

Wireless World

RADIO AND ELECTRONICS

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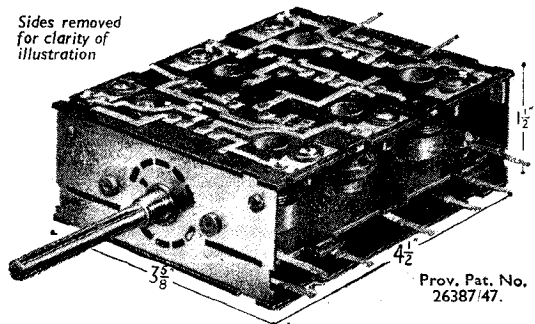


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Valves and their applications

EC91 GROUNDED-GRID AMPLIFIER

The sensitivity of a V.H.F. receiver is largely determined by the first valve. In most cases the design is a compromise between good performance and simplicity of construction, and a circuit which

meets both requirements is of great value. In this respect the EC91 grounded-grid triode is a very useful valve in the 100-250 Mc/s frequency range.

The signal-to-noise ratio which can be obtained in a receiver with a given signal voltage at the aerial terminals is limited by two factors, receiver noise and noise radiation external to the receiver. At frequencies up to 90 Mc/s the external noise level due to atmospheric interference or noise radiation from the Milky Way is relatively high and only a small increase in signal-to-noise ratio can be achieved by reducing the receiver noise below that of a modern high-frequency pentode such as the EF91 or the EF42. At higher frequencies, however, the receiver noise becomes progressively more important and improvements in the receiver will give an effective increase in signal-to-noise ratio.

The receiver noise at V.H.F. is almost entirely dependent on the equivalent noise resistance and input impedance of the first valve. The triode is superior to the pentode because (a) partition noise (due to random current division between screen-grid and anode) is absent, (b) negative feedback by an impedance in the cathode circuit has little effect on the signal-to-noise ratio, and (c) the signal-to-noise ratio is less critically dependent on the aerial coupling.

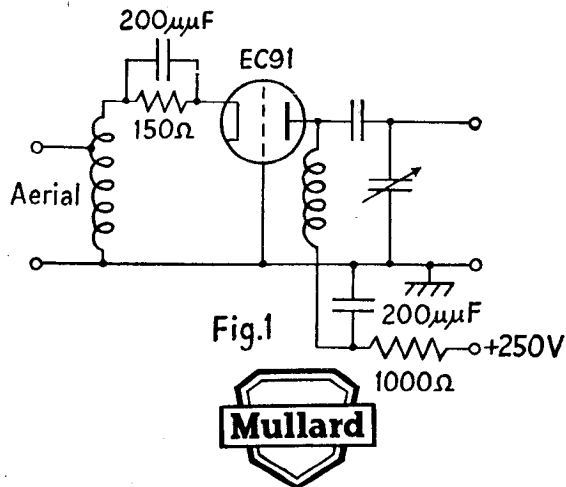
Various circuits have been devised to make a triode amplifier stable at high frequencies, and of these the most simple is the grounded-grid circuit in which the grid is used as a screen between input and output. The EC91 miniature triode (B7G base) has been designed for grounded-grid operation and is particularly suitable for the frequency range 100-250 Mc/s. Its static characteristics are summarised below:

V_h	6.3V	V_a	250V	r_a	12,000 Ω
I_h	0.3A	V_g	-1.5V	R_{eq}	400 Ω
c_{a-g}	2.5 μ F	I_a	10mA	w_a max	2.5W
c_{a-k+h}	<0.2 μ F	g_m	8.5mA/V	I_k max	15mA
c_{g-k+h}	8.5 μ F	μ	100		

The feedback input impedance of a grounded-grid amplifier is approximately $1/g_m$ when the load resistance is small. The input circuit may therefore have a very wide bandwidth (about 100 Mc/s) without the deterioration in signal-to-noise ratio which a damping resistor would produce. The receiver construction may be simplified by leaving the input circuit tuned approximately to the centre of the frequency range. The signal-to-noise ratio is not critically dependent upon the aerial coupling.

A typical EC91 circuit which has been used at 180 Mc/s is shown in figure 1. The aerial tap is about 2/3 of the tuning coil for an 80-ohm source. The measured sensitivity (for unity signal-to-noise ratio) expressed as an open-circuit signal voltage at the aerial terminals was $3.5\sqrt{\Delta f}$ μ V where Δf is the bandwidth in Mc/s. This figure was measured at the output of the last intermediate-frequency amplifier.

It may be noted in conclusion that a grounded-grid EC91 provides a satisfactory input circuit for a 90 Mc/s AM/FM receiver for the reception of the B.B.C.'s experimental transmissions.



Reprints of this report from the Mullard Laboratories together with additional circuit notes can be obtained free of charge from the address below.

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(MVM86)

Wireless World

VOL. LV. NO. 3

MARCH 1949

RADIO AND ELECTRONICS

Comments of the Month

INTERNATIONAL TELEVISION STANDARDS LESS than six months ago the Post Office, the Television Advisory Committee and the Radio Industry Council affirmed with complete unanimity their faith in the B.B.C. 405-line television standards and stated that the British system was to remain unchanged for a number of years. At the same time, strong hopes were expressed that these standards might be accepted to some extent internationally, if only for Western Europe. The advantages of doing this from the point of view of international programme exchanges are self-evident. Equally obvious are the benefits likely to accrue to industry through a form of standardization that would facilitate export trade.

Unfortunately, we are forced to admit that the efforts made in so many quarters to gain international acceptance of these ideas have so far failed. The French have adopted an 819-line system, while other countries are obviously inclining towards a considerably greater number of lines than in the B.B.C. system. Facing this, the British radio industry has evidently decided, if there cannot be a 405-line international standard, it is better to have some kind of standard. As a result, consultations have been held, and an agreement was announced between the main British manufacturers of transmitting equipment and the Dutch firm of Philips. Later, one of the British firms—E.M.I.—disclaimed any part in the agreement.

This agreement, which applied only to the mainland of Europe, provided for a 625-line definition, positive modulation, 25 frames a second with 2:1 interlacing, vestigial sideband operation and 6-Mc/s bandwidth.

At the present time, when the effectiveness of the agreement may well be negated by the non-participation of one of the principal British manufacturers, comment does not seem very useful. In any case, is the European adoption of any par-

ticular standard—other of course than our 405-line system—of any very great value to British industry, especially to makers of transmitting gear? Surely everyone abroad knows that our designers have the knowledge and ability to produce equipment for any reasonable standard. Also they know the commercial axiom "the customer is always right" is sufficiently valid to ensure that demands for television equipment of the kind to which their fancy inclines will be met.

At the same time, the art of the best kind of tradesman, who takes the long-term view, lies in leading the customer's fancy towards the thing that will ultimately be best for him. We still think that for large sections of the world, and especially for Europe, 405 lines is best. It is the system most suited for countries suffering from the economic aftermath of war and which are geographically capable of taking advantage of the relatively easy international exchange of programme material which the system makes possible.

