A RECORDING APPARATUS FOR THE ANALYSIS OF THE FREQUENCY OF RAPIDLY VARYING SOUNDS

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By the use of a series of band filters whose transmission regions are distributed over the whole range of acoustic frequencies an accurate and rapid analysis of the frequency of speech and other rapidly varying sounds is possible. An apparatus based upon this principle is described here, with which the Fourier spectrum of the sound to be investigated is made directly visible on the screen of an electron ray oscillograph, so that the variations of the spectrum can be filmed. The properties of the apparatus, especially the resolving power and recording speed attainable are discussed, as well as a number of particulars of construction and use. In conclusion several spectrograms of vowels filmed with the apparatus are reproduced as examples of its use, and they are briefly discussed.

In the case of electro-acoustic apparatus for the transmission of speech or music it is customary to describe the behaviour of the whole or of the separate elements by the way in which they react to sinusoidal oscillations of different frequency. This custom is based upon the fact that every sound vibration can be resolved into a number of sinusoidal components (Fourier components); so that from the frequency characteristics of an element (amplifier, cable or the like) it is possible to deduce immediately the form in which the complex vibration will be transmitted.

There are many numerical, graphic and instrumental methods for the separation of a vibration into its Fourier components, i.e. methods of harmonic analysis. For the most part, however, these methods are based upon the use of functions which are given in the form of a diagram or a table. If, therefore, it is desired to analyse sound vibrations by these methods, the sound vibrations must first be recorded. Moreover, the frequency spectrum of sound vibrations, that of speech for example, continually changes, and it is often just these changes in which we are interested. In order to investigate these changes a large number of strips of the recorded sound would each have to be analysed separately, which is a very laborious method scarcely deserving practical consideration.

With these objections in view special methods of separation have been developed for electro-acoustics which correspond better to the existing requirements and possibilities in this field.

A very obvious method is to make use of a band filter as an analysing element. If by means of a microphone the sound vibration is converted into an electrical A.C. voltage and this is fed successively to a number of band filters with different transmission regions, then from the occurrence or absence of an output signal it can be deduced in what frequency regions components of the vibration being investigated lie, and how strong they are. Obviously so many filters must be used that the desired fineness of structure of the whole acoustic spectrum is obtained.

In order to avoid the necessity of a large number of band filters, which make the apparatus complicated and expensive, the following device is usually employed. A sinusoidal voltage whose frequency can be continuously varied (auxiliary frequency), is fed, together with the signal to be investigated, to a mixing valve followed by which is a single band filter. If the difference frequency (or sum frequency) of the auxiliary frequency and one of the frequencies present in the signal falls exactly in the narrow transmission region of the band filter, an output signal is observed. From the values of the auxiliary frequency at which this occurs the Fourier spectrum of the signal can therefore be deduced.

Only one band filter is needed here, but two disadvantages are also involved: the measurement takes some time and as a result a fairly lengthy constancy of the voltage to be investigated is required. Therefore the auxiliary frequency method can indeed be used, for example, for determining the deformation which a given constant input signal undergoes in an apparatus, but not for the analysis of speech or other rapidly varying sound.

For an analysis of these sounds, therefore, recourse must be had to the fundamentally simpler method in which separate band filters are used for the different components of band filters required.

Equipped with such a set of analysing elements, it is now also possible to make the Fourier spectra visible directly in a simple way. For this purpose the outputs of the whole series of band filters are connected successively with an electron ray oscillograph by means of a rapidly rotating switch. By means of suitable connections the spectrum of the sound to be analysed can then be observed directly on the fluorescent screen, and its variations seen or recorded on a film.
An apparatus working on this principle and constructed in this laboratory will be described in this article \(^1\), while in conclusion some results obtained with the apparatus will be dealt with.

