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Frequency Response by AC Analysis on One-Pole Low Pass Filter for the Modification of Fixed Physically Constraints and to Manipulate Sound into a Different One

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Abstract: Passive filters are constructed using Resistor-Capacitor networks in low frequency applications up to 100 kHz. The capacitor lags increases as the input frequency increases and the circuit becomes more out of phase. Low pass filter has wide application in sound manipulation. Sound Sculpting had created a new environment for virtual musical instruments.

Keywords: Alternating Current Analysis, Filter, Frequency Response, Sculpting, Spice, Synthesizers

1. Introduction

Determined by the RC time constant a low-pass filter circuit passes the low frequency signals and attenuates the signal frequencies greater than the cutoff frequency [1]. To find the frequency response of a circuit using linear technology spice software, alternating current analysis is used. This analysis mode is useful for filters, networks, stability and noise considerations. The small signal alternating current analysis computes the alternating current complex node voltages as a function of frequency. After finding the direct current operating point of the circuit, linearized small signal models are found for this operating point [2]. Using independent voltage and current sources, the resultant linearized circuit is solved in the frequency domain over the specified range of frequencies. Since this filter has only one reactive component, the capacitor in the circuit hence it is also known as a first-order filter [3]. In sound sculpting single pure sound or number of layered sounds are creatively manipulated into a completely different one. Synthesizers are most commonly used for this. A filter is designed with the frequency response specification. The attenuation amount of frequency depends on the design of filter.

2. Scope of Approach

A low-pass filter circuit is used to block harmonic emissions in radio transmitters. It is very useful for the sculpting of sound created by synthesizers. Instead of fixed physically constraints sound sculpting makes it possible to design a musical instrument according to one's requirement or desired musical constraints. By subtractive synthesis the sound signal partials can be attenuated by filters so that the timbre gets altered.

3. Calculations

In a RC low-pass filter circuit the capacitor reactance varies inversely with the frequency, but the value of the resistor remains fixed as the frequency changes [4]. At low frequency the capacitive reactance of the capacitor will be more compared to the resistive value of the resistor. The voltage across the capacitor and resistor is

$$Vc\ (s) = \{(1/Cs) / (R+1/Cs)\}\ Vin(s) = \{(1) / (1+RCs)\}\ Vin(s)$$

$$Vr\ (s) = \{(R) / (R+1/Cs)\}\ Vin(s) = \{(RCs) / (1+RCs)\}\ Vin(s)$$

The transfer function of the voltage across the capacitor and resistor from the input is

$$Hc\ (s) = Vc\ (s) /Vin\ (s) = (1) / (1+RCs)$$

$$Hr\ (s) = Vr\ (s) /Vin\ (s) = (RCs) / (1+RCs)$$

The frequency response of the filter is flat for low frequencies resulting in a gain of nearly unity, until it reaches the cut off frequency [5]. The magnitude of gain across the two components is

$$Gc = \left| Hc\,(j\omega) \right| = \left| \{ Vc\,(j\omega) \} / \{ Vin\,(j\omega) \} \right| = (1) / \sqrt{\{ 1 + (\omega RC)^2 \}}$$

$$Gr = \left| Hr\,(j\omega) \right| = \left| \{ Vr\,(j\omega) \} / \{ Vin\,(j\omega) \} \right| = (\omega RC) / \sqrt{\{ 1 + (\omega RC)^2 \}}$$

