AUTOMATIC CALIBRATOR For Frequency Meters

A 126-tube electronic calibrator combined with adding machines records on paper tape the calibration data at 327 points for an Army SCR-211 two-band frequency meter, interpolates between these points, and automatically prints in the individual calibration book a five-digit dial number for 3252 frequency values

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IN THE MANUFACTURE of highly precise measuring instruments, it is sometimes found necessary, in order to obtain the required accuracy, to hand calibrate each individual instrument. In the particular case at hand, a two-band frequency meter known as the Army SCR-211 is required to maintain an accuracy of the order of 0.01 percent in the field. This frequency meter as manufactured by Philco Corporation consists of an electron-coupled variable-frequency oscillator, which can be checked at certain points of the dial against an internal fixed-frequency crystal oscillator.

Anyone who has had experience in the production of receivers having dials which read directly in frequency can appreciate the practical impossibility of making an oscillator track to a predetermined dial scale within 0.01 percent. For comparison, a good broadcast receiver has an accuracy of about ±10 kc, or about 1 percent. Variations such as inductance, capacitor plate contour and straightness, gear eccentricities, etc., require that the frequency meter be designed with a dial which reads in arbitrary units, and that a calibration booklet be prepared for each meter.

The two bands of the frequency meter cover a fundamental range of 125 to 250 kc and 2 to 4 Mc. The Army specifications call for a listing of calibration points every 0.1 kc on the low band and every 1.0 kc on the high band, or 3,252 calibration points in all. Each of these calibration points is recorded as a five-digit dial number.

Fortunately, it is found that by proper design of the tuning capacitor, the plot of dial reading versus frequency can uniformly be made sufficiently close to a straight line that only every tenth point printed in the calibration book need be hand calibrated, the remainder being interpolated linearly. Thus, hand calibration points are required every 1 kc on the low band, and every 10 kc on the high band, or a total of 327 points must be recorded by hand.

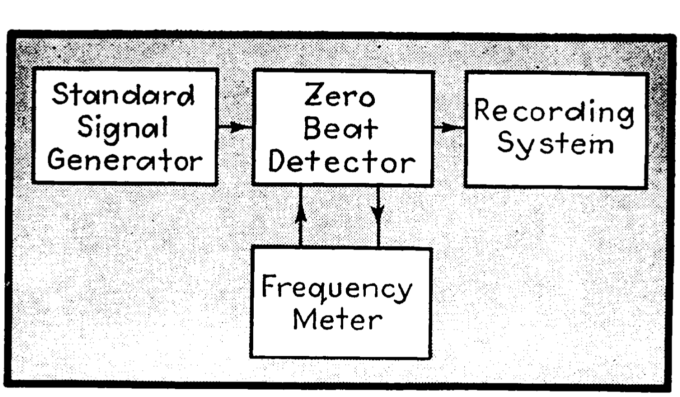
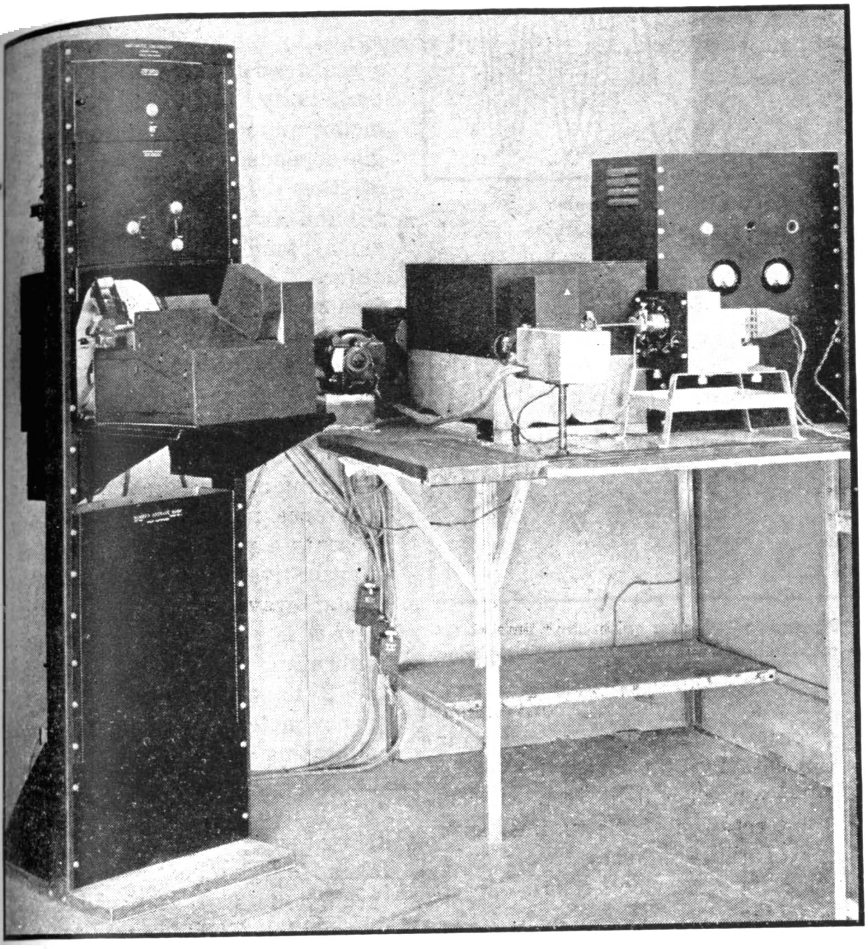


FIG. 1—Block diagram showing the three basic units of the automatic calibrator and their relation to the frequency meter being calibrated

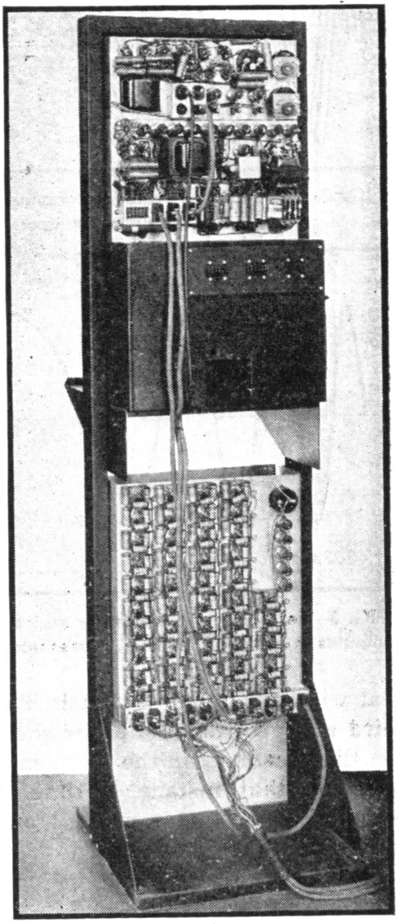
Calibration Time Is Shortened

It has been found that on a production basis an average of 2.5 hours was required to hand-calibrate one frequency meter, with another hour to compute the increments between adjacent calibration points so that interpolation could be made, another 1.75 hours to interpolate, and an additional 5 hours to type the calibration booklet. An average of several errors in each original hand calibration, plus additional errors in interpolating and typing, made necessary a very thorough checking of each frequency meter and calibration book, totaling 3.5 hours more.

Thus, to calibrate a frequency meter accurately by hand, including 2.25 hours additional for miscellaneous operations, required a



Overall view of calibrator, showing frequency meter on jig at right center, with dial coupled to zero beat detector. Printing rack is at left. Signal source is not shown



Rear view of printing system, showing number storage bank near bottom

total of 16 man hours. Furthermore, the fact that the frequency meter was under operating conditions during the 2.5 hours manual calibration period necessitated a temperature-controlled room for the calibration process, since ambient temperature changes, if permitted, could put kinks in the calibration curve, thereby rendering the meter inaccurate.

An automatic calibrating machine has been designed and constructed at Philco which, together with semi-automatic interpolating machines capable of typing the book directly, reduces the total calibration time from 16 hours to 6.5 hours. The actual direct time necessary to record the 327 calibration points of each frequency meter has been reduced from 2.5 hours to 16 minutes, and during these 16 minutes, the increments between adjacent calibration points are also automatically tabulated, thereby eliminating the previous hour required for manual computation. The short calibration time also eliminates the need for a temperature-controlled calibrating room. The automatic method has eliminated the human error, thus reducing the required checking time from 3.5 hours to 1.5 hours, which period is primarily devoted to insuring the stability of the frequency meter. Overall, since the equipment is being used on a 24-hour-a-day basis, over 140,000 man hours were saved in 1943.

