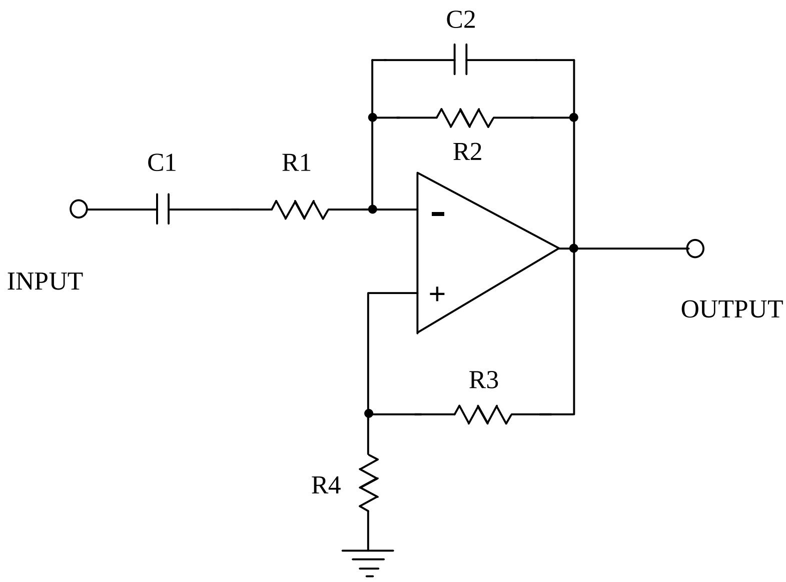
A BAND–PASS FILTER WITH Q AND FREQUENCY INDEPENDENTLY ADJUSTABLE

By Luiz Amaral
PY1LL/AC2BR

It is very common to find band-pass filters with fixed **Q** or with it frequency dependent.

This article shows a simple filter where the frequency adjustment is independent of the **Q** one, which may be a very great advantage for certain designs.

The suggested topology is shown in the Figure below.



Using the admittance matrix, it can be shown that the transfer function of that circuit is given by:

$F(S) = -[(R4+R3)/(R1.R3.C2)].S/[S^2+(R2.R3.C2–R2.R4.C1+R1.R3.C2).S+1/(R1.R2.C1.C2)]$ (1)

It can be easily seen that **$F(0)=0$** and that **$F(\infty)=0$**, showing that the filter is a band-pass one.

Rewriting **(1)** in the classic form **$F(S)= b.S/[S^2+(2.\omega_0/Q).S+\omega_0^2]$**, we have by comparison:

$\omega_0=\sqrt{[1/(R1.R2.C1.C2)]}$ (2)

$Q=2. R3.(\sqrt{R1.R2.C1.C2})/[R3.(R1.C1+R2.C2)-R2.R4.C1]$ (3)

We can see in **(2)** that the frequency **ω_0** is independent of **R3** and **R4**. But the **Q** still depends on all components.

If we make **R1=R2=R** e **C1=C2=C (4)**:

$\omega_0=1/(R.C)$ **(5)**

$Q=2.R3/(2.R3-R4)$ **(6)**

Now we see that the frequency ω_0 depends only on **R** and **C** and the **Q** only depends on **R3** and **R4**.

As for **R4=2.R3** the **Q** tends to infinity, with this value the circuit oscillates (in frequency ω_0) and, therefore, this value must not be chosen, being always kept **2.R3>R4**.

With **R4=0** or $R3\to\infty$, we have the case equivalent to that with the output directly connected to the +input of the operational amplifier, when then **Q=1**.

The gain in the frequency ω is the module of the transfer function $|F(j.\ \omega)|$. Let's calculate that value at the frequency ω_0 under the condition **(5)**:

$$F(S)=-[(R3+R4)/(R3.R.C)].S/\{S^2+[(2.R3-R4)/(R3.R.C)].S+1/(R^2.C^2)\} \quad (7)$$

That, with $S\to j.\ \omega$, results in:

$$F(j.\ \omega)=-[(R3+R4)/(R3.R.C)].\ j.\ \omega/[1/(R^2.C^2)-\omega^2+(2.R3-R4)/(R3.R.C).j.\ \omega] \quad (8)$$

And:

$$|F(j.\ \omega)|^2=[(R3+R4)/(R3.R.C)]^2.\ \omega^2/\{[1/(R^2.C^2)-\omega^2]^2+[(2.R3-R4)/(R3.R.C)]^2.\omega^2\} \quad (9)$$

If $w=\omega_0$, we have:

$$Ganho(\omega_0)=|F(j.\ \omega_0)|=(R3+R4)/(2.R3-R4) \quad (10)$$

Using **(6)** in **(10)**, one gets:

$$Gain(\omega_0)=(3.Q-2)/2 \quad (11)$$

In the case of **Q=1**, $Gain(\omega_0)=1/2$

The gain here is the ratio between the output voltage **Vs** and **Vi**, that of the input. As the first is limited by the symmetrical supply voltages $V_\pm$, for given input signal in the frequency ω_0[(1)], the gain at this frequency is limited by the amplifier saturation, that is, when the output voltage reaches, in its peak, the symmetrical supply voltages:

$$Gain\ max(\omega_0)=V_\pm/Vs=(3.Qmax-2)/2$$

or:

[(1)] **What indeed matters is the amplitude of the Fourier component at the frequency ω_0.**

$Qmax=[2.V_{\pm}/Vs+2]/3$ (12)

The expression **(12)** shows that, with an input **Vi**, the **Q** is limited to the value **Qmax** for keeping the circuit linear and, therefore, the filter properties without signal distortion. So, a greater supply voltage will permit a greater **Q** without saturation.

When **R4=2.R3**, as we saw before, the circuit oscillates, but, as in any ideal linear sinusoidal oscillator, the amplitude increases indefinitely, the output will not be sinusoidal but approximately square. To get a sinusoidal oscillator it is necessary to feedback the input with the output signal in a correct and limited manner, that is, the square signal limited enough by an external limiter, passes by the filter and its fundamental component is got at the amplifier output. The **Q** is limited by the amplifier saturation and, therefore, limited for the correct oscillator operation.

This article filter can be used in the analog detectors for **FSK** that, after the frequencies determination, have their **Q** adjusted for the best operational efficiency, that is, high enough to separate both **FSK** frequencies and eliminate the off-band noise, but low enough to accept the communication baud-rate.

If the reader wants to build an adjustable frequency filter, he can use a volume/tone potentiometer used with stereo amplifiers (two with the same axle) to vary simultaneously **R1** e **R2**, remembering that the linear frequency is given by **$F_0=\omega_0/(2.\pi)$**. As **R1** and **R2** cannot be zero, in series with each one we must have a fixed resistor to limit the maximum frequency. A similar limitation in the ratio between **R3** and **R4** also must exist to avoid oscillations.

Here a word on the independence of **Q** and frequency adjusts. Equation **(2)** shoes that the frequency is always independent of **R3** and **R4**, but **(3)** shows that **Q** depends on all components. So, when we impose the conditions **(4)**, small differences between **R1** and **R2** and between **C1** and **C2**, will result in small dependence on the **Q** at the chosen frequency. Obviously all of this are valid for an infinite open mesh gain, and this condition is never true, so that interdependence will always occur.

We must still remember that the maximum operation frequency of this type of circuit, as in any operational amplifier, is limited by its gain X frequency curve, that is, to the amplifier properties be valid, its open mesh gain must be as high as possible at the working frequency.