

UNIVERSITY OF NORTH CAROLINA AT CHARLOTTE
Department of Electrical and Computer Engineering

EXPERIMENT 6 – ACTIVE FILTERS

OBJECTIVES

The purpose of this experiment is to familiarize the student with active filter networks.

INTRODUCTION

A filter is a device that passes electrical signals at certain frequencies or frequency range while preventing the passage of others. Filter circuits are used in a wide variety of applications. In the field of communication, band-pass filters are used in audio frequency range (0 KHz to 20 KHz) for modem and speech processing. High-frequency band-pass filters are used for channel selection in telephone central offices. Data acquisition systems usually require anti-aliasing low-pass filters as well as low-pass noise filters in their preceding signal conditioning stages. System power supplies often use band-rejection filters to suppress the 60-Hz line frequency and high frequency transients. At high frequencies (greater than 1 MHz), all of these filters usually consist of passive components such as inductors (L), resistors (R), and capacitors (C). They are then called RLC filters. In the lower frequency range (1 Hz to 1 MHz), however, the inductor value becomes very large and the inductor itself gets quite bulky, making economical production difficult. In these cases, active filters become important. An active filter is one that incorporates gain, and is often based on an op-amp as the active device in combination with resistors and capacitors to provide an LRC-like filter performance at low frequencies. The principle advantages over passive counterparts are:

- They are able to perform complex filtering functions without the need for inductors.
- Because gain is available it is possible to implement filters with characteristics that could not be obtained from passive filters, such as negative impedance conversion.
- By including electronically adjustable gain blocks, filters with electronically variable characteristics are possible. Active filters are used extensively for signal conditioning in electronic circuits.

Poles and Zeros

In analog circuit theory, circuits and signals are often analyzed in the s - domain. This means that the Laplace transform is applied to signals and circuits, enabling them to be analyzed algebraically rather than as time varying signal. Conversion to the s -domain is achieved via Laplace transforms. In essence, this means that instead of having a time-varying signal $y(t)$ and a circuit impulse response function $h(t)$ that relates the output to the input, we have a signal $Y(s)$ and a transfer function $H(s)$ that gives the output signal as $H(s)Y(s)$. The signal $Y(s)$ is a complex entity, reflecting the frequency content of the signal. The domain for s is given by, $-j\infty < s < +j\infty$ (i.e., frequencies are mapped to the complex frequency domain). The transfer function $H(s)$ provides the complex amplitude gain for each frequency as the signal passes from the input to

the output. Poles and zeros determine the system function within a constant. In the time domain, coefficients of a differential equation represent the poles and zeros of a system.

The pole/zero method of analysis is useful for studying the behavior of linear, time-invariant networks. In pole/zero analysis, a network is described by its network transfer function that, for any linear time-invariant network, can be written in the general form

$$T(s) = \frac{a_m s^m + a_{m-1} s^{m-1} + \dots + a_0}{s^n + b_{n-1} s^{n-1} + \dots + b_0}$$

Where the coefficients a and b are real numbers and the order m of the numerator is smaller than or equal to the order n of the denominator. The order of the denominator is called the order of the network. In the factorized form the previous equation will be

$$T(s) = a_m \frac{(s - Z_1)(s - Z_2) \dots (s - Z_m)}{(s - P_1)(s - P_2) \dots (s - P_n)}$$

Where a_m is a multiplicative constant, $Z_1, Z_2 \dots Z_m$ are the roots of the numerator polynomial, called the transfer-function zeros or transmission zeros of the network, and $P_1, P_2 \dots P_n$ are the roots of the denominator polynomial, called the transfer-function poles or natural modes of the network. The poles and zeros can be either real numbers or complex numbers. However, since the a and b coefficients are real numbers, the complex poles or zeros must occur in conjugate pairs. That is, if $5 + j3$ is a zero, then $5 - j3$ also must be a zero. A zero that is pure imaginary ($+jz$ or $-jz$) causes the transfer function $T(j\omega)$ to be exactly zero at $\omega = +z$ or $-z$.