Automatic Equipment Used

The semi-automatic interpolating machines are similar to the adding machines used by finance companies for scheduling payments, the only difference between the two being the size of type employed. These machines are capable of carrying two totals, one of which may be added to the other and the new total printed, thereby enabling interpolation. Since this device is well known it will not be discussed further here.

The 126-tube automatic calibrating machine has several unusual features. Essentially, it consists of three parts, shown as a block diagram in Fig. 1. The first supplies a source of standard frequencies against which the meter is calibrated. The second provides a means of mechanically continuously driving the dial of the frequency meter and electrically generating a sharp pulse every time the frequency meter is tuned through zero

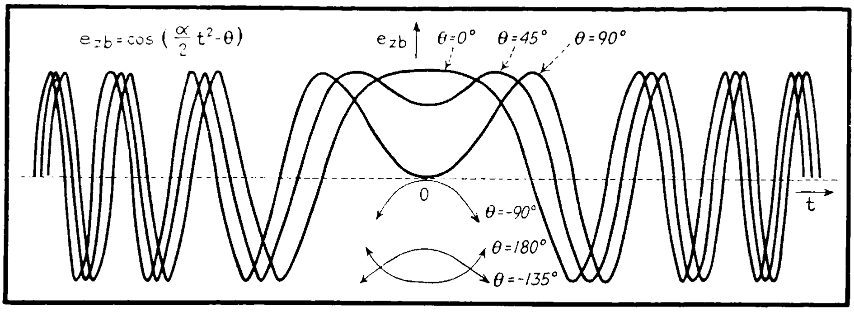


FIG. 2—A few of the infinite number of zero beat wave forms which can occur, depending upon the random value of Θ

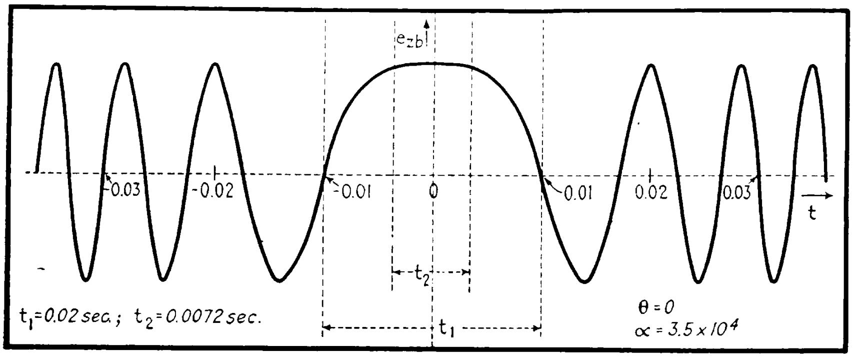


FIG. 3—Zero beat wave form for which $\Theta=0°$, and for which a calibrating time of six minutes is assumed. Here t_2 represents the maximum permissible limits for triggering

beat with the standard signal. The third unit records on a paper sheet the dial reading of the frequency meter at that instant of time at which the pulse is generated and also the difference between adjacent dial readings.

Standard Signal Source

As stated previously, calibration points must be taken every 1 kc on the low band and every 10 kc on the high band of the frequency meter. This requires accurate signals at 16-kc intervals for the low band (as will be explained later) and signals 10 kc apart for the high band. The method used for generating these signals employs two multivibrators, each locked in with a crystal oscillator that is continuously monitored against Bureau of Standards radio station WWV. Since this system is conventional, it need not be treated here.

Zero Beat Detecting Problem

Before describing the actual mechanism which was finally designed to detect the instant of zero beat, a short description of the problem involved will be given. That there is a problem at all is a result of the continuous drive applied to the frequency meter. If the drive were stopped at each calibrating frequency, conventional circuits could be used to determine zero beat within a very few cycles. However, as will be seen, the constant rotation of the tuning capacitor of the frequency meter introduces factors which require special consideration.

Let it be assumed that a constant-speed drive is being applied to the tuning capacitor shaft, so that the frequency meter is generating a signal which is changing continuously in frequency. Assume also a single standard signal whose frequency is constant and lies in the range of the frequency meter. And finally, let the outputs of the two signal sources be coupled into a mixer stage whose output circuit is responsive only to signals in the audio range. Then, as is shown in Appendix I, the wave shape of the audio signal developed across the output of the mixer will be as indicated in Fig. 2, where the origin for time ($t = 0$) is taken to be the instant when the frequency meter is exactly at zero beat with the calibrating signal.

It will be noted immediately that more than one form of the zero beat wave shape has been given. There are actually an infinite variety. The exact one taken depends on a quantity indicated as θ, and a constant α.

The equation of the zero beat voltage is

$$e_{zb} = E_{zb} \cos\left(\frac{\alpha}{2}t^2 - \theta\right)$$

where the amplitude E_{zb} is determined by the amplitudes of the two signals, α is the rate of change of periodicity, ω_f, of the frequency meter, and θ is a random phase angle dependent on the phase angles of the original beating signals. E_{zb} and α can be held constant, but θ may, and probably will, have a different value each time that the frequency meter passes through zero beat with a calibrating signal. Hence, any of the infinite variety of which six are shown in Fig. 2 can be expected to appear at one time or another.

It remains to be shown that the difference in zero beat wave form presents a problem. Figure 3 is a quantitative picture of one particular wave which might occur. Here θ is taken to be zero, and a calibrating time of six minutes on the 2 to 4-Mc band of the frequency meter is assumed. With this calibrating time, α, the rate of change of periodicity, is $2\pi(4\times10^6 - 2\times10^6)$ divided by 360 seconds. Under these assumptions the time t_1 taken for the beat voltage to go through its first zero value is approximately 0.020 second (Appendix II).

Accuracy Required

To achieve the overall accuracy of calibration mentioned in the introduction, it was desired that the error introduced by the zero beat detector itself should be unreadable on the dial of the frequency meter. Since the latter is graduated into fifty thousand vernier divisions, it was decided that an accuracy of better than plus or minus one part in one hundred thousand (half of one vernier division) would be acceptable. Assuming a linear scale, on the 2 to 4-Mc band of the frequency meter each vernier division represents 40 cycles change in frequency. Therefore it was required that the zero beat indicator trigger when the frequency of the signal under calibration was within 20 cycles of true zero beat. With a calibrating time of 360 seconds the frequency meter changes in frequency 20 cycles in a period of plus or minus 0.0036 second, or a total of 0.0072 second. This is indicated as t_2 on Fig. 3.

The problem of calibrating with sufficient accuracy arises from the fact that the time interval t_2 is relatively short compared with t_1. Over the latter interval the zero beat wave is marked by distinctive characteristics which could be used to trigger the calibrating equipment. But it might be difficult to make such triggering occur always in the shorter interval t_2.

Integration, A Step Toward Solution

Evidently the best means of utilizing the distinctive nature of the wave form near zero beat is an intergrating network. Such a circuit may be considered most simply as summing algebraically the area under the beat wave curve, adding areas above the zero axis, and subtracting areas below. At times remote from the instant of zero beat, alternate positive and negative areas are relatively small and nearly equal, so that their algebraic sum is small and builds up in amplitude very gradually. However, through the period t_1 Fig. 3) a large positive area is added. The integral or sum hence contains a large positive pulse, building up through the interval t_1. Such a pulse could be used to trigger the indicating equipment.

Referring to Fig. 2, if θ had been 90 deg instead of 0 deg, a very different pulse would have been developed, and in addition it would have begun to build up much earlier in time. It is therefore highly possible that triggering would have occurred outside the period t_2 which has been specified. Further, for other values of θ the pulse might be negative, or there might even be no appreciable pulse at all.

The integrated output voltage obtained for various values of θ is shown in Fig. 4, with the acceptable triggering period t_2 again indicated. These curves are also based on a six-minute calibrating time, and are in the form of Fresnel integrals. Tables of this integral are available.

It is apparent therefore that the major difficulty with the system outlined is the erratic nature of the pulse which would be obtained,

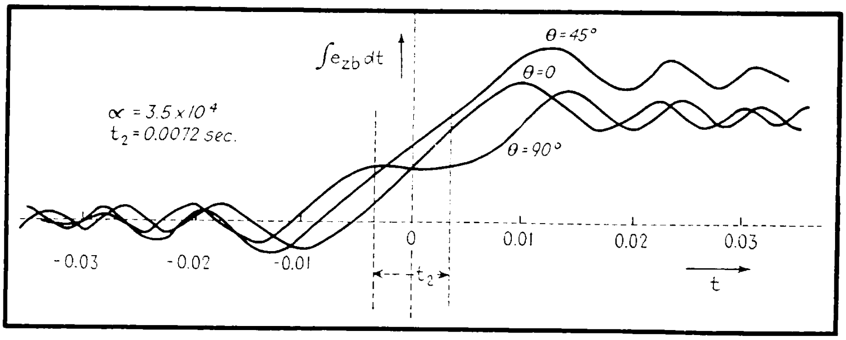


FIG. 4—Integrals of various zero beat wave forms. For $\Theta = -90°$, $-135°$, or $180°$, curves identical with those shown for $\Theta = 90°$, $45°$ and $0°$ respectively will be obtained, except that the two sets of curves will be of opposite polarity

with respect to amplitude, shape, and time of occurrence. Fortunately a means of overcoming this difficulty was found. The mathematical basis for the method devised will be stated here, and a proof of this particular case is given in Appendix III.

