GEORGIA INSTITUTE OF TECHNOLOGY
SCHOOL of ELECTRICAL and COMPUTER ENGINEERING

ECE 2025 Spring 2003

Lab #12: Design with Fourier Series

Date: 9–15 April 2003

Tuesday Lab sections only: Start Lab #12 on 15-April.

You should read the Pre-Lab section of the lab and do all the exercises in the Pre-Lab section before your assigned lab time. You **MUST** complete the online Pre-Post-Lab exercise on Web-CT at the beginning of your scheduled lab session. You can use MATLAB and also consult your lab report or any notes you might have, but you cannot discuss the exercises with any other students. You will have approximately 20 minutes at the beginning of your lab session to complete the online Pre-Post-Lab exercise. The Pre-Post-Lab exercise for this lab includes some questions about concepts from the previous Lab report as well as questions on the Pre-Lab section of this lab.

The Warm-up section of each lab must be completed **during your assigned Lab time** and the steps marked *Instructor Verification* must also be signed off **during the lab time**. After completing the warm-up section, turn in the verification sheet to your TA.

Forgeries and plagiarism are a violation of the honor code and will be referred to the Dean of Students for disciplinary action. You are allowed to discuss lab exercises with other students and you are allowed to consult old lab reports but the submitted work should be original and it should be your own work.

The lab report for this week will be an **Informal Lab Report.** It is only necessary to turn in Section 4 as this week's lab report. The report will **due the next time your lab meets: 16–22 April**.

1 Introduction & Objective

The goal of this laboratory project is to show that Fourier Series analysis and the frequency response $H(j\omega)$ are powerful methods for predicting the response of a LTI system when the input is a periodic signal. In this particular lab, we will use Fourier Series and the Fourier transform to analyze a frequency synthesis design problem in the frequency domain.

This lab uses a MATLAB GUI for continuous-time frequency response: **`CLTIdemo`**. This GUI provides a convenient way to visualize the sinusoidal response of LTI systems. When the input signal is an infinitely long sinusoid that extends over the range $-\infty < n < \infty$, the output is also a sinusoid. The frequency response tells us how the magnitude and phase of the output sinusoid can be calculated.

1.1 Background: Fourier Series Analysis and Synthesis

Recall the *analysis* integral and *synthesis* summation for the Fourier Series expansion of a periodic signal $x(t) = x(t + T_0)$. The Fourier synthesis equation for a periodic signal $x(t) = x(t + T_0)$ is

$$x(t) = \sum_{k=-\infty}^{\infty} a_k e^{jk\omega_0 t}, \tag{1}$$

where $\omega_0 = 2\pi/T_0$ is the *fundamental* frequency. To determine the Fourier series coefficients $\{a_k\}$ from a periodic signal, we must evaluate the *analysis* integral for every integer value of k:

$$a_k = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} x(t) e^{-jk\omega_0 t} dt \tag{2}$$

where $T_0 = 2\pi/\omega_0$ is the *fundamental* period. If necessary, we can evaluate the analysis integral over any period; in (2) the choice $[-\frac{1}{2}T_0, \frac{1}{2}T_0]$ was a convenient one, but integrating over the interval $[0, T_0]$ would also give exactly the same answer.

The Fourier Series representation is extremely useful when studying the effects of an LTI filter, because the output signal is also periodic. The Fourier Series coefficients of the output signal $\{b_k\}$ are obtained by **multiplying** by the frequency response:

$$b_k = a_k H(j\omega_0 k) \tag{3}$$

where $H(j\omega_0 k)$ is the frequency response of the LTI system evaluated at the harmonics, $\omega = j\omega_0 k$.

2 Pre-Lab: Run the Frequency Response GUI

2.1 Sinusoidal Response (CLTIdemo)

In this demo, you can select an input signal that is a sinusoid, and see the change created by the frequency response. This demo reinforces the concept that "sinusoid in gives sinusoid out." Figure 1 shows the interface for the `CLTIdemo`.