THE NEW ACT AT the time of writing the Wireless Telegraphy Bill is passing through its final stages and will presumably become law without any further amendments. The passing of the new Act will mark an important stage in the history of British radio, which, since its inception, has virtually been governed exclusively by the Act of 1904.

In spite of the great developments that have taken place, the Act introduces practically no new ideas beyond conferring on the Postmaster-General the right to control man-made interference. The main purpose of the measure is to define much more closely than before the powers vested in the P.M.G., which can now be described without exaggeration as sweeping. Almost all the activities with which this journal and its readers are concerned become—or are capable of becoming—what is legally defined as "wireless telegraphy."

P.C.M.

Being an Elucidation of the Mysteries

of Pulse { Code Count } Modulation

By THOMAS RODDAM

IT is not so very long since modulation meant, quite unambiguously, amplitude modulation. People were talking about frequency modulation, but that, we said knowingly, was all just some obscure racket: someone or another had proved that it didn't really do any good. Then frequency modulation turned out to be quite something after all, even if the mathematics were a bit tricky for some of the old stagers. Finally, if that is a safe word to use, everyone began on pulse modulation: they modulated the amplitude of pulses, they modulated the duration of pulses, they modulated the spacing between a pair of pulses and they modulated the pulse epoch, which is a way of saying the phase, though rather more pedantically. Fortunately for the sanity of those of us who have to plan radio systems, there is usually a good reason for choosing one or other modulation technique. The barriers are weakening, however, as the pressure rises: the amplitude modulation party is making better noise limiters, the pulse party is shaping its pulses for greater bandwidth economy. Soon it will cost us more to choose our design basis than to make the design itself!

Meanwhile, using the cycles* lavishly, a new modulation system has been under test in America. The original patent, filed in France by an Englishman, dates to before the war, but wartime developments in circuit technique have probably helped to bring the system to its present state. This system is p.c.m.: these letters at first stood for pulse code modulation, but since this article was first begun references to pulse count modulation have appeared.

Now this is not just another way of modulating a train of pulses. Although it is called p.c.m., the modulation is not at all the sort of thing we usually associate with the word modulation. P.c.m. is, in fact, half-way towards the "Vocoder": the message transmitted is essentially an instruction

to the receiving equipment, not a replica, in some direct form, of the signal. This must not be misunderstood; the breaking down of speech by the "Vocoder" has no obvious counterpart in the coding circuits of p.c.m. In both, however, there is a breaking down of the speech, the transmission by a means which does not contain the intelligence directly, and then a synthesis at the receiver.

To understand the principle of p.c.m. we can take a very simple analogy. The Editor has rung me

up to tell me that I have forgotten to enclose a curve, Figure X, for an article. As I cannot afford a stamp to send the curve by post, I give him the co-ordinates of a selected number of points on the curve, which he then causes one of his faithful stooges to plot, and a fine smooth line is drawn joining them. This, of course, is the idea we use in all pulse modulation systems: we sample the signal curve regularly and then smooth the resulting stepped output with a low-pass filter. P.c.m. involves two extra features. First of all, instead of modifying the transmitted pulse by an amount depending on the sample* number, a group of pulses is transmitted for each sample, and the group forms a sort of inner message giving the amplitude for that particular sample. It is as though instead of saying "the next ordinate is 2.54," I were to use the morse code.

This leads us to the second feature of p.c.m. As I pass the points on my curve over the telephone, I do not say 2.5437, but round the value to 2.54, or more probably to 2.5. If the curve is drawn through the modified points it will not be quite the right shape, and I choose the approximation closely enough for the error to be unimportant. In p.c.m. there is the same problem. To get reasonably good speech the signal must be sampled 8,000 times a second.

Each sample is a number, which is to be transmitted as a code signal, so that there are 8,000 groups a second. Each group is made up of a number of pulses; we shall see why later. Suppose that a group consists of only one "pulse." Then there may be an actual pulse, or it may be suppressed, and the sample size may therefore be either 0 or 1. We

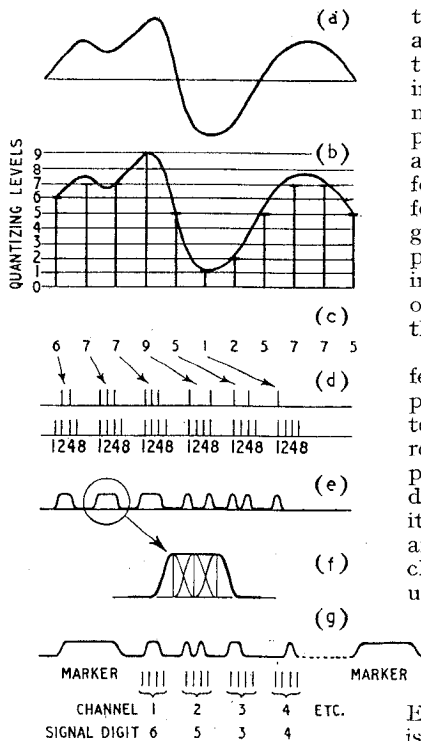


Fig. 1. Stages in the development of a p.c.m. signal: (a) original waveform (b) sampling at regular intervals to find nearest discrete step in amplitude (c) resultant message in digits (d) coded train of pulses with basic train for reference (e) rounding of pulses so that neighbours run together (f) detail of a pulse group (g) combination with pulse trains from other channels and marker pulses.

* To write "cycles per second" would be absurd, and the plural of "hertz" is awkward.

* "Sample" is here used as the statisticians use it, to mean a single thing, which in this case has a numerical magnitude.

P.C.M.—

could produce a signal of this kind by really savage peak chopping of the speech, then using the chopped speech to open a gate to let pulses through. Only when the speech happened to be passing through the zero amplitude axis would a zero be sent. If we use two pulses, the first pulse can stand for an amplitude of 2, and the second for an amplitude of 1. Then two pulses together mean 3; if the second is missing the amplitude is 2; if the first is missing but the second is there the amplitude is 1. This is what is known as operation in the scale of two, and anyone who has worked with thyratron or multi-vibrator counter circuits will recognize it. If we have n pulse positions, in which pulses either are there, or are omitted, we can send any digit from 0 up to $2^n - 1$; thus with 5 pulse positions we can send any whole number up to 31. Suppose that my curve in the example above was a valve characteristic, with a maximum current of 15 mA: then, using a five impulse code, I should have to give the current at any bias to the nearest 0.5 mA. By using a six digit code, I could give it to the nearest 0.25 mA. A little thought will show that near cut-off a line through these "nearest" points will look rather distorted, and that the more digits I use, the better it will be.

