A frequency-dependent compandor system for high-quality sound signal distribution

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THE BRITISH BROADCASTING CORPORATION
ENGINEERING DIVISION
A FREQUENCY-DEPENDENT COMPANDOR SYSTEM FOR HIGH-QUALITY
SOUND SIGNAL DISTRIBUTION

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A FREQUENCY-DEPENDENT COMPANDOR SYSTEM FOR HIGH-QUALITY SOUND SIGNAL DISTRIBUTION

SUMMARY

Compandors of the type used in telephony for the improvement of signal-to-noise ratio are unsatisfactory for high-quality sound circuits because of the objectionable effect produced by programme-modulated noise. In the case of 'white' noise, this difficulty can be largely overcome by splitting the audio-frequency range into two parts and applying the compandor to the upper band only, but considerable refinement in instrumentation is necessary to do this without degrading the programme quality in other ways.

The report describes an artifice which allows the principal advantages of such a two-channel compandor to be realized without the need for band-splitting filters in the programme circuit. This method, which involves the use of high-frequency pre- and de-emphasis in conjunction with a limiter controlled by a pilot tone, has been applied experimentally to a p.c.m. system, the pilot tone in this case being locked to half the sampling frequency. The possibility is discussed of simplifying the instrumentation of the system by taking advantage of certain characteristics of the ear.

1. INTRODUCTION

1.1. General

The compandor arrangement which is the subject of this report may be described as a combination of several known noise reduction methods, each of which, by itself, is in some way unsatisfactory for high-quality sound circuits; it is therefore appropriate to commence by discussing these various techniques individually.

1.2. High-frequency Pre- and De-emphasis

A well-known method of improving signal-to-noise ratio in cases where the noise is predominantly high-pitched is to modify the spectrum of the signal to be transmitted by a pre-emphasis network which increases the level of the high-frequency components; at the receiving end of the system, a de-emphasis network introduces a corresponding attenuation at high frequencies thus restoring the signal spectrum to normal, at the same time reducing the predominant components of the noise.

The above artifice depends on the fact that the high-frequency components of the signal contribute, on the average, only a small proportion of the total power and may therefore be selectively amplified without producing a corresponding increase in the overall level. However, in high-quality sound systems there are occasions, too numerous to be ignored, on which the amplitude of high-frequency components in the signal is so much greater than the average that the level after pre-emphasis temporarily exceeds the prescribed maximum. In these circumstances, according to the nature of the transmission system, either audible distortion may occur or interference may be produced in other communication channels; if, to avoid these effects, a quick-acting limiter is provided to restrict the maximum signal level, the sudden reduction in gain which accompanies a sound having a large high-frequency content is aesthetically objectionable. For these reasons, the increase in signal-to-noise ratio obtained by pre-emphasis in high-quality sound systems has to be restricted to between 3 dB and 7.5 dB, depending on the degree of occasional overmodulation or temporary gain reduction that can be tolerated. For many purposes, a greater improvement in signal-to-noise ratio is required and other devices have to be employed.

1.3. Syllabic Compandors

The effective signal-to-noise ratio obtainable from a sound signal transmission system can be increased by using a syllabic* compressor to raise the minimum signal level at the sending end without altering the maximum level. At the receiving end

* The term 'syllabic' is used to describe those automatic gain control devices whose gain returns to its 'no-signal' value within, say, two hundred milliseconds of the cessation of the signal, and which are thus able to follow level fluctuations up to syllabic frequency.
of the system, a syllabic expander having an output-
input characteristic inverse to that of the compressor
can then be used to restore the signal to its original
form, attenuating the noise in the process; this
combination of compressor and expander, known as
a syllabic compandor, has been in use for many
years in long-distance telephone circuits.