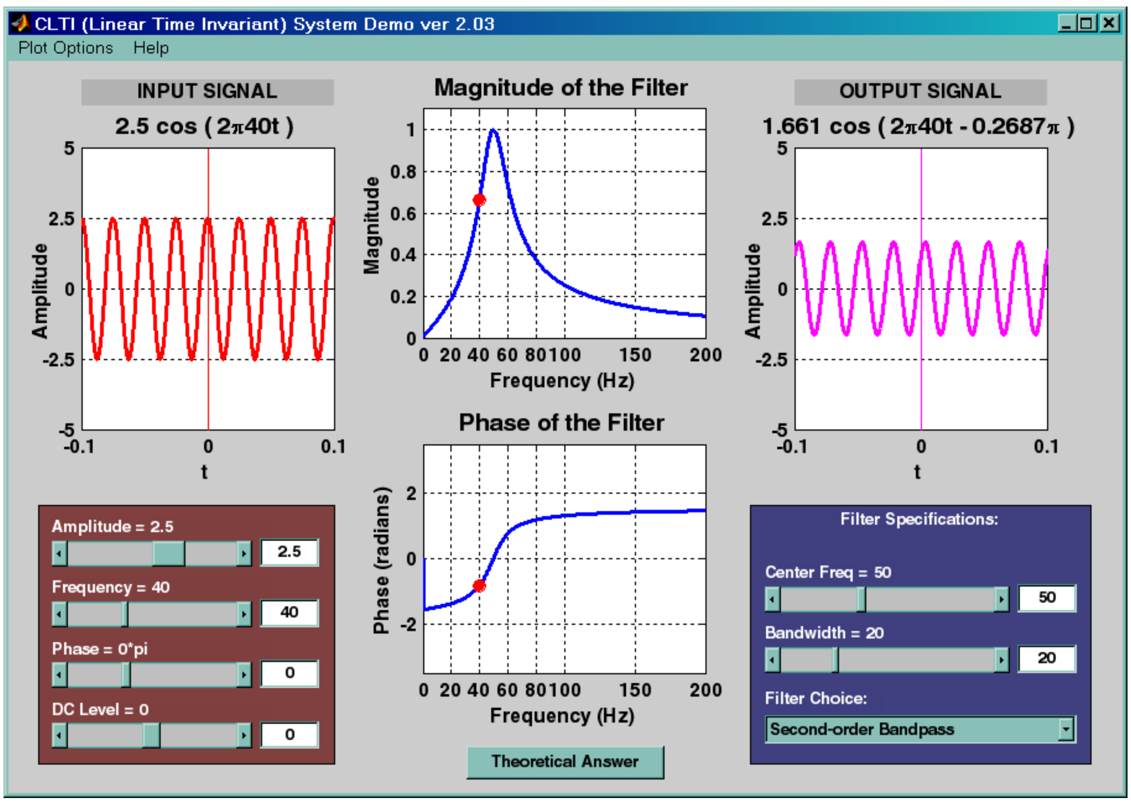

Figure 1: `CLTIdemo` interface for continuous-time frequency response.

We know that if the input to an LTI continuous-time system is a sinusoid of the form

$$x(t) = A + B\cos(\omega_0 t + \phi) \qquad -\infty < t < \infty \tag{4}$$

then the corresponding output is also a sinusoid:

$$y(t) = AH(j0) + B|H(j\omega_0)|\cos(\omega_0 t + \phi + \angle H(j\omega_0)) \qquad -\infty < t < \infty, \tag{5}$$

where

$$H(j\omega) = \int_{-\infty}^{\infty} h(t)e^{-j\omega t} d\omega \tag{6}$$

is the *frequency response* of the continuous-time LTI system. The **`CLTIdemo`** GUI illustrates this for a variety of simple analog filters.

(a) Use the `CLTIdemo` GUI to find the output of a first-order lowpass filter by selecting "First-Order Lowpass" from the menu and setting the cutoff frequency to 30 Hz. We have shown that the frequency response of this filter is

$$H(j\omega) = \frac{1}{j\omega + a} \tag{7}$$

where a is the cutoff frequency in rads/sec.

(b) Set the input to

$$x(t) = 1.0 + \cos(20\pi t).$$

Look at the output and compare its amplitude and phase to the input amplitude and phase. Click the box labeled "Theoretical Answer" to see a formula for the output $y(t)$.
Note: The GUI input frequencies are in hertz, which is $f = \omega/(2\pi)$; ω would have units of rad/s.

(c) Keeping the DC level and the amplitude of the cosine the same, use the slider to increase the input frequency and observe the change in the output. Keep increasing the slider until the frequency is $\omega = 80\pi$ rad/s (or $f = 40$ Hz). Compare the output in this case to the output at the original frequency of $\omega = 20\pi$.