**Description of the apparatus**

In fig. 1 the whole apparatus is shown diagrammatically. It contains 79 band filters \((F)\), whose successively scanned by a rotating switch \(E_1\). In this way the rectified voltage of each band filter in turn, which voltage is a measure of the corresponding component of the Fourier spectrum, is fed to the vertical deflection plates of a cathode-ray oscillograph (the elements \(G\) and \(M\) in fig. 1 will be disregarded for the present). At the same time a D.C. voltage in-

**Fig. 1. Diagram of the arrangement for the recording of Fourier spectra**. The sound to be analysed reaches the inputs of the 79 band filters via the pre-amplifier \(V_1\); the part of the signal transmitted by each band filter is rectified with a diode \(D\) and fed via the rotating switch \(E_1\) to the modulator \(M\), which supplies a modulated carrier wave of high frequency to the vertical deflection plates of an electron-ray oscillograph. With the rotating switch \(E_2\), a D.C. voltage increasing in steps is fed at the same time to the horizontal deflection plates. The length of the spectrum which appears in this way on the oscillograph screen is regulated with \(R_s\); \(V_2\) is a monitor amplifier which can be connected via resistances \(R_1\) by 79 switches \(S\) to the output of each band filter.

Transmission regions are distributed over the range of frequencies from 90 to 8000 c/s. The signal to be investigated is fed to the inputs of all the band filters connected in parallel. Since each filter can only transmit a very small part of the energy of the signal and the output voltage of each filter must, nevertheless, project sufficiently above the ordinary level of interference, the signal is amplified to the necessary level \((4 W)\) in the pre-amplifier \(V_1\), which because of inverse feedback causes only a very slight distortion.

Behind each filter there is a diode rectifier which rectifies the output voltage of the filter. Each of these rectifiers is connected to one of 79 contacts on a collector, which contacts are

\(^1\) An acoustic spectrometer constructed by Siemens & Halske is based on the same principle (E. Freystadt, Z. techn. Phys. 16, 533, 1935). In that apparatus, however, the division of the spectrum was less fine than in our case (namely 3 filters per octave).
amplitude is modulated in a modulator \( M \) with the rectified output voltages of the band filters. In this way vertical lines are traced on the fluorescent screen instead of separate points. In fig. 2 such a spectrum is shown. This method also has the advantage that the apparatus could be made less sensitive to interferences. Behind the modulator there is another band filter which transmits only a narrow frequency region around the generator frequency of 50 kc/s. Interfering voltages of other (not too low) frequency, which may be induced on the connections between the rectifiers and the modulator, are filtered out in this way, while any distortion products and the noise are rendered practically harmless.

The spectrum obtained on the screen, which in the case of speech for example continually varies, can be observed visually, or it can be photographed at short intervals on a moving film.

The action of the band filters can be checked by means of a monitor amplifier \( V_2 \) with loud speaker, which can be connected to each band filter in the place of the rectifier by means of a set of 79 switches \( S \). By reversing several switches or whole groups of them the sound in different frequency regions can be heard and the influence of the lack of certain frequencies on intelligibility can be studied.

After this brief description we shall now go into several important characteristics and structural details of the apparatus.

**Resolving power and recording speed**

It would be desirable to be able to determine exactly not only the frequency of each component but also the variations of its intensity with time, i.e. to be able to follow accurately the growth and disappearance of each component. The accuracy which can be attained here, however, is fundamentally limited by a kind of “relation of uncertainty”. The frequency of a sinusoidal vibration can only be determined precisely when the vibration lasts for an infinite time. With shorter duration it is impossible to speak of one definite frequency of the vibration, but it must be ascribed to a spectrum of finite width, as appears from the theory of Fourier integrals. This width becomes greater the shorter the vibration lasts. Therefore the more rapidly a spectrum varies, the less sharp will the frequencies be determined.

In measuring, of course, only a section of finite duration of the vibration in question can be considered, so that even with an infinitely long vibration we cannot determine the spectrum perfectly sharply. But, moreover, the time interval must expressly be chosen short when rapid variations of a spectrum are to be observed; the organ reacting to the vibration (ear, filter or general measuring instrument) must “forget” again the preceding effects quickly enough. From a consideration of band filters it is clear how in this case the antagonism mentioned occurs between the accuracy of the measurement of the frequency on the one hand (resolving power) and the recording speed on the other.