When the level of the incoming signal rises,
the gain of the compressor is automatically reduced
and that of the expander increased. Any noise
originating in the transmission system is therefore
subject to a variable attenuation, and the propor-
tion of it appearing at the output of the expander
rises and falls with the signal level. The common-
est manifestation of this phenomenon occurs with
"white" noise, which is heard as a high-pitched
sound, and, when amplitude-modulated by the
signal, gives a subjective effect commonly referred
to onomatopoeically as 'hush-hush.' In the case of
the relatively narrow-band telephone systems for
which compandors were originally designed, this
'hush-hush' noise is to a large extent masked by the
speech signal itself, and, even when audible, can
be tolerated as long as intelligibility is not affected.
However, in wide-band systems for the transmission
of broadcast-quality programme material, it fre-
quently happens that the predominant components
of the signal and of the programme-modulated noise
lie in widely separated parts of the spectrum, so
that very little masking occurs; in these circum-
stances, even a small amount of such noise can be
aesthetically objectionable, producing in some cases
an effect resembling high-order harmonic distortion.

1.4. Split-band Compandors

One method of avoiding the 'hush-hush' effect
is to divide the audio-frequency band, by means of
filters, into two or more parts, and to provide separate compandor systems for each frequency
region; each component of the noise is then
modulated only by those components of the pro-
gramme signal which are sufficiently close to it in
frequency to produce effective masking.

In cases where the noise components at middle
and low audio frequencies make a negligible con-
tribution to the audibility, it is possible to apply
the artifice just described in a simpler form by
dividing the a.f. band into two parts and companding
only the upper part. The action of the compressor
then introduces additional, level-dependent gain in
the upper-frequency region only, giving the effect of
a variable high-frequency pre-emphasis; the
compensating action of the expander can likewise
be regarded as a variable de-emphasis.

The split-band compandor systems described
above can be effective in reducing noise but have
the disadvantage of requiring crossover filters in
the programme chain at both sending and receiving
ends of the system. If these filters are not accurately
matched, the overall response/frequency charac-
teristics of the system will not be uniform in the
crossover region; further, if the gain changes
introduced by the compressor and expander are not
exactly complementary, the relative responses of
the system in the different frequency bands will
fluctuate during the programme. A high degree of
precision is therefore necessary in the construction
and maintenance of such compandors.

2. PRINCIPLE OF PROPOSED SYSTEM

2.1. General

The purpose of the present report is to de-
scribe a method of companding which enables the
principal advantages of a two-channel split-band
compandor to be realized in practice without the
necessity for band-splitting filters in the signal
chain. This method consists, in brief, of applying a much higher degree of pre- and de-emphasis than
would normally be allowable, restricting the maxi-
mum level of the pre-emphasized signal by a quick-
acting limiter, and using a pilot-tone-operated
automatic gain control at the receiving end of the
system to restore the signal level to normal. The
artifice is applicable to any system of transmission
in which the channel capacity is sufficient to allow
the introduction of a modulated pilot tone without
unduly narrowing the frequency range of the pro-
gramme signal. An experimental compandor of this
type has been constructed and tests have been
carried out in conjunction with a p.c.m. system of
a type that could be used for high-quality sound-
signal distribution; an effective noise reduction
of some 15 dB — equivalent to adding 2½ bits to the
pulse code — was achieved without the introduction
of any noticeable side effects.

Fig. 1 shows a simplified block diagram of the
companding system. The programme signal is
applied to a network which pre-emphasizes the
upper-frequency end of the spectrum and thence to
a band-stop filter BSF1 which removes all com-
ponents lying within a narrow-frequency band
centred on a pilot frequency \( f_0 \) — which for a 15 kHz
canal bandwidth could conveniently be, say,
14.75 kHz. At the output of the band-stop filter, a
steady tone of frequency \( f_0 \) is added to the pro-
gramme signal at a level some 20 dB below the
peak level of that signal. The composite signal
is then applied to the input of the transmission link
through a limiter L1, consisting of a variable-
gain amplifier controlled by rectified signals from
its own output. This limiter has zero dB gain for
small signals, but is set to reduce gain whenever
its input signal level exceeds the maximum which

* The construction and part of the development of the ex-
perimetal equipment was carried out by K. F. Lansdowne,
who also assisted in the ensuing tests.
may safely be applied to the link. The pilot tone is thus modulated in amplitude according to the variations of gain occurring in L1. The incoming signal level is such that in the absence of pre-emphasis, the maximum signal amplitude at the input of L1 would just reach the limiting point. In the presence of pre-emphasis, programme signals having a high proportion of their power in the upper part of the audio-frequency band will then cause this limiter to operate; the maximum degree of gain reduction produced by L1 in these circumstances will be equal to the maximum degree of pre-emphasis applied.

