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FILTER CIRCUITS FOR ELECTRONIC SOUND PRODUCTION

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FILTER CIRCUITS FOR ELECTRONIC SOUND PRODUCTION

Summary

Electronic composition and broadcast plays require low-pass, high-pass and band-pass filters having steep cut-off characteristics, and it must be possible to combine them as required. It is also desirable to be able to vary the cut-off frequencies readily. Filter circuits especially suitable for this purpose are discussed and directions are given for their construction.

1. General Requirements of the Filter Circuits Employed for Electronic Music

Two fundamental methods are employed in the technique of shaping for the production of sound in electronic music: firstly the so-called synthetic method, which composes the sound spectra from discrete sinusoidal oscillations of given mathematical proportions and secondly, the analytical method, which employs very wide line spectra produced by harmonic oscillators or wide band spectra originating from a noise generator as its raw material. One of the chief requirements of the analytical method is the analysis of the raw material into spectra of various types for the production of timbre. All known types of filter circuit, namely, low-pass, high-pass and band-pass, are used for the technical realization of this task.

In what follows, a few especially favourable filter circuits for this purpose are described and the special place of electronic music in filter technique is discussed.

For sound events which take in the entire audible range, filters with fixed limiting frequencies, i.e., with an invariable transmittance, are of no value. In contrast to

other filter situations, here the need for a variable filter transmittance is paramount. In view of the many filter channels required the expenses can be kept within tolerable limits only by utilizing the circuit elements in as many different ways as possible.

For characterization of the filter band width two quantities are important. Denoting the limiting frequencies of the pass range by ν_1 and ν_2, then

$\Delta\nu = \nu_2 - \nu_1$ is the absolute band width and

$\Delta\nu_r = \dfrac{2\Delta\nu}{\nu_2 + \nu_1}$ is the relative band width.

When the centre ν_o of the transmittance range is displaced along the frequency axis, either $\Delta\nu$ or $\Delta\nu_r$ can be kept constant, depending on the application.

A division of the audible range into channels of constant absolute band width would require too many individual channels at too great an expense, although this is the only way that the analysis of a harmonic spectrum could be carried out exactly. On the other hand, subdivision with constant relative width results in the known third and octave filters.

It will be shown that good results can be obtained using several filters which span the audible range and combine to form a dividing network - for the most part band-pass - which can be connected in selectively. Such a set of multi-channel band-pass dividing filters which can be switched in and out individually also includes the filters with low- and high-pass characteristic as well as the band rejection filters.

2. The Connecting of Filters in Parallel

The laws of quadripole theory do not give any direct information concerning the exact behaviour of n individual

quadripoles connected together to form a (2n + 2)-pole dividing network. The theory of dividing networks is difficult. However, where optimum output matching is not required, the following precautionary measures, if observed when connecting several filters in parallel, are entirely adequate for all practical purposes.

Depending on whether the filter inputs are Π-connected or T-connected, in the real or imaginary range care must be taken to see that the occurring zero points of the parallel wave resistances do not permit the total virtual input resistance to break down and thus to short circuit other transmittance regions. Fundamentally, if only for purposes of the power balance sheet, the wave resistance of a filter in the transmittance range is primarily real and in the rejection range is primarily imaginary. Von Brandt(1) has dealt completely with all possible wave resistance functions.

However, if the dividing system is fed with a sufficiently low generator resistance, then in order to obtain an optimum operational damping a corresponding generator resistance must be connected in front of the individual filters. This will serve at the same time as a decoupling resistance. For the sharpest decoupling - especially in the earth-asymmetrical filters which do away with certain circuit elements - a cathode stage is recommended in the case of input parallel connection to act as an impedance transformer. At an executed cathode stage with tube EF14 in triode connection, R_{in} = 5M Ω and R_{out} = 150 Ω were measured. Owing to the 100% negative feedback, disturbing distortions above 1% only occur at input voltages above 25 v.