As indicated in fig. 1, the filters consist of single \( L-C \) circuits which are so damped by a resistance \( R \) in parallel that at resonance the impedance is 20 000 ohms. This is also the value of the preceding resistance \( R_1 \). The ratio \( a \) between output and input amplitude (transmission factor) of a vibration of any given frequency \( f \) with such a filter is

\[
a = \frac{1}{2 + jq \left( \frac{f}{f_0} \right)^2 - 1}
\]

In this expression \( f_0 = 1/2\pi\sqrt{LC} \) is the resonance frequency of the filter and \( q = R/2\pi f_0 L \) is the so-called quality factor of the circuit (the loss resistance of the self-induction \( L \) is accounted for in the resistance \( R \)). The larger \( q \), the steeper the resonance curve given by (1) falls away on each side of \( f_0 \) (fig. 3).

Let us suppose that a sinusoidal voltage is suddenly applied to the input of such a filter. In addition to a forced oscillation whose intensity can be calculated from (1), there then occurs a free oscillation with the frequency \( f_0 \) which gradually dies out according to

\[
-\frac{1}{e^{t/RC}}
\]

The time \( \tau = 2RC \) after which the intensity of the free oscillation has fallen by a factor \( 1/e \) is called the decay time of the filter. It will obviously be useless to measure the output voltage with a varying input A.C. voltage of the filter at intervals which are not at least equal to \( \tau \) (preferably still much longer). Since the following is true:
\[ \tau = 2RC = 2 \frac{R}{2\pi f_0 L} \cdot 2\pi f_0 = \frac{q}{\pi f_0} \quad (2) \]

the more rapidly it is desired to record, i.e. the smaller the decay time \( \tau \) is to be made, the smaller the quality factor \( q \) of the filters must be made, i.e. the flatter the resonance curves of the filters must be. Therefore the less accurately is the frequency of the transmitted signal determined.

Fig. 3. Resonance curve \( a(f) \) of the band filter for different values of the quality factor \( q \).

From equation (2) it may at the same time be seen that at a given value of \( q \) the band filters have a shorter decay time for the high frequencies than for the low. For the high frequencies therefore a more rapid recording is possible than for the low frequencies. Since, when the whole spectrum is recorded with a single film, the same recording speed must be used for all frequencies, it would seem reasonable to construct all the filters with the same decay time, so that in the case of the filters for high frequencies it is possible to work with a high value of \( q \), i.e. a sharp resonance curve and correspondingly great resolving power. We have not done this, however, because the sharper the resonance curves of the filters are made, the greater the chance that a Fourier component whose frequency lies just between the resonance frequencies of two neighbouring filters will remain unobserved. Since this must be avoided as far as possible the intervening regions between the filters must be made smaller, i.e. with a sharper resonance curve a larger number of filters is necessary.

In order not to be compelled to use too large a number of filters we chose the relatively small value of 32 for the quantity \( q \). If we use a recording speed of 20 pictures per second, which will be sufficient for most investigations, the decay time will only be longer than the recording time for the filters with \( f_0 < 32.23 / \pi \approx 200 \text{ c/s} \); in the case of the lowest frequency with which we are concerned \( f_0 = 90 \text{ c/s} \), \( \tau = 1/9 \text{ s} \), so that rapid variations of the components in this neighbourhood will not be quite adequately brought out on the film. In practice, however, this is not a serious objection.

With the 79 filters already mentioned, whose resonance frequencies are distributed according to a geometrical series over the frequency range from 90 to 8000 c/s which is of importance for speech \(^2\) (there are then 12 filters to an octave and the frequency relation between two adjacent filters is therefore the same as between two adjacent notes on the piano (about 1.06)), it may be calculated according to equation (1) that the transmission factor of the filters and those frequencies \( f_m \) where two adjacent frequency curves intersect \( (f_m / f_0 = 1.03, \text{ see fig. 4}) \), amounts to \( a = 1/2 \sqrt{2} \). At resonance \( (f = f_0) \) \( a = 1/2 \). Thus if a component lies just between two filters, both of these filters give a certain output signal from which the intensity of the component can be found by adding the squares. But when a component falls exactly at the resonance peak of a filter, several neighbouring filters to the left and right also give an output signal which in magnitude is 44, 22, 14, \( \ldots \) per cent, etc. respectively, of the centre filter. This case — "excitation curve" in the case of a truly sinusoidal input voltage resonating with a filter—is illustrated in fig. 2. The spectrogram has a character similar to that of the excitation curve of the basilar membrane of the human ear, which can be represented as consisting of a similar series of damped resonators \(^3\).

