Project 2: Sampling Rate Conversion

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Objectives of this project

- To expose you to a fundamental concept in multirate signal processing and the effects of non-ideal implementation.
- To give you the opportunity to be creative and play around with audio signals in the context of rate change.

Please note that this is an *individual* project and each person can talk to others, but must ultimately conduct their own simulations and write their own report.

Sampling Rate Conversion

As discussed in the lectures, *sampling rate conversion* (SRC) is a process of converting a discrete-time signal at a given rate to a different rate. This process can be implemented with a linear time-varying system with a complexity that is a function of the SRC factor. In this project we will be dealing with SRC by a rational factor as discussed in class. In this type of system, a combination of upsampling, downsampling and linear time invariant (LTI) filtering is implemented. Typically, a signal will experience loss of information because the SRC process results in a signal that is sampled below Nyquist or the LTI filter (which is required to be an ideal lowpass to avoid aliasing artefacts) is non-ideal.

We will investigate the effects of SRC on sounds signals.

Project Guidelines

You have to obtain two types of sound signals – one music and one speech. You can create your own signals or use something you have obtained from another source (just please credit the source in the report – this is always good research practice).

Overall, you will implement various forms of decimation and interpolation and general SRC and quantitatively and qualitatively measure the effects on the two sound signals you have chosen.

Instructions

1. Acquire your sound samples and please make sure that their bit rate or sampling frequency results in a "good" perceptual quality signal. Your sources may be different, so we will call the associated sampling frequency of each sound signal F_s. It is okay to work with other people and use their sound samples, but please credit them. If you've created the signals, please make note of the conditions used to obtain the sound samples (e.g., what computer, what type of speaker –

built-in or microphone, using which software program, or where or from whom the files were obtained). Please state the bit rate and F_s. You should have two files in the end.

2. In this component you will study the effects of decimation. For each of your sound signals, as discussed in class implement a decimator. You will need an anti-aliasing filtering stage and then a downsampler.

    a. Select a filter design approach you are comfortable with in your simulation environment (e.g., MATLAB). Chapter 10 of the textbook discusses different filter design strategies, and in MATLAB the help files are useful. A "good" FIR filter can be created using the Remez algorithm to achieve a 0.1 dB ripple in the passband and a stopband gain that is down by at least 30 dB (e.g., if the passband gain is 0 db, then the stopband gain would be at or below -30 dB); a filter length of M=30 will achieve this for a cutoff frequency of $\pi/2$, for example. Note, however, that your cutoff will depend on D. Please detail how you've designed the filter and its specifications in the report.

    b. Using this filter design approach in your decimator, experimentally determine three downsampling factors $1 < D_1 < D_2 < D_3$ that produce increasingly worse sound quality, from perceptibly "slight" to "significant" distortion. In each case provide the value of D_i, the new sampling frequency and describe the types of audio distortions that you observe (e.g., hissing noise, synthetic hum). **Please save the decimated sound signals because you will need them for part 3.**

    c. Now, *relax* the filter specifications so you allow more ripple in the passband (e.g., 0.25 dB) and have higher gain in the passband (e.g., -20 dB). Repeat part b discussing the distortions in relation to your results in part a.

    d. Completely remove the anti-aliasing filter of part a. For the same decimation factors D_i as in part b, describe the sound quality once again.

3. Now you will study the effects of interpolation. For each of your *decimated* sound signals in part 2b (above), implement an interpolator to return the sampling frequency back to F_s. As illustrated in class, you will need an upsampler and a filtering stage to remove the repeated "images" in the frequency domain. Use the filter design approach you selected for the decimation part.

    a. For each result of part 2b, pass the sound samples through an interpolator that returns the sampling rate back to F_s. Thus, you will have three interpolation factors of $1 < I_1 < I_2 < I_3$ where $I_i = D_i$ for $I = 1, 2, 3$. Once again, please record your observations on sound quality in comparison to the original sound signal (also at rate F_s) as well as those of part 2b that are the inputs of your interpolators and explain the differences you observe and why you believe you have these differences.

4. As discussed in class, design a SRC with factor I_1 / D_3. Pass the original sound signal through the SRC and describe any differences and sound artefacts. What signal processing non-idealities do you think have contributed to the distortions, if any were observed?

5. Repeat part 4 for an SRC of I_3 / D_1.

Deliverables

Your results should be presented in a written report. There should be one report per person. You should submit your project report with the following components:

- A cover page with the following information: your name, student number, title of this course and project.
- Executive summary – you should briefly summarize your findings in this report and convey the main points of interest to you.
- Introduction – discuss how you obtained the sound signals and what software you used. Mention what type of filter design approach you applied and its various specifications.
- Results and Discussion – provide tables of results in as organized a manner as possible highlighting points of interest. Please also provide a discussion of the results as requested in the Instructions above.
- Appendix – list the code for your simulations here. Please make it in small font (if needed) to avoid this section being very long.

Grading of the reports will be based on the quality of your report: insights presented, accuracy, organization and completeness.