3. The Low-pass(2)

Along with a band-pass dividing network, which might constitute the core of the filter equipment in an electronic studio, there are also many possibilities of applying filters for

pure high- and low-pass characteristics. In front of the band-pass filters as prefilters they can increase the band-pass selection, where required, on one damping flank. Moreover, connecting high- and low-pass filters in series* results in the otherwise difficult production of band-pass with variable absolute band width for a constant mean transmittance frequency ν_o.

A few filter circuits suitable for variable limiting frequencies, will now be discussed, starting with the low-pass filters. The remarks about the greater steepness of attenuation will apply only to genuine quadripole circuits or other circuits capable of similar performance.

From among the known low-pass filters in bridge and branching connection, only those are chosen which require the minimum circuit means to produce a favourable damping characteristic, in particular those which eliminate the circuit elements for varying the limiting frequency. The limiting frequency of a filter showing no losses (reactance quadripole) is characterized by the theoretically exact transition from the rejection to the transmittance range. In the formulae the corresponding angular velocities $2\pi\nu_g = \omega_g$ are employed. Even with optimum matching, transition to the rejection range is made slower owing to the unavoidable losses, so that practically speaking the standard frequency $\Omega = \frac{\omega}{\omega_g} = 0.9$**, instead of 1.0 is the most that can be expected for useful transmittance range. If the attenuation maximum of the rejection range is not near $\Omega = \infty$, i.e., near zero or infinite frequency, but is a finite

*Such filters are already on the market, e.g. the filter in the "Sound Spectrum Analyser Type 1000 B" of the Western Electro-Acoustic Laboratory, and the "Sound Analayser Type 420-A" of H.H. Scott Inc.

**Only with special means for wave-resistance smoothing, otherwise less favourable.

frequency for an expansion of the attenuation function with the factor m as the attenuation pole, then this will be expressed by the standard frequency $1 < \Omega_\infty < \infty$.

From a constructional point of view it is more practical to switch condensers than coils. The 50 db. damping considered necessary in the rejection range demands at least a two-membered filter; for unexpanded members the standard frequency interval $\Omega = 2.5$ of the limiting frequency is needed.

The low-pass bridge circuit according to Fig. 1 permits an expansion of the attenuation increase without additional outlay. The dimensional data can be derived from the figure. A variation of the transmittance range by means of a variable condenser with constant self-induction of the coils requires a corresponding change in the matching resistance R.

Fig. 2 shows a monovalent differential filter developed from the bridge filter, doubled and connected in series like the radio drama compressor W49* in broadcasting. When the coils are kept constant the required proportional matching is carried out by means of a transformer with a differential winding. To use the differential transformer itself for the change in transmittance is impracticable owing to the unavoidable variable leakage, which is very disturbing for the differential formation. In practice, it is very desirable to be able to switch the filters either to parallel or series connection as required, like amplifiers with high ohmic input and low ohmic output. In addition, as already mentioned, provision

*This apparatus consists in the series connection of a monovalent, unexpanded low- and high-pass filter in differential connection with variable limiting frequency. Variation is accomplished by means of coils and condensers, as the wave resistances of both filter parts must always be adapted for any desired limiting frequencies selected.

should be made for variable amplification in every channel and each should be capable of being switched on and off without any clicking. Therefore, tubes are used to shut the filters off.

As Fig. 3 shows, for this purpose a cathode stage was placed at the input and a normal RC amplifier stage at the output, the latter capable of being switched on or off as desired by means of a non-clicking strong negative voltage fed to the control grid. The input virtual resistance of the cathode stage comes to several M Ω, while the virtual internal resistance of the filter side is only approximately 150 ohms. In practice, in the circuit, according to Fig. 3, the filter matching resistance R_i can be changed from 500 ohms to 20 kilo ohms without any appreciable change of amplification in the cathode stage, and the limiting frequency can then be varied accordingly.

4. The High-pass Filter[2]

Owing to the possibility of expanding the damping characteristic at the X-part without additional circuit means, the discussion of the high-pass filter will be based on a circuit such as that shown in Fig. 4a. For a variable high-pass filter the inverse resistance X-section with only one variable condenser, in accordance with Fig. 4b, is more advantageous. For practical execution, the elements of circuit 4a, or better still 4b, can be used in a single-valued differential circuit (W49) doubled in series in accordance with Fig. 2.