Principle of Zero Beat Detector

For certain types of functions, of which the integral of the zero beat wave given above is one, the sum of the squares of any two forms of the function which differ only in that they are 90 deg apart is always the same, regardless of the absolute phase of the two forms. Moreover, the resulting function is the square of the envelope of the original function plotted for all possible values of its phase angle.

A simple example of this can be given. If the original function is taken to be $A \sin(\omega t + \theta)$ where θ is any phase angle, then a second form, differing in phase by 90 deg, is $A \sin(\omega t + \theta + 90°)$. But this is the same as $A \cos(\omega t + \theta)$, and the sum of the squares of these two forms is A^2. If a plot of the original function is made for all values of θ, it is seen that the envelope is a straight line of amplitude A, and the square of this envelope is a line of amplitude A^2.

In Fig. 5 is shown the envelope of all the integral curves in Fig. 4, with the dashed line representing the square of this envelope. This dashed line is the pulse which can be derived from the zero beat wave regardless of what value the random angle θ may have. It is only required that a second zero beat wave be produced which differs in the angle θ by 90 deg. This can be done readily, as will be described later. The squaring action produces a sharper pulse than any of the integrals themselves. The acceptable triggering time is again shown as t_2.

This is the basic principle of the zero beat detector. The manner in which it was incorporated will next be described.

Superhet Circuit is Used

For the same fundamental reasons involved in the design of an

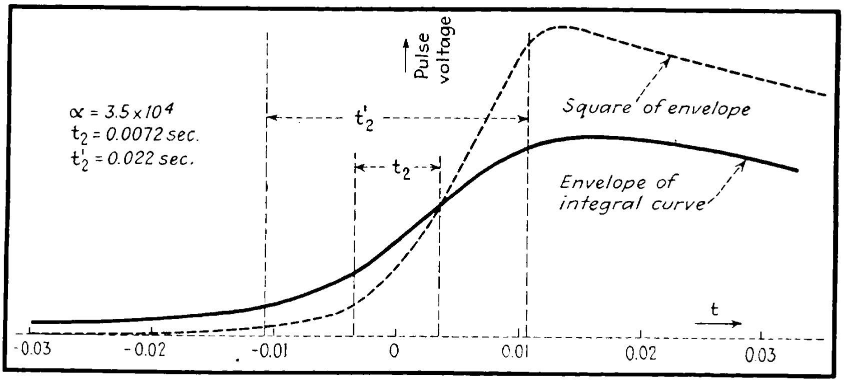


FIG. 5—Pulse form obtained at each zero beat by addition of squared integrals of quadrature zero beat wave forms. This curve is independent of random phase variations between the frequency meter and the standard signal. Note that practically the entire pulse occurs in the interval t'_2.

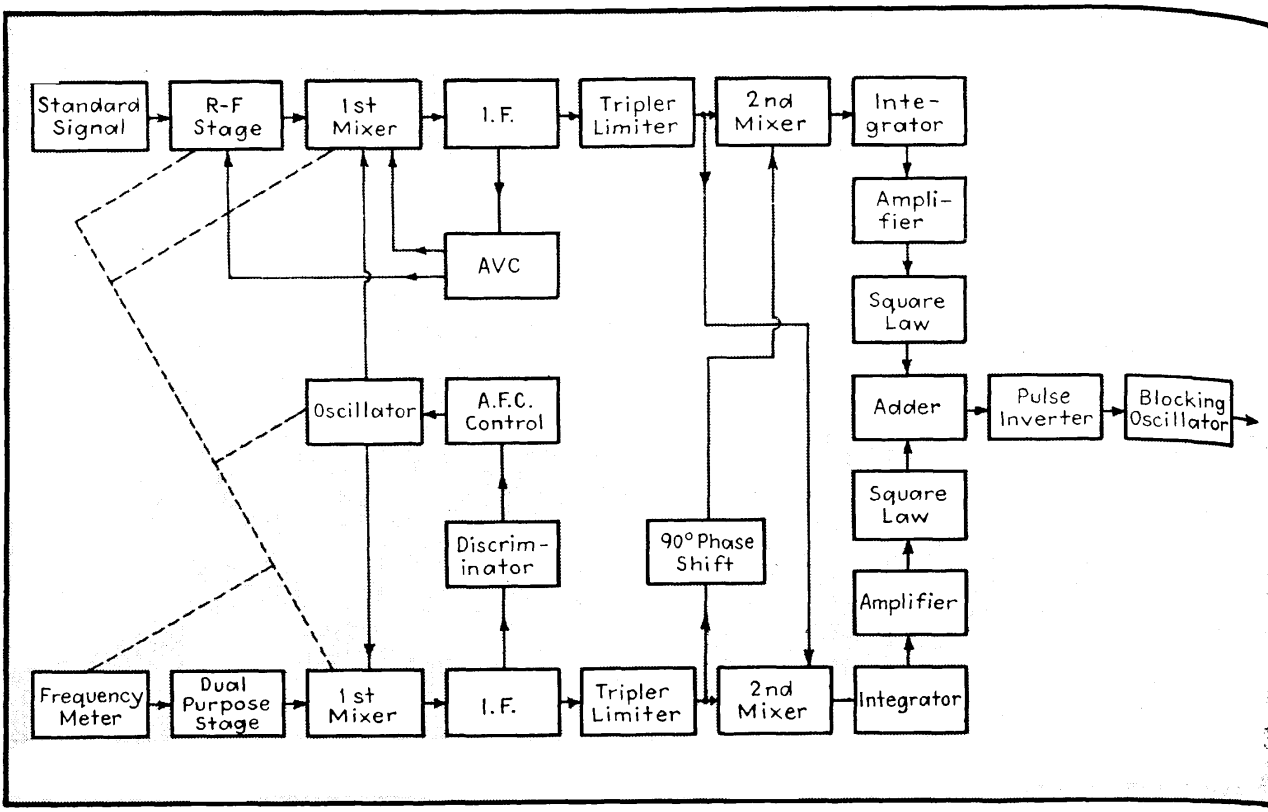


FIG. 6—Block diagram of complete zero beat detector

ordinary radio receiver, it was decided to use the superheterodyne type of circuit for the zero beat detector. There are two channels, one for the standard signal and one for the frequency meter, as shown in the block diagram in Fig. 6. The variable tuned portions of these and the frequency meter are all ganged together, on one motor-driven shaft. This ganging is indicated on the block diagram by the dashed lines, and is pictured in Fig. 7.

Each channel is fed from its own source, converted to an intermediate frequency by a common oscillator, and passed through its own i-f system. Both are then coupled to each of two mixers, the outputs of which are zero beat forms differing from each other in phase by 90 deg. These are each integrated and passed through square law stages, and the sum of the two taken. At this point the required constant-shape pulse has been obtained. The remainder of the detector is made up of circuits for obtaining a large-amplitude pulse suitable for operating the printing mechanism.

These channels will now be examined stage by stage, and those circuits of a unique nature described in some detail.

R-F Section

Channel 1 is fed from a standard signal source, which in this case is a multivibrator held in synchronism with WWV as heretofore mentioned. Since for high band calibration (which will be discussed first) signals are generated every 10 kc from 2 to 4 Mc, it was found desirable to introduce a tuned r-f stage to remove all but a few signals in the vicinity of the one desired. These selected signals are applied to one grid of the first mixer in channel 1.

Channel 2 is fed from the frequency meter through a dual-purpose stage, which for high band operation is effectively a unity gain untuned stage. It will be described in more detail later. The signal is then applied to one grid of the first mixer in Channel 2.

The second grid of each mixer is fed from a common oscillator stage, but from isolated points to prevent channel interaction.

These stages comprise the r-f section of the unit, and as mentioned above are ganged on a common shaft. All except the oscillator are adjusted to tune together and are so geared that their curves of frequency versus shaft rotation are the same as an average frequency meter. The oscillator is adjusted to track 480 kc higher at all points, a frequency which is entirely optional and determined only by design considerations. Because of the type of circuit used to obtain the quadrature signal, however, it is necessary that this frequency, once determined, be maintained exactly. To this end a conventional automatic frequency control circuit is used, operating from channel 2.

