Sapienza
Università di Roma

Fixed Gain Active Band Pass or Cut Filter Design

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Abstract

The aim of this project is to design a pass band filter which presents a fixed gain upon Q or ω_0 variations. After a theoretical treatment, the results obtained will be implemented on a Deliyannis active band pass filter. Next, a structural model for a reverse filter will be presented: theoretical mathematical results will lead to band cut filter with 0 dB gain at the resonance ω_0 and a fixed gain outside the cutting band. The project will consist in an first analytical analysis through the Matlab environment and subsequently in simulations through the OrCAD PSpice environment with real implementations of the mathematical results analytically obtained.

Contents

Chapter 1

Band Pass Filter

1.1 Theory of Biquads

Second order filters, commonly known as Biquads, are characterized by the following equation:

$$H(s) = \frac{a_2 s^2 + a_1 s + a_0}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2} \tag{1.1}$$

where ω_0 represents the system natural resonance pulsation, defined as the pulsation to which the system tends to oscillate in the absence of any driving or damping force, and Q represents the quality factor, which is defined a dimensionless parameter that describes how under-damped an oscillator or resonator is, and characterizes a resonator's bandwidth relative to its resonance pulsation. In other wards it describes the ratio of the energy stored in the oscillating resonator to the energy dissipated per cycle by damping processes and is related to the damping ratio by the equation:

$$Q = \frac{1}{2\zeta} = \frac{\omega_0}{\Delta\omega} \tag{1.2}$$

The damping ratio ζ is a dimensionless measure describing how oscillations in a system decay after a disturbance. Many systems exhibit oscillatory behaviour when they are disturbed from their position of static equilibrium, overshooting bounces while trying to return to their equilibrium position. Sometimes losses damp the system and can cause the oscillations to gradually decay in amplitude towards zero or attenuate at least. The damping ratio is then a measure of describing how rapidly the oscillations decay from one bounce to the next. The damping ratio characterizes the system making it undamped (ζ=0), under damped (ζ<1) through critically damped (ζ=1) to over-damped (ζ>1).
What has just been said describes the importance of having complex conjugate poles in the equation (1.1), as only in that case the system oscillates. These poles are therefore obtained by:

$$p_{1,2} = -\frac{\omega_0}{2Q} \pm \jmath\omega_0\sqrt{1 - \frac{1}{4Q^2}} \tag{1.3}$$

In order to obtain complex conjugate poles, the root argument must be positive:

$$1 - \frac{1}{4Q^2} > 0 \tag{1.4}$$

that leads to a constraint on quality factor Q:

$$Q > \frac{1}{2} \tag{1.5}$$

In general, sticking to the equations (1.1) and (1.3), depending on whether there are complex conjugate poles or not, we can distinguish different cases:

- **Low Quality Factor**, Q $< \frac{1}{2}$, is said to be ***overdamped***. Such a system doesn't oscillate at all, but when displaced from its equilibrium steady-state output it returns to it by exponential decay, approaching the steady state value asymptotically.
- **High Quality Factor**, Q $> \frac{1}{2}$, is said to be ***underdamped***. Underdamped systems combine oscillation at a specific frequency with a decay of the amplitude of the signal. Under-damped systems with a low quality factor, a little above Q=$\frac{1}{2}$, may oscillate only once or a few times before dying out. As the quality factor increases, the relative amount of damping decreases.
- **Intermediate Quality Factor**, Q $= \frac{1}{2}$, is said to be ***critically damped***. Like an over-damped system, the output does not oscillate, and does not overshoot its steady-state output. Like an under-damped response, the output of such a system responds quickly to a unit step input. Critical damping results in the fastest response (approach to the final value) possible without overshoot.

Indeed, according to (1.2), higher values of the quality factor Q mean lower values of the damping ratio ζ, increasing the peak response, and viceversa.

According on what has just been described it is possible to distinguish the different filter topologies.

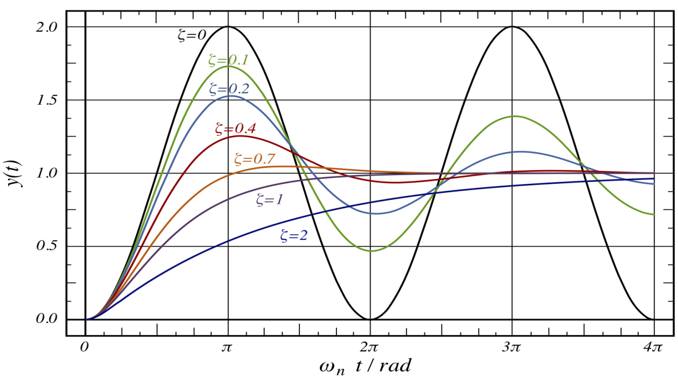

Figure 1.1. Damping Ratios

1.2 Fixed Gain Active Band Pass Filter

A band pass filter (BPF) is an electronic filter that passes frequencies within a certain range and attenuates the outside ones, and the attenuation amount of each frequency depends by the filter design. The ideal transfer function of a modelled band pass filter is:

$$BPF(s) = \frac{a_1 s}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2} \tag{1.6}$$

where the ω_0 corresponds to the peak pulsation ω_{peak}, and the passing band is equal to $\frac{\omega_0}{Q}$. Accordingly,the more Q value increases, the more the passband narrows. In a band pass filter design it is important to fulfil the system specifications in terms of ω_0 and Q. However, once these values have been fixed, the filter gain is ω_0 and Q dependent, unless the parameter a_1 is opportunely sized. The band pass filter gain Δ_v^{BPF} at the resonance ω_0 is thus given by:

$$BPF(s)\Big|_{s=\jmath\omega_0} = \frac{a_1 \jmath\omega_0}{-\omega_0^2 + \frac{\omega_0}{Q}\jmath\omega_0 + \omega_0^2} = a_1\frac{Q}{\omega_0} \tag{1.7}$$

Therefore, once known the requested gain value G_{dB} for the band pass filter, the a_1 value must then be:

$$a_1 = \frac{\omega_0}{Q}10^{\frac{G_{dB}}{20}} \tag{1.8}$$

The following graphs respectively represent the gain behaviour of an ideal band pass filter with or without the fixed gain request:

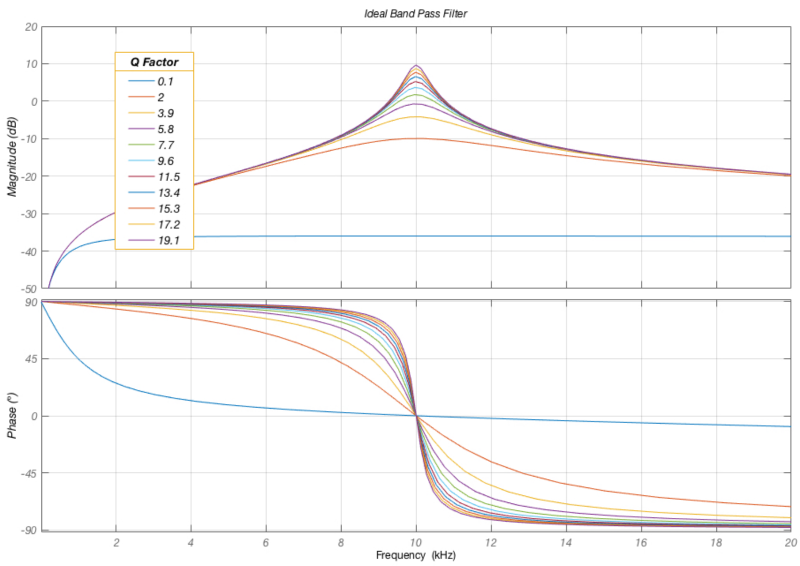

Figure 1.2. Ideal Band Pass Filter

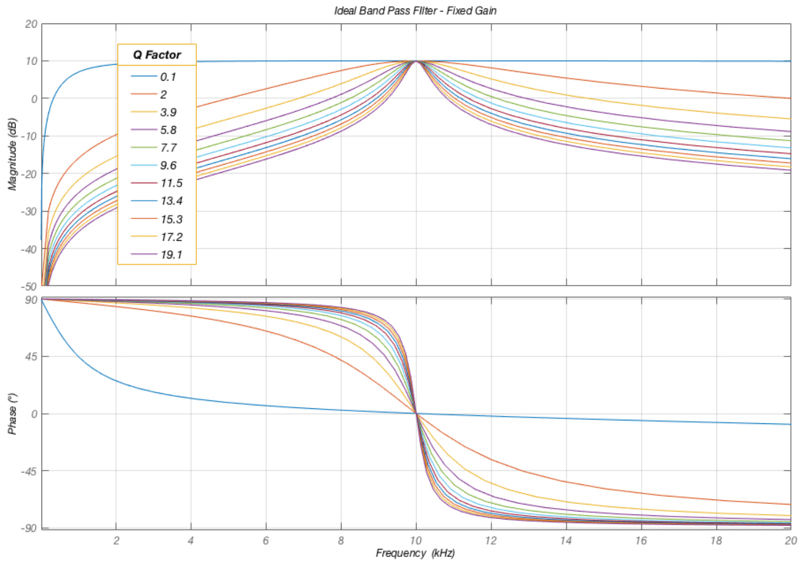

Figure 1.3. Ideal Band Pass - Fixed Gain

The obtained behaviour has now to be implemented on a real circuit and the Deliyannis band pass filter will be employed with no loss of generality [1]. The Deliyannis form has been chosen because of his parameters linear dependence, and its electronic circuit is described in the following figure:

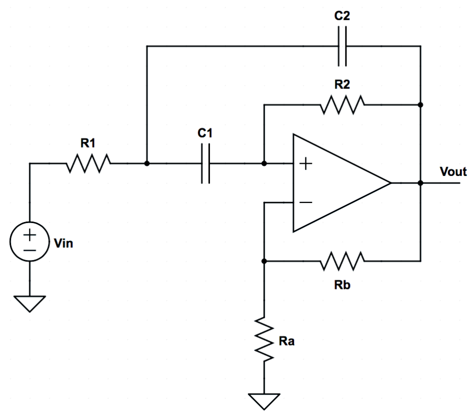

Figure 1.4. Deliyannis Band Pass Filter Schematic

The transfer function of the Deliyannis circuit is thus obtained:

$$BPF_{Deliyannis} = \frac{\frac{1+K}{C_2 R_1} s}{s^2 + (\frac{C_1+C_2}{C_1 C_2 R_2} - \frac{K}{C_1 R_1})s + \frac{1}{R_1 R_2 C_1 C_2}} \tag{1.9}$$

where K $= \frac{R_a}{R_b}$.

[1] Every other band pass filter might be suitable

Referring to (1.6), from (1.9) we obtain the parameter dependencies to proceed towards the filter sizing:

$$a_2 = \frac{1+K}{C_2R_1} \tag{1.10}$$
$$\frac{\omega_0}{Q} = \frac{C_1+C_2}{C_1C_2R_2} - \frac{K}{C_1R_1} \tag{1.11}$$
$$\omega_0 = \frac{1}{\sqrt{R_1R_2C_1C_2}} \tag{1.12}$$

In order to solve the fixed gain filter problem, it is necessary to make some assumptions that simplify the calculation dependencies: the requested gain is Δ_v^{BPF} and $C_1 = C_2 = C$, where C is a prefixed value for capacitors[1]. It was even possible to fix the resistor values, but tuning a capacitor is more expensive and more sensitive to parameters variations resulting then less reliable. Therefore, accordingly with the (1.7), the (1.8) and the (1.10) it is possible to write:

$$\frac{1+K}{CR_1} = \frac{\omega_0}{Q}\Delta_v^{BPF} \tag{1.13}$$

obtaining:

$$R_1 = \frac{(1+K)Q}{C\omega_0\Delta_v^{BPF}} \tag{1.14}$$

From the (1.12) we obtain the value of R_2:

$$R_2 = \frac{1}{\omega_0^2C^2R_1} \tag{1.15}$$

which replaced in the (1.11):

$$\frac{\omega_0}{Q} = \frac{2C}{C^2\frac{1}{\omega_0^2C^2R_1}} - \frac{K}{CR_1} \tag{1.16}$$

obtaining thus the K value:

$$K = CR_1(2\omega_0^2CR_1 - \frac{\omega_0}{Q}) \tag{1.17}$$

Replacing then the K value from (1.17) in the (1.13) we obtain the second order equation to solve for Deliyannis band pass filter:

$$R_1^2 - \frac{1+\Delta_v^{BPF}}{2C\omega_0Q}R_1 + \frac{Q}{2C^2\omega_0^2Q} = 0 \tag{1.18}$$

Solutions are given by:

$$R_1 = \frac{1+\Delta_v^{BPF} \pm \sqrt{(1+\Delta_v^{BPF})^2 - 8Q^2}}{4\omega_0CQ} \tag{1.19}$$

[1] It is a designer choice based on project constraints

A real resistor model implements parasitic elements which might lead to accept a complex solution by the equation (1.19), but since these have small values, their contribution compared to poles and zeros within the band under exam is negligible. Furthermore, as far as a negative resistor is achievable, it is far simpler as well as cheap to implement a positive one. Consequently, excluding both the complex solution and the negative one, the last acceptable is given by:

$$R_1 = \frac{1 + \Delta_v^{BPF} + \sqrt{(1 + \Delta_v^{BPF})^2 - 8Q^2}}{4\omega_0 CQ} \tag{1.20}$$

where, for a real solution

$$(1 + \Delta_v^{BPF})^2 - 8Q^2 \geq 0; \tag{1.21}$$

This limit imposes a constraint on the filter gain itself:

$$\Delta_v^{BPF} = 10^{\frac{G_{dB}}{20}} \geq Q\sqrt{8} - 1 \tag{1.22}$$

where G_{dB} is the filter gain value expressed in dB.
Finally, we obtain a set of equations that allow to size the band pass filter with a fixed gain and independent tuning of parameters: the value of R_1 is evaluated through the (1.20), known the value of Δ_v^{BPF} (or G_{dB}) which is constrained by the value of Q according with the (1.22); values of K and R_2 can be calculated through the substitution of the obtained value for the variable R_1 respectively in equations (1.17) and (1.15). Lastly we evaluate the value of R_b, fixing the value of R_a, through $R_b = KR_a$.

