ECE 300 Lab 5: Harmonic Distortion and SNR

## Objective

The objective of this lab is to measure the total harmonic distortion (THD) and signal-to-noise ratio (SNR) of a given system. The THD provides a measure of the linearity of a system and the SNR describes how “clear” the signal is relative to the noise.

## Prelab:

1. Find the Normalized Average Power in V2 and dBV of
2. 250mV amplitude sinusoid
3. 250mV amplitude ramp function (i.e. over one period x(t)=(500mV/T0)\*t for -T0/2 < t < T0/2)
4. Write a MATLAB function to convert a given power in dBV to
5. The normalized average power in V2
6. The amplitude of the sinusoid in V
7. The value of the magnitude of it’s Fourier series coefficient, |c1|

Use your function to convert -20dBV to the desired values.

1. Adjust your MATLAB script to take in a vector of power values in dBV and compute the harmonic distortion. The first value in the vector should be the power of the fundamental frequency. Convert the following vector of power values into a HD. p=[-15 -40 -20 -30 -40] Verify that your script gives you a HD=-4.5 dB.

# Settings and Parts Used

Other Equipment:

Laptop and usb memory with NTFS formatting

2 – 1.6kΩ resistors (note the difference from last week’s lab)

1 each - 10nF cap, 100nF cap (these are the same as last week’s lab)

## Background

### Harmonic Distortion

In this lab we will revisit linearity in the context of our recent knowledge of Fourier Series and Fourier Transforms. Shown in Figure 1 are examples of input-output relationships of two different memoryless systems (similar to what you plotted in Lab 1), one linear and the other non-linear. Note that when a signal goes through a purely linear system, Figure 1A, the only thing that can change about it is its amplitude. The basic “shape” of the signal will remain unaffected and the frequency will be exactly the same. As such the power spectra of the input and output waveforms will each have only one frequency, but they may have a different power level due to a change in amplitude. When the same signal is passed through a nonlinear system, Figure 1B, the system will change the “shape” of the signal. Looking at the input and output signals of the nonlinear system, it is clear that, while their *fundamental* frequencies are the same, the output will have additional harmonics that are not present in the input signal.

Using these additional harmonics of the output signal, we can quantify the difference between the input and output signals in terms of the “harmonic distortion”. If the input to the system is a pure sinusoidal tone, the *total harmonic distortion* (THD) of a system is the ratio of the power in all of the harmonics (k≥2) of the output signal relative to the power in the fundamental (k=1). Note that the power at DC is ignored because it represents a simple offset rather than a change in “shape” due to nonlinearities.

x

y(x)=Ax

y(t)

t

t

x(t)

x

y(x)= x2

y(t)

t

t

x(t)

A)

B)

*Figure 1: Examples of input-output relationships of two different systems along with the time varying input and output signals. A) y(t)=Ax(t) and B) y(t)=x(t)^2. The input-output relationship of each system is denoted by the bold curves in each of the plots. The input signals are shown by the rotated signals below each of the plots. The corresponding output signals are shown to the right of each plot.*

To get an equation for THD recall that the average power of the cosine at a single harmonic is given by



where k=1 represents the fundamental and k≥2 are the harmonics. Recall also that the total average power in the signal is given by the sum of the average powers at each harmonic.



As such the THD can be calculated by finding the total power in all harmonics divided by the power in the fundamental



In reality, there is no way to measure to the infinite harmonic, so we usually truncate the THD measurement to a certain harmonic. Note also that you need the Fourier coefficients for this measurement but that the spectrum analyzer provides values in dBmV. Recall that



Because the THD is a ratio of powers it is unit-less and is expressed as a percentage. It can also be expressed in dB as



Note that this is ***not*** equivalent to summing the dB values and ***you must first convert to the coefficients before computing THD***.