At the far end of the link, the programme signal passes through the variable-gain element of a second limiter L2. The gain of this limiter is controlled by the level of the pilot tone, which is separated from the programme signal by an appropriate band-pass filter BPF1 in the control chain. Matters are so arranged that an increase in the pilot tone level produces a corresponding decrease in the gain of L2, and that in the absence of a programme signal, the gain of L2 is depressed by an amount at least equal to the maximum gain reduction that could ever occur in L1; a sufficient reserve of gain is thus available in L2 to compensate for any gain reduction produced by L1. The programme signal leaving L2 passes through a band-stop filter BSF2 to remove the pilot tone and thence, through a de-emphasis network, to the output.*

As long as the level of the pre-emphasized signal at the sending end of the link does not exceed the prescribed maximum, the gains of L1 and L2 remain constant, the former at its maximum value and the latter at its minimum. In these circumstances the improvement in signal-to-noise ratio is that due to the pre- and de-emphasis. Whenever the programme material contains sufficient energy in the upper part of the audio-frequency band, the maximum signal amplitude at the input of L1 is such that in the absence of pre-emphasis, above the prescribed maximum input to the link, L1 operates, reducing gain by the amount necessary to avoid overloading the link; the resulting fall in the level of the pilot tone arriving at the far end is compensated by an increase in gain of L2, which in the process restores the level of the programme signal. For complete restoration of the signal it is, of course, essential that the maximum rate of change of gain should be such that the products generated by the modulation of the programme and the pilot tone at the sending terminal can be accurately reproduced at the receiving terminal.

In the foregoing it has been tacitly assumed that the peak amplitude of the pilot tone is negligible compared with that of the programme signal. If this is not the case, the addition of the pilot tone will raise the maximum level of the combined signal entering L1 to slightly above the limiting point - thus causing the compandor to operate - even when the programme signal contains only components which are too low in frequency to be affected by pre-emphasis. If, however, the peak amplitude of the pilot tone is 20 dB below that of the programme signal, low-frequency components alone can produce only 1 dB of companding and no audible 'hush-hush' effect results.

2.2. Comparison with Other Methods of Noise Reduction

The action of the above device resembles in some respects that of a two-channel split-band syllabic compandor. However, because the gain control at the sending end of the system is effected by a limiter rather than by a compressor of the normal type, reduction in gain - and the corresponding increase in noise at the output - does not occur until the signal level in the link reaches the maximum permissible value, when conditions for noise masking are most favourable. Moreover, neither the static nor the dynamic characteristics of L2 are critical provided that the loop gain can be made high enough to ensure substantially complete level restoration, and the operating time constants much shorter than those of L1.

It should be noted that the arrangement described above contains a number of features al-
ready known to the art. A compandor system operating on a pre-emphasized signal and using an amplitude-modulated pilot tone, carried by a separate channel, to control the r.m.s. level, has been developed for sound-film recording. The "Lincompex" system developed by the GPO for radio communication works on a similar principle but controls the peak value of the signal, and uses limiters to give an infinite compression ratio, with a frequency-modulated pilot tone for level control. Both of these systems, however, employ complementary gain-control devices at the sending and receiving terminals. The artifice of mixing the pilot tone with the signal at the compressor and using a closed-loop gain-regulating circuit rather than a complementary variable-gain element at the receiving terminal has been employed in an auxiliary gain-corrector forming part of a Siemens and Halske carrier-frequency compandor. The present combination of the various devices is, however, believed to contain an element of novelty.