First Mixers

The signals developed in the plate circuits of the two first mixers may now be considered. Since the frequencies of the oscillator and the frequency meter are varied together and held to a constant difference of 480 kc, the signal in the plate of the mixer in channel 2 is of course constant at 480 kc. In channel 1 this is not the case because

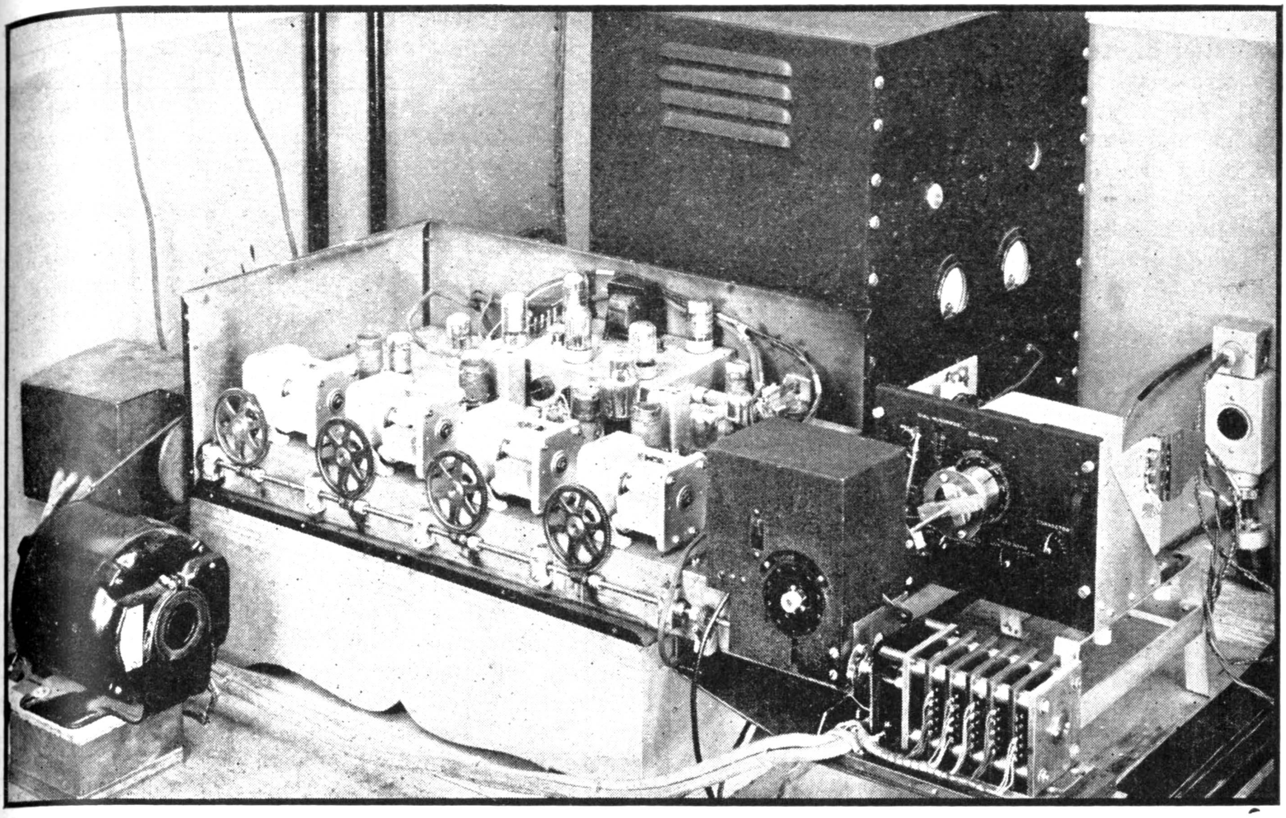


FIG. 7—Overall view of zero beat detector with r-f section open. Note ganging of tuning capacitors to the motor at left, and to revolution counter and frequency meter at right. The i-f channels and pulse-forming stages are in the rack at the rear, together with regulated power supplies for each channel

the signals against which the oscillator is beating are fixed, while the oscillator itself is varying at a uniform rate with the motor drive.

Assume that the oscillator is at 3.48 Mc at some instant. The mixer and r-f stages of channel 1 are then tuned to the particular standard signal which is at 3.0 Mc and the i-f is 480 kc. As the oscillator changes to 3.485 Mc, the beat from the 3.0-Mc signal becomes 485 kc. But at this time the beat from the next standard signal, which is 3.010 Mc, becomes 475 kc, and as the oscillator approaches 3.49 Mc, this beat approaches 480 kc.

Thus, as the oscillator is tuned through each standard signal, the i-f signal in channel 2 sweeps through the frequency 480 kc. If the band width of this i-f channel is held to less than plus or minus 5 kc, there will be only one signal present at any time. This is necessary to prevent spurious beats.

Limiters and Triplers

Proceeding through the two i-f channels, it will be noted that the final i-f stage in each case is a combined tripler and limiter. In order that the calibrating pulse may be of the same amplitude as well as the same shape for each zero beat, it is necessary that the two signals which are actually being combined be of the same amplitude (though not necessarily the same as each other) for each beat.

Limiting is therefore incorporated in the plate circuits of the two final i-f stages, and sufficient gain is provided in the i-f systems so that on the weakest standard signal and the lowest output frequency meter these stages are driven well past their limiting levels. Thus considerable latitude may be permitted in the amplitudes of the harmonics generated by the multivibrator and the output of the frequency meter without impairing in the least the accuracy of the equipment.

An avc system is also employed in the standard signal channel for the same purpose, inasmuch as this channel is particularly subject to wide variations in signal level. With these circuits included in the i-f systems, the amplitude of the signal at each mixer grid is maintained extremely constant.

The use of tripling in the final i-f stages is interesting in that it provides a simple means of tripling the accuracy. Recalling the equation for the zero beat wave, it is seen that the time taken for the beat to go through the distinctive period t_1 is proportional to α. If α be tripled, this time interval is reduced to one third. But α is proportional to the number of cycles of change in frequency per second, and hence if we triple the changing frequency and therefore triple the number of cycles of change per second, the time interval within which a triggering pulse is generated is reduced to one third.

The effect of this is shown in Fig. 5 by expanding the time scale 3 times so that the acceptable triggering interval is represented by t_2'. Channel 2 is tripled of course merely to keep its frequency the same as the center frequency of channel 1. It is obvious that quadrupling or even higher multiplication could be used.

Second Mixers

The output of the final i-f stage of channel 1 is now fed to one grid of each of the two second mixers. The latter are the stages in which the zero beats are actually developed. The actual frequency meter signals and standard signals are not being used directly to produce the zero beat, but since a common oscillator is used, when the i-f signals are identical in frequency it follows that the two r-f signals are also identical.

Note that a slight deviation of oscillator frequency from its proper value, such as might be caused by drag in the afc system, does not impair the accuracy. Such a deviation will of course change both intermediate frequencies by the same amount. Note further that this stepping down from r-f to i-f does not reduce the accuracy in the same way that tripling was seen to increase it; because in this case the reduction in frequency is obtained by subtracting from another frequency, and not by division. The rate of change of frequency, as represented by α, is unchanged.

The output of the tripler stage in channel 2 requires one more operation. It is necessary to obtain not only the third harmonic of the constant frequency, but also a second signal identical except shifted in phase by 90 deg. This is readily obtained by introducing a loosely coupled double-tuned transformer, and taking one signal from the primary and one from the secondary of the latter. With the input at the proper frequency, these signals are in quadrature with each other. It is to maintain input at proper frequency that afc is utilized. The quadrature signals are applied to the second mixers in channels 1 and 2.

The signals found in the plates of the two second mixers are now two identical zero beats between the fixed and varying i-f signals, except that the angle θ is different by 90 deg. (Appendix IV). Each is now ready for integration.

Checking Quadrature at Mixers

Before discussing integrators, a simple means of checking quadrature at this point might well be mentioned. If the outputs of the two second mixers be connected each to one pair of plates of a cathode-ray tube, and two r-f signals of approximately the same frequency applied to the inputs of the system, then an ellipse with axes at right angles should appear on the screen of the tube. The secondary tuning of the quadrature transformer may in fact be adjusted by this method. This should preferably be done with the two r-f signals as nearly of the same frequency as possible since this is the condition which is of interest.