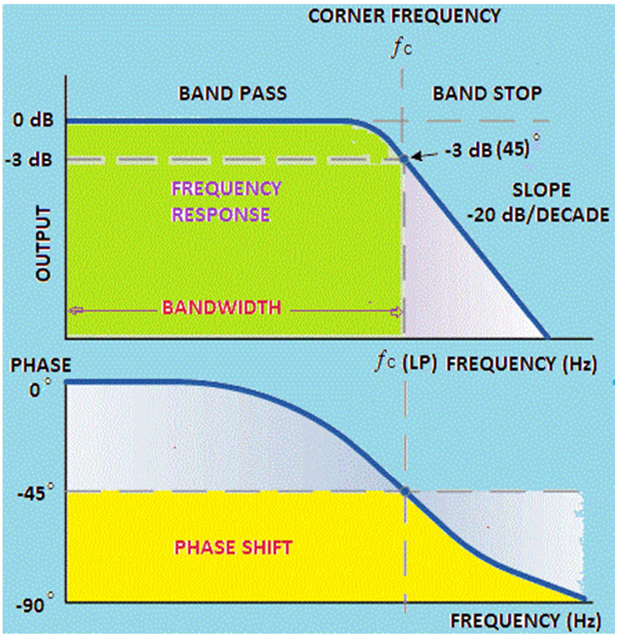

Figure 1. *Frequency Response of 1st order Low Pass Filter.*

Figure 1 shows the frequency response of the 1st order Low Pass Filter. The Cut-off frequency is the frequency point where the capacitive reactance and resistance are equal. The signal output will be attenuated to 70.7% of the signal input [6]. After cut off frequency the response of the filter circuit decreases to zero at a slope of -20dB/ Decade. The current in the circuit is

$$I\,(s) = \{Vin\,(s)\} / \{R + (1/Cs)\} = \{Cs / (1 + RCs)\}\, Vin\,(s)$$

A range of frequencies below a certain value can be allowed to pass through the circuit by selecting the correct resistor and capacitor values. The impulse response for the capacitor voltage and resistor voltage is

$$Hc\,(t) = (1/RC)\, e^{-t/RC}\, u\,(t) = (1/T)\, e^{-t/T}\, u\,(t)$$

$$HR\,(t) = \{\delta\,(t) - (1/RC)\, e^{-t/RC}\, u\,(t)\} = \{\delta\,(t) - (1/T)\, e^{-t/T}\, u\,(t)$$

The band pass zone also represents the bandwidth of the filter. Signal frequencies above fc are in the band stop zone. It takes some time to charge the plates of the capacitor as the input voltage changes. Due to the capacitor, the phase angle Φ of the output signal lags behind the input [7]. The phase angles are:

$$\Phi c = \text{angle } Hc\,(j\omega) = \tan^{-1}(-\omega RC)$$

$$\Phi r = \text{angle } Hr\,(j\omega) = \tan^{-1}(1/\omega RC)$$

At the -3dB cut-off frequency it will be -45° out of phase. As the input frequency becomes higher the capacitor lags increases and the circuit becomes more and more out of phase [8].

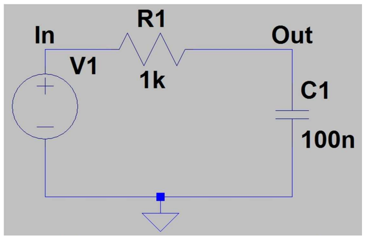

Figure 2. *First order low pass filter.*

4. Analysis

A custom made musical instrument has the ability of extension, compared to an existing base instrument. The sound of a base instrument is fixed for which it is designed and the required variation of sound is not possible according to the desired limits. More ever the frequent variation or modifications is not possible. The sound sculpting allows altering the existing sound output. The development of the physical dimensions of an instrument constitutes the development in the design of filter. Filter design have a great impact in the sound production and regeneration. The appropriate flow of desired signal can be achieved by using filters. A sound can be varied by increasing or decreasing the low frequency. Increasing the low frequency in a sound can make it much darker and heavier. A low-pass filter passes the signals which have lower frequency and attenuates the signals which have higher frequencies than the cutoff frequency [9]. It is determined by the RC time constant. In this filter circuit the value of the resistor remains constant as the frequency changes but the reactance of capacitor varies inversely with the frequency.