Fig. 4. With the value \( q = 32 \) and a distance of 1/12 of an octave between the resonance frequencies of successive band filters, the transmission factor \( a_m \) at the point of intersection of two adjacent resonance curves is \( 1/\sqrt{2} \) times as large as that at resonance (\( a_o \)). \(^{2}\) See for example R. Vermuelen, Octaves and decibels, Philips Techn. Rev. 2, 47, 1937, graph on p. 49.

\(^3\) See J. F. Schouten, The perception of pitch, Philips Techn. Rev. 5, 286, 1940, where on p. 290 such a model for the basilar membrane is discussed.
If the construction of the filters is based on a recording speed of 20 times per second, it is of course desirable that there should be no other elements in the apparatus which prevent recording at such a speed. Critical points in this respect are the outputs of the modulator, the input of the modulator and the electron ray tube. The rectified voltages are taken from a resistance $R_2$ inserted in every rectifier circuit, through which resistance the rectified currents flow, while a condenser $C_1$, in parallel with $R_2$, serves to smooth the voltage to the required degree. The time constant $R_2C_1$ of this circuit must be so small that the condenser $C_1$ can be practically discharged between two recordings. $R_2C_1$ cannot be made indefinitely small, since $R_2$ must be large enough to obtain a high resolving power. With a length of the spectrogram of 0.00011 sec, which is small enough in order not to affect the band filter by too high a consumption of current, and also since $C_2$ must be large enough to obtain the desired smoothing effect. Nevertheless, it was found possible to give the product $R_2C_1$ exactly the value $1/20$ sec, so that the recording of 20 pictures per second was not hindered by this. In fact, apart from the impossibility of making $R_2C_1$ indefinitely small, a much smaller value of $R_2C_1$ would not even be desirable, since then the $R_2C_1$ would not be as small as that of a short-lived change in the signal, which would disappear again so quickly that there would be a great chance that it would be unnoticed with a time interval of $1/20$ sec between successive recordings.

Similar considerations are also valid for the input of the modulator. As soon as the switch $E_1$ makes contact with a certain lamella of the collector, the input capacity $C_1$ of the modulator is loaded with the resistance $R_2$ in the connections between rectifier and collector. It was now found necessary to make $C_2$ large enough to limit the effects of the switching impulses occurring; at the same time, however, the charging time $C_1R_2$ of the modulator input must be made extremely small. For the recording of 20 spectrograms, each consisting of about 80 measured points, only $1/20$ sec is available for the scanning of each lamella; since there must also be sufficient space between the lamellae of the collector, the actual time of contact is only about half as long. In order to record the correct voltage value, the time $C_1R_2$ must therefore be chosen appreciably smaller than 0.0003 sec. With the values chosen of $R_2 = 0.25$ M$\Omega$ and $C_1 = 500$ $\mu$F, $R_2C_1$ became 0.00011 sec, which is small enough.

Finally there is the electron ray oscillograph. For tracing one vertical line in the spectrum, with the desired recording speed of 20 pictures per second, 0.0003 sec is available according to the above. With a maximum length of the lines of for instance 4 cm the tracing speed of the oscillograph therefore amounts to 4 cm/0.0003 sec $= 120$ m/s. Although in ordinary cases, for instance with the electron ray oscillograph 3122, measurements can easily be made with such a tracing speed, in this case this is not immediately true, since we must also pay attention to the resolving power. With a length of the spectrogram of about 8 cm the width of each of the 79 vertical lines may not be greater than $1/4$ mm in order that they may not overlap. With the required very fine fluorescent spot the necessary light intensity for exposures with the tracing speed mentioned could only be obtained by the use of an electron ray tube with post-acceleration $^3$.