In p.c.m. we are actually transmitting a point-by-point plot of

most interesting features of p.c.m., and both its virtue and its limitation. The limitation is, of course, the distortion. As we have

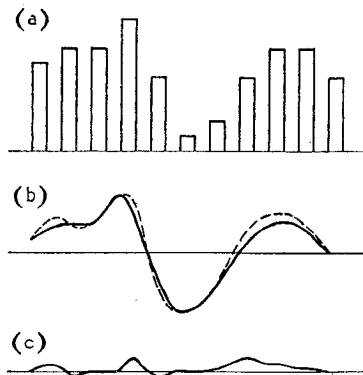


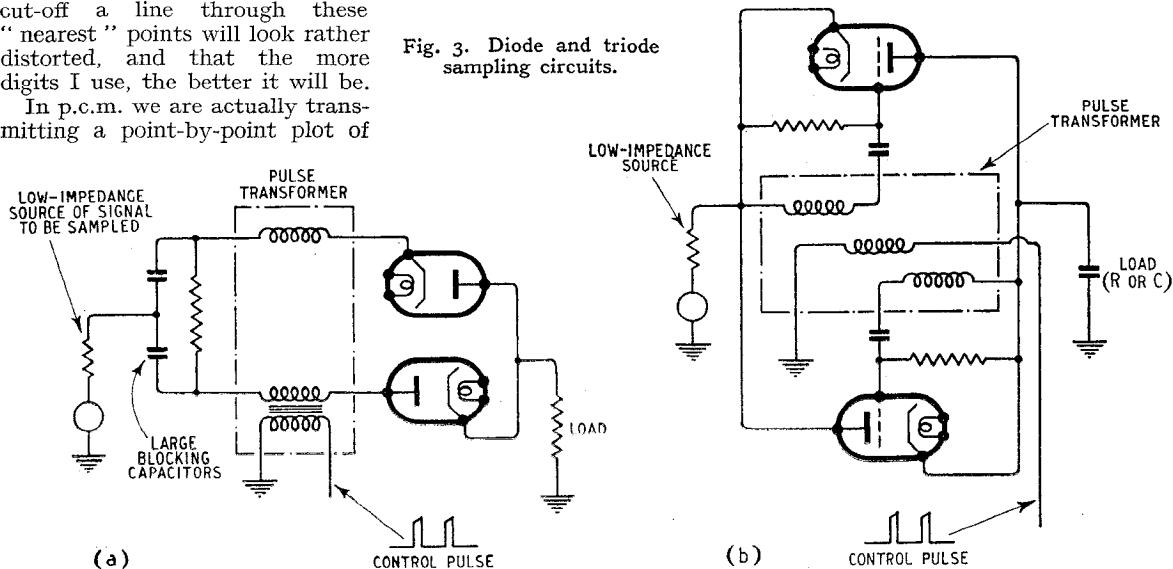
Fig. 2. Reception of pulse code modulation: (a) signal as reconstructed at the receiver (b) after smoothing (the dotted curve is the original signal) (c) quantizing false signal.

seen, we don't transmit the actual waveform, but an approximation to it. The difference is, in effect, a false signal, and is called "quantization noise": it is only 33 db below full signal level when a 5 pulse code is used, although even this means that 40,000

of pulses whose exact timing and amplitude are not significant. clean regular pulses can be produced by selecting the middle of the received signals, and so long as no noise impulse is large enough to produce a false triggering of the receiver circuits, the system is completely undisturbed by normal noise. Consequently lower power levels can be used, because the receiver no longer demands a large signal to override the internal first-circuit noise. With the rather low duty ratio involved in p.c.m. links and the high frequency which must be used to provide the bandwidth, there is at present some advantage in using low power for transmission even at the cost of complication.

The number of steps needed is reduced by a most ingenious arrangement. Very roughly, the ear is sensitive to fractional changes of level, not absolute changes of level. That is why the loudness scale is a logarithmic one. It is therefore reasonable to make the voltage steps not equal but increasing in magnitude from step to step. In this way the half-step size, which is the error in specifying the instantaneous voltage, is proportional to the

Fig. 3. Diode and triode sampling circuits.



the waveform, and, of course, we cannot do it exactly. This taking of the measured points to the nearest fixed step is called "quantizing"; it is one of the

pulses a second are being used to transmit 4,000 c/s of speech. The virtue of quantizing is that circuit noise can be eliminated completely. As a message is made up

actual magnitude of the voltage, and the result is a constant "quantizing fractional error" for all signal magnitudes. This can be done by applying the audio-

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frequency signal to a potential divider which includes a non-linear resistor and thus acts as an instantaneous compressor. Equal sizes of step after this compressor correspond to a graded series before the compressor.

Let us recapitulate the essentials of the system. The input audio-frequency signal is first passed through a low-pass filter to keep the bandwidth down to the standard telephony bandwidth. Then it is applied to the instantaneous compressor, which

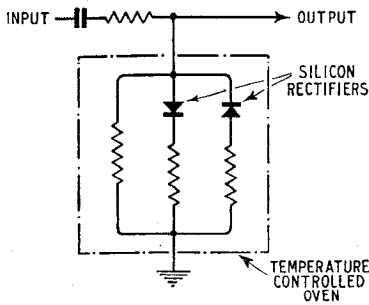


Fig. 4. Circuit of instantaneous compressor.

squashes down the larger amplitudes. Somewhere before the compressor a constant-volume amplifier, or "Vogad," may be fitted, to bring the speech level to that required on the circuit. Then the speech is applied to the sampler, which decides which of the 2ⁿ levels is the appropriate one at the instant of measurement. This information goes to the coder, which sends out a group of *n* pulses or spaces between pulses, corresponding to the number given by the sampler. The signal is then fed to the combining unit, where the interlacing of a number of channels takes place, and the whole complex pulse system is applied to the magnetron modulator.

At the receiver the pulse groups belonging to the different channels are sorted out, and each pulse group is then examined by the decoder. This produces a pulse, the amplitude of which is dependent on the actual code group. These amplitude-modulated pulses are sent through the instantaneous expander, which opens up the larger amplitudes, and then through a low-pass filter which

removes the pulse structure, leaving only the audio-frequency signal. This sequence is shown in Fig. 2.

This description of the process is rather over-simplified and does not represent the latest system. As described, the signal consists of groups of separate pulses. This would mean an unnecessarily large bandwidth, as all that is needed is the information that a pulse is or is not there. The pulses are therefore rounded off, so that two or more pulses in succession run together to form a single long pulse. This gives a useful reduction in the bandwidth required, and there is a very good reason for reducing the bandwidth. Although there is plenty of room in the microwave region, the video-frequency amplifiers will have an amplification, which is inversely proportional to bandwidth, so that halving the bandwidth enables the same gain to be obtained with half the number of stages. Another requirement is that the receiver should be able to pick out groups of pulses from the continuous train which it receives. A marker pulse is therefore added at the transmitter so that the receiver can be synchronized.