(d) Repeat the previous part with the filter set to "Ideal Lowpass" with a cutoff frequency of 30 Hz. Start with the input signal from part (a).

(e) Set the frequency of the input back to $\omega = 20\pi$ and change the filter to "First-Order Highpass" with a cutoff frequency of 30 Hz. Observe the output as the frequency is increased. What is the DC component of the output? Does the amplitude of the output sinusoid get bigger or smaller as the frequency is increased?

(f) Convince yourself that the following frequency response is a first-order HPF:

$$H(j\omega) = \frac{j\omega}{j\omega + b}$$

where the parameter b is the cutoff frequency of the HPF in rad/s.

2.2 Sinusoidal Synthesis from Fourier Series Coefficients

The primary point of this Pre-Lab section is to recall information about Fourier Series and also to show that you can adapt your existing MATLAB functions to a new situation quickly.

In this project, we will use the Fourier Series coefficients to predict the response of a LTI system. It will be necessary to have two functions: one to calculate the Fourier Series coefficients, and another to synthesize a signal from a given set of Fourier Series coefficients. We will restrict our attention to the case of square wave signals whose definition over one period (T_0) is:

$$x(t) = \begin{cases} 1 & \text{for } 0 \le t \le T_1 \\ 0 & \text{for } T_1 < t < T_0 \end{cases} \tag{8}$$

In this case, you can derive a general expression for the Fourier Series coefficients as

$$a_k = \begin{cases} T_1/T_0 & \text{for } k = 0 \\ \dfrac{\sin(\pi k T_1/T_0)}{\pi k}\, e^{-j\pi k T_1/T_0} & \text{for } k \neq 0 \end{cases} \tag{9}$$

If you are mathematically inclined, you might want to work out the Fourier Series integral for $x(t)$ in Eq. (8) by hand to show that the $\{a_k\}$ values are samples of a "sinc" function.

(a) Write a MATLAB function that will evaluate the $\{a_k\}$ coefficients for a square wave using (9). The function call should look like:

```
ak = ak4sqwave( krange, T0, T1 )
```

The argument `krange` should be a vector that gives the set of k indices for evaluating the $\{a_k\}$ Fourier Series coefficients, e.g., `krange=[-5:5]`.

(b) For synthesis, you should modify a function that you have already developed in Lab #2 (`makecos.m`). Recall that the Fourier Series synthesis using $2N + 1$ coefficients would be

$$x_N(t) = \sum_{k=-N}^{N} a_k e^{j2\pi k f_0 t}$$

and $x_N(t)$ would be real-valued if the coefficients satisfy the conjugate-symmetry property, $a_{-k} = a_k^*$.

Write a function called `ak2sig` that will produce a time signal from a given set of Fourier Series coefficients. Here is the format for the function call:

```
[xx,tt] = ak2sig( ak, krange, Tperiod, tstart, dur, fs )
```

where the vectors `ak` and `krange` must have exactly the same length because `ak` will be the set of $\{a_k\}$ coefficients at the indices given in `krange`. The parameters `tstart` and `dur` define the starting time and duration of the time interval over which the signal will be synthesized, `Tperiod` is the period of the synthesized signal, and `fs` the sampling rate.

(c) Demonstrate that your MATLAB functions written in the previous two parts work correctly by generating $x_7(t)$ in the middle panel of Fig. 3-17 in the *SP-First* textbook.

3 Warm-up: Periodic Signal Filtered by an Analog Bandpass Filter

3.1 Square Wave Spectrum

In this part of the warm-up, the objective is to make a plot of the spectrum versus frequency for the square wave defined above.

(a) The first step is to get the Fourier Series coefficients. Utilize the M-file `ak4sqwave` to evaluate the $\{a_k\}$ coefficients for the square wave over the range of indices $k = -N, \ldots, -1, 0, 1, 2, \ldots, N$. This M-file returns the $\{a_k\}$ coefficients as a vector containing $2N+1$ elements.

Evaluate the Fourier Series coefficients $\{a_k\}$ given in Eq. (9) for $x(t)$ defined in Eq. (8) for the case where the parameters of the square wave are $T_0 = 0.05$ secs. and $T_1 = 0.02$ secs. Determine the $\{a_k\}$ coefficients for $N = 7$ and make a stem plot of the magnitude of the coefficients versus k.

