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DESIGNS REFERENCE DATA

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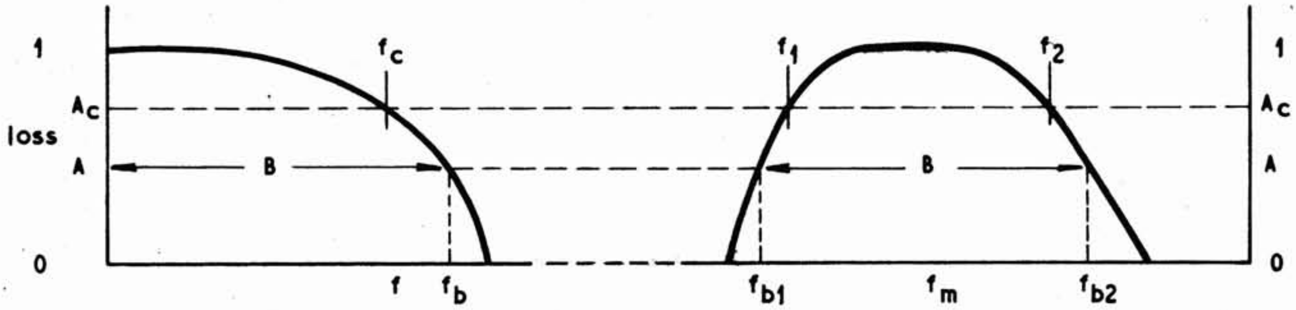
Written by: L.E. Weaver

LOW PASS-BAND PASS TRANSFORMATION

Any low pass filter may be transformed to an equivalent high pass, band stop or band pass filter, although in practice the last of these is by far the most important. The definition of equivalence in this sense can be seen from the sketch:



At any value of insertion loss A the bandwidth B is the same in the two equivalent filters, and incidentally the insertion phase shift through the filters is also the same at the frequency f_b as at f_{b1} and f_{b2}.

The procedure for the design of a band pass filter from a low pass original is as follows. Suppose a given insertion loss A_c is required at frequencies f_1 and f_2 in the bandpass; a low pass filter of a suitable type and the same nominal impedance is then designed, which has the same loss A_c at a frequency $f_e = (f_2 - f_1)$. In most instances f_1 and f_2 will be nominal cut-off frequencies, but not necessarily so. The mid-band frequency f_m is calculated from the relationship $f_m^2 = f_1 f_2$, then to obtain the transformed band pass filter each low pass capacitor is shunted with an inductor, and each low pass inductor has a capacitor connected in series with it, where in every instance $\omega_m^2 LC = 1$

When the band pass elements have been found in this manner it is possible that some of the elements may have awkward values, or the configuration of the circuit may be such as to aggravate difficulties arising from stray reactances. Recourse should then be had to equivalent circuits and/or to impedance transformations in order to optimise the element values and the configuration. It should also be borne in mind that a performance comparable to that of the low pass original will require elements with considerably higher Q values; for the effect of dissipation to be parallel in the two cases the band pass Q can be approximately found by multiplying the low pass Q by the factor fm/fq, where fq is the frequency at which the low pass Q is specified. This follows from the need to keep L/R and CR constant for the same dissipation. The factor fm/fq will obviously become very large when the relative bandwidth is very small.

The transformation given above is equivalent to a replacement of the low pass angular frequency ω' by $\omega_m\left(\frac{\omega}{\omega_m}-\frac{\omega_m}{\omega}\right)$, from which it is clear that the resulting filter is geometrically symmetrical about the mid-band frequency f_m, so that for every frequency f of the low pass filter there will be corresponding upper and lower frequencies f_u and f_e in the band pass filter given by $f_u f_e = f_m^2$. This geometric symmetry is a feature of all band pass filters derivable from a low pass configuration, and is often undesirable for practical purposes. A common instance is the case of a band pass filter for the selection of an amplitude or frequency modulated signal.

It is worth briefly investigating the extent of the asymmetry. If we replace $\frac{\omega}{\omega_m}$ by $\frac{\omega_m+\Delta\omega}{\omega_m}$ then $\omega' = \omega_m\left[\frac{\omega_m+\Delta\omega}{\omega_m} - \frac{\omega_m}{\omega_m+\Delta\omega}\right]$. When the second bracketed term is replaced by its binomial expansion we have

$$\omega' = \omega_m\left[\frac{2\Delta\omega}{\omega_m} - \left(\frac{\Delta\omega}{\omega_m}\right)^2 + \left(\frac{\Delta\omega}{\omega_m}\right)^3 - \cdots\right]$$

from which it is apparent that if $\Delta\omega$ is small enough for terms of second order and higher to be ignored, that is if the relative band width is sufficiently low, the band pass filter can be taken to have arithmetic symmetry.