Integrating Circuits

A standard integrating circuit consisting of a large series resistance and low-reactance capacitor is given in Fig. 8, while the actual circuit used is shown in Fig. 9. The difference is occasioned by the fact that it is necessary to incorporate a blocking capacitor between the mixer plate and amplifier grid, and it is also necessary to supply a d-c return for the latter. The actual constants used are determined by two factors. In Fig. 8, the higher the value of R and the lower the reactance of C, the more nearly will the output approach a true integral. However, it is necessary in Fig. 9 that the time constant in the amplifier grid be short enough to permit the grid to return to its normal no-signal bias between pulses. The values given proved to be a satisfactory compromise.

Square-Law Stages and Adder

The two integrals thus obtained are passed through amplifiers to the square-law stages and the adder, shown in Fig. 10. Each square-law stage consists of two type 7B7 pentodes biased to a point where the second harmonic distortion is greatest. They share a common plate resistor, and their grids are driven with opposite polarity from the plate and cathode of a type 7A4 tube serving as phase inverter. The fundamental and third harmonic are balanced out and the voltage across the common plate resistor is predominantly second harmonic, or the square of the input voltage.

Addition of the two squared signals is accomplished by connecting a potentiometer from one pair of plates to the other and taking output from the arm at the center. At this point the signal voltage is the average of the two plate voltages, or one-half their sum. A potentiometer is used so that any unbalance in the square-law characteristics of the pentodes resulting in less output from one pair may be adjusted by moving the arm slightly nearer to that pair. Unbalance between each tube of either pair is first adjusted by means of the variable cathode resistor of the phase inverter.

Here again, cathode-ray tube checks simplify the adjustment of the equipment. R-F signals are

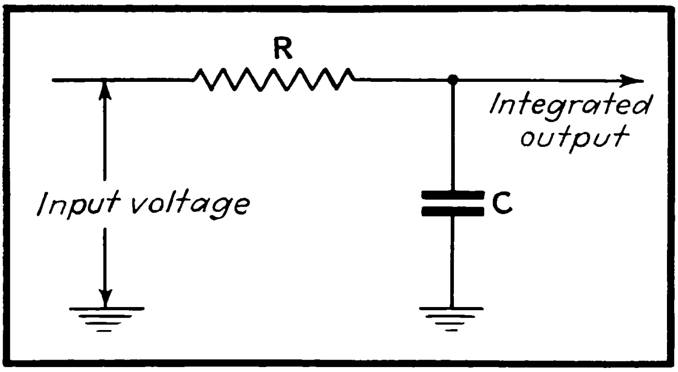


FIG. 8—Conventional integrating network

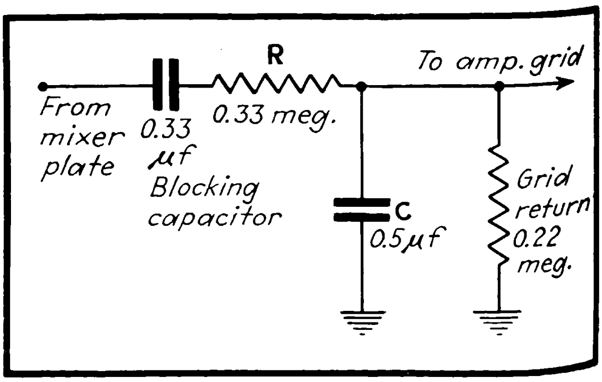


FIG. 9—Schematic of actual integrating circuit used

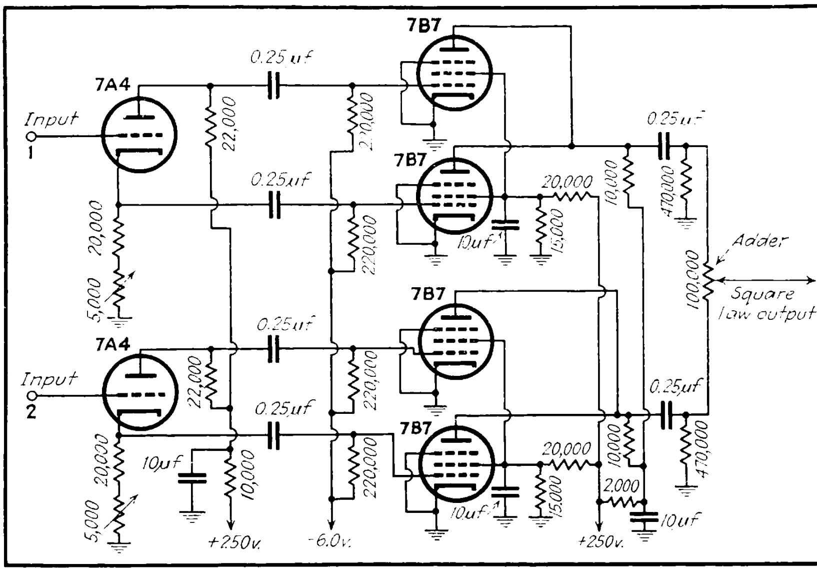


FIG. 10 (above)—Schematic of square-law stages and adder

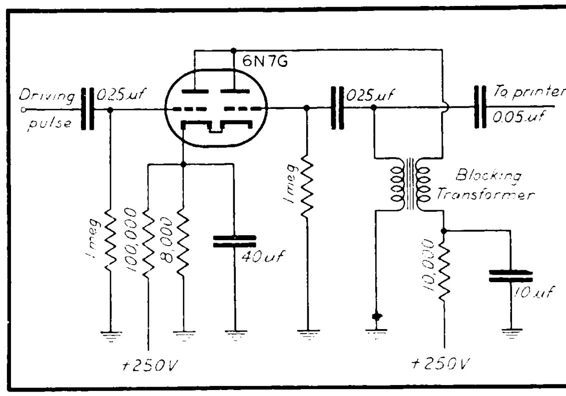


FIG. 11 (right)—Schematic of blocking oscillator which is triggered by pulse from adder and provides final pulse used to actuate printer

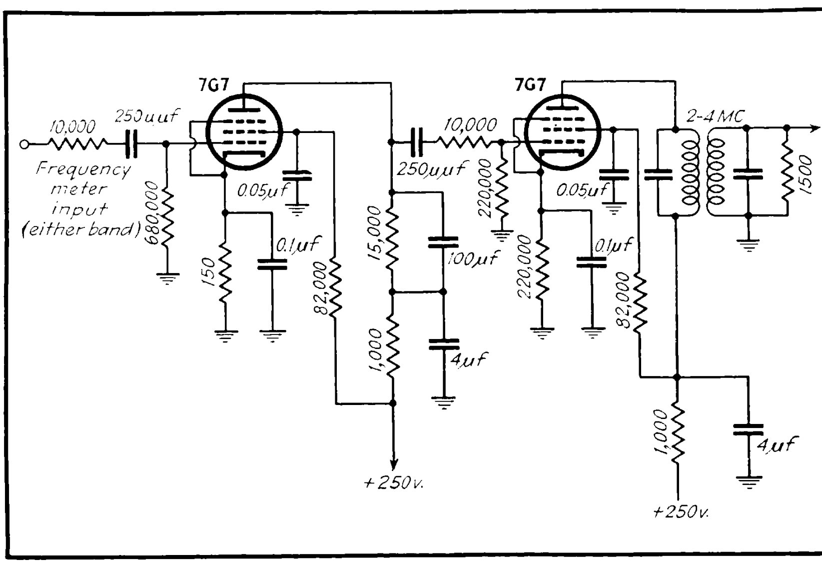


FIG. 12—Schematic of dual-purpose input stage for channel 2. This stage accepts a signal from either band of the frequency meter without switching, and supplies a 2 to 4-Mc signal to channel 2. The latter signal is the fundamental of the high band or the sixteenth harmonic of the low band

applied to the inputs of the equipment, and one phase inverter triode removed. The cathode-ray tube plates are connected to the input of the other phase inverter and to the plates of the corresponding pair of square-law pentodes. Since the audio signal and the square of the latter are thus compared on the cathode-ray tube, there should appear a "hair pin" or U. The resistor in the cathode of the phase inverter is then varied until the tips of the hair pin are of equal amplitude. This indicates equal square-law output from the two pentodes. The operation is then repeated on the other square-law stage.

Since the r-f signals are fixed, the signals at the inputs to the square-law stages are sinusoidal and are the same except for differing in phase by 90 deg. They may be represented by $A \sin (\omega t + \theta)$ and $A \cos (\omega t + \theta)$, and the sum of their squares is A^2, or in other words, a d-c component only. The signal at the arm of the adding potentiometer should thus be a straight line, and of zero amplitude if the oscilloscope does not pass direct current. The vertical plates of the cathode-ray tube are therefore connected to the arm of the potentiometer, and the latter adjusted for a minimum of audio ripple. Some of the latter will be observed, since a certain amount of fourth and higher even harmonics will be present in the output of the square-law tubes.

