REAL-TIME APPLICATIONS OF DIGITAL SIGNAL PROCESSING
EXPERIMENT 4 - PRELIMINARY WORK

1. a. The duration of the sound is 5 seconds.

   b. The duration of the sound is 10 seconds. The pitch of the chirp sound has been lowered, since we have decreased the sampling frequency in "sound" command, which does the work of a D/A converter. Since the sampling period is increased, the duration is twice as much. Since the sampling frequency is decreased, the frequency components lowered, thus the pitch is lowered.

   c. The duration of the sound is 2.5 seconds. The pitch of the chirp sound has been raised, since we have increased the sampling frequency in "sound" command, which does the work of a D/A converter. Since the sampling period is decreased, the duration is half of the original. Since the sampling frequency is increased, the frequency components are higher, thus the pitch is raised.

   d. The duration of the sound is 10 seconds. The pitch of the chirp sound has been lowered, since we have increased the number of samples. Then, twice as much samples enabled twice as much duration with the same sampling frequency, which is 10 seconds. This sound is similar to the sound in part b.

   e. The duration of the sound is 5 seconds. The pitch of the chirp sound did not change, since we have increased both the number of samples and the sampling frequency at the same rate. Then, twice as much samples enabled twice as much duration and twice as much the sampling frequency enabled half as much duration, which does not change the original duration. Thus, this sound is similar to the sound in part a.

   f. The duration of the sound is 5 seconds. The pitch of the chirp sound did not change, since we have decreased both the number of samples and the sampling frequency at the same rate. Then, half as much samples enabled half as much duration and half as much the sampling frequency enabled twice as much duration, which does not change the original duration. Thus, this sound is similar to the sound in part a.

```matlab
clear; clc; close all;
% Q3a
fs = 8000;
T = 1/fs;
t = 0:T:5;
mychirp = chirp(t,500,5,1000);
sound(mychirp,8000);

% Q3b
sound(mychirp,4000);

% Q3c
sound(mychirp,16000);

% Q3d
y = resample(mychirp,2,1);
sound(y,8000);

% Q3e
sound(y,16000);

% Q3f
y = resample(mychirp,1,2);
sound(y,4000);
```

2. Implementation of decimation and upsampling functions is done in LabVIEW and the results are shown in Figures 1, 2, 3 and 4.

   - "Decimate (continuous).vi" and "Upsample.vi" correspond to downsampler/compressor and upsampler/expander respectively.
   - We give the low cut-off frequencies of the filters as "1/(2*decimating factor)" and "1/(2*upsampling factor)" where sampling frequency is 1, since for interpolation and decimation operations, there is a need for lowpass filters with cut-off frequencies π/L and π/M respectively in radians, which are expected in Hertz in LabView. This conversion is done.
   - By changing the constant 1 (sampling frequency) to 1000 for both filters, we observe no change in output arrays. This is because cut-off frequency did not change. We have both multiplied the cut-off frequency and the sampling frequency, thus we have observed no effect.