The band pass group delay can be obtained from $\omega' = \omega_m\left(\frac{\omega}{\omega_m} - \frac{\omega_m}{\omega}\right)$ by differentiating the two sides. Then we have

$$\frac{d\beta}{d\omega} = \frac{\omega^2+\omega_m^2}{\omega^2}\cdot\frac{d\beta}{d\omega'} \doteq 2\left(1-\frac{\Delta\omega}{\omega_m}\right)\cdot\frac{d\beta}{d\omega'},$$ ignoring higher terms.

Hence, the band pass delay will only exactly equal twice the low pass delay if $\left(\frac{\Delta\omega}{\omega_m}\right)$ is negligible compared with unity. Otherwise there will be a slope of the group delay across the band as a result of this term.

Work done by: P. Denby

A Note on the Measurement of Audio Level, Volume and Noise

Small power levels in the BBC are usually quoted in decibels relative to 1 milliwatt. Such levels are designated dBm. A sine wave signal dissipating a power of 1 milliwatt in a 600Ω resistor produces across it a voltage of 0.775 volts r.m.s. BBC A.C. Test Equipment measures voltage and whether it responds to the average or the peak (i.e. crest) value, it is calibrated in terms of r.m.s voltage expressed as decibels with respect to 0.775 volt.

Where 600Ω audio equipment is used under matched conditions, the power gain and the voltage gain are identical. However, increasingly today, audio amplifiers are voltage gain devices having a high input impedance and a low output impedance and are not used under matched conditions. For this reason, audio measurements made in terms of voltage should not be quoted in dBm.

Such voltage measurements are made with respect to a reference level of 0.775 volt r.m.s. There is no special symbol agreed for such decibels; they are usually simply referred to as decibels (dB) in the BBC, but where it is necessary specifically to identify them, they would be referred to as dB (with respect to 0.775V). The need for a more compact expression has led the IEC SC29B to propose the term dB (0.775V). The widely accepted dBV, decibels with respect to 1.0V r.m.s., and its derivative dBµV, are not used for audio measurements in the BBC.

A sinewave signal of 0.775 volt or OdB is referred to as a zero level signal. A correctly calibrated BBC peak programme meter reads "4" on this signal. A zero volume programme applied to the same peak programme meter has maximum excursions upto "6", that is the "peak" programme excursion is +8dB or 1.95 r.m.s. In the BBC (though not necessarily elsewhere) line-up tone is set at a level 8dB below maximum level. Hence for a zero volume programme (peaks to 6 on ppm) line-up level is zero level (4 on ppm).

In operational areas noise measurements are usually made using a test programme meter and treating the noise as if it were programme. A zero volume programme peaks to "6" or a zero level tone peaks to "4", when the TPM dials read '0'. If the noise is now connected and the dials adjusted until it peaks to "6", the sum of the dial readings is said to be the noise level but it is important to remember that this relates to a particular peak reading, r.m.s. calibrated, BBC device. If necessary the figure in (usually negative) dB's should be qualified with (TPM peaked to 6). Alternatively, the change of dial readings between the noise measurement and the programme volume measurement is often referred to as the peak signal to peak noise ratio.

Weighted noise measurements are made in a similar way but through an aural sensitivity network. This should be switched to "line-up" for measuring the line-up tone (peaked to 4) and to "ASN" for the noise measurement (peaked to 6). As before the change of dial readings gives the weighted peak signal/peak noise ratio.

Written By:- L. E. Weaver

TERMINATION OF IMAGE PARAMETER FILTERS

It is standard practice to terminate image-parameter (Zobel) filter networks in m-derived half sections in order to obtain a better match to resistive terminations. The classical value of m for this purpose is 0.6, but while this provides the best approximate match over the greatest proportion of the passband, it is perhaps worth pointing out that it is not the optimum value where the best possible match is required over a reasonably large fraction of the passband, e.g. with delay networks. For such purposes a value of m=0.6835 is preferable. A comparison of the two values is given in the table below.

f/f_c	z/z_o (m =.6)	z/z_o (m=.6835)
0	1	1
.2	.994	.999
.4	.979	.998
.5	.970	1.001
.6	.962	1.010
.7	.961	1.035
.8	.984	1.098