### Noise

Noise is a funny signal. Noise has no definite formula such as sin(t) to model it, and every time you measure it, you get different values. As such, it has to be modeled as a random variable, so that it is described with a mean and standard deviation about the mean. In particular, “white noise” has a zero mean with a flat spectrum over all frequencies. Because noise is not a definite function, how can you provide a measure of how much noise there is in a signal? What is the noise “amplitude” relative to the signal amplitude? You can’t use the time average value, because the mean is zero even though there is clearly noise present in the signal. You can’t use the peak value because it is constantly changing and you can never be sure if you have the “maximum” value. This is why noise is measured in terms of its average signal power, which is defined as



where is the noise voltage. Note that this is also equal to the variance of the signal if it has zero mean and is also the same as . All of these various quantities refer to the time-average power of the noise. Because noise is not periodic, to get the average power, you have to actually integrate out to infinity, which is impossible. In practice we take a very long recording of the noise and then compute the average power.

Even though the noise power can be defined solely in the time-domain, interpreting noise measurements cannot be done without consideration of the frequency domain. Ideal white noise contains all frequencies out to infinity with equal power and random phase. If you take a white noise signal and put it through a filter with a passband gain of 1, the noise at the output of the filter will have less power because the power at some of the frequencies has been eliminated. In other words, the  (variance or “amount”) of the filtered noise will be smaller than the variance of the unfiltered input signal. The more narrow the bandwidth of the filter becomes, the smaller the output noise power (“amount”) will be. The problem is that every practical system is a filter which often has an unknown bandwidth. What makes this really tricky is that the “system” includes all measuring instruments in addition to the device that is being measured. Therefore, in order to make noise measurements, you must pass the noise signal through a system with a known and fixed bandwidth. We will measure the noise power by putting the noise signal through a filter with a bandwidth that is much smaller than either the Signals Board or the measurement equipment. Then, since we know the bandwidth of the filter, we can measure the output noise with the DMM by measuring its RMS value and then squaring that value to get power.

The SNR is used to measure the signal power relative to the noise power, where the higher the SNR, the smaller the “amount” of noise in the signal. The SNR is defined as



and is often expressed in dB as



We have discussed how to measure the power in the noise. If the signal is a single sinusoid, then the signal power is equivalent to 2|ck|2. Note that, because we are measuring voltages in this case, PSig is the normalized power, which has units of  not dBmV, so you cannot simply use the measurement from the SA or scope. How would you measure the total normalized signal power for a square wave or a triangle wave or some other arbitrary periodic waveform?

## Part 1: Measuring Harmonic Distortion

1. Scope Ch1 = SIG

Scope Ch2 = V\_OUT

Set the Scope to Acquire in Averaging Mode with 8 averages

FG Ch1 = SIG: Sine, 1Vpp, 1kHz

1. Adjust the scope so that you can clearly see approximately 20-30 periods of the input and output signals and also the FFT of the board’s output signal. Make sure that you can clearly see the fundamental frequency through the 5th harmonic (i.e. k=1-5) in the FFT.
2. Looking at the input and output signals on the scope, adjust the gain potentiometer on the board so that the input stage has a gain of 2.
3. Notice that for a small amplitude of the input signal the spectrum of the output signal is a single spike. As you increase the amplitude of the input signal, harmonics will begin to appear in the output signal. Verify that the first non-zero harmonic starts to show up when the input amplitude is approximately 2.5 Vpp.
4. Increase the amplitude of the input signal until the output begins to clip. While looking at the scope, adjust both higher and lower the amplitude, frequency, and phase of the input signal. Notice what affects the ***number*** of harmonics that are visible on the output and the power levels of those harmonics. Notice what affects the ***frequencies*** of the harmonics.
5. Reset the input signal back to what was given in Part (a).
6. **Your Experiment:**
   1. Adjust your MATLAB script from the prelab to calculate harmonic distortion given a vector of power measurements in dBV for all non-zero frequency components from the fundamental through the 5th harmonic (i.e. P=[P1 P2 P3 P4 P5]).
   2. For five different amplitudes, measure the fundamental and harmonic power levels and use your script to calculate the harmonic distortion in dB. Each increase in amplitude should show ***at least*** a 3dB increase in distortion from the one before it.
   3. Record your data in the table at the end of the lab and plot the harmonic distortion with respect to amplitude.
   4. Also record your scope captures for the data with the smallest amplitude and that with the largest amplitude.

