Modifying Fs an Qt of an Electro-dynamic loudspeaker

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Theory

An Electro-dynamic loudspeaker has a transfer function of a second order high pass filter. The normalized form of this transfer function is:

$$F(s) = \frac{s^2}{s^2 + \frac{w_{0S}}{Q_{TS}} s + w_{0S}^2} \qquad (1)$$

For a loudspeaker $w_{0S} = 2 \ f_s$ and $Q_{TS} = Q_{ts}$.

Sometimes fs and Qts do not have the values we desire. Our goal is a transfer function with a different $Q_{ts} = Q_{NS}$ and a different $f_s = f_{NS}$ of the form:

$$F(s) = \frac{s^2}{s^2 + \frac{w_{NS}}{Q_{NS}} s + w_{NS}^2} \qquad (2)$$

We can get this from our existing loudspeaker with transfer function (1) by putting an electronic filter in front of the power amplifier that has a transfer function:

$$F(s) = \frac{s^2 + \frac{w_{0S}}{Q_{TS}} s + w_0^2}{s^2 + \frac{w_N}{Q_N} s + w_N^2} \qquad (3)$$

This function can be rewritten as the sum of a high pass, a band pass and a low pass filter as follows:

$$F(s) = \frac{s^2}{s^2 + \frac{w_N}{Q_N} s + w_N^2} + \frac{Q_N w_{0S}}{Q_{TS} w_N} \frac{\frac{w_N}{Q_N} s}{s^2 + \frac{w_N}{Q_N} s + w_N^2} + \left(\frac{w_{0S}}{w_N} \right)^2 \frac{w_N^2}{s^2 + \frac{w_N}{Q_N} s + w_N^2} \qquad (4)$$

The gain factor for the low pass filter then becomes:

$$K_1 = \left(\frac{w_{0S}}{w_N} \right)^2 = \left(\frac{fs}{f_N} \right)^2 \qquad (5)$$

The gain factor for the band pass filter:

$$K_2 = \frac{Q_N w_{0S}}{Q_{TS} w_N} = \frac{Q_N f_S}{Q_{TS} f_N} \qquad (6)$$

The component values for the filters depend solely on the requirements of the target transfer function (2) and not on the loudspeaker properties. The filter is tuned to a particular loudspeaker by just two gain factors. The resulting block-diagram becomes the following:

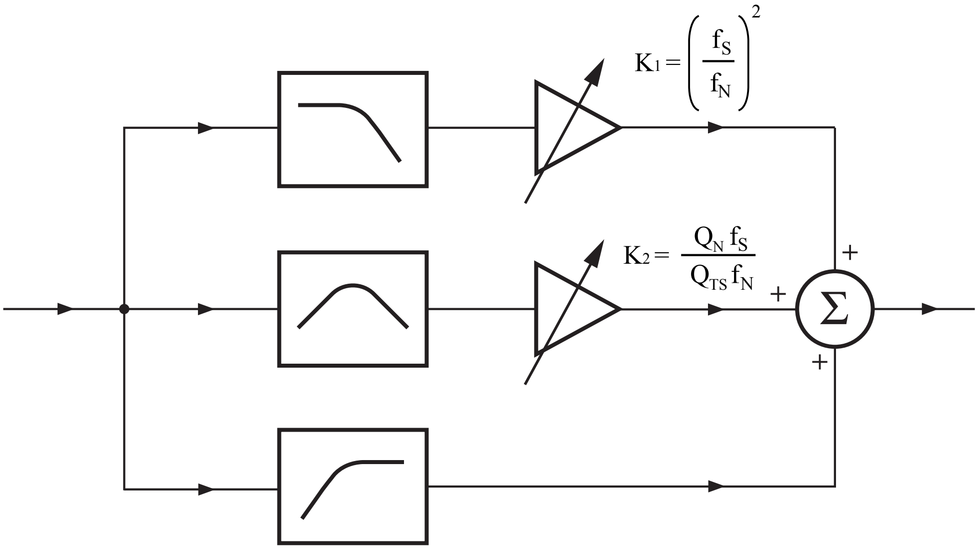


Figure 1. Realization of LS correction filter

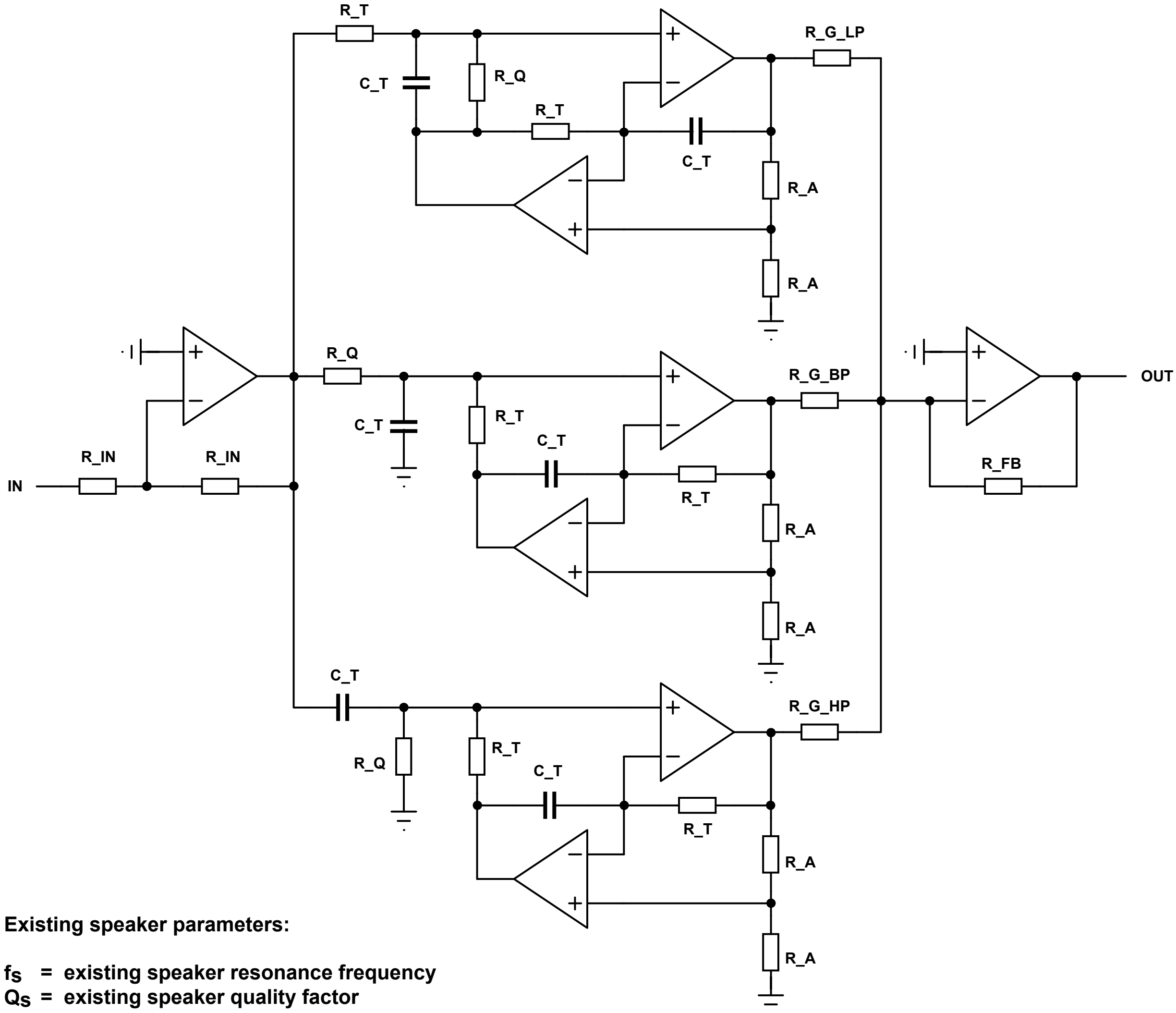
Due to aging, the fs of cone loudspeakers usually lowers during time. By making the gain factors adjustable by trim potentiometers, the filter can easily be re-adjusted. When this filter is used for extending bass response the gain of the low pass filter need to be higher than one and depending on the Q-factors, the gain of the band pass filter usually also needs to be higher than one. Keep in mind that active filters (especially the band pass filter) can have inverting responses.

Alignment

Given a Fs and Qts of the actual speaker and a F_N and Q_N to be wanted, first dimension the Lp, Bp and Hp filters for F_N and Q_N. Now define K_1 and K_2. When K_1 and K_2 are made adjustable by use of trim potentiometers, they can be trimmed accurately. The sum of the Lp and Hp has a sharp null in the response at Fs. Temporarily remove the band pass branch from the summing point. Now adjust K_1 for a minimum at Fs. (Not at F_N !). Reconnect the Bp branch and disconnect the Hp and Lp branch. Adjust the gain (K_2) at F_N to the value given in (6). As a more accurate alternative K_2 can be adjusted at Fs to give the response value:

$$\left|A_{Fs}\right| = \frac{1}{Qs\sqrt{\left(1\ \frac{F_N^2}{Fs^2}\right)^2 + \left(\frac{F_N}{Fs\ Q_N}\right)^2}} \qquad (7)$$

BiQuad LS Correction filter using Fliege filters



Existing speaker parameters:

f_s = existing speaker resonance frequency
Q_s = existing speaker quality factor

Target parameters:

f_n = new resonance frequency
Q_n = new quality factor

Component values for a nominal gain of 1:

Chose R_IN = input resistance
Chose R_T = filter tuning resistance
Chose R_FB = gain setting resistance
Chose R_A = all R_A are equal

$$C\_T = \frac{1}{2\pi f_n \times R\_T} \qquad R\_Q = R\_T \times Q_n$$

$$R\_G\_LP = 2 \times R\_FB \times \left(\frac{f_n}{f_s}\right)^2 \qquad R\_G\_BP = 2 \times R\_FB \times \frac{Q_s f_n}{Q_n f_s} \qquad R\_G\_HP = 2 \times R\_FB$$