Exp05 - Multirate Signal Processing - Preparation

- **Name:** **Lab Date:**
- **Student No.:** **Day of the week:** **Time:**

1. Try to look back on Experiment 1, where you explored downsampling. Assume now that your system is using 48KHz as sampling rate, and that you have three downsampling stages placed sequentially, and each is downsampling by a factor of two. Draw below a frequency domain graph, indicating the positions of the original sampling rate and Nyquist limit, as well as the resulting sampling rates and the Nyquist limits for each of the stages. Include all values as well.

2. Assume now that you have designed an FIR low pass filter, for a very sharp cutoff frequency at 12KHz. In your design you used 48KHz as the sampling rate to calculate the time domain coefficients (impulse response). Now assume that along your system you place this *same* filter after two downsampling stages, each downsampling by a factor of two. What will happen to the filter operation? If you have a 4KHz, 1Vpp sinusoid as input, what would you expect to see at the output of your system, and why? (assume that you are recombining the signal properly after the decimation/downsampling stages).

3. Consider now what you learn in FIR filter design. You are *given* a low pass FIR filter designed for a 48KHz sampling rate, order 20 and cutoff at 12KHz (that is, at *half* of the full bandwidth of your system). Without using any filter design tool, describe how would you design a *high pass* FIR filter of order 20, cutoff at 12KHz and 48KHz sampling rate? Suggestion: read up on Quadrature Mirror Filters (QMF).

4. As you consider the *firmware* implementation of the two FIR filters of the previous question – that is, how fast you can do things on the DSP – write below two advantages of using two FIR filters simultaneously to split the bank in half. If you write a C function to implement this filtering routine on a separate piece of paper, you get an extra mark. Remember, you take in one sample at a time (type `float`) and will produce *two* outputs.

5. Sketch a block diagram for a system that uses two FIR filters to split the band in half, downsamples *only the lower half* by a factor of 2 and then recombines all paths to produce an output. What else would you need to recombine all signal paths to create a meaningful output?