The experimental system* was a twelve channel system; that is, twelve audio-frequency circuits were combined in the one radio circuit. This number was chosen because it corresponds to present-day multi-channel carrier telephony practice; most telephone systems are now planned on a basis of primary trunk groups of twelve channels. As each speech band extends up to 3,400 c/s a sampling repetition rate of 8 kc/s is used, requiring 96,000 signals per

second for a 12-channel group. A seven-digit code is used, so that the number of pulses per second is 672,000. The pulse length used within the terminal equipment is 0.4 μ sec, but the pulses radiated are longer than this.

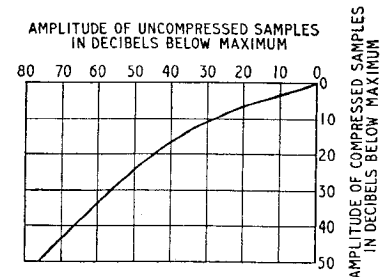


Fig. 5. Characteristic of instantaneous compressor.

As we have already seen, the speech for transmission is first passed through a low-pass filter, to eliminate frequencies above 3,400 c/s. After the filter a peak-chopper circuit is used to restrict the amplitude applied to the following equipment. To this point the twelve channels remain separate: they are now sampled and joined together. The sampling

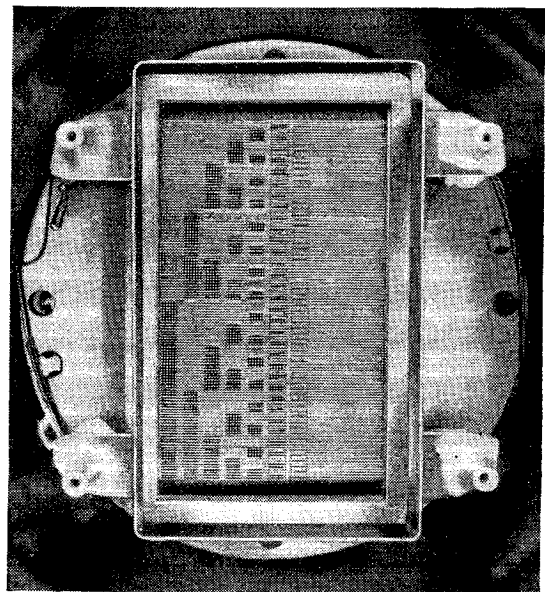


Fig. 6. Interior of coder tube viewed from gun end.

* "An Experimental Multichannel Pulse Code Modulation System of Toll Quality," by L. A. Meacham and E. Peterson, *Bell System Technical Journal*, Vol. XXVII, No. 1, Jan., 1948.

circuit consists of two diodes, shown in Fig. 3(a). (This is Fig. 12 of *B.S.T.J.*, Vol. XXVII,

No. 1). When the control pulse appears at the pulse transformer both diodes conduct and connect the input to the load: the blocking capacitors are charged up, and at the end of the pulse the charge remains to keep the diodes cut off for the relatively low signal voltage. The next pulse again opens the diode gate. The triode circuit in Fig. 3(b) operates in the same way, except that the gate is opened by the pulse applied to the grid while the signal path is from cathode to anode. The output of the sampling circuit is a series of pulses, the amplitude of which is the peak signal amplitude at the instant the pulse is on. By connecting one of those circuits to each speech channel, supplying a

some ways. The pulses are applied to the Y plates and the Y deflection is then held constant while the beam is swept across by a horizontal sweep circuit. The beam is then cut off until the next pulse deflects it again. As the

takes place: the resulting electrons go to the collector and produce a bias which moves the

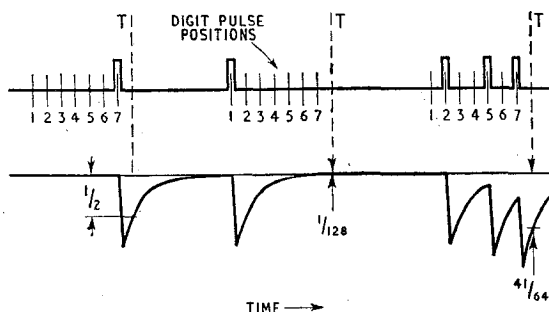


Fig. 8. Decoding waveforms from simple R-C circuit.

beam sweeps in the X direction it passes over an aperture plate, which has a pattern of slots in it.

beam into the space between two wires. This is actually a feedback amplifier circuit and the equivalent of 20 db of feedback is provided. The collector, the quantizing grid and the aperture plate are all visible in Fig. 6. The collector is the box structure which frames the aperture plate. The output takes the form shown in Fig. 7 (Fig. 9, p. 56 of *B.S.T.J.*, Vol. XXVII, No. 1).

The circuits which follow are fairly straightforward. The decoding circuit used at the receiver is perhaps the only other circuit of special interest. The input to this circuit is, of course, a group of up to 7 pulses. A resistance-capacitance circuit is used: if a charge is put on the capacitor, the voltage across it falls during some time t to 50%. During the next interval t , it falls to 50% of the value at the beginning of the

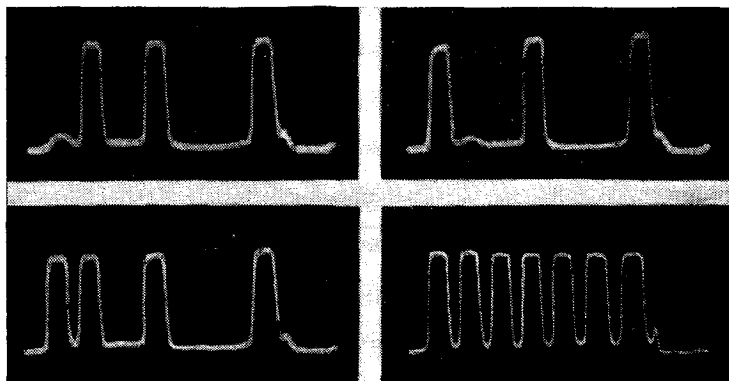


Fig. 7. Typical pulse code outputs.

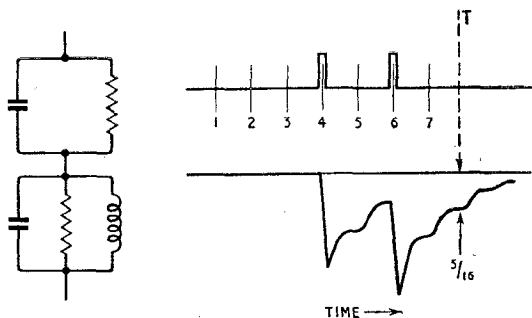
train of "12-way interlaced" pulses to each gate and paralleling the outputs a single set of interlaced amplitude-modulated pulses is provided which contains all the intelligence from the twelve channels.