Blocking Oscillator

At each zero beat there appears at the arm of the adding potentiometer a pulse which is constant in shape and amplitude. It is, however, negative in polarity because of the nature of the square-law stages, and is not of sufficient amplitude to actuate the printing mechanism. To obtain a sufficiently large pulse, the blocking oscillator shown in Fig. 11 is used. The negative pulse is applied to the latter through a polarity inverter, which is merely a resistance-coupled amplifier stage.

It will be noted that positive bias obtained from a voltage divider across the main plate voltage supply is applied to the cathode of the blocking oscillator double triode. The amount of this bias is sufficient to cut off both sections of the 6N7G so that it is normally quiescent. Its free running period (without bias) is long compared to that of the driving pulse, but somewhat less than the time interval between pulses.

When the positive driving pulse appears at its first grid for a short instant, it begins to pass through one cycle of a normal blocking oscillation, and the usual sharp pulse of great amplitude (over one hundred volts) is developed across

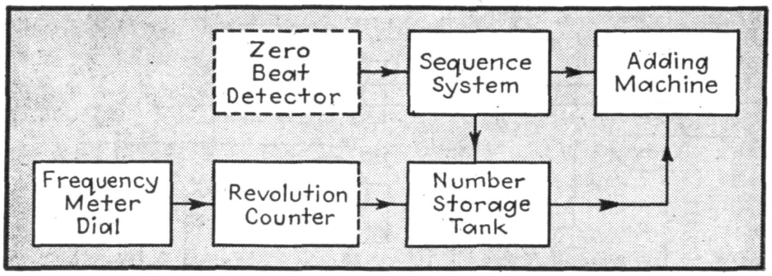


FIG. 13—Block diagram of printing system

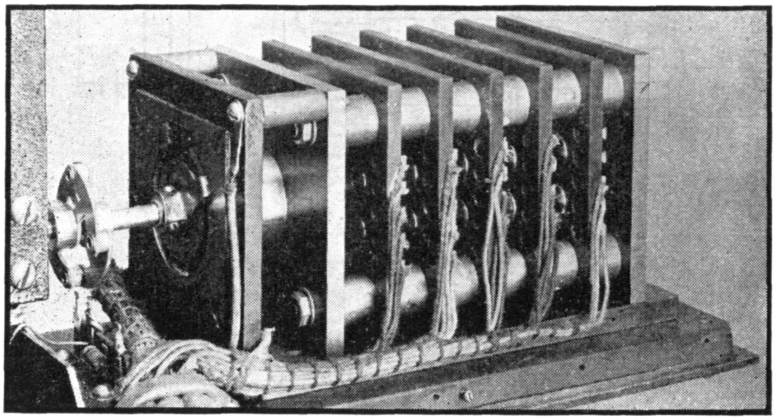


FIG. 14—Open view of revolution counter switch. Note the 5 decks of 10-point switches, and the "disconnect" section at front

the grid winding of the transformer. The second grid immediately blocks and remains cut off for the remainder of the natural period of the oscillator.

The system is thus unresponsive to spurious pulses of any sort during the greater part of the interval between true pulses. By the time the oscillator would normally begin its second cycle the driving pulse has long since disappeared, so that only one output pulse is obtained for each zero beat. This is the final output of the zero beat detector.

Low Band Calibration

Calibration of the low frequency band proved to be more of a problem than had been anticipated. It was originally intended to divide all frequencies by 16 in conformance with the frequency meter itself. Standard signals every 1 kc from 125 to 250 kc, with an i-f value of 30 kc, were to be used. Gradual multiplication through the i-f system was to bring the final frequency again to 1440 kc, so that the balance of the equipment would be the same for both bands.

Such a system was actually constructed, and was found to be unsatisfactory simply because of the difficulty in eliminating phase modulation from the source of 1-kc signals. In order to synchronize such a source on the 100-kc crystal oscillator, it was necessary to use at least one intermediate multi-vibrator at 10 kc. As a result it was found that while the total number of cycles per second remained correct, the starting time of each cycle was subject to slight variations. This is the equivalent of erratic phase modulation of the 1-kc signals. Because of the enormous multiplication involved (in the worst case, from 1 kc to 250 kc and from 30 kc to 1440 kc) the phase modulation became several cycles of frequency modulation. The zero beat detector is of course very sensitive to frequency modulation, so that spurious beats were obtained in nearly every case.

After considerable investigation a satisfactory means of making the calibration was devised. It will be noted that the sixteenth harmonic of the low band varies from 2 to 4 Mc. By multiplying the signal from the frequency meter by sixteen, therefore, it may be run through the high band calibrator directly. Since calibration points are required every 1 kc of the fundamental, the sixteenth harmonic must pass through zero beat every 16 kc on the calibrator, and a multivibrator producing standard signals every 16 kc from 2 to 4 Mc is required. It will be seen that 126 calibration points will thus be obtained, which is the number required.

To obtain synchronization for the 16-kc multivibrator, an 80-kc multivibrator was locked on the fourth harmonic of the 100-kc standard. The fifth harmonic of the 16-kc multivibrator was then synchronized on the 80-kc signal.

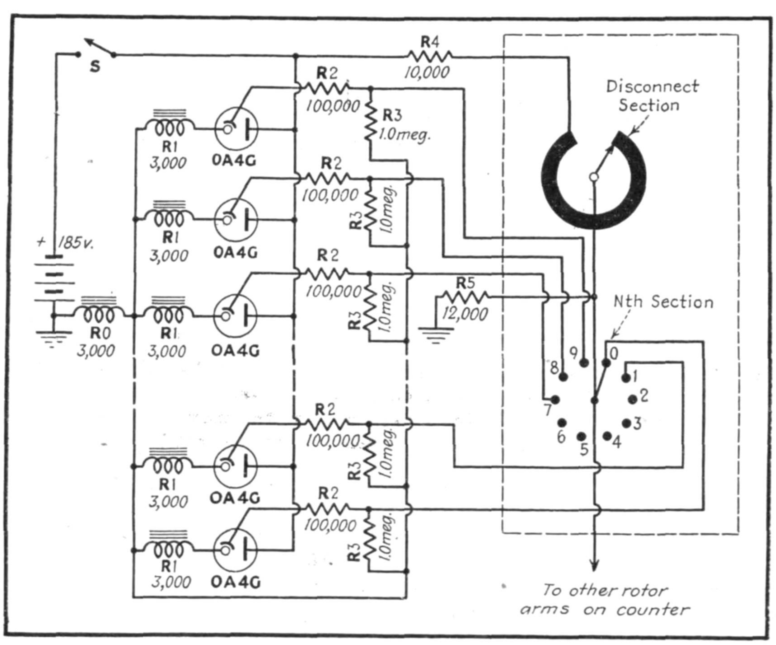


FIG. 15—Schematic of one column of number storage bank. The dashed rectangle at the right encloses a representative section of the revolution-counting switch

Dual-Purpose R-F Stage

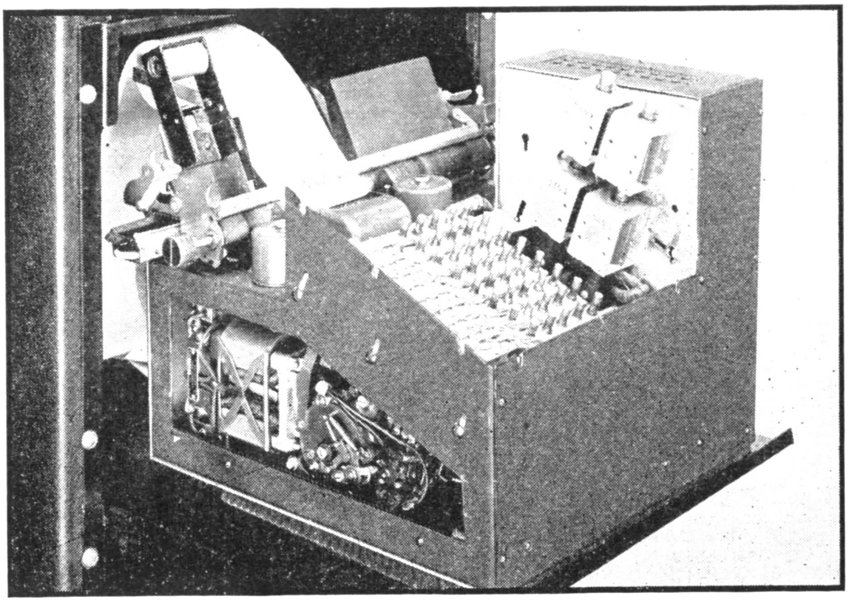
The dual-purpose two-band stage located between the frequency meter and the first mixer of channel 2 is shown in Fig. 12. It was designed to accept signals from either band of the frequency meter without switching, passing the high band directly, and multiplying the low band by 16. The first 7G7 provides considerable gain on the low band, but relative attenuation on the high band because of the capacitor shunting the plate load. The second 7G7 is a straight band-pass amplifier on the high band, and a multiplier to select the sixteenth harmonic of the low band. As a multiplier, the gain of this stage is of course much less than as an amplifier. At the output the signal from either band is of the same order of magnitude, and the overall gain is therefore approximately one.