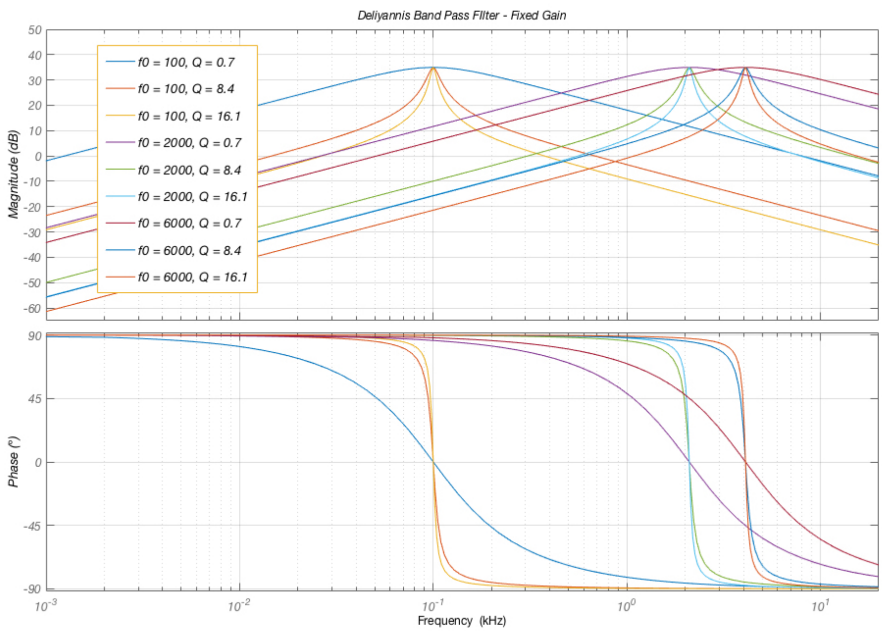

Figure 1.5. Deliyannis Band Pass - Fixed Gain

The following step is to integrate the obtained equations in a real electronic simulation CAD, which also keeps non-idealities into account. The Deliyannis band pass filter schematic has been designed through OrCAD PSpice Designer, as in the following picture.

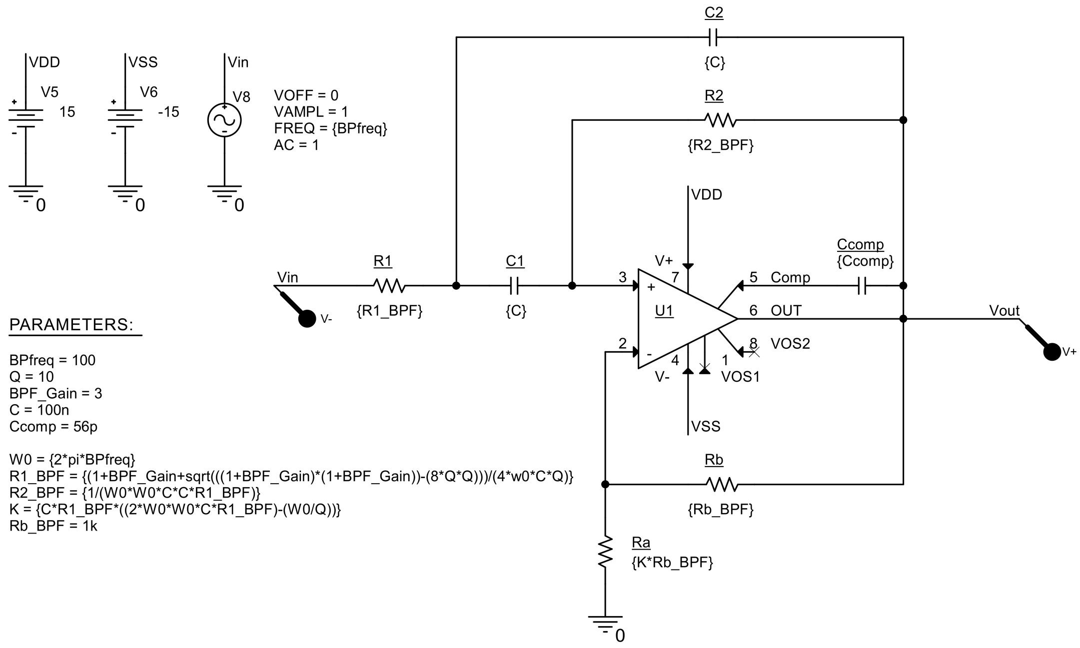

Figure 1.6. Deliyannis Band Pass Filter - OrCAD PSpice

All equation variables have been defined as parameters in order to set up the desired values during parametric sweep simulation. As first step, we are going to compare the ideal behaviour obtained through Matlab simulation with the PSpice one, which has to be set up with the same ω_0 and Q steps, in order to evaluate the non-idealities effect. The Δ_v^{BPF}, here expressed as 'BPF_Gain', has been set to 33.25 dB (46 in linear gain), according to the gain lower limit imposed by the equation (1.22).

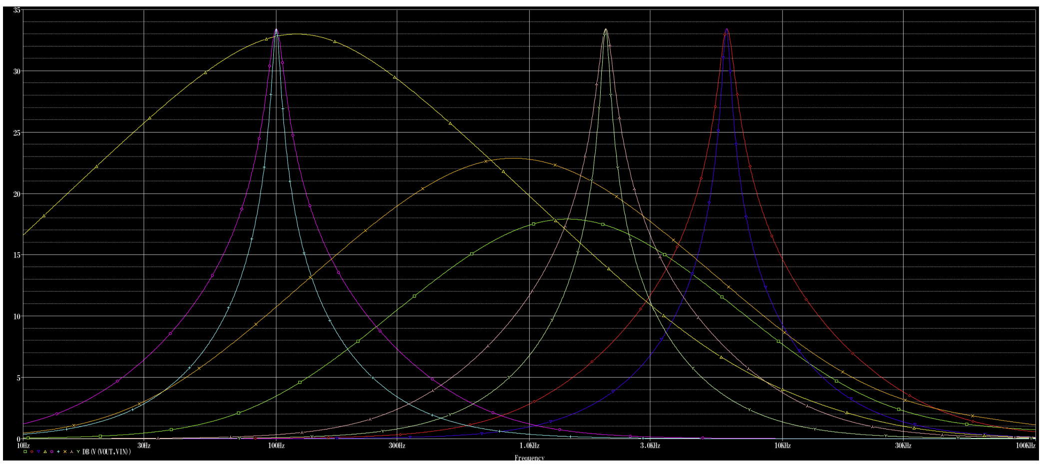

Figure 1.7. Deliyannis Band Pass Filter - OrCAD PSpice Simulation

As we can see the ideal behaviour is not respected for Q=0.7, and worse as the ω_0 increases, while for higher values of Q the ω_0 and gain are well fixed around the values that have been imposed. In order to evaluate the real behaviour of the band pass filter when the Q changes, we fix the frequency around 1kHz, and run a parametric sweep on the Q factor between 0.1 and 20, with steps of 0.2. For the equation (1.22), the gain of filter has been set to 56, that means 35 dB.

