ECE280 - Lab 8: Voice Scrambler/Descrambler

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I have adhered to the Duke Community Standard in completing this assignment.

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Contents

1 Objectives

1. Use Simulink to explore filters
2. Use Simulink to create a model that scrambles input signals
3. Use Simulink to create a model that descrambles scrambled input signals

2 Background

Multiplying a signal and sine function in the time domain is equivalent to convolving their Fourier transforms in the frequency domain. Since the Fourier transform of the sine function is two impulse functions, the frequency content of the other signal gets centered at the frequency of the sine function. The voice Scrambler system uses this shift in frequency when multiplying with the sine function to swap low and high frequencies. Furthermore, low pass filters are used to prevent aliasing.
The voice Decrambler system multiplies the scrabmled signal with another sine function of the same freqeuncy in the time domain to restore the original unscrambled input signal, then passes the output through a low pass filter to prevent aliasing. The objectives of this lab include studying the described systems in MATLAB's Simulink, understanding how these system's work, and testing the systems.

3 Results and Discussion

- Filter and Design

  What happens with low frequency values (lower than 2 kHz)? What happens with high frequency values (higher than 3 kHz)? How are your observations related to the cutoff frequency?
  Frequency values above 3000 Hz get highly attenuated. Frequencies below 2000 Hz are audible. This is related to the cutoff frequency of the low pass filter which is 2000 Hz; the higher the frequencies are from the cutoff frequency, the higher the attenuation.

- Scrambler

  Describe the design of your scrambler. Include the block diagram and state/discuss important block parameters.
  The block diagram is displayed in figure 1. The Scrambler system Simulink model consists of a From Multimedia File block used to input a voice signal which passes through a Filter Designer block set to be a low pass filter (LPF) with cutoff frequency of 2000 Hz to limit frequencies in the incoming signal in order to prevent future overlap in copies in the frequency spectrum when high and low frequencies are swapped. After passing through the LPF, the signal is multiplied with a sine wave of frequency 2000 Hz in the time domain resulting in negative and positive (left and right) shifting of the frequency content in the spectrum by 2000 Hz. The shifted signal passes through another LPF of frequency 2000 Hz to ensure that low and high frequencies are swapped while cutting off extra high frequencies. Spectrum Analyzer blocks are used to view the spectrum of the signal before being multiplied with the sine function and that of the output. The Audio Device Reader block is used to play the processed signal.

  What limitations did you consider when designing the system and how did you address them? (e.g., why were filters necessary and how did you choose the cutoff frequencies?).
  Limitations considered include aliasing in the output signal. This was addressed by using two low pass filters (LPFs). The first LPF with cutoff frequency of 2000 Hz was added before the signal is multiplied with the sine function to limit frequencies in the incoming signal to 2000 Hz in order to prevent future overlap in copies in the frequency spectrum when high and low frequencies are swapped. The second LPF with the same cutoff frequency was added before the signal is played to ensure that low and high frequencies are swapped while cutting off extra high frequencies.

For the chirp and sinusoid test, explain whether or not each is a good test for this system. Why or why not? How does this compare to the music and speech signals?
Both the chirp and sinusoid signals were good inputs to test the system as they displayed no aliasing, inverted signals in the spectrum analyzerm, and scrambled voice output. The speech signals (sweep.wav file and hello.wav) were also effective tests. The music signal was hard to analyze but still displayed the expected patterns.

Include an image captured from the oscilloscope that verifies the frequency inversion. Comment on what the figure is showing.
The captured image is displayed in figure 2. The left oscilloscope image displays the input signal after passing through a low pass filter; the right one displays the fully processed signal. As visible, low and high frequencies are inverted after the signal is fully processed.

Questions from the methods:

- What does the frequency spectrum of this signal look like when you multiply the sinusoid and the message signal in the time domain, what will the resulting frequency spectrum look like?
  Suppose the frequency spectrum of the input signal looks like two delta functions at -100 Hz and 100 Hz. Multiplying the sinusoid and the message signal in the time domain will result 4 delta signals at frequencies -2.1 kHz, -1.9 kHz, 1.9 kHz, and 2.1 kHz in the frequency spectrum.
- Given that your sinusoid has a frequency of 2000 Hz, what is a reasonable value for the cutoff frequency?
  2000 Hz is an appropriate cutoff frequency because it blocks the repeated values at frequencies beyond 2000 Hz, effectively preventing future overlap in frequencies (aliasing).
- Given your previous observations regarding the frequency spectrum of the original signal, what should the cutoff frequency of the low pass filter be set to?
  2000 Hz is an appropriate cutoff frequency because it blocks extra copies at high frequencies and allows for swapping of high and low frequencies.