(b) Make the same stem plot of the magnitude of the Fourier Series coefficients as in part (a), but convert the horizontal axis into frequency, so that it becomes a plot of the spectrum of the square wave.

Instructor Verification (separate page)

3.2 Fourier Synthesis

In this part, you must synthesize approximations to $x(t)$ using a finite number of Fourier Series coefficients $\{a_k\}$. Let N denote the largest index used, so that $2N+1$ is the number of terms used to form the signal.

(a) Use the `ak2sig` function from the Pre-Lab to create $x_N(t)$ for $N = 7$ and $N = 15$. Assume that the $\{a_k\}$ coefficients are those derived for the square wave with $T_0 = 0.05$ secs. and $T_1 = 0.02$ secs. For each of these synthesized signals, make a plot showing the synthesized signal and the original $x(t)$ on the same plot. Use a two-panel subplot to show the two cases in one figure window.
Note: The time interval for the plot should be three periods of the signal from $t = -T_0$ to $t = 2T_0$. The sampling rate should be quite a bit higher than the Nyquist rate which is dictated by the highest harmonic frequency. Over-sampling will be needed to "see" convergence.

(b) Explain how you are getting convergence as N increases. Where does the approximation error seem to be largest?

3.3 Frequency Response of an Analog Filter

In the lab project, you will use a continuous-time LTI system for filtering. In this section of the warm-up, we will investigate the following frequency response:

$$H(j\omega) = \frac{jb\omega}{(\omega_c^2 - \omega^2) + jb\omega} \tag{10}$$

where ω_c is the center frequency, and the parameter b controls the bandwidth of the filter.

(a) Make a plot of the magnitude and phase of $H(j\omega)$ versus ω in rad/s. Pick the parameters of the frequency response to be $\omega_c = 40\pi$, and $b = 20\pi$. In order to get values for the plot, you should evaluate the $H(j\omega)$ formula directly for a dense grid of frequencies. Use a range of frequencies that extends from -500 rad/s to $+500$ rad/s.[1] From the plot of $|H(j\omega)|$ versus ω, determine what kind of filter $H(j\omega)$ is.

Instructor Verification (separate page)

(b) Determine the peak value of the magnitude (frequency) response and the location of the peak. Use the algebraic form of the frequency response formula $H(j\omega)$ to explain that the peak value is correct.

3.4 Sinusoidal Response of BPF

The **`CLTIdemo`** GUI can implement the BPF defined by (10) if you choose the filter named "Second-Order Bandpass."

(a) Use the `CLTIdemo` GUI to create a second-order bandpass filter by selecting "Second-Order Bandpass" from the menu and setting the center frequency to 20 Hz, and the bandwidth frequency to 10 Hz.[2] This should be the same frequency response as in Section 3.3.

(b) Set the input signal to

$$x(t) = 1.0 + \cos(40\pi t)$$

Look at the output and compare its amplitudes and phases to the input amplitudes and phases. Click the box labeled "Theoretical Answer" and record the result.

(c) Now change the input signal to $x(t) = \cos(80\pi t)$, and record the numerical values of the output signal's amplitude and phase. Repeat for $x(t) = \cos(120\pi t)$, again recording the amplitude and phase of the output signal.

[1] You can plot the frequency response versus frequency in hertz or radians/sec. Either way is acceptable, but make sure that you label the horizontal axis.

[2] The `CLTIdemo` GUI will convert the frequencies from hertz to rad/s.

(d) Now consider the case where the input signal $x(t)$ is the square wave whose spectrum was plotted in Section 3.1. Use the information from the previous part (along with the values of $\{a_k\}$) to write the output $y(t)$ as a sum of cosines.

$$y(t) = B_0 + B_1 \cos(\omega_0 t + \psi_1) + B_2 \cos(2\omega_0 t + \psi_2) + B_3 \cos(3\omega_0 t + \psi_3) + \ldots$$

Use the values of the frequency response and the $\{a_k\}$ coefficients from Section 3.1 to determine the numerical values for ω_0, B_1, B_2, B_3, and ψ_1, ψ_2, and ψ_3.