Recording Problems

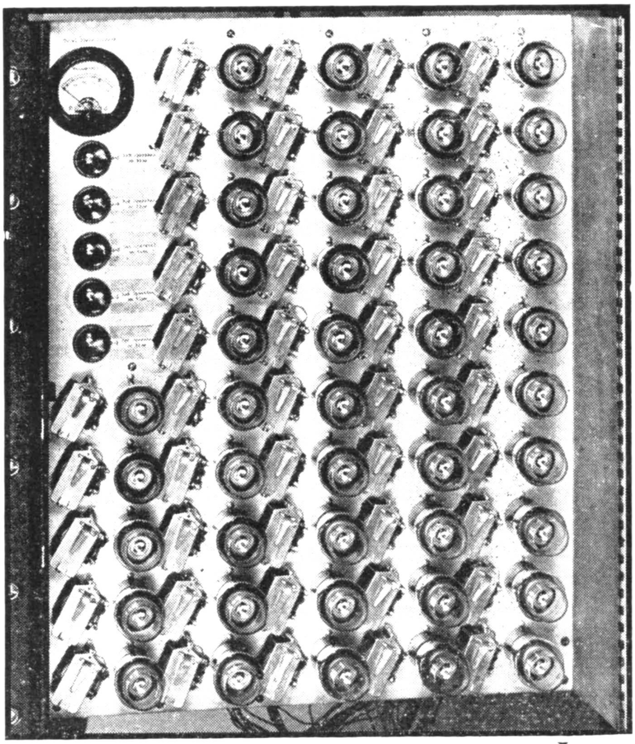
The zero beat detector serves to generate a pulse at the instant that the frequency meter is tuned to a calibration frequency. The remaining problem is that of accurately recording the dial reading at this instant. Since the dial is driven through 50,000 vernier divisions in approximately 6 minutes, 139 vernier divisions are passed in a single second. To record accurately the dial reading to the nearest vernier division while the dial is rotating at this speed requires special care. An ordinary revolution printer which momentarily presses a piece of paper against a set of revolving drums carrying the dial reading would cause appreciable blur.

Such a printer could be modified by having the drums stand still while the imprint is being made, after which the revolutions lost by such standstill would have to be made up through a differential or spring storage system. The mechanical complexity of such a recorder, however, is such that frequent maintenance would be required. A Strobotac photograph might be employed as an alternative but fast emulsions would have to be used, either necessitating production delay in development

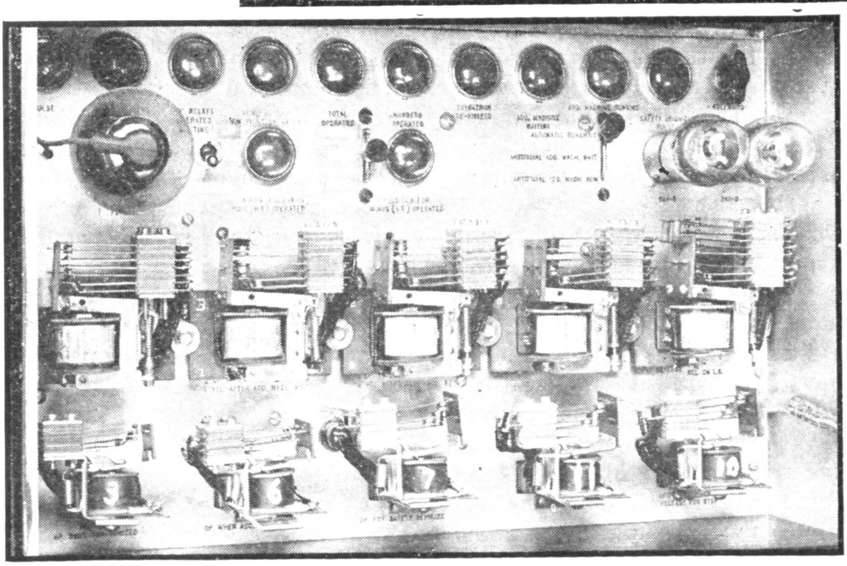
(*Continued on page* 342)



ABOVE
FIG. 16—Open view of adding machine, showing solenoids for actuating keys



RIGHT
FIG. 17—Number storage bank, showing 45 of the OA4G tubes, the associated relays, and monitoring lights and meter at upper left



BELOW
FIG. 18—Sequence system, showing thyratron *S* at left, relays, and monitoring lights for rapid indication of correct sequence of operations

Automatic Calibrator

(Continued from page 107)

by established photographic concerns, or requiring that the radio firm manufacturing the frequency meter go into the photographic business.

Recording System Used

To overcome these objections, it was decided to use an adding machine as the printer by providing a means of setting up the dial reading of the calibration point on the keyboard.

The complete recording system is shown in the block diagram in Fig. 13. The frequency meter dial is mechanically coupled through a gear drive of suitable ratio to an electrical revolution counter. This counter (see Fig. 14) consists of 5 decks of 10-point selector switches, each deck being geared to its adjacent deck by a 10:1 intermittent stepdown so that every time the "tenths" rotor rotates between 9 and 0, a gear link advances the "units" rotor by one stator segment. Every time the "units" rotor passes between 9 and 0 another gear link advances the "tens" rotor by one segment, etc.

The stator contacts selected by the rotors at any given instant, therefore correspond to the numerical dial reading at that instant. If, at the particular instant at which a reading is to be taken, the "tenths" rotor happens to be between contacts, then the next point to be contacted will be recorded.

For this reason and because of the practical impossibility of adjusting the "tenths" rotor so that it will break contact with the "9" point exactly simultaneously with the breaking of the rotor contact in the units section, a "disconnect" section is added. The rotor of this section is directly coupled mechanically to the "tenths" rotor, and the stator segment is so arranged that the rotor and stator are in contact throughout approximately 330 deg of rotation. The remaining 30 deg is a little greater than the angle throughout which the "tenths" rotor is not in contact with either the "9" stator contact or the "0". Electrically this "disconnect" section is in series with the voltage supplied to the remaining rotor sections, so that the switch is rendered electrically inoperative during the mechanical throw-over period.

Each stator contact of the revolution counter is connected to the starter anode of a cold-cathode gas discharge tube (OA4G) as in Fig. 15, which shows the schematic of one column of the number storage bank. The circuit is so arranged that when switch *S* is open, all voltages are removed from the tubes, leaving them de-ionized.

When the reading on the counter is to be recorded, switch *S* (which is actually a thyratron ionized by the pulse from the zero beat detector) is closed, supplying 185 volts to the anodes of all the tubes. Resistors $R4$ and $R5$ supply approximately 100 volts to the starter anode of the first tube to be selected by the rotor of the counter switch.

Under these conditions, the tube selected will ionize and reduce its anode-cathode potential to 70 volts. Since $R1=R0$, 57.5 volts will appear across $R0$ and will act as a positive bias on the cathodes and starter anodes of the remain-

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ing tubes. Further rotation of the counter switch rotor will then supply an effective starter anode-to-cathode voltage of only 42.5 volts to any subsequent tubes selected, and this value is insufficient to ionize any of them.

After S is closed, the first tube to be contacted by the counter switch becomes ionized and will remain ionized until switch S is opened. (This "opening" will actually be a deionization of the thyratron being employed for S, after storage of the number is no longer required).

Thus the number storage bank constitutes an electrical means of storing the dial reading of the frequency meter at any calibration point for any desired length of time. This reading is transferred from the storage bank to the keyboard of an adding machine by having the plate current of each ionized OA4G tube operate a relay (coil represented by $R1$ in Fig. 15) which in turn operates a corresponding solenoid that presses the proper number key of the adding machine. A view of the adding machine and solenoids with cover off is shown in Fig. 16, and the entire number storage bank with the front door open is shown in Fig. 17.

Sequence of Printing Operations

After the dial reading of the calibration point is set up on the adding machine, it is required that the reading be printed and that the difference between adjacent readings also be tabulated. This necessitates the following sequence of operations:

(a) Press the "non-print" button.

(b) Press the "+" button if the frequency meter dial is revolving to increasing numbers.

(c) After the machine starts running, release all pressure.

(d) When the machine comes to rest, press the total button.

