

SAMPLE: EXPERIMENT 10

Bandpass Filter / Fourier Analysis

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Electric Experiments 2

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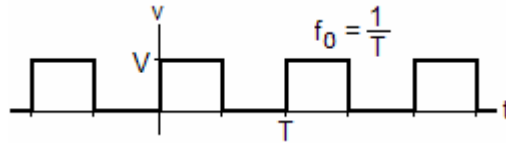
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Experiment 10: Band Pass Filter / Fourier Analysis

Introduction

According to Fourier analysis, a square wave may be considered to be a superposition of an infinite number of odd harmonic frequencies whose amplitudes decrease inversely with frequency. The fundamental, or lowest frequency, is the frequency of the square wave. The Fourier series of a square wave voltage, whose peak to peak amplitude is V , and average value is V_{AVE} , may be expressed as:

$$v(t) = V_{AVE} + \frac{2V}{\pi} \sum_n \frac{1}{n} \sin(2\pi n f_0 t),$$



where $n = \text{odd}$, and the fundamental frequency is f_0 .

If a square wave is applied to a band-pass filter, the wave's frequency components will be affected by the filter's frequency response. The filter's output frequency spectrum will be different than the input spectrum.

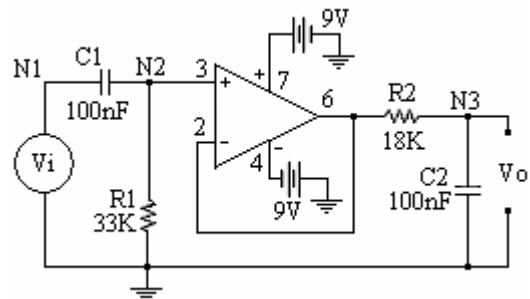
In Fourier analysis, the concepts of "time-domain" and "frequency-domain" are important to understand. The square wave input to the filter and output from the filter is considered to be expressed in the time-domain when its amplitude is expressed as a function of time. An oscilloscope normally plots waveforms in the time-domain. When the amplitude of the waveform is expressed as a function of frequency, it is expressed in the frequency-domain.

Using Fourier analysis, it is possible to convert time-domain expressions into the frequency-domain, and frequency-domain expressions into the time domain. This may be done by a computer "algorithm" called the "Fast Fourier Transform", or "FFT" for short, which is built into many analysis software, including *OrCad* (PSPICE), and *Maple*. The objective of this experiment is to investigate the relationship between time-domain analysis and frequency domain analysis. Acquired time-domain data will be compared to simulation data using *OrCad* and also compared to analysis using *Maple*. If GPIB or DAQ instrumentation is available, additional analysis may be performed using software such as *National Instruments LabVIEW* or *Tektronix Open Choice*.

The filter circuit used in this exercise is the same as used in experiment 9. The time domain results of experiment 9 can be compared to the frequency domain results of this experiment.