These pulses are now applied to the instantaneous compressor. In the simplified description it was stated that this was before the sampler, but that was purely to help to indicate the action. The compressor leaves unaffected the lower level signals, but reduces the higher levels considerably so that an increase of 15 db in the input produces only 5 db rise in output. The circuit and its characteristics are shown in Figs. 4 and 5 (Figs. 13 and 14, *loc. cit.*)

The compressed amplitude-modulated pulses are now applied to the coding valve, which both quantizes and codes. This valve resembles a cathode-ray tube in

The electrons which pass through slots fall on a pulse plate and produce impulses in the output circuit. The impulse pattern depends on the slot pattern at the particular value of Y and the slots are so arranged that they give the required code. The aperture plate, with its slots, is shown in Fig. 6 (Fig. 19, *loc. cit.*). To make sure that the beam sweeps across at a definite "quantized" level, a parallel wire grid and a rectangular collector electrode are mounted in front of the aperture plate, and feedback is provided from this collector to the vertical amplifier. If the beam falls on a wire, secondary emission

Fig. 9. Shannon-Rack decoder and modified waveform.



second interval, that is to $\frac{1}{4}$ of the initial value. By making this time t equal to the pulse interval, the contribution to the voltage across the capacitor made by the

P.C.M.—

last pulse will be $\frac{1}{2} \frac{Q}{C}$ where Q is the charge, and the voltage is measured one pulse period after

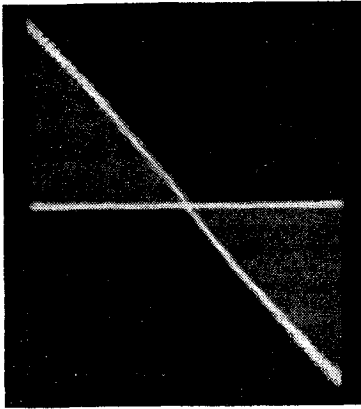


Fig. 10. Overall characteristic of coder (horizontal) and decoder (vertical) including compressor and expander.

the last pulse. The last but one pulse will contribute $\frac{1}{4} \frac{Q}{C}$ and earlier pulses $\frac{1}{8}, \frac{1}{16} \dots \frac{1}{128}$,

all multiplied by Q/C . If a full set of 7 pulses is present, the final voltage will be $127/128$ and any intermediate multiple of $1/128$ can be produced. The decoding circuit is arranged so that each digit pulse delivers a standard charge to the capacitor, and the result is the waveform shown in Fig. 8 (lower half of Fig. 21 *loc. cit.*). By adding a tuned circuit, as shown in Fig. 9 (Fig. 22 *loc. cit.*) a "flat" is put on the waveform so that accurate timing in sampling the decoded wave at T is not so necessary.

The input-output characteristic of the system is shown in Fig. 10 (Fig. 25 *loc. cit.*). The tapered steps can be seen quite clearly and the overall linearity gives an indication of the accuracy of matching between the compressor and the expander, which is an amplifier with a compressor in the negative feedback path.

In measurements of noise on ten links in tandem, a maximum audio signal-to-noise ratio of 58 db was obtained. Crosstalk from any one channel to any other was 66 db down and perfect reception was obtained when the radio signal-to-noise ratio was 18 db.

MANUFACTURERS' PRODUCTS

Television Magnifiers

AN increase of effective picture size of $2\frac{1}{2}$ times is claimed for the Magnavista television lenses made by Metro Pex, Ltd., 71, Queens Road, Peckham, London, S.E.15. Sizes suitable for 9-in and 10-in tubes are available, either for mounting directly to the receiver, or on separate telescopic stands. Prices range from £6 6s to £8 18s 6d.

Modified Oscilloscope

AN alternative cathode-ray tube having a blue trace with yellow-green afterglow can now be supplied by Taylor Electrical Instruments, 419-424, Montrose Avenue, Slough, Bucks, in their Model 30A oscilloscope. The price of the instrument, which is known as "30A with persistent trace," is £33 10s.

New Drive Mechanisms

TWO new condenser drive mechanisms have been added to the range made by Jackson Bros., Kingsway, Waddon, Surrey. Both

models are fitted with three-wave-band coloured scales, calibrated with station names on medium and long waves and framed in florentine bronze escutcheons. Type SL8 at 25s has a cord drive with spin-wheel control knob, and type SL5 at 24s is fitted with a reverse vernier control.

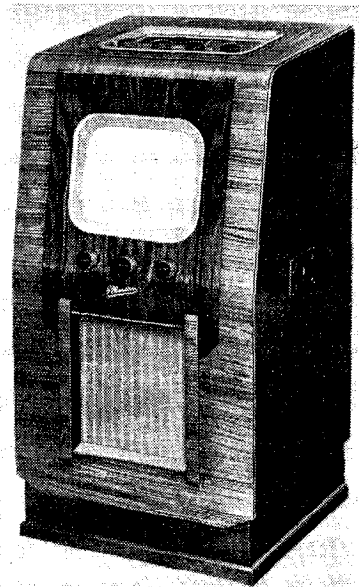
New Domestic Receivers

A BATTERY set giving power output and quality comparable with those of mains-operated receivers has been introduced by Murphy Radio, Welwyn Garden City, Herts. The four-valve superhet circuit covers short, medium and long waves, and a QP25 push-pull output valve feeds an 8-in permanent magnet loudspeaker. The type number is B141 and the price is £21 4s 9d including tax. The new Murphy A122M for a.c. mains is a modified version of the A122 with a redesigned cabinet and costs £26 17s 9d including tax.

Two new a.c./d.c. superhets (4 valves plus rectifier) have been developed for the export market by Vidor, Ltd., West Street, Erith, Kent. Both are housed in walnut

and sycamore inlaid cabinets with "mirror" finish. Model CN385 covers 13.5-32m, 32-100m, 190-550m and 1,000-2,000m, while Model CN 386 covers 10-12m (band-spread), 13.5-32m, 32-100m and 190-550m.

A special c.r. tube giving enhanced picture brightness, and coupled contrast and brightness controls are features of the new Marconiphone VRC52A and H.M.V. Model 1805 television receivers. Automatic interference suppression on sound and a vision interference limiter are provided and the picture size is $8\frac{3}{4}$ in \times $6\frac{1}{2}$ in. The H.M.V. Model 1805 costs £89 5s and the



Marconiphone VRC52A television and radio receiver.

Marconiphone Model VRC52A, which includes a radio receiver for short, medium and long waves with preset tuning for two stations in medium waves, costs £110, both prices including purchase tax.

Metal Detector

AN industrial electronic metal detector has been produced for Cinema Television, Ltd., Worsley Bridge Road, Lower Sydenham, London, S.E.26. This has been designed for use in conjunction with a conveyer-belt system for the examination of chocolate and other food products which are carried through a search head which detects any metal particles, operates a suitable alarm and automatically stops the conveyer belt in the presence of either ferrous or non-ferrous particles.