Instructor Verification (separate page)

The calculation above amounts to an analysis of how you can "filter" the periodic input signal (the square wave from Section 3.1) through a continuous-time LTI system whose frequency response is given in Section 3.3. Since this is an analog system, we cannot do the actual filtering in MATLAB; instead, we can only calculate what the output signal would be by finding the Fourier Series of the output.

4 Lab Project: Design of a Frequency Synthesizer

Generating a perfect sinusoid can be difficult if we insist that the wave shape be exact. In fact, it is extremely rare to find hardware that can produce values that match $\cos(\omega t)$ perfectly. On the other hand, it is relatively easy to generate an "on-off" periodic signal with a switching circuit that is driven by a clock circuit. It turns out that we can filter any periodic signal (such as a square wave) to make a sinusoid. In this lab exercise, you will analyze a frequency synthesizer and design its parameters by using methods in the frequency domain.

The basic idea of the system is to use a bandpass filter to "clean up" the output of a "cheap" signal generator (Fig. 2). The signal generator only needs to be capable of making a periodic signal with the correct period.[3] Ideally, the bandpass filter will remove all the harmonics in $x(t)$ except one and make the output signal $y(t)$ a pure sinusoid.

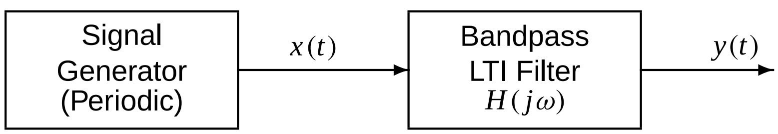

Figure 2: Block diagram representation of a Frequency Synthesizer.

The bandpass filter will be specified in the frequency domain by giving its frequency response[4] as:

$$H(j\omega) = \frac{j\omega G}{(\omega_c^2 - \omega^2) + j\omega b} \quad (11)$$

where the three parameters G, ω_c and b can be manipulated to change the characteristics of the frequency response. There are three characteristics of interest: location of the passband (center frequency), width of the passband, and maximum height of the passband (most likely at the center frequency). This particular frequency response corresponds to a resonant electric circuit that is relatively easy to build.

4.1 Fourier Analysis of the Input Waveform

Assume that the signal generator produces a periodic square-wave signal that is defined over one period as in (8). In order to do the system design in the frequency domain, we need to derive the Fourier Series of the square wave, and plot its spectrum.

[3] For example, the signal generator might produce a triangle wave or a square wave or a sawtooth wave for $x(t)$.

[4] $H(j\omega)$ is also the Fourier transform of $h(t)$.

(a) Use the Fourier Series function written previously (`ak4sqwave`) to evaluate the Fourier Series coefficients for $x(t)$ in (8) for the case where $T_0 = 25 \times 10^{-6}$ secs (25 μs) and $T_1 = 10 \times 10^{-6}$ secs (10 μs). Find the $\{a_k\}$ coefficients for $N = 7$ and make a stem plot of the magnitude of the Fourier coefficients versus k,

(b) Repeat the foregoing magnitude response plot, but make it versus frequency f in Hz.

(c) Finally, as a confirmation that you understand how to analyze this waveform, you should also derive the mathematical expression for the Fourier Series coefficients. Consult the Fourier Series presentation in Chapter 3 for a similar example that grinds out the integrals. Give your formula for the $\{a_k\}$ coefficients in polar form with a magnitude and phase.

4.2 Frequency Response of the Bandpass Filter

Now we must exhibit the frequency response characteristics of the bandpass filter in terms of its parameters: G, ω_c and b which were defined in (11).

(a) Write a MATLAB function that will evaluate $H(j\omega)$ for the BPF in (11). The function should have at least four input arguments: G, ω_c and b, as well as a frequency vector for ω. Use the following template for the function call in the M-file:

```
HH = BPFresponse(wc,b,G,wrange)
```

where `wrange` is a vector containing the set of frequencies where $H(j\omega)$ will be evaluated.