The system shown in Fig. 1 is not fully equivalent to a split-band compandor, for a price has to be paid for the elimination of the band-splitting filters from the programme chain. Without these filters, the gain reduction produced by L1, and the corresponding gain restoration effected by L2, are both applied to the whole audio-frequency band. A situation could therefore be envisaged in which the arrival of a signal having its energy concentrated in the upper part of the audio-frequency range would increase the audibility of low-frequency noise. This effect is not likely to be serious in transmission systems which produce only white noise. In these systems the ratio of signal to low-frequency noise components is usually well in excess of requirements, and in exchange for a certain increase in signal-to-noise ratio at high frequencies, the signal-to-noise ratio at low frequencies may safely be reduced by the same amount; it should be noted that this condition is already tacitly assumed to apply in 'music-in-band' systems, in that the standard 'pre-emphasis' characteristic for such circuits is designed not only to raise the signal level in the upper part of the a.f. band, but to lower it by a comparable amount in the lower part. This characteristic is shown in Fig. 2 curve (a); the dotted curve (a') shows the corresponding de-emphasis.

3. DESIGN CONSIDERATIONS

3.1. General

The experimental compandor system to be described was designed primarily for use with a p.c.m. system, referred to in Section 5, which transmitted audio-frequency signals with a nominal upper limit of 14 kHz. It was decided in the first instance to restrict the upper-frequency limit of the programme signal to 13 kHz and to make the frequency of the pilot tone 13.75 kHz; allowing for the imperfections of practical filters, this would allow about ± 250 Hz for the sidebands produced by the modulation of the pilot tone.

![Fig. 2 - (a) and (a')]: CCITT characteristics for pre- and de-emphasis respectively (b) and (b'): Pre- and de-emphasis characteristics respectively, as modified for pilot tone compandor

3.2. Sending Terminal

The bandwidth required to accommodate the amplitude-modulated pilot tone is in practice determined by instrumental considerations, and here it is necessary to consider in greater detail the action of L1 and to depart from the simplified arrangement shown in Fig. 1.

To avoid audible distortion due to momentary overloading of the link on the arrival of a high-level signal, the limiter L1 as shown in Fig. 1 would have to be capable of carrying out any necessary gain reduction within 100 μs or less. Such a high rate-of-change of gain would, however, introduce audible modulation products into the programme signal. It would also produce sidebands of the pilot tone occupying a total bandwidth comparable with that of the programme signal; unless this wide-band control signal were transmitted intact over the link, it would not be possible to cancel all the audible modulation products by the action of L2.

It is, however, possible to avoid momentary overloading of the transmission link without demanding an excessive rate-of-change of gain in L1, by recourse to an artifice, originally employed in a
general-purpose limiter and described in an earlier report, by which unwanted modulation products can be reduced to negligible proportions. Fig. 3 shows the essential features of the circuit. The incoming signal is applied to two identical variable-gain amplifiers VG1 and VG2, in the latter case via a delay network T1. The signal-output of VG1, rectified and smoothed, is fed back to the gain-controlling terminal of VG1 in such a way as to produce the required output/input characteristic; the operating time of the control loop thus formed can be made as short as may be required. The control signal which operates on VG1 is also applied to VG2 through a low-pass filter LPF1, which restricts the rate-of-change of gain sufficiently to avoid the generation of audible modulation products; the delay in network T1 is such as to offset the finite operating time of the VG1 control loop plus the time lag introduced by LPF1 and thus to avoid signal overshoot occurring under transient conditions. The signal to be transmitted over the link is then taken from the output of VG2.

![Diagram](image)

**Fig. 3** - Detailed block diagram for limiter L1 giving restricted operating speed without signal overshoot.

In the original general-purpose limiter to which Fig. 3 refers, the cut-off frequency of LPF1 was about 1.5 kHz and the delay T1 about 1/3 ms. If the limiter L1 in Fig. 1 were of this type, the frequency band required to accommodate the modulated pilot tone would extend some 1.5 kHz above and below $f_0$. For the present purposes, therefore, the bandwidth was narrowed to the ±250 Hz allowed in Section 3.1, by reducing the cut-off frequency of LPF1 to 250 Hz — thus restricting the maximum rate-of-change of gain of VG2 — and the increased time-delay in LPF1 was allowed for by increasing the delay time of T1 to 2 ms. This process could in principle be carried still further, at least up to the point at which the anticipatory action of L1 — and hence of L2 — led to an audible rise in noise at the output of the system before the arrival of a signal. In practice, however, the maximum value of the delay T1 would be restricted by economic considerations to a few milliseconds.