TELEVISION "GOODNESS FACTOR"

Why More Lines May Mean a Worse Picture

By R. W. HALLOWS,

M.A. Cantab, M.I.E.E.

FROM discussions that I have heard and from questions that I have been asked I know that not a few people find the statement that a larger number of scanning lines may result in a worse image on the screen of the television receiver a dark saying. This article attempts two things. It aims at giving the initiated some useful figures and it offers them a way of answering the dread seeker after enlightenment. Further, the article suggests straightforward methods of determining whether or not a given television system is capable of providing a balanced image (the term "balanced" will be explained in a moment), of finding the extent of the unbalance, should this be present, and of calculating the number of scanning lines that can be employed to the best advantage for any range of modulation frequencies genuinely transmitted and received.

When a person suffering from ocular astigmatism looks at two crossed wires, or two black straight lines crossing on a white background, he may see them as shown in Fig. 1 (a). If he focuses the horizontal line AB sharply, then PQ is muzzy. The angle θ at which the defocusing of PQ is most marked varies from eye to eye and is mainly dependent on the departure from simple spherical form of the cornea, and not as a rule the lens of the eye. The oculist's part is to discover the degree of inequality between the horizontal and vertical focusing powers of the eye and the angle at which this inequality is at its worst. He then prescribes a corrector in the form of a spectacle lens with both "sphere" and "cylinder" elements. If the spherical element enables the eye to obtain a sharp focus horizontally, the cylindrical element, placed at the proper angle, counteracts the deformity of the living optical system and enables it to provide sharp focusing in the vertical sense as well. Aided by the correcting lens, the eye now

sees both of the crossing lines equally clearly as in Fig. 1(b).

In other words, the living optical system, plus the correcting lens, gives perfectly balanced focusing in both directions. The term "balance" as used in this article means that resolution of an image is equally good in the vertical and the horizontal directions. Without such balance no image can be perfectly clear. It

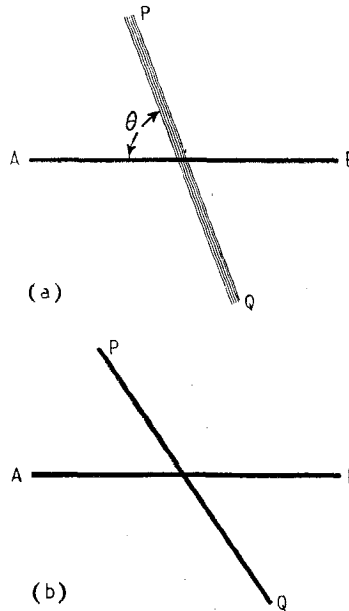


Fig. 1. (a) Crossed wires as seen with the unaided eye by a sufferer from optical astigmatism. The angle θ at which the defocusing of PQ is at its worst depends upon the deformity of the individual optical system. Diagram (b) shows the same wires as seen by an eye with no astigmatism or by the astigmatic eye looking through a correcting lens. At (a) horizontal and vertical definition are unbalanced; at (b) they are balanced.

may be acceptable if the unbalance is slight; but so long as there is any unbalance it is not so good as it might be. The greater the

unbalance, the worse the image.

In television the angle at which unbalance may occur is fixed. It is always a right angle.* The scanning lines slice the image into horizontal strips and thus determine the degree of definition in the vertical sense; the changing brightness of the scanning spot divides the image into minute light and dark patches as it moves from side to side of the c.r.t. screen and so limits the horizontal definition. Perfect balance in the transmitted and received image cannot be ensured unless the system in use is capable of slicing the image into vertical strips as narrow as the horizontal.

Don't be confused by the terms "horizontal" and "vertical" as applied to the slicing of the image and the corresponding sense of definition. A moment's thought will show that the number of horizontal cuts (scanning lines) determines the top-to-bottom, or vertical definition. The greater the number of these, the narrower are the alternate black and white horizontal lines that can be resolved: if the image is built up by 377 active scanning lines interlaced, it would be possible for it to show $188\frac{1}{2}$ white lines and the same number of black. To make the horizontal definition equally good it must be possible to resolve vertical lines as narrow as the narrowest horizontal lines that the system can handle. Since the aspect ratio of the B.B.C. transmission is $5/4$, it follows that for its transmission with 377 active scans per image this number of alternate vertical black and white lines is $377 \times 1.25 = 471\frac{1}{4}$.

When a pattern consisting of alternate black and white vertical lines is scanned by the electron beam of the Emitron the output of the head amplifier for each scanning line is a.c. of the form shown in Fig. 2(b). These a.c. voltage waveforms are used as modulation frequencies in the

* Or as nearly a right angle as makes no matter. Actually, the scanning lines are of course, not quite horizontal, but have a slight downward slope.

Television "Goodness Factor"— transmitter. In the receiver a.c. at vision frequency appears at the output of the video amplifier stage and is converted into d.c. voltage fluctuations for application to the grid of the c.r.t. by the action of the d.c. restorer.

It will be seen that a change from black to white or *vice versa* requires one half-cycle of a.c. For this change to be as sharp as it is between the white background of this page of *Wireless World* and the outlines of the letters printed upon it a.c. of square waveform would be required. The rise from *p* to *q* and the fall from *q* to *p* would have to be perfectly vertical. In other words these rises and falls would occur in no time at all, or in zero wavelength, which corresponds to infinite frequency. Hence to produce the perfectly square waveforms required for absolutely sharp contrasts the modulation would have to contain all frequencies up to infinity. Actually, there is no need for this since the resolving powers of our eyes are limited to something not much better than one minute of angle. Examine a printed page with a powerful magnifier and you will find that the transitions from black to white are not what the unaided eye reports them to be. Sinusoidal waveforms such as those in Fig. 2(b) where the middle of each white vertical line results in a "crest" and the middle of each black line in a trough give the quite acceptable reproduction on the c.r.t. screen indicated in Fig. 2(c).