As a low-cost two-valued expanded filter, adequate for most purposes, the circuit according to Fig. 5a with differential coils is recommended. For a high-pass filter, of course, the circuit shown in Fig. 5b can simply be employed without adding a third variable condenser, using standard coils of simple construction. The advantages of the tube circuit of Fig. 3 are still entirely valid for the high-pass filter, of course, and combining them with the circuits of 5a or b appears

to be the best solution for the requirements of an electronic studio.

5. General Considerations Concerning Variable Band Filters

(a) Damping and wave resistance curve[1,3]

A few of the many possible band-pass branching circuits are shown in Fig. 6. The equations for dimensioning the inductances of all band-pass filters in branching circuits contain, in addition to the constant expansion factors, the following two basic elements:

$$L_T = \frac{Z}{2\pi\Delta\nu} \quad \text{and} \quad L_{II} = Z \frac{\Delta\nu}{2\pi \times \nu_o^2} .$$

For a small absolute band width $\Delta\nu$, $L_T > L_{II}$ and hence also the cyclic quality vary greatly. The advantage of the Class 1 band-pass filter for the case of small $\Delta\nu$ lies in the fact that only inductances according to L_T or L_{II}, are found in a single section for the same cyclic quality, while all Class 2 branching filters contain both basic inductances.

The wave form of the operational damping in the transmittance region which is obtained on actual matching with a wave-resistance curve of Class b, is adequate for the requirements of electronic music, so that a wave-resistance flattening with half sections of Class d is generally unnecessary. A wave-resistance curve of Class b_i is sufficiently symmetrical only for the case of symmetrical distribution of the damping poles. A more exact investigation of the matching conditions is always to be recommended in these cases with a view to avoiding too great a pulsation factor in the band damping. Class b_i wave resistance, however, provides the advantage, as the filters of damping Class 1^x in Fig. 6/VII and Fig. 6/VIII show, that coils can be replaced by condensers. The so-called zigzag filter of

Class 2_x in Fig. 6/IX benefits from this, whereas the half section according to Fig. 6/X does not. The zigzag filter according to Fig. 6/XII, on the other hand, eliminates some condensers. In Class I band-pass filters, or in the case of expansion according to Class 1^x, the damping has only one infinity point in the upper or lower suppression range (Figs. 7a and b), so that for a symmetrical damping curve normally two sections must be connected together with supplementary characteristics.

The use of this Class 1 or 1^x filter is recommended for smaller band-pass widths, starting with thirds, since its damping increases in the suppression range with smaller band-pass width and the band damping due to cyclic losses remains sufficiently small compared with Class 2 or Class 2^x filters. For relatively wide bands, especially from octave filters up, only those of Class 2 are feasible.

(b) Investigation of the possibilities of displacing the transmittance range of the band-pass filter along the frequency axis, given the condition L = const.[2].

For constant relative band width in the double logarithmic system of coordinates, a straight line inclined at 45° is always obtained. For both basic inductance equations, Z would have to increase proportionally with ν_o, if L is to remain constant.

The majority of simple filters I to VIII of Classes 1 and 1^x, as well as the zigzag filters XI and XII of Class 2, still contain the factors $n = \sqrt{\nu_1/\nu_2}$ in the dimensioning equations and only at constant relative band width does this remain unchanged during a variation of ν_o. In the longitudinal and lateral meshes of the simple filter Class 1 and Class 1^x, however, in each case only inductances L_{T_1} and L_{T_2} or L_{Π_1} and L_{Π_2} occur in pairs, as more exact formulae show.

Since in the pairs of equations n occurs alternately in the numerator and the denominator, then in the simple filter in which the factor n plays a part a variation is possible only if n = const. i.e., for a constant relative band width. Therefore, practically speaking, a change in the band width of the simple filter according to any law other than that of constant relative band width by corresponding variation of the expansion factor n, is out of the question.