## Part2: Measuring SNR

1. FG Ch1 to SIG: Sine, 500mVpp, 200Hz, zero offset  
   FG Agilent 33220A to ZIN: Noise, 8Vpp, make sure Hi-Z

Connect scope Ch1 = SIG

Connect scope Ch2 = ZIN

Connect scope Ch3 = V\_OUT   
Scope Ch MATH = FFT of V\_OUT

Set the scope to Acquire in Normal Mode

DMM Tektronix 4020 = AC V on V\_OUT

### Measure the unfiltered SNR

1. Start by verifying your ability to measure signal powers with the DMM. Switch the ZIN offset switch to 0 and make sure that the YIN Gain switch is set back to 1 so that only the signal is going through the board unaltered. Measure the voltage with the DMM and verify that you get the same measurement as the dBV value shown in the frequency domain on the scope. **Record this signal power in dBV in your data memo.**
2. Now you’re going to turn the signal off and measure the noise power without the signal. Turn the signal off on the FG and switch the ZIN offset switch from 0 to ZIN to add the noise signal. Using the measurement from the DMM, verify that the noise power is approximately -18 dBV. **Record this value of the noise power in dBV in your data memo.**
3. Turn the signal back on so that both the noise and signal are going through the board together. Calculate the SNR using
4. **Your experiment:**
   1. Mathematically determine the input signal amplitudes that will give you SNRs of approximately +10dB and -10dB. Record these amplitudes in the table at the end of the lab. The noise power isn’t changing, so to verify that your SNR is correct, flip the ZIN switch back to 0 to measure the signal only power. Then calculate the SNR using the value that you measured from Part (c) for the noise power.
   2. Provide the screen captures from the scope for each of these SNRs.

### Measure the SNR of the signal after it goes through a filter.

1. Reset all of the signals to what is shown in Part 2(a) above.
2. Put in the 1st order filter passive elements as you did in the previous lab using the 1.6kΩ resistors and 10nF capacitor that were mentioned in the parts list for this lab. Switch over to filtering the signal with the 1st order filter.
3. Adjust the FG buttons and ZIN offset switch as you did previously to measure the signal power by itself and the noise power by itself. **Record these values in your data memo and calculate the SNR.**
4. **Your Experiment:** 
   1. Mathematically determine the input signal amplitudes that will give you SNRs of approximately +10dB and -10dB. If the function generator will not produce a small enough amplitude for the -10dB SNR, you should use an attenuator (supplied in lab) and adjust the FG amplitude accordingly. Ask the instructor if you do not know how to use the attenuator. Record these amplitudes in the table at the end of the lab. Use the noise power from Part (h) and your measurements of the signal-only powers to verify your calculations.
   2. Provide the screen captures from the scope for the SNR=+10 and SNR=-10 signals.

### Measure the SNR of the signal after it goes through a different filter

1. Reset all of the signals to what is shown in Part 2(a) above.
2. Keep the resistors the same, but replace the 10nF capacitor with the 100nF capacitor.
3. Adjust the FG buttons and ZIN offset switch as you did previously to measure the signal power by itself and the noise power by itself. **Record these values in your data memo and calculate the SNR.**

### Measuring the SNR with a triangle wave as the input signal

FG Ch1 to SIG: Ramp, 500mVpp, 200Hz

Change the filter capacitor back to the 10nF capacitor

1. Adjust the FG buttons and ZIN offset switch as you did previously to measure the signal power by itself and the noise power by itself. You must use the DMM to measure the signal power for this case. **Record these values in your data memo and calculate the SNR.**
2. **Your Experiment:**
3. Mathematically determine the input signal amplitudes that will give you SNRs of approximately +10dB and -10dB. Again use an attenuator if necessary to get the signal amplitude small enough for the -10dB SNR. Record these amplitudes in the table at the end of the lab. Use the noise power from Part (m) and your measurements of the signal only powers to verify your calculations.
4. Provide the screen captures from the scope for the SNR=+10 and SNR=-10 signals.