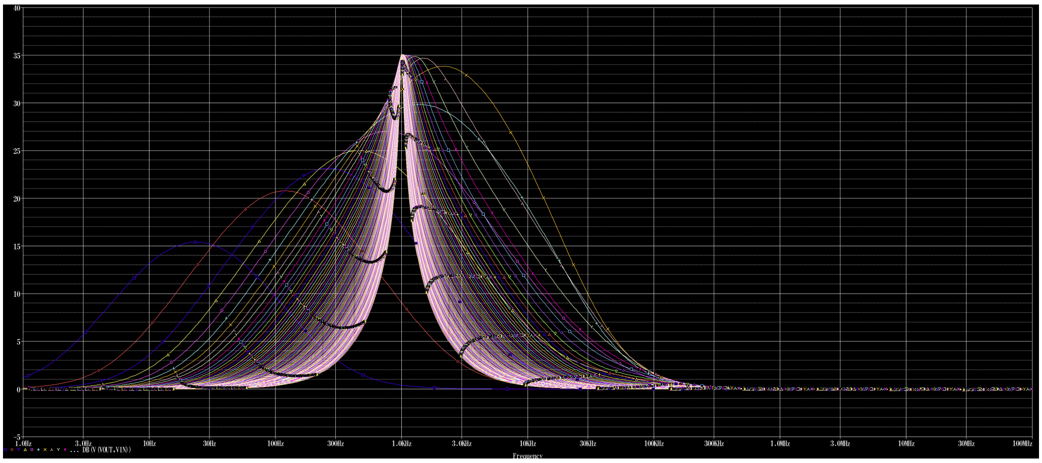

Figure 1.8. Deliyannis behaviour to Q changes - OrCAD PSpice Simulation

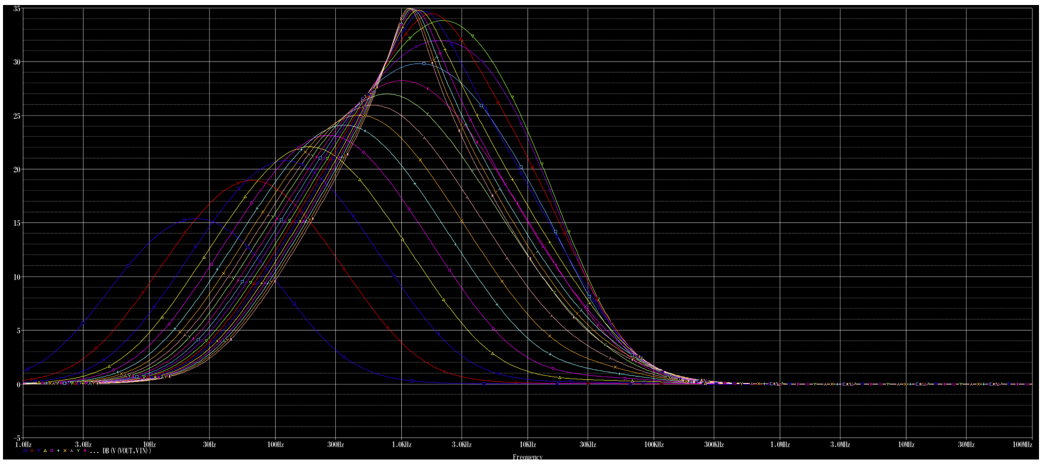

Figure 1.9. Deliyannis behaviour to small Q changes - OrCAD PSpice Simulation

The ω_0 and gain deviations are due to an imposed value too high for small Q curves. It is then possible to divide the fixed gain value in steps according to the requested overall Q range. In the following figures the fixed gain value has been reduced to 15.5 dB for filter curves within the Q range 0.5÷1, leaving unchanged the previously set gain value for Q>1 curves.

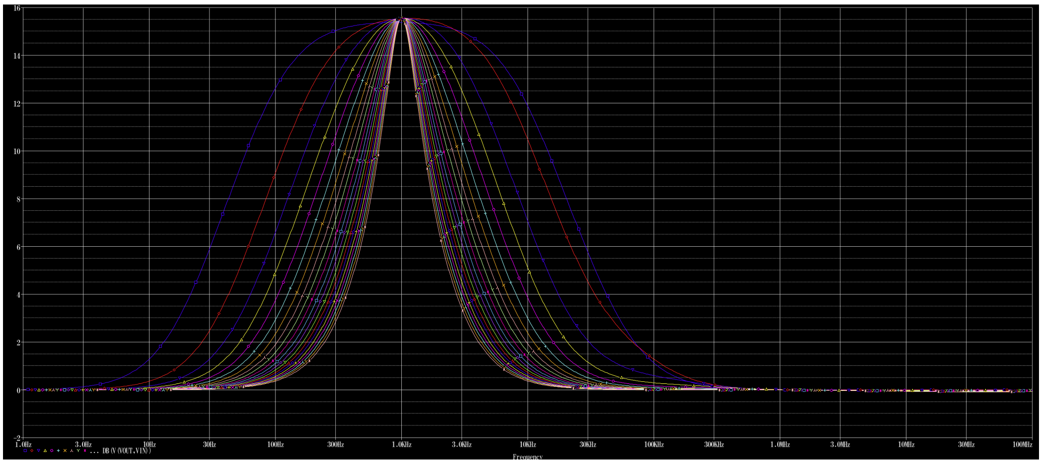

Figure 1.10. Deliyannis behaviour for Q within $0.5 \div 1$ - OrCAD PSpice Simulation

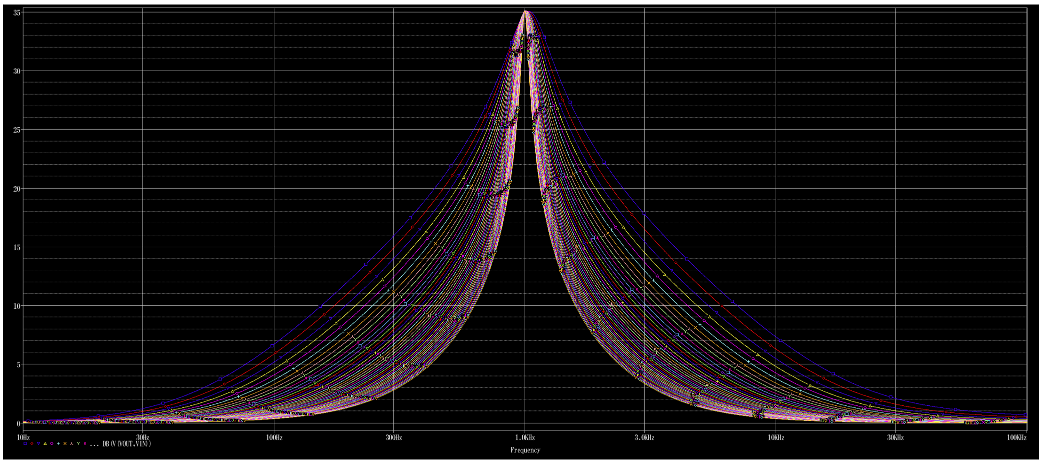

Figure 1.11. Deliyannis behaviour for Q>1 - OrCAD PSpice Simulation

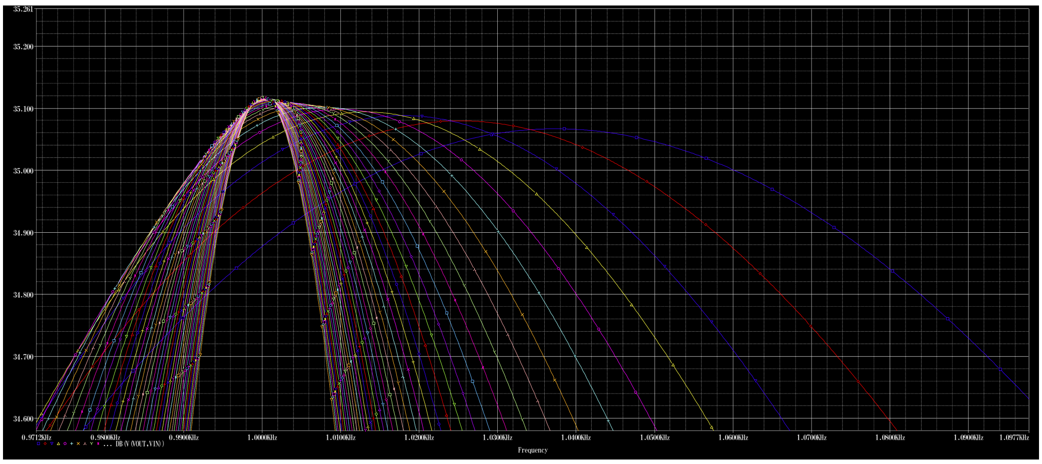

Figure 1.12. Zoom around ω_0 centering for Q>1 changes - OrCAD PSpice Simulation

It is possible to observe in figure 1.12 how this solution leaded to a better centering of the ω_0 during Q transitions in a fixed gain filter system: for high values of Q the ω_0 error is within the 4%.
The result of dividing in two gain steps the Q excursion, as in the following figure, confirms the better behaviour of the overall filter compared to the one step gain filter.