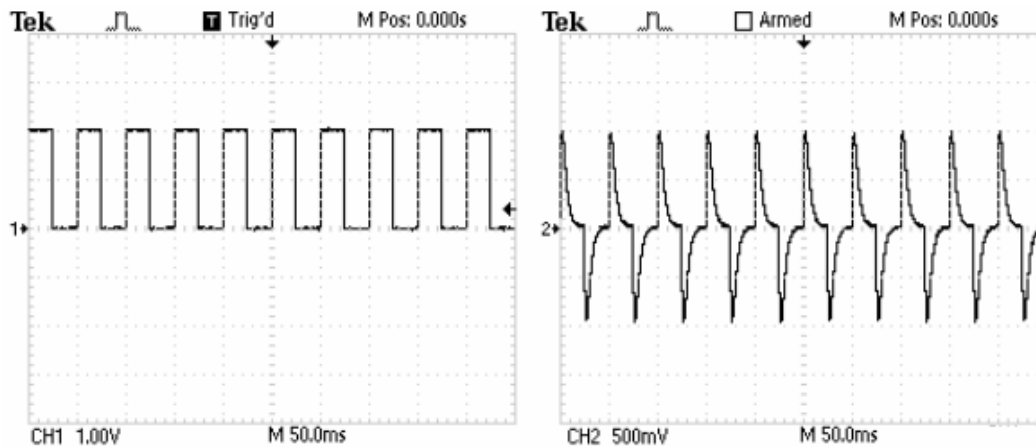
Procedure

The circuit on the right is the same as used in experiment 9. Input and output connections are shown for digital data acquisition. This experiment requires computer data acquisition equipment. The procedure below uses the Tektronix TDS1002 oscilloscope and Tektronix *Open Choice Desktop* software.



1. Connect the circuit on the right.
2. V_i is the function generator set to produce a 20 hertz, 2 volt peak to peak, square wave with a 1 volt offset (goes between 0 and 2 volts).

3. Connect the oscilloscope channel 1 to node N1 and Channel 2 to node N3. Set the oscilloscope to display 10 cycles of the input and output waveforms as shown below.



The time domain step response of the filter was analyzed in experiment 9. The response will be analyzed in the frequency domain in this lab exercise. It may be helpful to read the section on displaying the Fourier spectrum of your instruments' user manual.

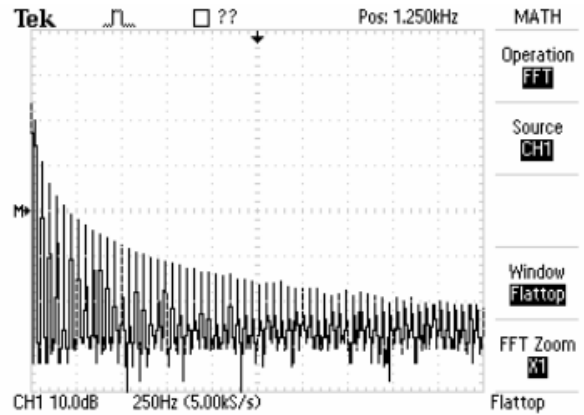
The TDS1002 uses a "Math FFT" algorithm to generate the Fourier spectrum from the acquired time domain data. The time domain waveform needs to be set up carefully to get an accurate display of the spectrum. Please note the following:

- The FFT spectrum is calculated from the center 2048 points of the time domain waveform. Since there are 2500 points in the 10 divisions of the entire display, the FFT is calculated from about the center 8 divisions.
- There is a tradeoff between frequency resolution and the bandwidth of the displayed spectrum (due to aliasing).
- The highest frequency (and bandwidth) that can be measured accurately by a digitizing oscilloscope is one half the sample rate (the Nyquist frequency).
- The TDS1002 transforms 2048 time domain points to 1024 frequency domain points resulting in a spectrum whose bandwidth is equal to the Nyquist frequency.
- The amplitude is displayed in dB where 0 dB is equal to 1 volt rms.