6. The Tuned Double Filter as an Octave Filter

For division of the audible range into octave sections a band-width progression with constant relative band width is required. For such a wide band a damping curve according to Class 1 or Class 1^x is unfavourable. The most useful filter for this task has been found to be the tuned double filter of Class 2 or Class 2^x (Fig. 6/IX, X). If the variable octave filter be designed with fixed condensers, then Z must fall with rising octaves (Fig. 7).

7. Control of Band Width Where the Band Centre Remains Constant[5,6]

On studying the equations for determining the dimensions of various audio-frequency band-pass filters we find that only in the unexpanded simple filter according to Fig. 6/III can this problem be solved by simultaneous variation of Z and C. By leaving a filter according to Fig. 6/III under no load between tubes without closing Z we then obtain a so-called radio filter circuit which enables the band width to be controlled for ν_o = const. merely by varying the coupling condenser or a mixed inductance-capacitance coupling. An altogether satisfactory method of solving this problem, and one which is practical also for band widths that are not too small, consists in the possibility mentioned at the beginning, of connecting suitably variable high- and low-pass filters in series.

If band-pass filters constructed on other principles are already available in an electronic studio, the negative feed-back known from amplifier technique makes it possible to attain an improvement in selection. Accordingly, the filtered out voltage would have to be returned by means of negative feed-back to the filter input again. It is necessary, however, for the negative feed-back to remain free from phase shift for each feed-back factor. This condition can be exactly satisfied in the negative feed-back circuit only with a tube which has a suitably dimensioned anode resistance and affords a suitable choice of the two coupling capacitances. According to preliminary tests this negative feed-back method appears to be the most favourable in the audio-frequency region for a control factor up to 1 : 20. Even if band-pass filters with a width of only a few c.p.s. are desired, this method is the only possible one.

8. Requirements of a Filter for Partials

The shaping of the timbre in a given raw material which is rich in overtones should be as free as possible. It is necessary, therefore, not merely to be able to select the individual overtones but also to be able to vary their relative intensities within wide limits. The channel means for separating individual harmonic overtones is determined by the necessity of having a filter with constant absolute band width. These variable band position filters can be made very advantageously as multiple-valued bridge filters with constant coil and condenser variation for a constant wave-resistance curve(7). If it were desired to carry out this method exactly and to separate the 25th and 26th harmonic, for example of a fundamental in the range of 100 to 200 c.p.s., it would be necessary to have an uneconomically large number of filter channels. In any case, it is musically more interesting to be able to separate the lower harmonics individually, while taking the higher ones together. A compromise solution would therefore

be to have a channel scheme in which absolute band widths were provided with the band range position rising by groups (Fig. 8). However, a single variable filter by no means satisfies the practical need for a free sound variation, since each individual spectrum would have to be laboriously composed by montage in chronological sequence. An improvised mixing of partials by ear is quite impossible. Therefore, the possibility was provided of connecting all filter channels or,at least, in the case of variable filters, several channels simultaneously in parallel, with the possibility of switching them off and on at will without clicking. Push buttons controlling relays serve to switch these on and off, while simple rotary potentiometers are employed for amplitude control.

The fact that the filter amplifiers must be switched each time to attain the desired effect, but at the same time are used in many more different ways, is not found to be troublesome.

As Fig. 9 shows, the output resistances of the in-out stages are a.c. parallel. In order to prevent the total operational resistance applied to the amplification from becoming too small, only 10 stages can be operated in this manner in parallel. Where there are many more channels, separate decoupling stages must be placed before the final stages. Such a stage with tubes of closed off "filter amplifiers" with high ohmic input and low ohmic output can be connected together in accordance with the usual studio technique employing a few filters of a similar kind as well as amplifiers and control elements. The wave resistances of the filters no longer appear externally, so that the dimensioning of the oscillation resistances in the meshes can also be reduced, without other limitation, to the most economic solution.