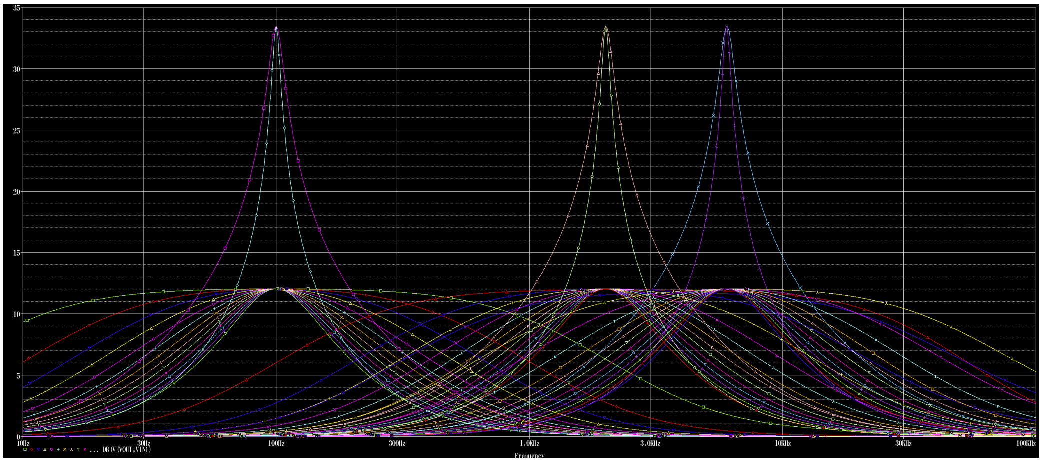

Figure 1.13. Deliyannis behaviour with two gain steps - OrCAD PSpice Simulation

Chapter 2

Band Cut Filter

2.1 Theory of Reverse Filters

With the word Reverse Filter we mean a system acting in a reciprocal way compared to the simple filter structure. In other words, using the same filter, the gain Δ_v, resonance pulsation ω_0 and the quality factor Q sized for filter, must recur for the Reverse Filter configuration, but with a reciprocal excursion (compared to the 0 dB level). This solution, as we will see, is useful in order to implement cut filters with a flat behaviour in the outside filter band, due to the presence of the amplifier on the open loop. The system structure is described in the following block diagram:

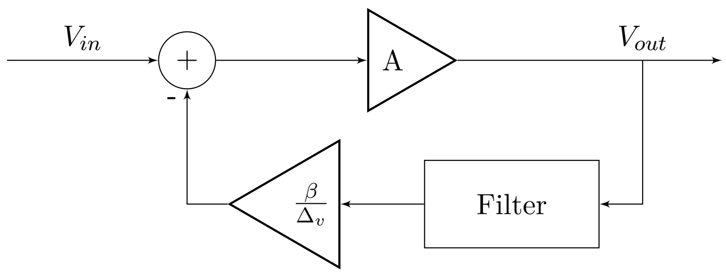

Figure 2.1. Inverse Filter System

The presence of A amplifier, not only justifies the flat behaviour in the outside filter band, but it also allows to redistribute gain between the two system amplifiers: therefore, in a cut function, the total amount of gain is halfway distributed between the two, in order to amplify by half gain all the audio band and cut by half gain the independent tunable filter passing band, thus justifying the negative feedback in the system block diagram and obtain the total amount of cut gain required. The gain distribution purpose is also for limiting the noise and distortions introduced by OPAs in high gain configuration (required for high Q filters), and to achieve a more efficient 'One Stage Only Cut Filter'. Indeed, the described structure allows, as far as just said, to build an independent parameters tunable cut filter, realized with one gain stage only, due to the negligible influence of the feedback gain and filter OPAs because of their 0 dB system driving or, in other wards, because of their

only role to select the fractional band to be cut. The presence of the $\frac{\beta}{\Delta_v}$ amplifier is in order to control the feedback gain: in fact the feedback filter gain is fixed to a certain value, due to the constraint on the Q factor as previously said, which is made unitary through the following gain stage dividing by the gain of the filter itself $\Delta_v(\omega_0)$, and then amplified by a desired β factor.
The inverse filter system, assuming a H(s) filter function, is then described through the following equation:

$$RFS(s) = \frac{A}{1 + A\frac{\beta}{\Delta_v}H(s)} \tag{2.1}$$

As previously said about the gain that is halfway distributed between the two gain stages, it is correct to assign A=$\frac{G}{2}$ dB, because it represents the overall band gain, or better, the out filter band gain that the system must achieve. Consequently the equation (2.1) becomes:

$$RFS(s) = \frac{10^{\frac{G}{40}}}{1 + 10^{\frac{G}{40}}\frac{\beta}{\Delta_v}H(s)} \tag{2.2}$$

We just said that the most important thing, on which the whole project is based, is that at the resonance pulsation the system output must assume a 0 dB value. Consequently the β value should be calculated congruently with previous observations concerning the gain distribution: given the gain of G dB for the overall system, and assuming that $H(s)|_{s=\jmath\omega_0} = \Delta_v$, we obtain the following equation:

$$RFS(s)|_{s=\jmath\omega_0} = 1 = \frac{A}{1 + A\beta} \tag{2.3}$$

through which β is finally found:

$$\beta|_{A=\frac{G}{2}} = \frac{A-1}{A} = \frac{10^{\frac{G}{40}} - 1}{10^{\frac{G}{40}}} \tag{2.4}$$

We obtained the fundamental equations regulating a general reverse filter: if the feedback filter is a tunable filter, consequently the entire system would become a tunable reverse filter.

2.2 Fixed Gain Active Band Cut Filter

According to the results obtained in the section above, we can now realize a Band Cut Filter whose block system is described in the following figure:

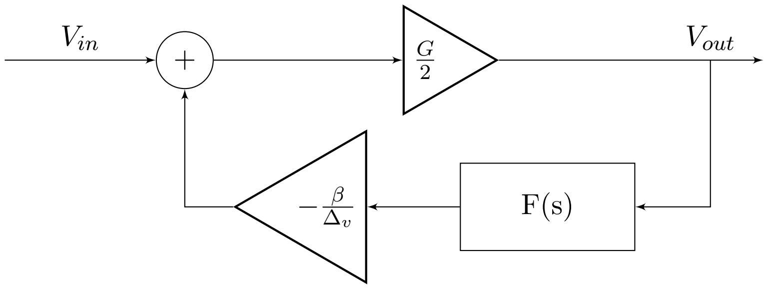

Figure 2.2. Band Cut Filter System

The open loop gain A is equal to $\frac{G}{2}$ due to the gain distribution, as mentioned before. The Band Pass Filter phase is supposed to be within a [-90°, +90°] range, confirming a non inverting behaviour and justifying the inverting $-\frac{\beta}{\Delta_v}$ gain in order to implement the negative sum in a positive feedback summing node. The following schematic is an implementation of the Band Cut Filter, where the band pass filter is a passive one, without loose of generality[1].

