ECE 280L: Introduction to Signals and Systems
Lab 6: Voice Scrambler/Descrambler

Nathan Lewis

Cemal Yağcioğlu

November 16, 2017

Duke University

Instructor:	Professor L. Collins
TA:	Chance Fleeting
Section:	ECE 280L-04L

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

Contents

1 Objectives **1**

2 Background **1**

3 Results and Discussion **1**

3.1 Filter Design 1

3.2 Scrambler 1

3.3 Descrambler 4

4 Conclusion **5**

1 Objectives

Upon completion of this lab one should have designed a filter according to certain specifications such as cutoff frequency and implemented it in SIMULINK. One should have also designed a system that scrambles a signal using methods such as filters and sine generation within SIMULINK. In addition to this one should also design and implement another system within SIMULINK that is capable of descrambling the signal produced from the scrambler from earlier. In terms of understanding one should leave know the effect of multiplication on the frequency representation of a signal as well.

2 Background

To understand how the scrambler/descrambler works you first must know that signals in the time domain are convolved in the frequency domain. Usually convolution is hard to show graphically and mathematically, however in this case we are using the product of a signal and another sinusoid input - which in the frequency domain only has two impulses at the fundamental frequency (positive and negative). This makes the convolution easier to show mathematically as essentially during convolution we are simply shifting the frequency content of the signal to the location of the frequency impulses created by the sinusoid. This is what happens during scrambling to our signal, the same process when applied to the resulting signal will revert this effect essentially descrambling the scrambled signal.

3 Results and Discussion

3.1 Filter Design

What happens with low frequency values (¡2kHz)? What happens with high frequency values (¿3kHz)? How are your observations related to the cutoff frequency parameter, Fc?

With frequencies below 2kHz we hear a clear signal with an increasing pitch as we increase the frequency (an expected behavior). As we increase the frequency beyond 3kHz the resulting signal becomes inaudible. This is directly related to the cutoff frequency as we have utilized a low pass filter so frequencies beyond the cutoff frequency aren't present in the output signal, therefore can't be heard.

3.2 Scrambler

Describe the design of your Scrambler. Include the block diagram and state/discuss important block parameters.

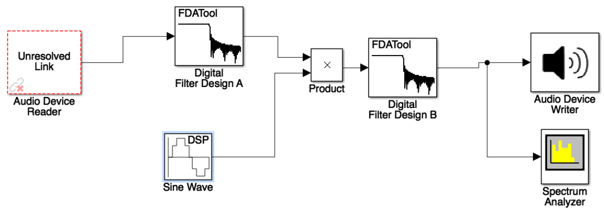

Figure 1: Block diagram of our scrambler

In our block design we first pass our message signal (from the Audio Device Reader block) into a Digital Filter design block which we set appropriately to act as a low pass filter with a cutoff frequency of 2Hz. We then use a product to take the resulting signal and multiply it with the sinusoid signal we produce from Sine Wave generator block where we produce a sin wave with an amplitude of 1 and a frequency of 2kHz. The output signal from the product block is then passed through another low pass filter with a cutoff frequency of 2kHz (the same block as used at the beginning and eventually passed to the two output blocks - audio device out and spectrum analyzer.

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?)

Filters were necessary part of this design as when multiplication of the signals occurred we needed to make sure there was no overlap (in the replicated signals). This meant we had to choose a cut off frequency at the same frequency as the fundamental frequency of the sinusoid we used in the multiplication.

For the chirp and sinusoid test signals, explain whether or not each is a good test for this system. Why or why not? How does this compare to music and speech signals?

The sinusoid test signals work well as you can control the frequency increments and can see the effect of the initial low pass filter on frequencies greater than the cut off frequency and how the frequency has been inverted - as you increase the frequency and the pitch lowers. You can come to the same conclusions using the chirp signal, as that progresses through the same frequencies, except not controlled by you. The music and speech provide good proof of the scrambling actually occurring, as we already know what the input signal is supposed to sound like so when we hear a scrambled output it provides proof of the systems functionality.

Include an image captured from the oscilloscope that verifies the frequency inversion. Comment on what the figure is showing

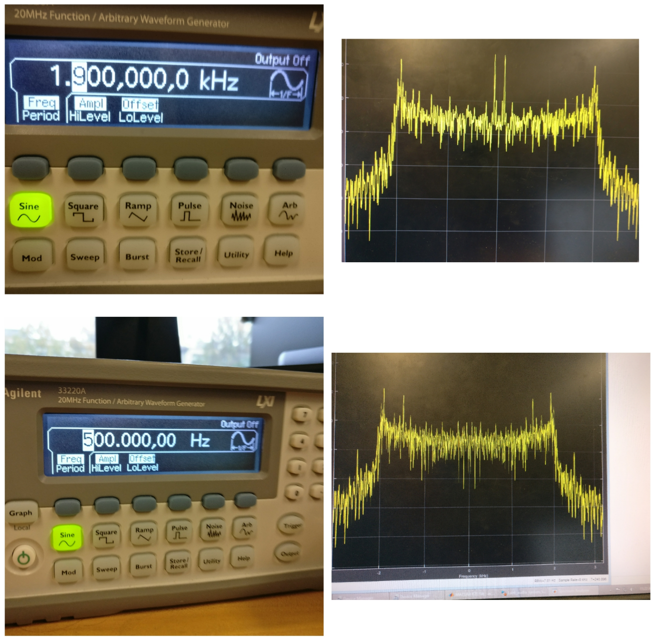

Figure 2: Pictures showing sinusoid frequency and spectrum analyzer output of scrambler system

In these pictures we can see that when the frequency of the sinusoid input is high (1.9kHz) there are two distinct peaks at a much smaller frequency of 0.1kHz. Then when the frequency is reduced to 0.5kHz the two peaks now appear around 1.5kHz. This is due to frequency inversion.