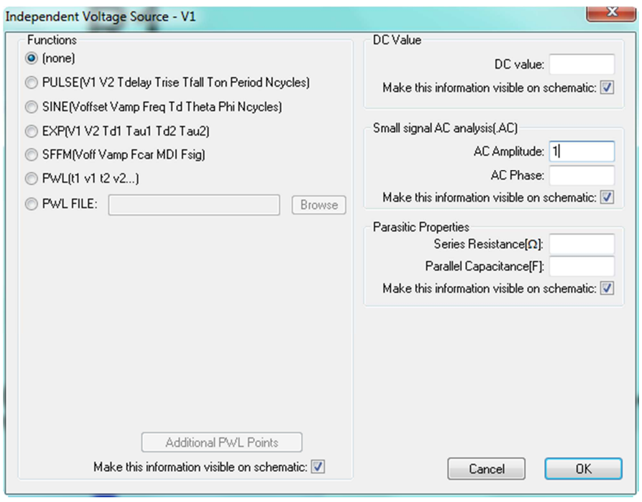

Figure 3. *Independent voltage source at alternating current analysis mode.*

When the frequency is low, the capacitor's capacitive reactance will be more compared to the resistive value of the resistor. Hence the voltage across the capacitor will be much greater than the voltage drop developed across the resistor. The transition band width depends upon the slope of reduction of signal by the filter. For an acoustic piano the depth will be at 100Hz whereas for a bass guitar the depth will be at 80 Hz. In the same way an electric guitar hum at 50Hz and fullness at 240Hz [10]. The transfer function of one pole filter is the reciprocal of that of a one zero. It can be used to turn those signals which are noisy. Pole represents the single frequency point for raising the gain above and zero represents the single frequency point for the gain to fell below.

Using linear technology spice software the first order low pass filter is drawn as shown in figure 2. In/Out labels are marked at the nodes and the ground terminal is applied to the circuit. The value of resistor is selected at 1 kilo ohm and capacitor at 100 nano farad. The voltage source is selected and the function is set to none. Figure 3 shows the Value of independent voltage source used at none mode. The direct current value is used by the software to calculate the operating point before linearization [11]. Direct current value will be 0 if it is left empty. In the small signal alternating current analysis, the alternating current amplitude is kept at 1. If one pole low pass filter is fed with impulse it will result in an exponential decay. In a guitar amplifier the RC filters are also used to decouple pre-amp stages [12]. The noise signals can be rejected.

When the source is set the alternating current analysis is then selected as shown in figure 4. The smooth curves depend upon the number of points per decade. So the number of points per decade should be kept at sufficient value for smooth curves. The start frequency should be at least 1 decade lower than the theoretical low cut off frequency and the stop frequency should be at least 1 decade higher than the theoretical low cut off frequency. The type of sweep is selected at decade, number of points per decade at 100, start frequency at 10 and stop frequency at 1 Meg. A filter's frequency response is represented by bode plot, rate by roll off and characterized by cut off frequency [13].

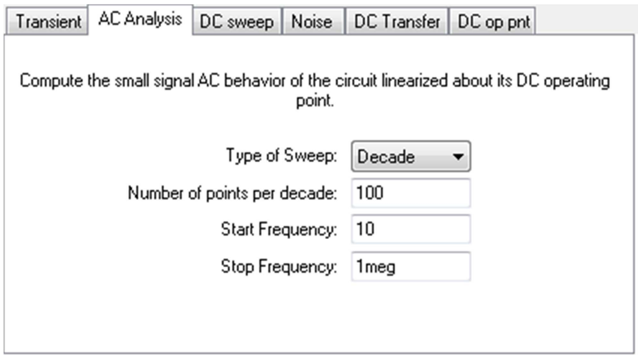

Figure 4. *Alternating current analysis.*

In the expression editor shown in figure 5, the attached cursor is selected to 1st. The algebraic expression to plot is fed as V[out]/V[in]. Now the simulation is run by selecting V[out]/V[in]. At high frequencies the capacitor behaves like a short circuit and the output drops to zero [14]. When the time constant of the circuit is very long it behaves like an integrator, because the output voltage is directly proportional to the integral of voltage input [15]. The first order filters magnitude bode plot will be a horizontal line below cut off frequency and above cut off frequency it will be a diagonal line [16].