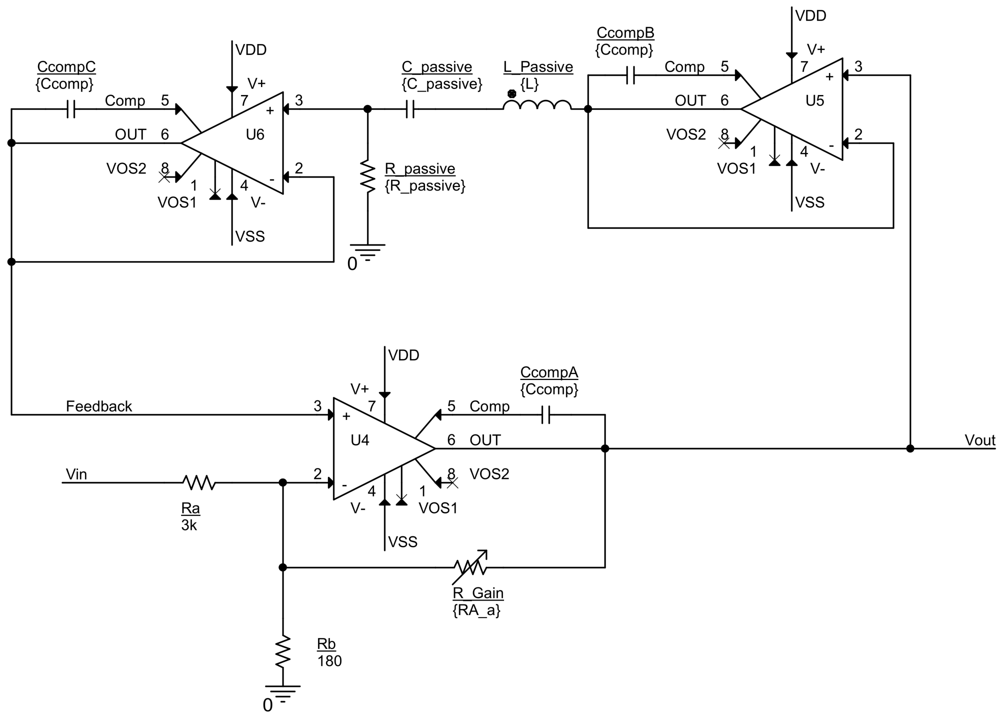

Figure 2.3. Deliyannis Band Cut Filter Schematic

[1]A Fixed Gain Active Filter could be used, but with a $\frac{1}{\Delta_V}$ factor implementation

The use of a passive filter means that at the resonance $\omega = \omega_0$ the filter gains 0 dB. Consequently there is no more need of the $\frac{1}{\Delta_V}$ factor which has not more to be implemented and opportunely sized, simplifying the project under exam. The band pass filter has been decoupled, in order to obtain its ideal behaviour, through its input and output buffers, with a 0° phase shift around the resonance pulsation ω_0. The open loop OPA is an inverting configuration for the input signal, which gain is ideally controlled through the equation:

$$A = -\frac{R_{Gain}}{R_A} \tag{2.5}$$

Thus the band pass filter output is 180° phase shifted by the V_{in} signal, being possible to implement the negative feedback summing node on the non inverting input of the open loop OPA.
The resistor R_b ideally drives the feedback gain through the expression:

$$\beta = 1 + \frac{R_{Gain}}{R_b} \tag{2.6}$$

The ideal case is approximatively achieved through distant values of the two input partition resistors $R_a = 3\ k\Omega$ and $R_b = 180\ \Omega$. The feedback gain resistor R_b^{Real} remains almost its value:

$$R_b^{Real} = \frac{R_b R_a}{R_b + R_a} = 169.81\ \Omega\ \approx R_b \tag{2.7}$$

Furthermore, applying the Thévenin's theorem to the input node, the R_b influence to the open loop gain is negligible:

$$V_{Thévenin} = \frac{R_b}{R_b + R_a} V_{in} = 0.06\ V_{in} \tag{2.8}$$

$$R_{Thévenin} = \frac{R_b R_a}{R_b + R_a} = 169.81\ \Omega\ \approx R_b \tag{2.9}$$

Consequently, being the R_{Gain} value in the $1\ k\Omega \div 10\ k\Omega$ range, the β gain is about equal to the open loop gain: $A \approx \beta \approx \frac{R_{Gain}}{R_b}$, except the sign change. Thus, the gain distribution in this case is equal for the feedback gain β and the open loop gain A, due to the feedback loop configuration of this circuit, which means that at the resonance ω_0 the closed loop gain is almost 0 dB since the OPA has a buffer behaviour. It's almost 0 dB due to the non-zero attenuation of the passive band pass filter, where the resistor value varies in order to tune the ω_0 and consequently the variable attenuation might not be negligible[1]. In addition to this the gain partition error might affect the outside band gain, since it influences the ratio between the open loop gain and feedback loop one which may no longer be unitary. Nevertheless the latter is negligible with distant values between resistors R_a and R_b, making the partition ratio closer to the ideal case, as shown in (2.7) and (2.9). However, an advantage of this configuration is the presence of the input divider which allows the band cut filter to accept signals with higher dynamics, and its attenuation ratio,

[1]This does not happen with an active fixed gain band pass filter implementation.

according to the (2.8), can be tuned through the resistor R_a. As far as has been said, since there is a constraint on gain resistor values, which means $A \approx \beta$, the resistor R_{Gain} must drive the overall gain requested outside the cutting band.
The Reverse Filter System equation (2.1) for this configuration outside the cutting band, since it is a buffer stage at the resonance, becomes:

$$BCF(s) = \frac{\alpha A}{1 - A\beta H(s)} \approx \frac{-\frac{R_b}{R_b+R_a}\frac{R_{Gain}}{R_b}}{1 + \frac{R_{Gain}}{R_b}(1 + \frac{R_{Gain}}{R_b})H(s)} \tag{2.10}$$

where H(s) is the passive band pass filter transfer function and α represent the division factor of the input signal, as in (2.8). Consequently, outside the cutting band, the gain product $A\beta \approx A^2$ must equal the band pass filter attenuation value:

$$A\beta \approx A^2 = \left(\frac{R_{Gain}}{R_b}\right)^2 = Z_{H(s)|_{\omega \neq \omega_0}} \tag{2.11}$$

in order to obtain an open loop gain, or a gain outside the cutting band, equal to αA, which means:

$$R_{Gain} \approx \frac{10^{\frac{G}{20}}}{\alpha} R_b \tag{2.12}$$

where G is the overall gain in dB outside the cutting band. Therefore G must assume the half of the overall cut gain requested.

The simulation performed through PSpice, shown below, confirms the analytically obtained results. Outside the cutting band the simulation gain is in the 0.5 $dB \div$ 10.5 dB range, which means a cut gain range of 1 $dB \div 21$ dB, according to what has been said in the previous section. Thus, according to (2.12), the resulting R_{Gain} is in the 1 $k\Omega \div 10$ $k\Omega$ range, as previously stated.