(b) Use the `BPFresponse` function to make a plot of the magnitude and phase of $H(j\omega)$ defined in (11) over the range $-300 \leq f \leq 300$ kHz, or equivalently $-6\pi \times 10^5 \leq \omega \leq 6\pi \times 10^5$ rad/s. Use the following values for the parameters of the BPF:

$$\omega_c = 2\pi(40 \times 10^3) \text{ rad/s} \qquad G = b = 2\pi(20 \times 10^3) \text{ rad/s}$$

4.3 Find the Output Signal and Spectrum

The Fourier coefficients of the output signal are $b_k = a_k H(jk\omega_0)$, because the theory of the frequency response tells us how to determine the exact output of the lowpass filter by tracking each sinusoidal component through the filter. If we use the $2N+1$ term approximation for synthesizing the output signal, the approximate output is

$$y_N(t) = \sum_{k=-N}^{N} b_k e^{jk\omega_0 t} = \sum_{k=-N}^{N} a_k H(jk\omega_0) e^{jk\omega_0 t} \tag{12}$$

where $\{a_k\}$ are the Fourier coefficients of $x(t)$, and $\{b_k\}$ are the Fourier coefficients of $y(t)$.

(a) *Frequency Domain:* Make a three-panel plot showing the ***spectrum*** of $x_N(t)$ in the top for $N = 7$; the magnitude of $H(j\omega)$ in the middle; and the ***spectrum*** of $y_N(t)$ in the bottom plot (for $N = 7$ also). For $H(j\omega)$ use the specific values for ω_c, G and b given in the previous section. Use the same horizontal frequency scale for all three plots so that they line up. Record the frequency where the peak is located in the frequency response, and explain how the peak location is related to the parameter ω_c. Explain how the output spectrum is produced from the input spectrum.

(b) *Time Domain:* Next, you should make a plot of the output signal in the time domain for $N = 7$, i.e., plot $y_7(t)$ versus t over the range $-T_0 \leq t \leq 2T_0$.

This requires that you evaluate the $\{b_k\}$ Fourier coefficients numerically and use `ak2sig` to synthesize $y_N(t)$. In this approach, use the Fourier coefficients $\{a_k\}$ that were evaluated numerically, and

then evaluate the frequency response $H(j\omega)$ at the appropriate frequencies. Recall that the Fourier synthesis of the output is given by Eq. (12).
Is the output signal a sinusoid? How close is $y(t)$ to being a sinusoid?

(c) Use your knowledge of the input Fourier series and $H(j\omega)$ in (11) to write a mathematical formula for the first-harmonic component of the output in terms of b, G and ω_c but don't assume that $b = G$. The result should be a sinusoid.
Note: Assume that the frequency response $H(j\omega)$ has its center frequency ω_c aligned with the fundamental frequency of $x(t)$.

(d) While keeping $b = 2\pi(20 \times 10^3)$, determine a new value for G so that the first-harmonic component of the output signal has an amplitude of 100.

Now we turn our attention to the design of the BPF to produce the desired output sinusoid with control over the amount of distortion.

4.4 Evaluate the Output Distortion

The objective of the design is to produce an output $y(t)$ that is a pure sinusoid at one given frequency. The approach is to use the system in Fig. 2 and design the bandpass filter to pass the first harmonic of $x(t)$ and reject all others. However, the frequency synthesizer will exhibit some distortion because the BPF in (11) can never be an ideal filter.

Therefore, we need to define a measure of the distortion in the output signal $y(t)$. There are many possible distortion measures, but we will use the worst-case percentage error between the desired output sinusoid $y_{\text{des}}(t)$ and the actual output. First of all, we define the *relative output error signal* as

$$e(t) = \frac{y(t) - y_{\text{des}}(t)}{A_{\text{des}}} \times 100\% \tag{13}$$

where $y(t)$ is the actual output of the BPF, $y_{\text{des}}(t)$ is the desired output signal, and A_{des} is the desired maximum amplitude of $y_{\text{des}}(t)$. The desired output signal is a pure sinusoid consisting of the first harmonic only, with a desired amplitude of $A_{\text{des}} = 100$, but an unspecified phase.

The *worst-case percentage error* is the maximum value of $e(t)$ in (13)

(a) For the output signal from Section 4.3, make a plot of the *relative output error signal* over three periods. This should be carried out by taking the difference between the desired sinusoidal output and the actual synthesized output which can be produced via a Fourier Series synthesis for $N = 7$. Mark the location(s) and value of the worst-case error on the plot.

(b) Now, determine the distortion as measured by the *worst-case percentage error.* Use the parameters given in Section 4.3.