What this comes to is that to be able to resolve adequately the 471 vertical lines of (1) we need

$471/2 = 235\frac{1}{2}$ cycles . . . (2)
in the active portion of each scanning line. But the active portion of the line lasts only 83.5 μ sec and the time for the whole line is 99 μ sec. Hence the number of cycles required for the whole line is approximately

$235.5 \times 1.2 = 282.6$ cycles (3)
The total modulation bandwidth needed to give balanced definition in a 405-line system with 25 images per second of 5/4 aspect ratio is:

$$f = \frac{282.6 \times 405 \times 25}{2.86 \text{ Mc/s}} \dots \dots (4)$$

We can tidy up the results of

(1), (2), (3) and (4) into the handy form:

$$f_{\text{min}} = \frac{L'}{2} \times arLn \times 10^{-6} \text{ Mc/s} \dots \dots (5)$$

where

- f_{min} = minimum modulation bandwidth to give balanced definition;
- a = aspect ratio
- r = ratio of whole line to active portion
- n = images per second
- L = total scanning lines
- L' = active scanning lines:

From (5) we can obtain a means of ascertaining the maximum

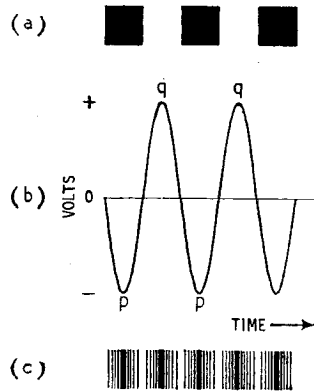


Fig. 2. (a) Consecutive elements of a scanning line for an image consisting of alternate black and white vertical lines. (b) Corresponding a.c. voltage waveform at output of head amplifier. (c) Resulting reproduction on screen of receiver cathode-ray tube.

permissible number of scanning lines for any given modulation bandwidth. A slight reshuffling is necessary, for both L and L' are unknowns. Since $L' = pL$, we have:

$$L_{\text{max}} = \sqrt{\frac{2f_{\text{mod}}}{arnp}} \dots \dots (6)$$

where

- L_{max} = greatest permissible number of lines
- f_{mod} = actual modulation bandwidth
- p = ratio of active lines to total scanning lines

and the remaining factors a , r , n are as before.

Further, we can obtain what may be termed a Definition Ratio, or "goodness factor" for any television system, or for a particular transmitter. Calling this factor D , we have

$$D = \frac{f_{\text{mod}}}{f_{\text{min}}} \dots \dots (7)$$

For the present B.B.C. system

$$D = \frac{2.7}{2.86} = 0.944.$$

My suggestion is that television systems should be described not just by the number of their scanning lines, but by this number *and* the definition factor. The B.B.C.'s, for example, would be termed a 405-line, 0.944 system. This method could also be applied to receivers in test reports and possibly in manufacturers' specifications; in this case, of course, D , would be the ratio of the maximum frequency adequately dealt with by the set and the maximum modulation frequency of the transmission. If a British receiver cut off at 2.4 Mc/s it could be described as having $\frac{2.4}{2.7} = 0.89$ or 89 per cent. definition.

From (5) and (6) the truth of the statement that more lines may mean a worse image is readily seen: they certainly will unless the range of the modulation frequencies is increased accordingly. Two other statements, both rather puzzling at first sight, may also be shown to be true. These are (1) with the standard 405 lines . . . increasing the bandwidth to some 4-5 Mc/s results in very noticeably higher definition;† and (2) no one has yet seen a real 405-line picture.‡

The figure arrived at in (5) gives the *minimum* modulation bandwidth for balanced definition. The resolution resulting is of the kind shown in much exaggerated form in Fig. 2(c). Black shades off into white through greys, which may be regarded as paler and paler blacks and less and less dark whites. Make the frequency higher and the waveforms become squarer, with the result that the transitions are more abrupt and vertical lines more clearly painted on the receiving screen. Hence a bandwidth of 4-5 Mc/s is bound to result in a noticeable improvement in definition.

This is one reason why we have never yet seen a 405-line image as it might be. There are at least two others of importance.

† Television Standards, *Wireless World*, Oct. 1948, p. 382, Col. 1, line 20.
‡ *ibid.*, "Editorial Comment," p. 351, Col. 2, line 28.

With 377 active lines the greatest bandwidth that can be usefully employed is probably not more than 5 Mc/s. This is due partly to the limited resolving powers of our eyes and partly to the fact that a slight blurring of outlines may be necessary to aid persistence of vision in producing the illusion of movement from the observation of a rapid succession of still images. Such lack of sharpness is certainly to be found in individual frames of a cine film. But why should only 377 of the 405 lines be active and 28—nearly 7 per cent—be used for the frame sync pulses? Some systems use only 5 per cent; some, possibly even less. An increase in the number of active lines would improve the vertical definition and the bandwidth might then be put up to a little over 5 Mc/s to give a corresponding improvement in the horizontal definition.

The second point is that there are minute gaps between the scanning lines: it is as though the image were sliced horizontally with a rather coarse saw rather than with a razor blade. To eliminate the gaps is a difficult problem, but a solution will no doubt be found. When "lineness" has been removed the television image will be much more pleasing.

It is interesting to see how the balance of definition works out for 525-line television. There are now a great many 525-line systems in operation in the United States,

but there appear to be no generally accepted standards of bandwidth or number of frame sync pulses. The only standards adopted by all in addition to 525 scanning lines are an aspect ratio of approximately 4/3 and an image frequency of 30 per second. The last is probably a liability rather than an asset. It is demanded by the standard 60-c/s periodicity of American a.c. mains supplies; but since an image frequency of 25 per second gives all the necessary steadiness, the greater number means that part of the modulation bandwidth available must be employed in serving no very useful purpose.

Let us take the case of a 525-line system with a bandwidth of 3 Mc/s and 94 per cent of the scanning lines and 84 per cent of each line active. Then L' works out at 488.25, which we may take as 489, since interlaced scanning demands an odd number of active lines; $a = 1.33$, $n = 30$, $r = 1.19$. From (5) we have:

$$f_{\min} = \frac{489}{2} \times 1.33 \times 1.19 \times 30 \times 525 \times 10^{-6} = \text{approximately } 6.1 \text{ Mc/s}$$

From (6) we have a surprising result:

$$L_{\max} = \sqrt{\frac{6 \times 10^6}{1.33 \times 1.19 \times 0.94 \times 30}} = 365$$

Here, indeed, the increased number of scanning lines means a poorer image than that given by our 405-line system, even though the bandwidth is assumed to be greater by 0.3 Mc/s. From (7)

the definition ratio is $\frac{3}{6.1}$, or 0.492

as compared with 0.944. If a 525-line system were used with a 5/4 aspect ratio in a country where there is a 50-c/s a.c. supply, f_{\min} would come down to 4.75 Mc/s. Assuming an actual modulation bandwidth of 3 Mc/s, the definition ratio would be 3/4.75 or 0.632.

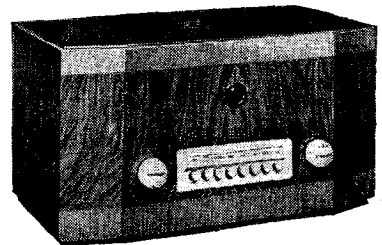
It seems clear that it is of little use to go beyond about 405 lines unless and until transmitters, relay links and moderately-priced televisors can deal adequately with modulation bandwidths well above 5 Mc/s. It is actually disadvantageous to do so owing to the inferior definition of the images.

HIGH QUALITY REPRODUCER

TWIN elliptical diaphragm loudspeakers fed from an 8-watt push-pull output stage with negative feedback form the backbone of the new H.M.V. Model 2000 record reproducer. The automatic record changer is fitted with the latest

H.M.V. lightweight pickup and there are independent tone controls for bass and treble—the latter with five positions to give the best compromise between h.f. response and surface noise on records of all types and ages. A low-impedance input of 5 ohms is provided for coupling to a radio receiver, a power of 0.5 watt being required for full audio input.