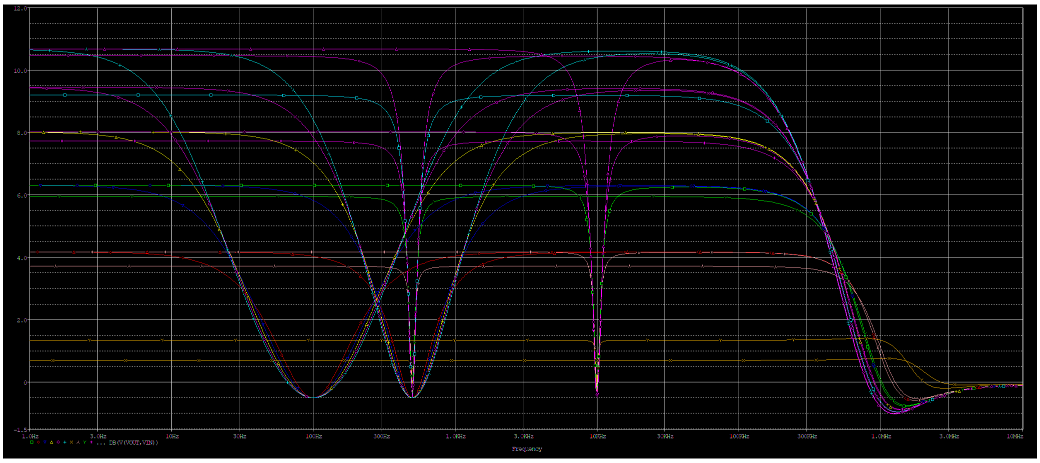

Figure 2.4. PSpice Simulation BCF - Passive Feedback BPF

Finally, the Matlab simulation is reported for a Band Cut Filter which implements a Deliyannis Fixed Gain Band Pass Filter with half distributed gain between open loop gain A and feedback loop gain β, as described in the previous section, and the $\frac{1}{\Delta_V}$ factor for the Deliyannis filter gain itself.

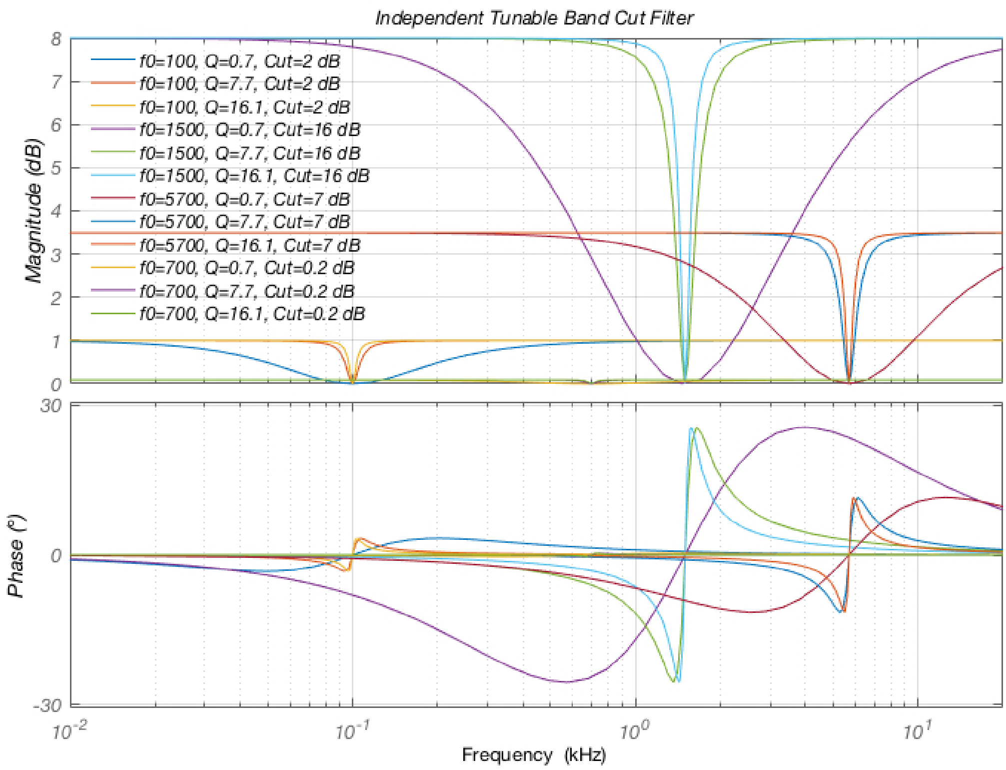

Figure 2.5. Deliyannis Band Cut Filter - Fixed Gain

Appendix

Band Pass Filter Matlab Script

breaklines

```
clear all;
clc;

s=tf('s');   % Defines 's' Transfer Function

% Global Parameters

f  = 10000;    % Define BIQUAD center frequency
a2 = 1;        % Define Biquad a2 parameter
w0 = 2*pi*f;   % Define Biquad w0 parameter
Q  = 1;        % Define Biquad variable Q parameter
Av_dB = 35;    % Define the desired fixed gain value

Av = 10^(Av_dB/20);  % Linear Conversion dB Value

C = 1e-6;      % uF
R = 10e3;      % 10 kOhm
Rb = 1e3;      % 10 kOhm

%% Ideal Band Pass Filter   --  Variable Gain

figure('Name','IdealBandPassFilterVariableGain','NumberTitle','off');
    % Q from 1 to 20, step of 1
    for IncrementQ = 1:1:20
        Q = IncrementQ
        BandPass = (a2*s)/(s^2+s*w0/Q+w0^2)
        hold on;
        bode( BandPass, {0, 2*pi*20000}, 'b' );
    end

%% Delyiannis Band Pass Filter   --  Variable Gain

figure('Name','DelyiannisBandPassVariableGain','NumberTitle','off');
    % Q from 1 to 20, step of 1
    for IncrementQ = 1:1:20
        Q = IncrementQ
        R2 = 1/(w0^2*C^2*R)
        K = R*C*(2*w0^2*R*C - w0/Q)
        Ra = K*Rb
        Delyiannis = (((1+K)/(C*R))*s)/(s^2+((2/(C*R2))-(K/(C*R)))*s
            +1/(R*R2*C^2))
        hold on;
        bode( Delyiannis, {0, 2*pi*20000}, 'r' );
    end
```