In principle, the gain recovery time of L1, like the operating time, could be made about 2 ms; however, the gain could then vary appreciably during each cycle of a signal containing components at low audio-frequencies, thus producing waveform distortion. The degree of precision in the subsequent gain restoration which would be necessary to remove this distortion is much higher than that normally required in a syllabic compandor — for example, while a voltage gain error of ±5% fluctuating at syllabic rate may be tolerable, a corresponding amplitude modulation within one cycle of the programme signal could produce audible distortion. It is important therefore that the recovery time of L1 should not be too short. If, on the other hand, the recovery time were made too long, noise from the link would be audible for a short time after cessation of a signal, when the gain of L1 remains low and that of L2 high. In practice a value of the order of 150 ms gives an adequate compromise.

![Diagram](image)

**Fig. 4** - Detailed block diagram of receiving terminal of experimental compandor.

### 3.3. Receiving Terminal

In the simplified circuit illustrated in Fig. 1, the maximum speed of operation of L2 is restricted by the time delay in BPF1, which is included in the control loop. In order to achieve rapid gain restoration, the arrangement shown in Fig. 4 was used at the receiving end of the experimental system; it is based on the artifice illustrated in Fig. 3. The modulated pilot tone of frequency $f_0$, separated from the programme signal by the band-pass filter BPF1, is applied to one of two identical variable-gain amplifiers, while the programme signal, freed from the pilot tone and its sidebands by the band-stop filter BSF2, is applied via a delay network T2, to the other. Rectified and smoothed pilot tone from the output of the first of the two variable-gain amplifiers provides a common control voltage for both, so that the required gain-regulating action is duplicated in the programme circuit. Filter BPF1 is no longer within the control loop of L2, so that the delay which it introduces need not restrict the speed of operation. The delay in T2 is made 2 ms — sufficient to offset the time lag produced by BPF1 and by the smoothing applied to the rectified control voltage. Comparing Fig. 4 with Fig. 1, it will be noted that the programme signal and pilot tone do not pass through the same variable-gain element, so that the possibility of intermodulation between them at this stage is avoided; the same
artifice could be applied, if it were necessary, at the sending terminal to avoid intermodulation between programme and tone in L1, by providing a third variable-gain element operated by the same control voltage as VG2 in Fig. 3.

3.4 Pre- and De-emphasis Characteristics

The pre- and de-emphasis characteristics adopted for the experiments are shown in Fig. 2, curves (b) and (b') respectively. These are identical in form with the standard characteristics of curves (a) and (a'), but the circuit gain is increased by 7 dB at the sending end of the system and reduced by the same amount at the receiving end; the resulting pre- and de-emphasis has little or no effect on components of the signal below 400 Hz. The de-emphasis characteristic of Fig. 2 curve (b'), when applied to white noise, reduces the level as judged subjectively, by about 15 dB; the improvement in signal-to-noise ratio thus obtained is equivalent, in a p.c.m. system, to an extra 2½ bits per sample.

The characteristics (b) and (b'), adopted arbitrarily in the first instance, were found, by subsequent experiment with the p.c.m. system referred to earlier, to represent a rough optimum. Displacement of the curves downwards in frequency by one octave, thus allowing lower-frequency components to actuate the variable-gain system, produced audible programme-modulated noise; displacement towards the high frequencies naturally increased the residual noise from the system without any corresponding advantage. The de-emphasis curve (b'), which gives an 18 dB step in attenuation, was found to give a reduction in subjective noise within 1 dB of that obtained when the height of the step was increased indefinitely; there was therefore no need to increase the height of the step, and hence the operating range of the variable-gain elements at either end of the system, beyond 18 dB.* The effect of increasing the slope of the pre- and de-emphasis curves was not investigated; on the basis of the results already obtained it was considered that any advantage thus obtainable would be marginal and certainly insufficient to outweigh the difficulty of making the two networks accurately complementary.