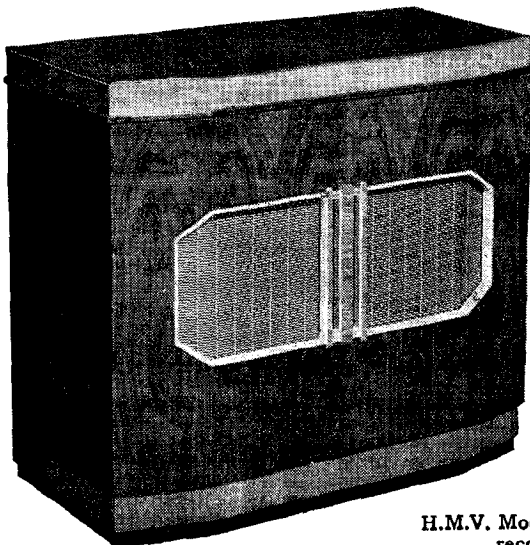
The Model 2500 high-quality receiver unit has been designed to work



H.M.V. Model 2500.

in conjunction with the Model 2000 reproducer and has the requisite 5 ohms output impedance. It is a push-button superhet with an r.f. stage and wide-band intervalve couplings. A specially stabilized oscillator has been provided to minimize frequency drift. There are five push buttons for medium-wave stations and two for long waves. A three-position "fidelity" switch gives variable bandwidth in the i.f. coupling. The receiver has an output of 2½ watts and can be used with a loudspeaker direct, where the extra power of the Model 2000 reproducer is not required.

The makers are the Gramophone Co., Hayes, Middlesex, and the prices are: Model 2000, £112 17s 6d; Model 2500 £45 19s 10d, both prices including purchase tax.



H.M.V. Model 2000 "Celebrity" record reproducer.

SINGLE SIDEBAND RADIO-TELEPHONY

First Use of the System in Marine Communications

By H. D. B. KIRBY

(Standard Telephones and Cables)

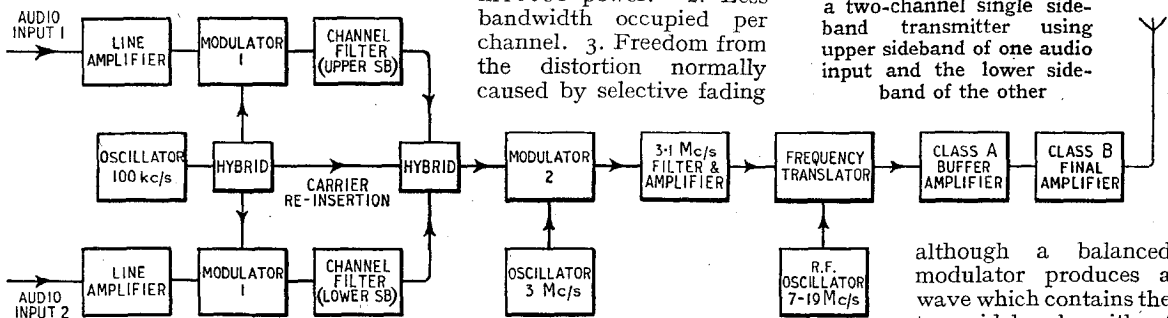
THERE has been since the war a general extension of the use of single-sideband operation on long-distance point-to-point radio-telephone links throughout the world. Now the use of this system has been extended to the long-distance ship services with the installation of single-sideband equipment in the new Cunard White Star liner *Caronia*.

What exactly is this system, and what advantages can it offer over the normal double-sideband method of transmission? To answer these questions it is necessary to consider first what happens when a speech wave is used to modulate an r.f. carrier. This has been discussed at some length in previous issues of this journal,^{1,2} so it will simply be stated here that the ordinary process of amplitude modulation produces a wave which may be analysed into three groups of sine waves, the upper sideband, the lower sideband and the carrier. If the carrier has a frequency f_c and the modulating signal consists of a number of frequencies between 100 and 5,000 c/s then the upper

sideband is, as it were, a mirror image of the other, so that each contains the same intelligence as the other. Consequently the only medium necessary to convey the modulating signal from the transmitter to the receiver is one of the sidebands. Unfortunately, however, although the intelligence is present in the sideband, it cannot be extracted at the receiver without the use of the carrier. In the ordinary d.s.b. system the carrier is transmitted together with both sidebands, and when these are applied to a non-linear impedance in the demodulator stage of the receiver a number of new frequencies are produced among which are the original signal frequencies. In the s.s.b. system the carrier is still necessary for the process of demodulation, but since it does not vary with modulation it can either be generated at the receiver or transmitted at a very low level and amplified separately.

The advantages of the system may be summarized as follows:

1. Improved signal-to-noise ratio at the receiver for a given transmitter power.
2. Less bandwidth occupied per channel.
3. Freedom from the distortion normally caused by selective fading



Block schematic diagram of a two-channel single sideband transmitter using upper sideband of one audio input and the lower sideband of the other

sideband will comprise a band of frequencies between $(f_c + 100)$ and $(f_c + 5,000)$ c/s and the lower sideband frequencies between $(f_c - 100)$ and $(f_c - 5,000)$ c/s.

The frequency, phase and amplitude of the carrier remain unaltered whatever the modulating signal, and it therefore contains no intelligence. Moreover, each

and multi-path propagation. What do these advantages mean in the field of marine radio-telephony?

Since the radio equipment carried on board ship is necessarily limited in size, weight and power consumption, it has been the practice in the past to attempt to improve communication between ship and shore by using more

powerful transmitters and more sensitive receivers on land. This is not effective above a certain point, at least in the ship to shore direction, since noise then becomes the limiting factor. By the use of s.s.b.; however, it is possible to concentrate the whole of the available output power in the one intelligence-bearing sideband. The effect of this is that the signal at the receiver is equivalent to that which would be received from a d.s.b. transmitter about four times as powerful. This will double the distance at which the level of the received signal will be satisfactory for public use.

The decrease in bandwidth will reduce the noise picked up at the receiver since it will be possible to tune the filter circuit more sharply. This again will increase the useful range of the equipment.

Thirdly, much of the unpleasant distortion usually caused by selective fading is considerably reduced, so that a link which is quite unworkable on d.s.b. may well provide a reasonable speech circuit under similar conditions with s.s.b. operation.

We do not yet know of a satisfactory method of producing a single-sideband signal directly,

although a balanced modulator produces a wave which contains the two sidebands without the carrier. Thus a very simple transmitter could be used to transmit such a suppressed carrier signal, but the receiver would have to be very complicated since for demodulation a locally generated carrier would be required not only of exactly the same frequency as the original carrier, but also of the same phase. Since this would