Band Pass Filter - Fixed Gain Matlab Script

breaklines

```
clear all;
clc;

s=tf('s');             % Defines 's' Transfer Function

% Global Parameters

f   = 10000;           % Define BIQUAD center frequency
a2  = 1;               % Define Biquad a2 parameter
w0  = 2*pi*f;          % Define Biquad w0 parameter
Q   = 1;               % Define Biquad variable Q parameter
Av_dB = 35;            % Define the desired fixed gain value

Av = 10^(Av_dB/20);  % Linear Conversion dB Value

C = 1e-6;             % uF
R = 10e3;             % 10 kOhm
Rb = 1e3;             % 10 kOhm

%% Delyiannis Band Pass Filter  --  IDEAL Fixed Gain

figure('Name','DelyiannisBandPassIdealFixedGain','NumberTitle','off');
    % Q from 1 to 20, step of 1
    for IncrementQ = 1:1:20
        Q = IncrementQ
        R2 = 1/(w0^2*C^2*R)
        K = R*C*(2*w0^2*R*C - w0/Q)
        Ra = K*Rb
        a2 = (Av*w0)/Q;
        % Ideality Fixed Gain
        Delyiannis=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R)))*s+1/(R*R2*C^2))
        hold on;
        bode( Delyiannis, {0, 2*pi*20000}, 'g' );
    end

%% Delyiannis Band Pass Filter  --  Real Fixed Gain

figure('Name','Delyiannis Band Pass Filter Real Fixed Gain', 'NumberTitle', 'off');
    % Q from 1 to 20, step of 1
    for IncrementQ = 1:1:20
        Q = IncrementQ
        R1 = (1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
        R2 = 1/(w0^2*C^2*R1)
        K = C*R1*(2*w0^2*C*R1 - w0/Q)
        Ra = K*Rb
        a2 = (1+K)/(C*R1);
        % Real Fixed Gain
        Delyiannis_Fixed=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R1)))*s+1/(R1*R2*C^2))
        hold on;
        bode( Delyiannis_Fixed, {0, 2*pi*20000} );
    end

%% Variable w0

figure('Name','DelyiannisRealBPF_FixedGain,Variable_w0','NumberTitle','off');
        % f0 from 100 Hz to 15 kHz, step of 5 kHz
        for f0 = 100:5e3:15e3
            % Q from 1 to 10 kHz, step of 3
            for Q_Increment = 1:3:10
                Q = Q_Increment;
                w0 = 2*pi*f0;
                R1 = (1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
```

```
                R2 = 1/(w0^2*C^2*R1)
                K = C*R1*(2*w0^2*C*R1 - w0/Q)
                Ra = K*Rb
                a2 = (1+K)/(C*R1);
                % Real Fixed Gain
                Delyiannis_Fixed_w0_Variable=(a2*s)/(s^2+((2/(C*R2))
                    -(K/(C*R1)))*s+1/(R1*R2*C^2))
                hold on;
                bode( Delyiannis_Fixed_w0_Variable, {0, 2*pi*20000},
                    'y' );
            end
        end
```

Band Cut Filter Matlab Script

breaklines

```
clear all;
clc;

s=tf('s');                  % Defines 's' Transfer Function

%% Global Parameters

f  = 10000;    % Define BIQUAD center frequency
a2 = 1;        % Define Biquad a2 parameter
w0 = 2*pi*f;   % Define Biquad w0 parameter
Q  = 1;        % Define Biquad variable Q parameter
Av_dB = 35;    % Define the desired fixed gain value

Av = 10^(Av_dB/20);  % Linear Conversion dB Value

C  = 10*10e-9;    % 10 nF
R  = 10e3;        % 10 kOhm
Rb = 10e3;        % 10 kOhm

%% Test Plot

figure('Name','Cut Band FIlter Real','NumberTitle','off');
    %A
        CutBand_Gain_dB=2;
        % Define linear value of OPA open loop gain
        OpenLoop_OPA_Gain=2*10^(CutBand_Gain_dB/40);
         % Define feedback gain control (divided by Av, BPF gain)
        Feedback_Gain=((OpenLoop_OPA_Gain-2)/(Av*OpenLoop_OPA_Gain));
        f0 = 100;
            % Q from 0.7 to 16.1, step 7.7
            for Q_Increment = 0.7:7.7:16.1
                Q=Q_Increment;
                w0=2*pi*f0;
                R1=(1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
                R2=1/(w0^2*C^2*R1)
                K=C*R1*(2*w0^2*C*R1 - w0/Q)
                Ra=K*Rb
                a2=(1+K)/(C*R1);
                Delyiannis_BPF=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R1)))*s
                    +1/(R1*R2*C^2));
                % Define Cut Band Filter Transfer Function
                CBF=(OpenLoop_OPA_Gain/2)/(1+(OpenLoop_OPA_Gain*
                    Feedback_Gain*Delyiannis_BPF/2))
                hold on;
                bode(CBF,{0,2*pi*20000});
            end
    %B
        CutBand_Gain_dB=16;
        % Define linear value of OPA open loop gain
        OpenLoop_OPA_Gain=2*10^(CutBand_Gain_dB/40);
        % Define feedback gain control (divided by Av, BPF gain)
```

```
        Feedback_Gain =((OpenLoop_OPA_Gain-2)/(Av*OpenLoop_OPA_Gain));
        f0=1500;
            % Q from 0.7 to 16.1, step 7.7
            for Q_Increment = 0.7:7.7:16.1
                Q=Q_Increment;
                w0=2*pi*f0;
                R1=(1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
                R2=1/(w0^2*C^2*R1)
                K=C*R1*(2*w0^2*C*R1 - w0/Q)
                Ra=K*Rb
                a2=(1+K)/(C*R1);
                Delyiannis_BPF=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R1)))*s+1/(R1*R2*C^2));
                % Define Cut Band Filter Transfer Function
                CBF=(OpenLoop_OPA_Gain/2)/(1+(OpenLoop_OPA_Gain*Feedback_Gain*Delyiannis_BPF/2))
                hold on;
                bode(CBF,{0,2*pi*20000});
            end
    %C
        CutBand_Gain_dB=7;
        % Define linear value of OPA open loop gain
        OpenLoop_OPA_Gain=2*10^(CutBand_Gain_dB/40);
        % Define feedback gain control (divided by Av, BPF gain)
        Feedback_Gain =((OpenLoop_OPA_Gain-2)/(Av*OpenLoop_OPA_Gain));
        f0=5700;
            % Q from 0.7 to 16.1, step 7.7
            for Q_Increment = 0.7:7.7:16.1
                Q=Q_Increment;
                w0=2*pi*f0;
                R1=(1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
                R2=1/(w0^2*C^2*R1)
                K=C*R1*(2*w0^2*C*R1 - w0/Q)
                Ra=K*Rb
                a2=(1+K)/(C*R1);
                Delyiannis_BPF=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R1)))*s+1/(R1*R2*C^2));
                 % Define Cut Band Filter Transfer Function
                CBF=(OpenLoop_OPA_Gain/2)/(1+(OpenLoop_OPA_Gain*Feedback_Gain*Delyiannis_BPF/2))
                hold on;
                bode(CBF,{0,2*pi*20000});
            end
    %D
        CutBand_Gain_dB=0.2;
        % Define linear value of OPA open loop gain
        OpenLoop_OPA_Gain=2*10^(CutBand_Gain_dB/40);
        % Define feedback gain control (divided by Av, BPF gain)
        Feedback_Gain=((OpenLoop_OPA_Gain-2)/(Av*OpenLoop_OPA_Gain));
        f0=700;
            % Q from 0.7 to 16.1, step 7.7
            for Q_Increment = 0.7:7.7:16.1
                Q=Q_Increment;
                w0=2*pi*f0;
                R1=(1+Av+sqrt((1+Av)^2-8*Q^2))/(4*w0*C*Q)
                R2=1/(w0^2*C^2*R1)
                K=C*R1*(2*w0^2*C*R1 - w0/Q)
                Ra=K*Rb
                a2=(1+K)/(C*R1);
                Delyiannis_BPF=(a2*s)/(s^2+((2/(C*R2))-(K/(C*R1)))*s+1/(R1*R2*C^2));
                % Define Cut Band Filter Transfer Function
                CBF=(OpenLoop_OPA_Gain/2)/(1+(OpenLoop_OPA_Gain*Feedback_Gain*Delyiannis_BPF/2))
                hold on;
                bode(CBF,{0,2*pi*20000});
            end
```