Data Memo for Lab 5

Names:

Date:

Section:

# Part 1: Harmonic Distortion

### Table 1: Harmonic Distortion Data

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
| Harmonic | Frequency (Hz) | Power @ Amp 1  (dBV) | Power @ Amp 2  (dBV) | Power @ Amp 3  (dBV) | Power @ Amp 4  (dBV) | Power @ Amp 5  (dBV) |
| 1 |  |  |  |  |  |  |
| 2 |  |  |  |  |  |  |
| 3 |  |  |  |  |  |  |
| 4 |  |  |  |  |  |  |
| 5 |  |  |  |  |  |  |
| HD (dB) |  |  |  |  |  |  |
| Amp (Vpp) |  |  |  |  |  |  |

1. MATLAB Plot of HD in dB versus amplitude in Vpp
2. Screen captures of input and output signals at the largest and smallest amplitudes

|  |  |
| --- | --- |
| Screen capture of smallest amplitude signal | Screen capture of largest amplitude signal |

1. Explain why it is not practical to measure HD with a triangle wave or other non-sinusoidal periodic signal as the input signal. Your answer should be based on the frequency domain.

# Part 2: Signal-to-Noise Ratio

### The Unfiltered Signal

1. Recorded signal power of the noise and the input sinusoid with amplitude of 500mVpp

|  |  |  |
| --- | --- | --- |
| Signal power at 500mVpp amplitude (dBV) | Noise power (dBV) | SNR (dB) |
|  |  |  |

1. Signal amplitudes and scope captures for signal with SNRs of ±10dB

|  |  |
| --- | --- |
| Signal amplitude in V for +10dB SNR = | Signal Amplitude in V for -10dB SNR = |
| Scope screen capture showing +10dB SNR | Scope screen capture showing -10dB SNR |

### The filtered signal with 10nF capacitor

1. Recorded signal power of the noise and the input sinusoid with amplitude of 500mVpp

|  |  |  |
| --- | --- | --- |
| Signal power at 500mVpp amplitude (dBV) | Noise power (dBV) | SNR (dB) |
|  |  |  |

1. Signal amplitudes and scope captures for signal with SNRs of ±10dB

|  |  |
| --- | --- |
| Signal amplitude in V for +10dB SNR = | Signal Amplitude in V for -10dB SNR = |
| Scope screen capture showing +10dB SNR | Scope screen capture showing -10dB SNR |

### The filtered signal with 100nF capacitor

1. Recorded signal power of the noise and the input sinusoid with amplitude of 500mVpp

|  |  |  |
| --- | --- | --- |
| Signal power at 500mVpp amplitude (dBV) | Noise power (dBV) | SNR (dB) |
|  |  |  |

### The filtered triangle signal with 10nF capacitor

1. Recorded signal power of the noise and the input triangle wave with amplitude of 500mVpp

|  |  |  |
| --- | --- | --- |
| Signal power at 500mVpp amplitude (dBV) | Noise power (dBV) | SNR (dB) |
|  |  |  |

1. Signal amplitudes and scope captures for signal with SNRs of ±10dB

|  |  |
| --- | --- |
| Signal amplitude in V for +10dB SNR = | Signal Amplitude in V for -10dB SNR = |
| Scope screen capture showing +10dB SNR | Scope screen capture showing -10dB SNR |

### Questions

1. With regard to the data tables in Parts (a), (c), and (e) of the Data Memo explain why the signal power remained the same but the noise power changed when the filters were added. Think in terms of how the filter is affecting the frequencies of the signal versus the noise.
2. Explain why the triangle signal looked almost like a sinusoid when the SNR was -10dB.
3. Explain in frequency domain terms why the noise power dropped by 10dB when the 10nF capacitor was replaced by the 100nF capacitor.
4. You can clearly see the difference between a high and low signal to noise ratio in the time domain. What is the difference between a high and low SNR in the frequency domain?