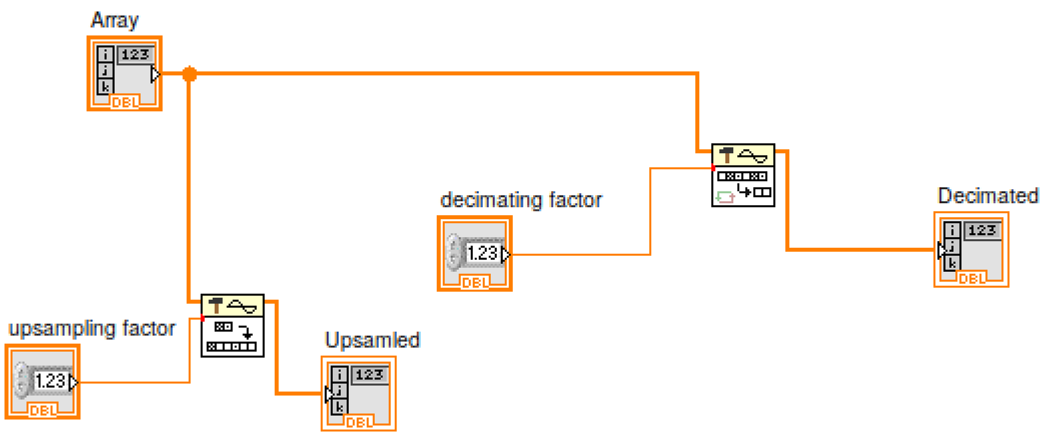

Figure 1: Screenshot of front panel for decimation and upsampling

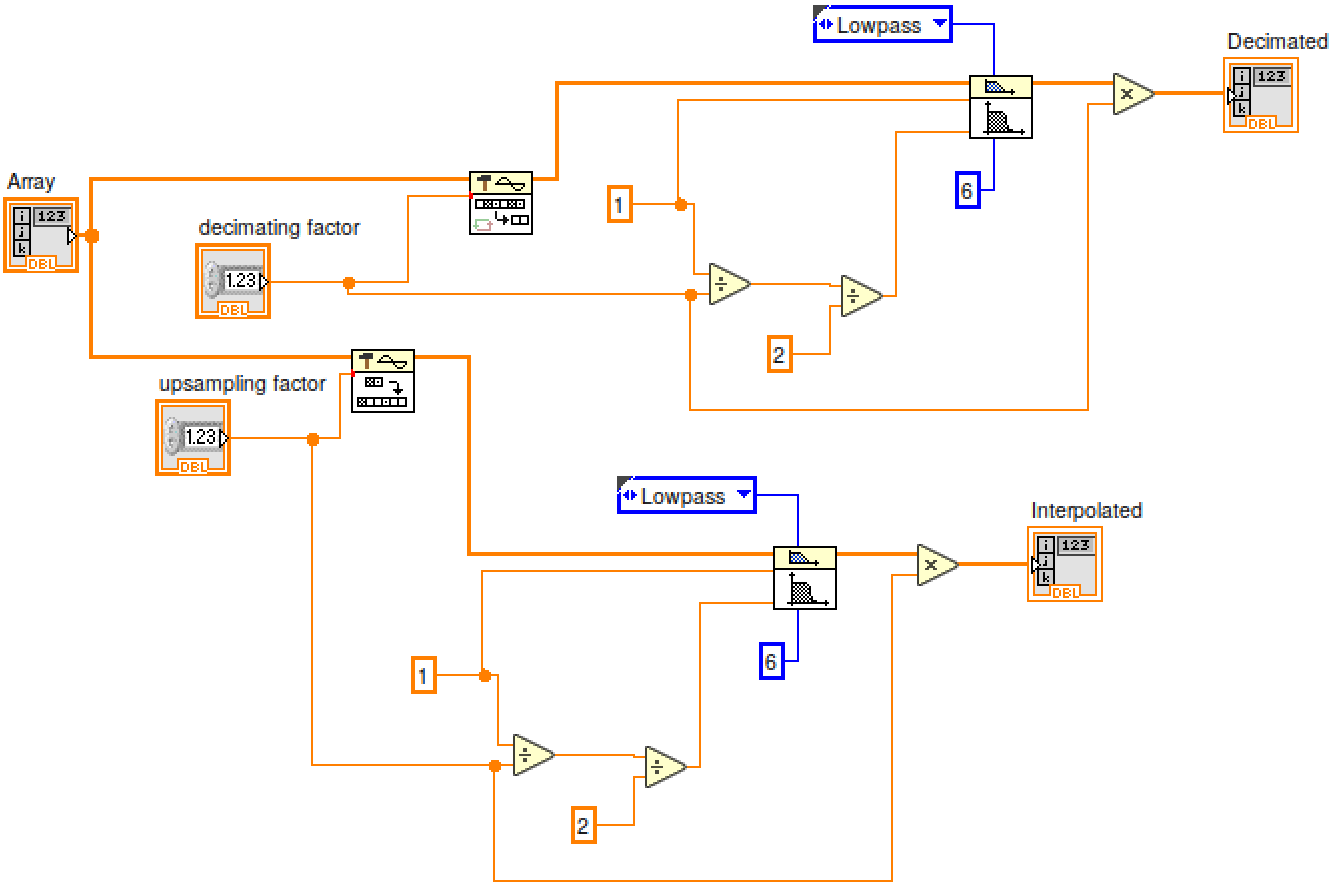

Figure 2: Screenshot of front panel for decimation and upsampling with low-pass filters

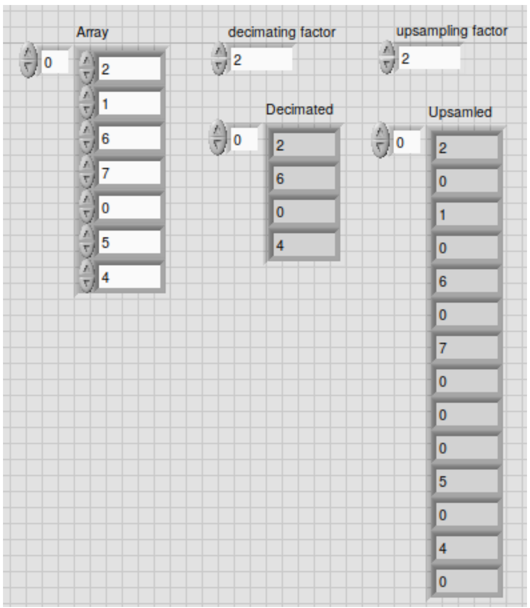

Figure 3: Screenshot of block diagram for decimation and upsampling

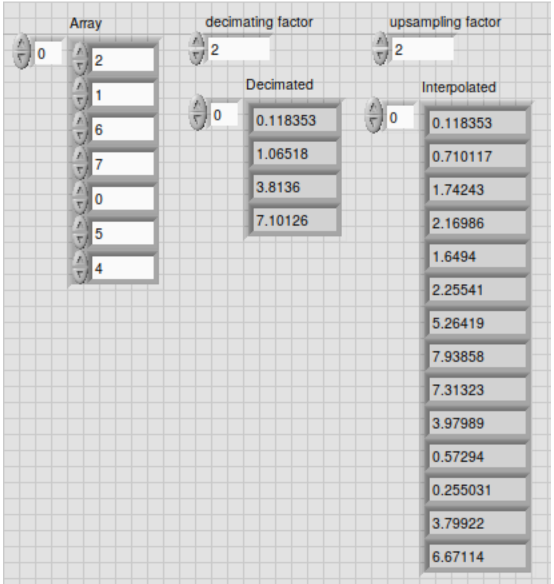

Figure 4: Screenshot of block diagram for decimation and upsampling with low-pass filters