4.5 Design in the Frequency Domain to Meet Distortion Specs

Now we can perform a general design of the frequency synthesizer for any specification on the amplitude of the output sinusoid and its distortion. Suppose that our design objective is to generate an output sinusoid whose specifications are:

$$\text{Amplitude} = 100 \qquad \text{Frequency} = 40 \times 10^3 \text{ Hz} \qquad \text{Distortion} < 3\%$$

The design specifications for the output frequency and the output amplitude above will determine the value of ω_c, and will tie the parameters b and G together.

When you made the plot of $y_7(t)$ in the previous section, you should have observed a distortion in the time-domain signal. In other words, the output was not a perfect sinusoid. The distortion in the output shows up as extra bumps in the waveform, and is due to all the higher harmonic terms in the output Fourier Series, but it is mostly due to the terms for $k = \pm 2$.

$$\begin{aligned} y(t) &= b_0 + \left(b_1 e^{j\omega_0 t} + b_{-1} e^{-j\omega_0 t}\right) + \sum_{k=2}^{\infty} \left(b_k e^{jk\omega_0 t} + b_{-k} e^{-jk\omega_0 t}\right) \\ &= B_0 + B_1 \cos(\omega_0 t + \psi_1) + B_2 \cos(2\omega_0 t + \psi_2) + \text{higher harmonic terms} \end{aligned}$$

Therefore, the output can be well approximated by considering the signal $y_2(t)$ that contains only the first two sinusoidal terms plus DC.

(a) You should derive the mathematical formula for $y_2(t)$ in terms of the unknown parameters G and b, assuming that ω_c has been chosen to pass the fundamental frequency. When you assess the quality of the frequency synthesis (below), you only need the mathematical formulas for B_1 and B_2 which are the magnitudes of the first two non-zero sinusoidal terms in $y_2(t)$ (the phases are not important).

(b) If we assume that all of the error is caused by the second harmonic alone, then we can write an expression for the "worst case" error in terms of G and b. Likewise, the desired amplitude specification on the first harmonic leads to a second equation for G and b to produce the correct amplitude. These two equations (in two unknowns) can be solved for the parameters of the BPF.

(c) Solve the two equations from the previous part for the specifications above.

(d) For the parameters found in the previous part, use MATLAB to find the Fourier Series coefficients of the output signal and make a plot of the spectrum versus frequency.

(e) Next, make a plot of the output waveform over three periods by doing a Fourier Series synthesis with $N = 7$. In addition, make a plot of the relative error signal over three periods. Put these two plots into a single figure window with `subplot()`. Mark the worst-case error on the error signal plot. Evaluate the Fourier coefficient b_2 for the second harmonic and write this value on the plot. Explain how the size of the error is related to the Fourier coefficient b_2.

(f) Does the output signal look like a pure sinusoid with the desired amplitude of 100? Explain why or why not.

(g) Discuss the validity of using one Fourier coefficient for the design. In other words, is your actual error signal exactly the size predicted by the design formula? Make a recommendation on whether or not you should include the *third* harmonic in the design. Do you think it would be easy to extend the design procedure to include the third harmonic? If you think you can derive these more accurate design equations, then describe how to do it (or go ahead and carry out the math).

Lab #12
ECE-2025
Spring-2003
INSTRUCTOR VERIFICATION PAGE

For each verification, be prepared to explain your answer and respond to other related questions that the lab TA's or professors might ask. Turn this page in at the end of your lab period.

Name: ______________________________ Date of Lab: __________

Part 3.1 Determine the numerical values of the Fourier coefficients $\{a_k\}$ for a square wave. Plot the spectrum of the square wave versus frequency.[5]

Verified:____________________ Date/Time:__________

Part 3.3 Plot the magnitude and phase of the frequency response of the continuous-time filter, $H(j\omega)$, defined in Eq. (10).

Verified:____________________ Date/Time:__________

Part 3.4 Find the $\{b_k\}$ Fourier Series coefficients of the output signal. Plot the spectrum versus frequency for the output signal. List the values of ω_0, B_k and ψ_k in the table below.

	$\omega_0 =$	
$k = 0$	$B_0 =$	
$k = 1$	$B_1 =$	$\psi_1 =$
$k = 2$	$B_2 =$	$\psi_2 =$
$k = 3$	$B_3 =$	$\psi_3 =$

Verified:____________________ Date/Time:__________

[5] It might be natural to plot a_k versus k, but when you show the spectrum the horizontal axis must be frequency (in Hz or rad/s).