(e) After the machine starts running, release the pressure. (During the ensuing cycle of the machine, the difference between the present calibration point and the previous one will be printed.)

(f) When the machine comes to rest, again press the "number keys, as still determined by the number storage bank.

(g) Press the "−" button.

(h) When the machine starts running, open switch S of Fig. 15 by deionizing the thyratron used as S, thereby deionizing the cold-cathode tubes of the number storage bank so as to prepare it for a new calibration point; also remove pressure from all keys. (During this cycle of the adding machine, the dial reading of the calibration point will be printed.)

If the dial of the frequency meter is being driven from high numbers to low (on one band it is driven in one direction, and in the opposite direction on the other), then operations (b) and (g) above are reversed.

All of the above operations are performed by means of conventional relay switching practice, with each operation initiated by the closing or opening of a contact placed on the adding machine in such a manner that the contact is closed while the machine is cycling and the contact is opened when the machine comes to rest. The entire sequence system is shown with cover open in Fig. 18.

Contact-Failure Detector

A safety device is incorporated in the sequencing system to insure against incorrect calibrations in case of failure of any one of the contacts of the revolution counter. (Such failure might be expected to be rather frequent since the fastest revolving rotor of this switch makes over 100 contacts per second. In actual use 24 hours a day, the particular design employed in the revolution counter gives a contact failure about once every two months, or after about 500 million contactings.)

This safety device operates upon the time interval between the instant that the pulse is received from the zero beat detector and the instant that the one OA4G tube in each column of the number storage bank becomes ionized. If this interval is less than approximately 0.01 second, then the calibration point is printed in the normal manner; but if, through a contact failure of the revolution counter, the interval exceeds 0.01 second, then no calibration point is printed. This is arranged by a suitable relay system consisting of one relay coil of a fast-operating relay inserted at $R0$ of Fig. 15 for each column of the number storage bank, and a slower-operating relay connected from the plates of the OA4G to ground.

If the slow-operating relay closes after the five fast relays have operated (the difference in operating time between the two types of relays is 0.01 second) then the contacts of the revolution counter are apparently in proper working order and the circuit is arranged to permit the normal printing sequences previously described to be executed. If, however, the revolution counter skips a contact point at the instant that a dial reading is to be recorded, the OA4G which should have been ionized does not become ionized and hence the slow-acting relay will be operated before the fast-acting one. Through suitable additional relay circuits, this causes the sequence

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system to omit the calibration reading entirely and to leave two blank spaces on the recording tape.

Use of Calibration Tape

In this manner, then, the dial readings at the calibration points are recorded. The actual frequency corresponding to any given dial reading is determined from the known frequency at the start of the calibration tape and the number of readings intervening. In practice, after the calibration tape has been completed by the machine, it is placed upon a ruled table which has marking lines corresponding to certain reference frequencies. In this manner, key frequencies (such as those which start each page of the finished calibration booklet) can be marked off.

The ruled calibration tape is then used to set up manually the interpolating machines, which supply ten interpolated calibration points for each one appearing on the tape and which simultaneously print the pages of the completed calibration book.

Credit is acknowledged to Mr. D. B. Smith and Mr. E. S. Brotzman who directed the development of the zero beat detector and the recording system respectively, and to each of the following for their in-

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valuable contributions: W. Frear, F. Reed, W. Newbold, H. H. Harris, M. Ames, H. Brough, M. Riccardi, E. O. Thompson, I. Stephan, W. Mattison, A. F. Nickel, D. Karcher, C. Woll, W. Miller, A. Miller, H. Seher, W. Bareiss and F. Kotulka.

Appendix I

e_f = instantaneous voltage output of frequency meter

E_f = peak value of e_f

e_s = instantaneous voltage output of standard signal

E_s = peak value of e_s

e_{zb} = instantaneous value of zero beat voltage

E_{zb} = peak value of e_{zb}

θ_f = instantaneous value of the argument of the sine function representing e_f

ω_f = instantaneous periodicity of e_f $= \frac{d\theta_f}{dt}$

ω_s = periodicity of standard signal

α = rate of change of $\omega_f = \frac{d\omega_f}{dt}$ (assumed to be constant)

ϕ_f = phase angle of frequency meter signal (value of θ_f at instant of zero beat)

ϕ_s = phase angle of standard signal

$\theta = \phi_s - \phi_f$

t = time measured from the instant of zero beat

$$e_f = E_f \sin \theta_f = E_f \sin \left(\int \omega_f\, dt\right)$$

$$\omega_f = \omega_s + \alpha t$$

$$\int \omega_f dt = \omega_s t + \frac{\alpha t^2}{2} + \phi_f$$

$$\therefore e_f = E_f \sin \left(\omega_s t + \frac{\alpha}{2} t^2 + \phi_f\right)$$

$$e_s = E_s \sin (\omega_s t + \phi_s)$$

The product of these two signals, as obtained in a mixer, neglecting all but audio-frequency terms, is

$$e_{zb} = E_{zb} \cos \left(\frac{\alpha}{2} t^2 - \theta\right)$$

Appendix II

If $\theta = 0$, then

$$e_{zb} = E_{zb} \cos \frac{\alpha}{2} t^2$$

for a calibrating time of 6 minutes,

$$\alpha = \frac{2\pi (4 \times 10^6 - 2 \times 10^6)}{360} = 3.5 \times 10^4$$

Let T = value of t for which e_{zb} goes through its first zero values. Then

$$\frac{\alpha}{2} T^2 = \frac{\pi}{2}; \; T = \pm\, 0.01 \text{ sec. and } t_1 = 0.02 \text{ sec}$$

Appendix III

Given the function $f_1(x) = \int \cos (x^2 + \theta)\, dx$ (1)

A second form of the function differing in phase by 90 deg is

$$f_2(x) = \int \cos \left(x^2 + \theta + \frac{\pi}{2}\right) dx = \int \sin (x^2 + \theta)\, dx \qquad (2)$$

It is desired to show that

$$[f_1(x)]^2 + [f_2(x)]^2 = f_3(x)$$

where $f_3(x)$ is independent of θ and its amplitude is equal to the square of the envelope of $f_1(x)$ plotted for all values of θ.

Expanding Eq. (1)

$$\begin{aligned} f_1(x) &= \int \cos x^2 \cos \theta dx - \int \sin x^2 \sin \theta dx \\ &= \cos \theta \int \cos x^2 dx - \sin \theta \int \sin x^2 dx \\ &= C \cos \theta - S \sin \theta \qquad (3) \end{aligned}$$

where $C = \int \cos x^2 dx$, and $S = \int \sin x^2 dx$

Similarly

$$f_2(x) = C \sin \theta + S \cos \theta \qquad (4)$$

$$[f_1(x)]^2 + [f_2(x)]^2 = C^2 + S^2 = f_3(x) \qquad (5)$$

and $f_3(x)$ is obviously independent of θ.

Let Eq. (4) represent the general function. To determine the amplitude of the envelope at any point, x, the function must be maximized with respect to θ, holding x constant:

$$\frac{d}{d\theta}\left[C \sin \theta + S \cos \theta\right] = C \cos \theta - S \sin \theta = 0$$

$$\tan \theta = C/S \quad (6) \qquad \sin \theta = \frac{C}{\sqrt{S^2 + C^2}} \quad (7)$$

$$\cos \theta = \frac{S}{\sqrt{S^2 + C^2}} \quad (8)$$

Substituting (7) and (8) in (4)

$$\tan \theta = C/S \qquad (6)$$

$$\sin \theta = C/\sqrt{S^2 + C^2} \qquad (7)$$

$$\cos \theta = S/\sqrt{S^2 + C^2} \qquad (8)$$

Substituting Eq. (7) and (8) in Eq. (4) gives

$$\text{Envelope} = C \frac{C}{\sqrt{S^2 + C^2}} + S \frac{S}{\sqrt{S^2 + C^2}} = \sqrt{S^2 + C^2} \qquad (9)$$

The square of Eq. (9) is equal to Eq. (5), which was to be proved.

Appendix IV

In Appendix I, let the constant frequency signal change in phase by 90 deg. Then

$$e_s = E_s \sin (\omega_s t + \phi_s + 90°).$$

Thus

$$e_{zb} = E_{zb} \cos \left(\frac{\alpha}{2} t^2 - \theta - 90°\right) = E_{zb} \cos \left(\frac{\alpha}{2} t^2 - \theta'\right), \text{ which}$$

is the same zero beat wave except that $\theta' = \theta + 90$ deg.

PAY FOR RADAR RATINGS in the British Royal Navy has been increased, ranging from 2s. a day for instructors to 3d. a day for new ratings. Special badges are not issued to the officers and ratings in the radar branch.