Relation between the length of line on the screen and the amplitude of the Fourier component

The rectified output voltage of the band filters can be modulated on the “carrier wave” of 50 kc/s in different ways. There are, however, two requirements: firstly that with a modulating voltage of $v_m = 0$ the amplitude $V_d$ of the carrier wave should also become equal to zero; secondly that the relation between $V_d$ and $v_m$ should have a certain character. By itself a linear relation would seem most obvious. There is, however, the objection that because of the great differences occurring in the intensity of the sound, the weak Fourier components would quickly become insignificant compared with the strong ones. If $V_d$ increases less than proportionally with $v_m$, as for instance in the case of a logarithmic relation, then all intensities are dealt with equally, but there is the disadvantage that a peak in the Fourier spectrum is even more flattened than already results from the damping of the band filters (see fig. 2). An intermediate way, in which the relation between $V_d$ and $v_m$ begins approximately linear and then curves off towards a sort of logarithmic relation, is the most suitable.

Fig. 5. Connections of the modulator. The anode alternating currents of the two valves which are excited by the carrier wave voltage of the generator $G$ are adjusted to equal magnitude with the resistance $R$, so that the band filter $BF$ receives no input voltage. By means of a D.C. voltage $v_m$ which is fed from $E_1$ to the control grid of one valve, the equilibrium is disturbed and a carrier wave of a certain amplitude is therefore passed on to the oscillograph $O$.

Such a relation is now realized by the modulator connections shown in fig. 5, in which at the same time the first condition mentioned, namely that $V_d = 0$ when $v_m = 0$, is also satisfied. The voltage of 50 kc/s of the generator $G$ is fed in the same phase to the control grids of two valves in push-pull connection working with inverse feed-back. Each valve furnishes a certain anode alternating current, but due to the compensation of the two currents

$^3$) Actually, of course, the fluorescent spot moves 16 times as fast, since it makes a vibration with a frequency of 50 kc/s along the line. It comes to practically the same thing, however, for photography, whether the spot describes the line once with a given velocity or 16 times with a velocity 16 times as great.

$^3$) See J. de Gier, An electron ray tube with post-acceleration, Philips Techn. Rev. 5, 245, 1940.
in the output transformer — which compensation can be precisely adjusted by the regulatory resistance $R$ — no carrier wave is ordinarily transmitted to the band filter $BF$ ($V_d = 0$).

If by means of the rotating switch $E_1$ a D.C. voltage $v_m$ is applied to the control grid of one of the two valves, the operating point of this is displaced on its characteristic to a point with a steeper slope, the anode alternating current of this valve becomes larger, the compensation is sufficient and the excess is transmitted as output signal (with 50 kc/s) to the band filter. If the matter is considered more carefully, in the case of the valve with inverse feed-back the relation between the anode alternating current $i_a$ and the grid A.C. voltage $v_g$ is given by

$$i_a = \frac{s}{1 + sR} v_g,$$

where $s$ stands for the slope and $R$ for the resistance in the cathode connection which effects the inverse feed-back. The slope $s$ is chiefly determined by the grid D.C. voltage $v_m$ and (at least in the beginning) increases proportionally with $v_m$. The expression $s/(1 + sR)$, however, increases less rapidly than in proportion to $s$ and finally approaches the constant value $1/R$. From the cooperation of these two functions, upon suitable choice of the point on the characteristic at which the valve operates with $v_m = 0$, exactly the desired relation between $V_d$ and $v_m$ is obtained, approximately linear at first and later curving.

Actually it is not a question of the relation between $V_d$ and $v_m$, but of that between the length of line on the fluorescent screen and the intensity of the corresponding Fourier component. Now the whole apparatus is about linear, thanks in part to the use of diodes (EA 50) in the rectifiers. Because of the large number of rectifiers (79) it would have seemed preferable to use blocking-layer valves, but owing to the linearity mentioned this was not done (moreover, the diode has the advantage of being less sensitive to overloading). Since the relation between the length of line on the screen and the deflecting voltage $V_d$ on the cathode ray tube is also satisfactorily linear, the relation between the length of line and the input voltage of the whole apparatus has practically the same form as the relation between $V_d$ and $v_m$. This relation as determined by measurement is reproduced in fig. 6.

