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ECE 101 Lab 6

Introduction:

The purpose of this lab was to sample a continuous time sinusoid of frequency Ω0 and process it with a lowpass filter of frequency Ωs. I analyzed sampling of signals with increasingly larger values of Ω0 until the sampling frequency was no longer twice the frequency of the input sinusoid, at which point over-aliasing occurred and distorted the recovered signal. I then analyzed the effect of a quadratic term inside the input argument, which had the effect of increasing the frequency over time, and thus pushing the input frequency past the cutoff frequency.

Results:

Problem a,b – For these two parts, I created a sampled discrete time signal from the continuous time sinusoid and analyzed the discrete time plot as well as the interpolated continuous time plot using the first 50 samples of the signal. Overall, the interpolated signal matched a continuous time sinusoid quite well, although it was clear that a 1st order hold was used as the interpolating function.

Problem c – For this part, I took the Fourier transform of the recovered signal and analyzed the magnitude and phase of the transform, as shown in the graphs. The graph of the transform included a delta function at -Ω0 and a negative delta function at Ω0, although the included graph shows the absolute value, which is more similar to a cosine of the same frequency. The phase graph clearly shows a zero value only when X(jΩ) = 0, which is expected.

Problem d – The above method was repeated for values of Ω0 = 2π(1500) and Ω0 = 2π(2000). Both of these graphs and transforms produced the proper reconstructed signal after sampling. The reason being that the frequency of both inputs were still less than half the frequency of the sampling frequency, Ωs.

Problem e – The sounds produced by the signals created in part d had a successively increasing pitch, which is to be expected. If analyzed in the frequency domain, the delta impulses simply shift out further towards the cutoff frequency, representing higher frequency sinusoids.

Problem f – The same process used for part d was used for this part with values of Ω0 = 2π(3500), Ω0 = 2π(4000) , Ω0 = 2π(4500), Ω0 = 2π(5000), and Ω0 = 2π(5500). The frequencies above 2π(4000) all exceeded the cutoff frequency as thus, as analyzed in the frequency domain, the impulses representing the sinusoid “crossed” and tended towards lower values. The effect is that the output was similar to an inverted sinusoid of a lower frequency. This can be heard as the pitch was lower for each successive frequency, Ω0.

Problem g,h – For this part, I created a “chirp” signal by adding an additional term to the input sinusoid argument. This additional term was a quadratic which had the effect of producing an ever increasing instantaneous frequency with respect to time. When played on a 1 second interval, the effect was a tone which increased in pitch as time progressed.

Problem i – For part i, I solved, mathematically, the approximate time at which the chirp signal would have its maximum pitch. To do this, I set the instantaneous frequency, Ωinst to half that of the cutoff frequency, Ωs, and solved for the variable t. I calculated the time value to be roughly 3.5 seconds, corresponding with a time sample of about 23,400.

Problem j – Using the same process as above, but with a time interval of 10 seconds, I produced a longer duration sound with which to analyze the chirp signal. I confirmed what I mentioned in part i, that the maximum frequency would occur at roughly 3.5 seconds. At the time where twice the input frequency, Ω0, and the cutoff frequency, Ωs, were equal, the representation in the frequency domain would be two delta pulses stacked on each other with opposite signs, in effect cancelling each other out and providing a frequency of 0.