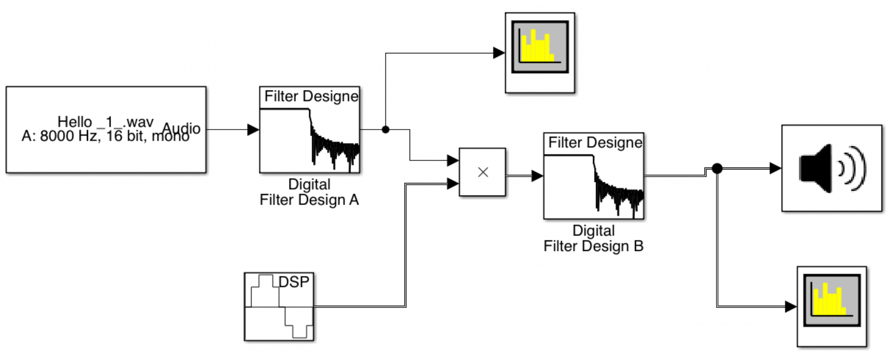


Figure 1: Block diagram of the scrambler model

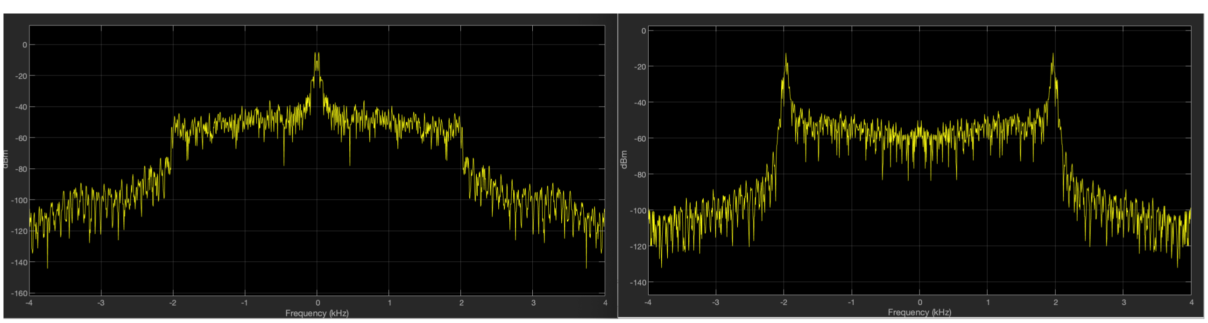


Figure 2: Image displaying frequency inversion

- Descrambler.

  Describe how the descrambler works to recover your original signal?
  The Descrambler system is similar to the Scrambler system model. The scrambled voice signal passes through a low pass filter (LPF) using the filter designer block with cutoff frequency of 2000 Hz to limit frequencies in the incoming signal in order to prevent future overlap in copies in the frequency spectrum when high and low frequencies are restored to their original positions. After passing through the LPF, the signal is multiplied with a sine wave of frequency 2000 Hz in the time domain resulting in negative and positive (left and right) shifting of the frequency content in the spectrum by 2000 Hz. The shifted signal passes through another LPF of frequency 2000 Hz to ensure that low and high frequencies are restored while cutting off extra high frequencies.

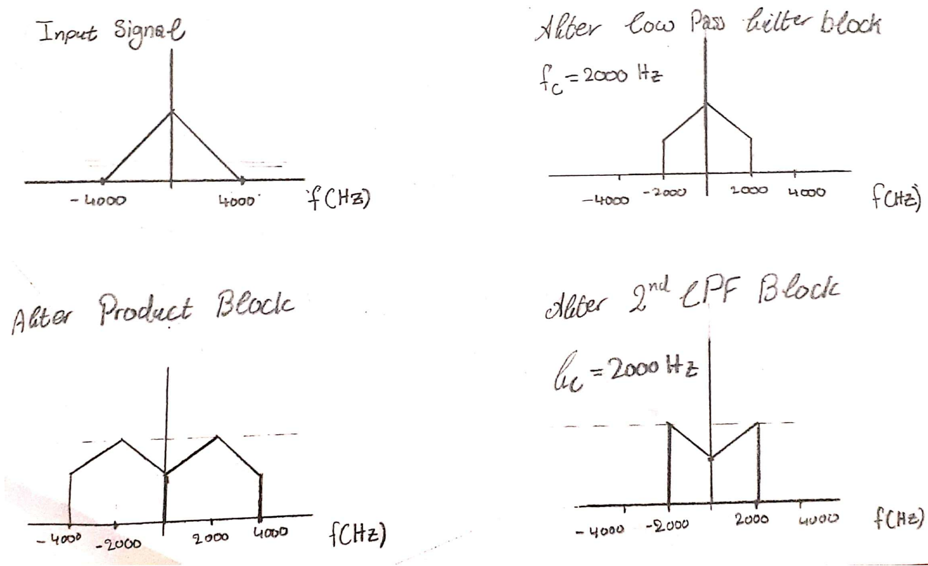
  Why can the descrambler have the same block diagram as the Scrambler?
  Both the Scrambler and Descrambler system perform the same operation: swapping high frequencies with low frequencies without aliasing. If the input to the Scrambler system is a scrambled system, then the swapping of frequencies will result in a descrambled output and vice versa; hence, they have the same block diagram.
  Did you obtain the results you expected for the chirp, sinusoidal, speech, and music signals? Which ones worked best?

  Yes, the expected results were obtained for the sinusoidal, speech, and music signals. The hello.wav, sweep.wav, and sinusoidal signal worked best and sounded almost identical the original.
  Does the scrambler reproduce the original signal perfectly? Describe any differences you observed between the original and recovered signals and discuss what these differences are due to.

  No, the scrambler does not perfectly reproduce the original signal. Observed differences include a slightly muffled and lower quality sound output. This is due passing the signal through multiple filters. As seen in previous laboratories, low pass filters applied on images result in blurs. Similarly, filtered voice signals can't sound identical to the original voice signals as the quality deteriorates.
  What happened when you eliminated the second low-pass filter in the descrabmbler?

  When the second low-pass filter in the descrabmbler was eliminated aliasing occured. The sound output had an aliasing effect and was lower in quality.

- Assume that your original input signal has a triangular frequency spectrum from -4000 Hz to 4000 Hz. Sketch the spectrum of the signal at each point (i.e., after each block in your model) in your Scrambler/Descrambler system. Figure 3 shows the spectrum of the signal at all wanted points specified in the lab manual.



Scanned with CamScanner

Figure 3: Spectrum at all wanted points

4 Conclusions

In this laboratory, a simple filtering system was built and studied in Simulink. Futhermore, a Simulink model that scrambles input voice signals was examined in detail. A Simulink model designed to descrabmle the scrambled voice signals was created and studied. Lastly, differences between the original and descrambled signals were examined.