What does the frequency spectrum of the message signal look like? When you multiply the sinusoid and the message signal in the time domain, what will the resulting frequency spectrum look like?

We get a similar frequency spectrum as the first one except replicated at the frequencies +- 2Hz.

The resulting spectrum will be the same as the one we observed above (replicated at the fundamental frequency) as multiplication in the time domain is convolution in the frequency domain.

Given that your sinusoid has a frequency of 2 kHz, what is a reasonable value for the cutoff frequency of this filter?

A reasonable value would simply be 2kHz, this is to achieve the anti-aliasing effect required to avoid replica signals overlapping as previously described.

Given your previous observations regarding the spectrum of the signal at the output of the Product block relative to the spectrum of the original signal, what should the cutoff frequency of the low pass filter be set to?

As we previously applied a low pass filter of a cut off frequency of 2kHz, upon multiplication of signals to avoid the higher frequencies becoming part of signal and stopping frequency inversion from occurring we should use a cut off frequency of 2kHz again.

3.3 Descrambler

What happened when you eliminated the second low pass filter in the Descrambler? How does the output of your system change? Can you understand the descrambled message? Does it sound different from the output of the model with the low pass filter? If there are differences, what are they due to? If there are no differences, why is this the case?
Second low pass filter causes a 'steep drop' in amplitudes of frequencies beyond +-2kHz. This plunging effect is observed on the frequency spectrum output. After second low pass filter is removed, amplitudes of frequencies beyond 2kHz limit increased. With this, aliasing is introduced, and as expected the output includes some frequencies as faint humming that does not exist in the input. This is observed due to lack of second low pass filter to remove any frequencies beyond the frequency range. One might argue that filter can lead to loss of some data of input, however as this is the second filter in the system with the same frequency limit, this filter only removes 'incorrect' signals outside of the boundaries already set by the first filter.

Describe how the Descrambler works to recover your original signal.
Descrambler works in the same way as scrambler. Both scrambler and descrambler are reversible linear systems. Scrambler filters the input, convolves it with a sinewave with set frequency in frequency domain, and filters the output again. Essentially, as also observed, scrambler reverses the signal within the set frequency. An increase in input frequency makes the output signals maximum pulses get closer to x=0 at the graph. Taking the output of the scrambler and using it as an input to another scrambler (which is essentially also a descrambler) causes this reversed signal to reverse again and therefore recovers the original where increase in frequency leads to increase in x axis locations of impulses.

Why can the Descrambler have the same block diagram as the Scrambler?
To simplify the already stated reason, both work like a switch to switch the signal from given phase to the other possible phase. Thus adding them and to end results in a signal that is switched back to its original signal.

Did you obtain the results you expected for the chirp, sinusoidal, speech, and music signals? Which ones worked best?
Sinusoidal signal works like expected and descrambler accurately gives back the original signal. This is an optimal base case test to see if the descrambler works as expected. Chirp and music signals are recovered with less accuracy due to their frequencies that overpasses frequency limits as expected. Speech is recovered with an accuracy that can be understood. Thus, sinusoidal signals and speech signals worked the best.

Does the descrambler reproduce the original signal perfectly? Describe any differences you observed between the original and recovered signals and discuss what these differences are due to.
The output signal has lower quality than the input signal. This is highly expected for input frequencies higher than the 2kHz as those signals will be losing their amplitudes through the filtration. Thus, when a music is played, the music can be heard but with a lower quality and with introduction of some unwanted noise. This differences are due to filtration of high frequencies, and also possibly due to having an 'unideal' filter that cannot fully prevent aliasing.

4 Conclusion

Conclusions

Signals are the main tool of information transfer in every electronical device as they are fast and reliable. However, one of the main concern in signal transformation would be the security of the information. To protect the information, a system should modify a signal by passing it through a reversible system before sending it to the receiver to be reversed back to original signal by the receiver. This creates the backbone of the end-to-end encrypted security system in modern technology and one way to observe a rather simple example of this is using amplitude modulation.
Through this laboratory, signal is scrambled by modifying the amplitude distribution of the signal's frequencies within a given frequency limit of 2kHz. This modification is done by, filtering the signal with low pass filter, convolving it in a frequency domain with specific frequency, and filtering the output of that signal again with low pass filter. This system satisfies the general expectance of a low level encryption: system is reversible, signal is hard to understand without reversing, reversing can only be done by the receiver that knows the 'reversing' system. Reversing system in this occasion is the system itself. In another saying, scrambler has the same structure as descrambler as both acts as a switch between reversed-sampled-filtered signal and original(twice reversed)-sampled-filtered signal (filtration is done twice)

In this system, transferred information should be limited within the frequency limits of the filters. As information is sent through systems with memory, a continuous signal has to be sampled and therefore a band limit for transformation is always required. The first filter in the system makes sure that the input signal is within the expected limit. The second low pass filter (with the same frequency limit) takes only one of the reverse replicas created by convolution in frequency domain. The system works in the same way for descrambler as well to reverse back to the original signal.

A reversible scrambler and a system of signal encryption can be achieved by using amplitude modulation in MATLAB. A system with correct frequency limits can scramble and descramble the input signal, if the input signal is within the frequency limits of the filters.

References

Duke University ECE Department, Lab 6: Voice Scrambler. Retrieved from Sakai, 2017.

Wikipedia, Scrambler. Retrieved from `en.wikipedia.org/wiki/Scrambler`, 2017.

University/School Laboratory Report Template. (2016, March 25). Retrieved from Chance Fleeting September 2017.