4. SIMPLIFICATION OF CIRCUIT

4.1 General

From Figs. 1, 3 and 4 it will be seen that the only parts of the equipment peculiar to the company system are the limiter L2** at the receiving terminal, the filters BSF1, BSF2, BPF1, and the delay networks T1, T2. Of these items, the delay networks are the most expensive; it is therefore of interest to know how far the delay which they provide may be reduced below the value required for complete restoration of the signal, without appreciably impairing the performance of the system.

4.2 Receiving Terminal

If the delay in network T2 were reduced, then any increase or decrease in the gain of limiter L1 would raise or lower the overall gain of the system for a short period before the compensating action of L2 had time to take effect; during this period, therefore, the signal level at the output of the system would be too high or too low.

For reasons given in Section 3.2, the maximum rate of increase in the gain of L1 has to be made slow enough to avoid 'cycle-following' at low audio-frequencies; in these circumstances, the compensating gain reduction in L2 can readily be made to follow so rapidly that the residual error which would appear even in the absence of the 2 ms delay in T2 is negligible.

Consider, however, the sudden arrival, at limiter L2, of a signal which has caused the gain of limiter L1 to fall. Any gain reduction in L1 must, for reasons given in Section 3.2, be effected within 2 ms, the time delay in T1. The compensating gain increase at the receiving terminal is however delayed by the finite operating time of L2 (which depends on the amount of smoothing required in the pilot tone rectifier circuits to keep unwanted components from modulating the programme signal) and by the time lag in BPF1. In the absence of sufficient delay at T2 therefore, the signal at the output of L2 will initially be below its proper level. It was found, however, in the course of listening tests covering a variety of programme material, that T2 could be completely removed, without audible deterioration in reproduced quality.

4.3 Sending Terminal

After this encouraging result, the next step was to reduce to a minimum the time delay in the network T1 associated with the limiter L1 at the sending terminal; to avoid transient signal overshoot at the output of L1 it was necessary to make a corresponding reduction in the time delay produced by LPF1 in Fig. 3. As indicated in Section 3.2, the lower limit for these two delays is set at about 1/3 ms by the generation of unwanted modulation products, the nominal cut-off frequency of

* Some increase in operating range would, however, be necessary if the limiter were required to give protection against errors of incoming level; if these errors are liable to be greater than, say, 4 dB, it would be preferable to introduce, at the input to the system, a separate limiter having a recovery time* long enough to prevent syllabic gain fluctuation.

** Even without a compandor it is in general desirable — and in the case of a p.c.m. system essential — to provide a protective limiter L1 at the sending terminal.
LPF1 is then 1.5 kHz and the attenuation 10 dB at 2-3 kHz. With the system in this condition, listening tests were carried out with various types of programme material, including recordings of a number of percussion instruments. Some change of quality — comparable with that introduced by magnetic recording — was observed in the case of isolated xylophone notes; this, however, was judged to be mainly due to low-frequency intermodulation products arising from the residual non-linearity (which corresponded to a maximum of 1/2% total harmonic distortion) of the experimental equipment used, the effect of which was accentuated by the 18 dB de-emphasis. Apart from this, no audible deterioration in programme quality was apparent.

It will be noted that with the cut-off frequency of LPF1 increased from 250 Hz to 1-5 kHz, both the sidebands of the modulated pilot tone were extended beyond the range for which the rest of the system was designed; in particular, the lower sideband components could intrude into the programme spectrum unless the latter were correspondingly restricted. Since the additional sidebands of the pilot tone were not utilized at the receiving terminal, but were removed by BPF1 (Fig. 4), it would have been logical to eliminate them at the sending end of the system. This could have been done by providing separate variable-gain channels for the programme and pilot tone, as already suggested in Section 3: the latter signal could then have had its bandwidth appropriately restricted before being mixed with the programme. However, the results of the listening tests on the system suggest that the intrusion of the unwanted pilot-tone components into the programme spectrum does no harm in practice, so that the instrumental complication necessary to remove them would not be justified. This conclusion was confirmed by an experiment in which, in the absence of programme, the level of the pilot tone fed to L1 was changed abruptly by switching a 12 dB attenuator in and out of circuit, thus producing a rate-of-change greater than the maximum that could occur under working conditions. The unwanted components were heard at the output of the system as a slight 'click' which, it was judged, would be readily masked by the type of signal that would call for the maximum rate-of-change of gain.