9. Description of an Executed Partials Filter with Six Octave Filters connected in Parallel

Although at the very beginning it was clear that the division of the filter ranges into six octaves from 200 to 12,800 c.p.s. in many cases is still too coarse, nevertheless because of the expenditure the octaval division according to Fig. 9 was decided upon as the initial equipment for the Cologne Electronic Studio. Variability within the individual filters was not required for the six channels connected in parallel. The corresponding damping curve, which also gives an advantage over the trinomial ground chain, is shown in Fig. 10. The amplitudes of the partials can be amplified in the individual channels up to 20 db. and can be attenuated by 30 db; in the most unfavourable case an additional external voltage interval of 55 db. could be measured. The circuiting principle of an octave filter is shown in Fig. 10b. In calculating the basic elements determined according to the formulae given below the figure, L_T, L_{II}, C_T and C_{II} should be multiplied by the factors given in the circuit.

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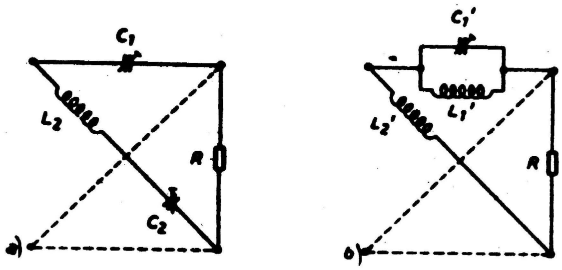

Fig. 4

(a) High-pass filter; dimensioning formulae of the unexpanded X-member:

$$C_1 = \frac{0.6}{R \times \omega_g} \qquad C_2 = \frac{0.6}{R \times \omega_g} \qquad L_2 = \frac{R}{\omega_g \; 0.6}$$

(b) Resistance-reciprocal high-pass filter; dimensioning formulae of the unexpanded X-member:

$$L_1' = \frac{R}{\omega_g} \qquad L_2' = \frac{R}{\omega_g} \qquad C_1' = \frac{1}{R \times \omega_g}$$

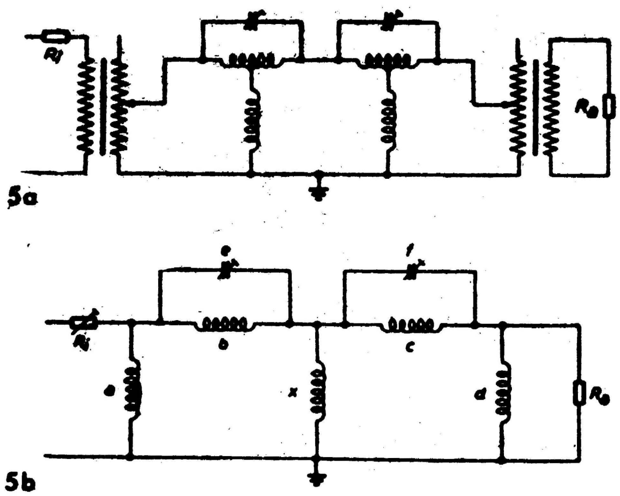

Fig. 5

(a) Two-membered expanded high-pass filter with differential coils for variable limiting frequency from $\Omega = 1.9$; $b \geqq 50$ db.

Region of variation and expansion factors as for Fig. 3.

(b) Two-membered expanded variable high-pass

$$a = \frac{1}{m_1} \times L_2 \qquad e = \frac{1}{2} \times \frac{1}{m_1} \times C_1$$

$$b = 2\,\frac{m_1}{1 - m_1^2} \times L_2 \qquad f = \frac{1}{2} \times \frac{1}{m_2} \times C_1$$

$$c = 2\,\frac{m_2}{1 - m_2^2} \times L_2$$

$$d = \frac{1}{m_2} \times L_2 \qquad x = L_2 \times \frac{\frac{1}{m_1 \times m_2}}{\frac{1}{m_1} + \frac{1}{m_2}}$$

(a and d connected in parallel).