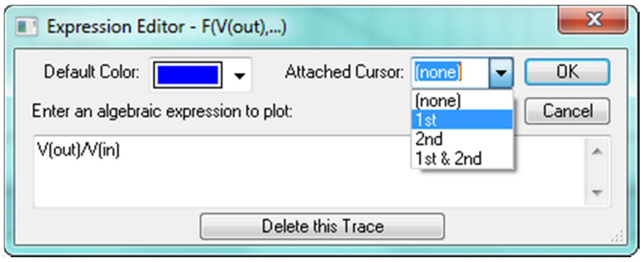

Figure 5. *Expression editor.*

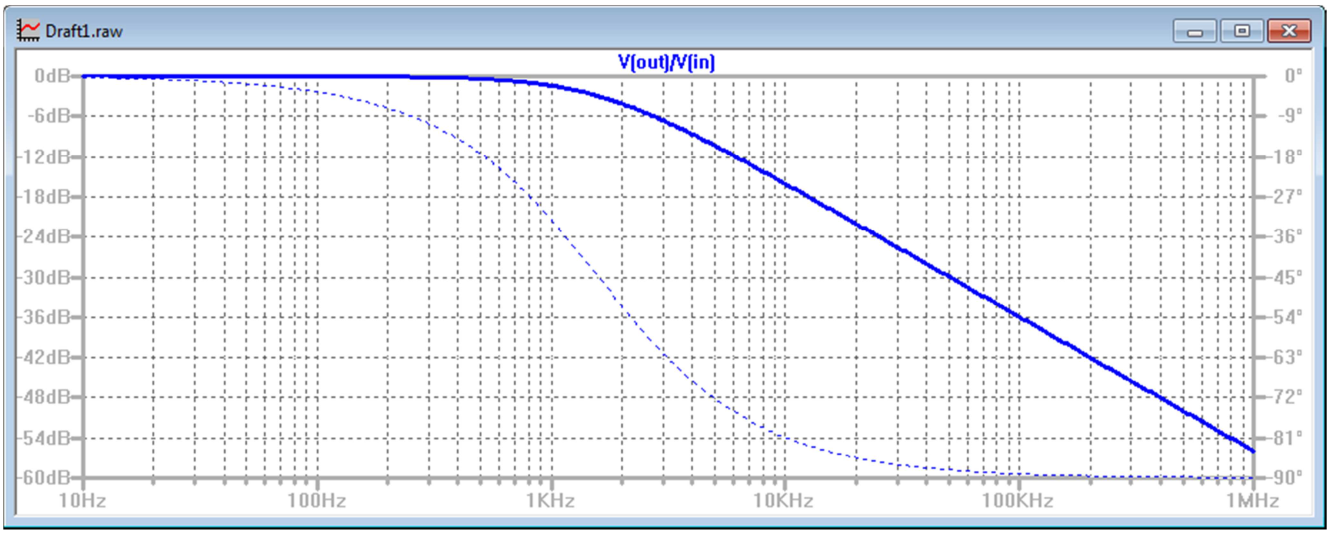

Figure 6. *V out/ V in.*

Figure 6 shows the ratio between the output signal voltage and input signal voltage of the RC low pass circuit. The plot displays the signal voltage ratio and the relation between the phases of the two signals. The magnitude of -1.44507 db is at 1 KHz along with phase of -32.1419° and group delay of 71.6962µs. By changing the value of resistor and capacitor, various effects on output can be seen and the required output can be achieved. To reduce or increase the signals above or below the set frequency, shelving filters can be used [17]. It implements a first order response and allows the boosting or cut of frequency above or beneath a certain point.

5. Conclusion

The frequency response of the filter is flat for low frequencies resulting in a gain of nearly unity. After the cut off frequency, the response of the filter circuit decreases to zero. On simulating at 10 Hz, the magnitude is -171.449µdB, phase is -359.995m° and group delay is 99.9961µs whereas at 100 Hz, the magnitude is -17.1115mdB, phase is -3.59527° and group delay is 99.6067µs. At 100 KHz, the magnitude is -35.9647dB, phase is -89.0882° and group delay is 25.3373ns. Combination of different values can be used for passing and manipulation of the desired low frequency signals from audio or musical instruments.

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Biography

Mr. Abilash is involved in R&D with various research papers in IJSRD, IJRASET, IRJET, IJNN, IJAEAS, IJAPST and SIIJ. He did MTech-VLSI, BTech-ETE, advanced studies in IE, IA, DBME, SAHC, CSHAM, FPM, IEEE-Mechatronics, IIT-ETFET, ITP and IPD. He is LMIETE, LMIEI, advisory and editorial board in AIS-IJEECS, JRCVG, IJAREEC, IJLEMR, IJVSP, IJCTT, IRJET, IAETRJ, AJSMR, JMSCR, IJSTRE, AJASET, BIJ, IERJ, ISJ, IJEMR, IJRG and reviewer in GJRE, AJNRA, IJERS, CI, ASTESJ, IJCT, IJAEGT, OPJ. His research area includes digital systems, artificial intelligence, sound engineering, mechatronics, nanotechnology, medical and polymer science.