It is clear from the foregoing that, by taking advantage of the tolerance of the ear and the nature of the signal, it is possible to simplify considerably the design of a compandor system of the type described with little or no effect on programme quality.*

5. APPLICATION TO A P.C.M. SYSTEM

The companding circuit described above is in principle readily applicable to a p.c.m. system. Quantizing noise has a spectrum similar to that of white Gaussian noise, so that low-frequency components are unlikely to cause trouble; moreover, the amplitude/frequency response of a p.c.m. system is uniform and stable, so that a pilot tone located near the upper end of the range can be used to give an accurate indication of gain changes applied to all the other signal components.

Tests were carried out using the compandor system described in Sections 2 and 3, in conjunction with an experimental binary p.c.m. system having a sampling frequency of 31.25 kHz and capable of operating with various numbers of digits up to 11.

The equipment functioned satisfactorily except that some interference was audible at frequencies equal to the difference between harmonics of the 13.75 kHz pilot tone and 31.25 kHz sampling frequency. The production of these difference components is a necessary consequence of the quantizing carried out in the p.c.m. process; the amplitude of such components is comparable with that of the quantizing noise, but the subjective effect was found to be much more objectionable. The interference was, however, eliminated by modifying the equipment to operate with a pilot tone having one-half the sampling frequency,** i.e. 15.625 kHz, and phase-locked in such a way that the crest value of the tone in each half cycle was sampled. Fig. 5

Fig. 5 - Sending terminal of experimental compandor for use with p.c.m. system

* Some reservations should perhaps be made in the case of two channels of a stereophonic programme system. It is known that the directional impressions received by the listener are conditioned by the initial transients of the sounds concerned; further investigation would be required to discover how far these transients may be modified without producing anomalous effects.

** Following a suggestion by E.R. Rout.
shows in schematic form the filtering and sampling arrangements at the sending end of the p.c.m. system in which, for reasons explained in Section 4.3, the cut-off frequency of LPF1 has been made 1-5 kHz and the delay time of T1, 1/3 ms. Fig. 6 illustrates the relevant portions of the spectrum of the sampled signal resulting from a programme input extending up to 14 kHz, assuming very narrow sampling pulses. Since the sidebands of the modulated pilot tone can extend over ±1-5 kHz the spectrum of the signals applied to the sampling unit of the p.c.m. system, shown in curve 6(a), extends up to 1-5 kHz above the half-sampling-frequency point (i.e. to 17-125 kHz), while the lower sideband of the sampling frequency, shown in curve 6(b), which represents the same spectrum displaced and inverted, extends down to 1-5 kHz below this point (i.e. to 14-125 kHz). As the upper and lower sidebands of the modulated pilot tone are symmetrical, corresponding components from the two overlapping spectra coincide in frequency, and it can readily be shown that their amplitudes are additive. It will be seen that this arrangement, instead of sacrificing a portion of the audio-frequency spectrum to accommodate the modulated pilot tone, utilizes for the purpose a region, around the half-sampling-frequency point, which, because of the imperfections of filters, is usually wasted. There seems no reason why the same principle should not be applied to companding in other systems in which the signal is periodically sampled.

At the receiving end of the system the output of the digital-to-analogue converter is applied to a circuit similar to that shown in Fig. 4 but, for reasons explained in Section 4.2, the delay T2 is omitted.

6. CONCLUSION

By judicious use of pre- and de-emphasis circuits in conjunction with a compandor system employing limiters and a pilot tone, it is possible to realize many of the advantages of a split-band compandor in a high-quality sound signal transmission link without the need for band-splitting in the programme circuit. This arrangement is suitable for use in connection with a time-division-multiplex system provided that the pilot tone is locked to one-half the sampling frequency, and has been found to operate satisfactorily with a p.c.m. system.

7. REFERENCES