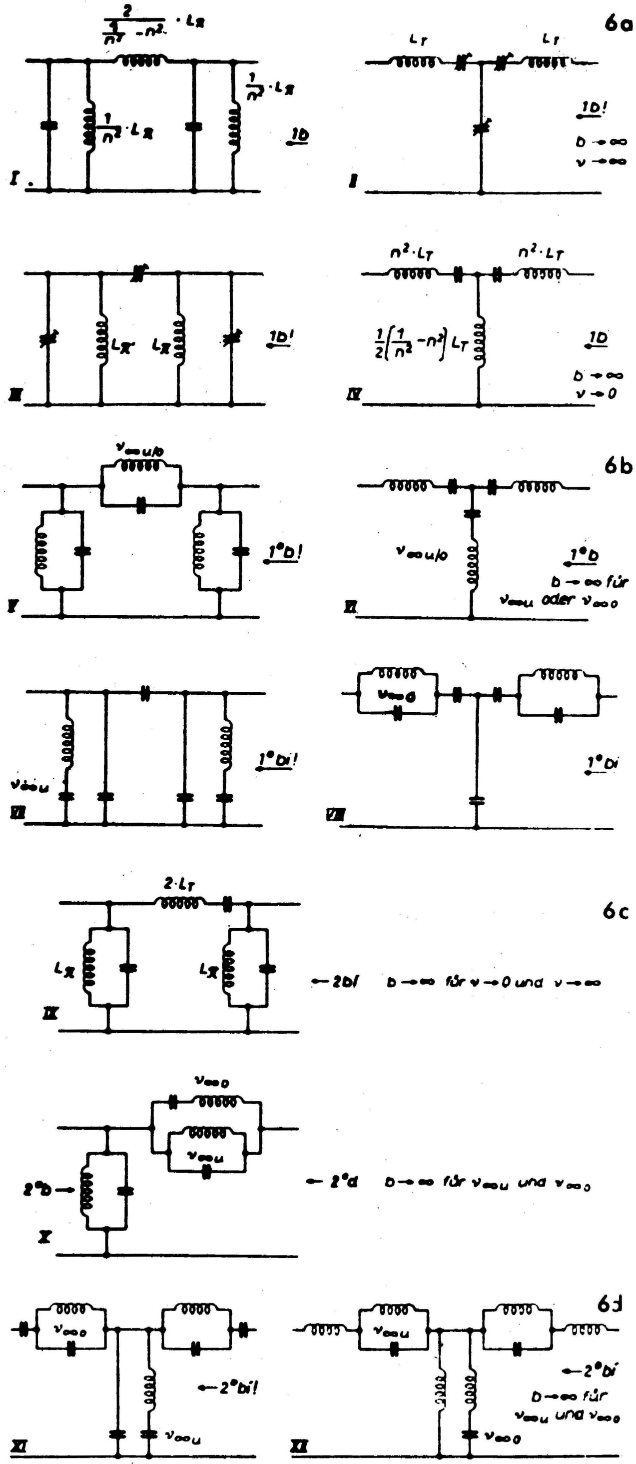

Fig. 6

Several band-pass filter types in branched circuits arranged in order of damping and wave resistance. Recommended circuits are indicated by a "!". At higher frequencies examples V and XI have the advantage that the coil capacitities can be organically compensated.

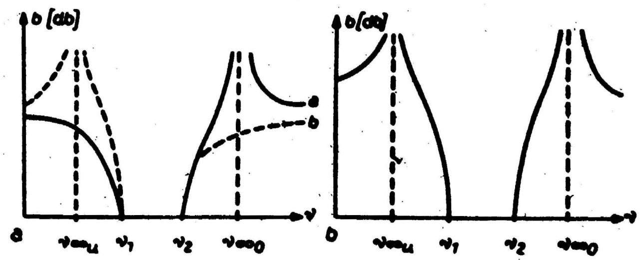

Fig. 7

Damping curve of the filters shown in Fig. 6.

(a) According to damping class 1^x.

(b) According to damping class 2^x.