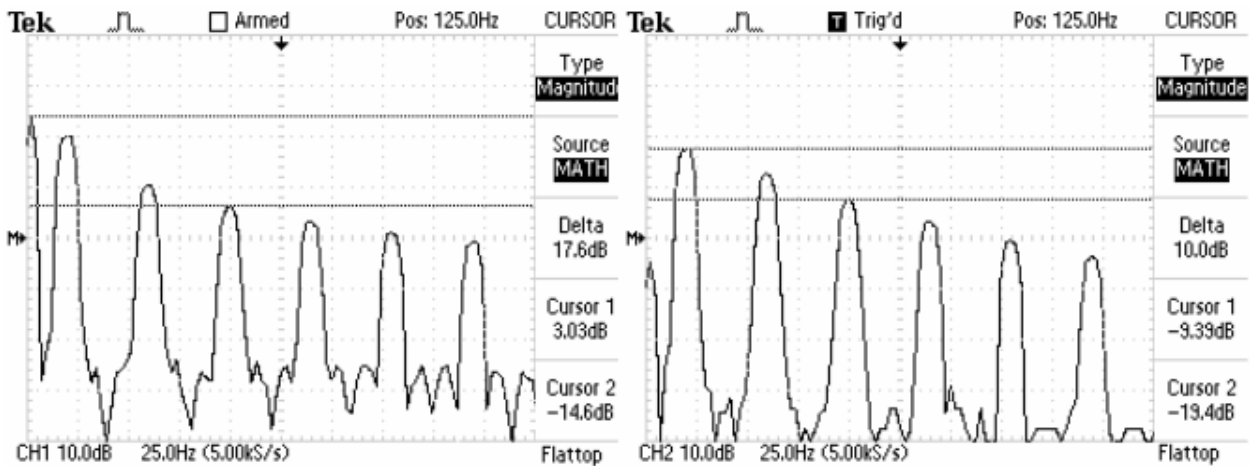
$$\text{dB}_{\text{VRMS}} = 20 \log \frac{V_{\text{RMS}}}{1\text{Vrms}} .$$
- Use the cursors to determine the amplitudes of the harmonics.
- Use data capture such as the TekXL toolbar for greater accuracy

- To obtain the FFT spectrum of channel 1:
Push the “MATH MENU” button,
select “FFT”, set the MATH FFT source
to channel 1.

Note the resulting display on the right.
The frequency scale is 250 Hz/Div.
The amplitude scale is in dB, where
0 dB = 1 volt rms.



- The frequency scale needs to be expanded by a factor of 10 in order to accurately measure the frequency components. This could be done with the time base (sec/div) control. However, this would reduce the bandwidth from 2500 Hz to 250 Hz. The TDS1002 has an FFT zoom control that provides a zoom up to 10. A zoom factor of 10 was used to obtain the input (CH1) and output (CH2) frequency spectrums shown below.



Cursors should be used to measure the amplitudes. The input frequency spectrum (CH1) above shows an rms amplitude value of 3 dB at 0 Hz. This corresponds to 1.414 volts rms, which is equal to a 1 volt average value (DC offset). The fifth harmonic at 100 Hz shows an amplitude of -14.6 dB. This corresponds to a peak voltage of 0.254 volts. The output frequency spectrum shows the fifth harmonic amplitude to be -19.4 dB.

- Display the first six harmonics of the input and output spectra of your band pass filter. Record the amplitude of the DC component and the first three harmonics of the input and output spectra (dc, 20 Hz, 60 Hz, 100 Hz).
Optional: Use the TekXL toolbar to obtain the data on your filter circuit.

V_{iDC} : _____ dB V_{i20} : _____ dB V_{i60} : _____ dB V_{i100} : _____ dB

V_{oDC} : _____ dB V_{o20} : _____ dB V_{o60} : _____ dB V_{o100} : _____ dB

Analysis

Note: Use a spreadsheet as applicable to do the analysis below.

1. The Fourier series of the input waveform is that of a 2 volt peak to peak amplitude square wave with a 1.0 volt dc component (average value). Determine the theoretical magnitude of the waveform's DC component, fundamental, third harmonic and fifth harmonic. Compare your measured values to the theoretical values by calculating the percent difference between them. You need to convert the dB values of your measurements to peak voltages to do this comparison.
2. Calculate the transfer function of the filter. Use phasor analysis to calculate the steady state output of the filter for the dc component, fundamental, third harmonic, and fifth harmonic (dc, 20 Hz, 60 Hz, 100 Hz). Compare your measured values to the theoretical values by calculating the percent difference between them. You need to convert the dB values of your measurements to peak voltages to do this comparison.
3. Simulate the circuit with *OrCad-Pspice* and use the FFT (Fast Fourier Transform) feature of *Probe* to obtain the frequency spectrum of the output. Compare the simulated results to your calculated results from step 2 above.
4. Simulate the circuit with *OrCad-Pspice* and use the frequency sweep analysis to obtain a Bode Plot of the filter's response (similar to plot shown below). Compare the results to analysis steps 2 and 3 above.