First Order Filter

The general first-order transfer function is

$$T(s) = \frac{a_1 s + a_0}{s + \omega_0}$$

Where $-\omega_0$ is the location of the real pole, and ω_0 is called the pole frequency. The values of a_0 and a_1 determine the type of the filter. When a_1 is zero we have a first-order low-pass network with the transfer function

$$T(s) = \frac{a_0}{s + \omega_0}$$

In this case the dc gain is a_0/ω_0 , and ω_0 is the corner or 3-dB frequency. This transfer function has one zero at $s = \infty$. On the other hand, when a_0 is zero we have a first-order high-pass network with the transfer function

$$T(s) = \frac{a_1 s}{s + \omega_0}$$

Here we have one zero at $s=0$ and a 3-dB or corner frequency of ω_0 .

PRELAB

1. Derive the voltage transfer functions for the circuit in Figure 8-1.

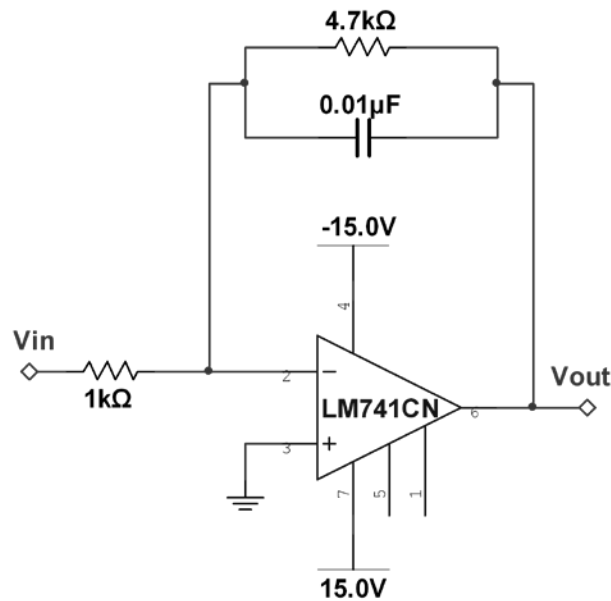


Figure 6-1 Low-pass filter configuration

2. Using MATLAB, PSPICE or Multisim plot the voltage transfer function vs. frequency for the network in Figure 6-1.
3. Design a high-pass first-order filter with a pass-band gain of -1.5 , and select a cutoff frequency between 1kHz and 10kHz. Use any standard components available in your laboratory for the design.
4. Using MATLAB, PSPICE or Multisim plot the voltage transfer function vs. frequency for the high-pass first-order circuit you designed.

PROCEDURE

1. Connect the low-pass circuit shown in Figure 6-1.
2. In the NI ELVIS board connect both FGen and AI0+ to Vin of the circuit. Then, connect AI0- to pin 3 of the LM741. AI0 is the stimulus channel.
3. Connect AI1+ to Vout and connect AI1- to pin 3 of the LM741. AI1 is the response channel.
4. Use the bode analyzer in ELVIS to implement a frequency sweep and obtain the bode plot. Record the cut-off frequency and save the bode plot.
 - a. Use a start frequency of at least one-tenth of the filter cut-off frequency.
 - b. Use a stop frequency of at least ten times the filter cut-off frequency.



Figure 6-2 NI ELVIS Bode Analyzer

5. Connect the high- pass first-order filter you designed in the prelab. Repeat steps 2 to 4 of the lab procedure.

POST-LAB

Post-Lab questions must be answered in each experiment's laboratory report.

1. Explain any differences between the experimental results and the calculated results for both the high-pass and the low-pass filters.
 - a. Compute the percent error between the cut-off frequency predicted from the transfer function and the actual cut-off frequency measured in the lab.
 - b. Compute the error between the theoretical and measured pass band gain.
 - c. What could be the cause of these errors?
 - d. At high frequencies the gain curve of the high-pass filter data may roll off or even turn back up. This behavior is not predicted by the transfer function for this circuit. Can you explain what might cause this behavior?

Be sure to include all items from the post-lab exercise above in your written lab report.