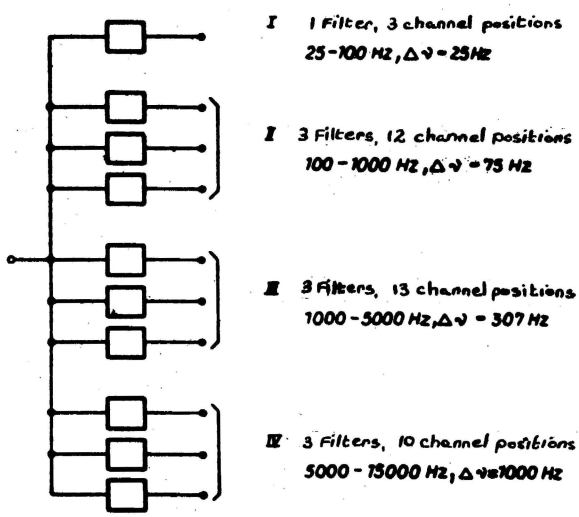

Fig. 8

Channel plan for a partials filter consisting of 10 parallel separately variable band-pass filters with grouped constant absolute band width in 38 channels.

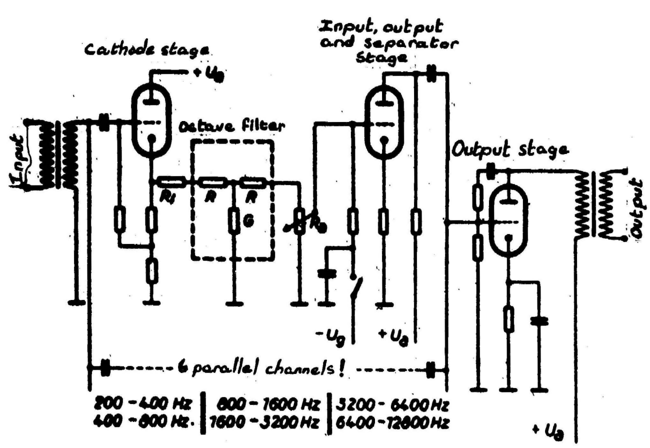

Fig. 9

Partial filters with six octave filters connected in parallel. (200 to 12,800 c.p.s.)

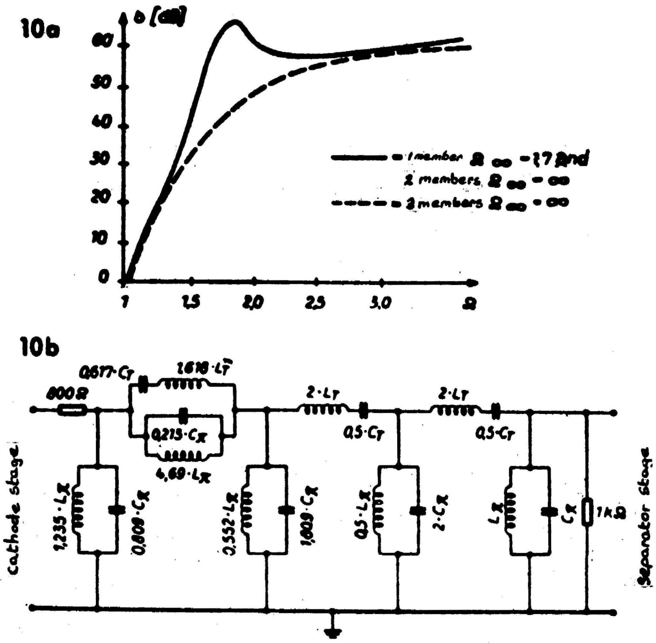

Fig. 10

(a) Standardized damping curve of the octave filter used in the partials filter.

(b) Basic circuit of the octave filter compressing two basic members and a full member Ω ∞ = 1.7.

$$L_T = \frac{Z}{2\pi \times \Delta\nu} \qquad L_{\Pi} = \frac{\Delta\nu}{2\pi \times \nu_o^2} \;;$$

$$C_T = \frac{1}{Z}\,\frac{\Delta\nu}{2\pi \times \nu_o^2} \qquad C_{\Pi} = \frac{1}{Z} \times \frac{1}{2\pi\Delta\nu}$$

$$(R = 1\ \text{k}\,\Omega; \quad Z = 0.8\ \text{k}\,\Omega)$$