The recording of the spectrograms

In order to make an “instantaneous exposure” of the changing sound the spectrum on the screen of the oscillograph may be photographed with a camera whose shutter must be opened for exactly one revolution of the rotating switches $E_1$, $E_2$. If it is desired to make a series of such exposures, and thus to “film” the sound, the film should be shifted by the height of one picture after each revolution of the switches. With the construction of the collectors here chosen, in which only as much space is allowed between the first and the last of the 79 contacts as is necessary to prevent the occurrence of short-circuiting of the source of voltage for the horizontal deflection of the spot (see fig. 1), the time available for the shift is too short, so that in each exposure the beginning of the spectrum would have to be missed or every other
Fig. 8. In the upper part of the wheeled cabinet may be seen a large number of boxes in which the 79 band filters are housed. Underneath are the rows of switches $S$ for listening to the filtered-out spectral parts of the sound being investigated. On top of the cabinet on the left is the electron ray oscillograph (GM 3152), on the right the film camera.

A complete revolution of the switches would have to be omitted. A better and simpler solution was therefore to allow the film to move continuously in the direction of the vertical spectrum lines with permanently open camera shutter. When this is done of course a uniform movement in the direction of the film is superposed on the motion of the fluorescent spot, so that the spectra traced are pulled into an oblique position (see fig. 7), while, moreover, the ends of each spectral line are slightly less sharp. These features, however, constitute no disadvantage in the analysis of the diagrams.

Photographing with continuously moving film sets a sharp limit on the time of phosphorescence of the fluorescent screen. While with a discontinuously shifted film the permissible time of phosphorescence is determined by the recording time of a complete spectrogram (1/20 sec), with a continuously moving film the only permissible time of phosphorescence is that which is determined by the lack of sharpness to be tolerated at the ends of the lines, and which is therefore of the same order as the time necessary for tracing one vertical line in the spectrum (i.e. 1/1600 sec). In the case of the screen of the electron ray tube used by us the phosphorescence time was found to be sufficiently short.

Fig. 9. On the bottom of the wheeled cabinet may be seen the motor which drives the two collectors $E_1$ and $E_2$ and, via a tooth-wheel transmission and a vertical axis, the film camera. In order to prevent acoustic disturbances the motor is mounted on a thick rubber plate. The collectors consist of a ring of high-frequency "Philite" in which a ring of conical copper pins is driven; with this construction wear is restricted and at the same time with little upkeep only a very low transition resistance at the contacts is obtained. In addition, on the bottom of the cabinet are the supply apparatus for the amplifiers, the modulator, the plates for horizontal deflection of the oscillograph, etc. Above are the 79 diode rectifiers. On top of the cabinet may be seen in the foreground the coupling in the driving shaft of the film camera, which may be connected and disconnected. The lead on top of the electron ray oscillograph supplies the post-acceleration voltage.
During one revolution of the rotating switches the film movement must be such that even with the largest amplitudes occurring successive spectra must not overlap. In order to ensure this once and for all at different speeds of revolution of the switches, the camera is driven, via a tooth-wheel transmission and a flexible shaft, by the same motor (a D.C. motor with variable speed) to the axle of which the switch arms are fastened. The camera is set in motion or stopped by means of a simple coupling mechanism.

As described in detail above, a maximum recording speed of 20 spectrograms per second can be attained. For many purposes a lower speed, for instance 10 pictures per second (thus half the speed of revolution of the switches) will suffice, and the recording speed then corresponds to the longest decay time of the filters occurring (1/9 sec with the filter for 90 c/s).

As a conclusion to this description, in figs. 8 and 9 two photographs of the apparatus constructed are given. Several structural details are pointed out in the text below the figures.

**Several results obtained with the apparatus**

In fig. 10 a number of filmed spectrograms are given which were made with the apparatus. They are the spectra of several sounds of speech, namely the vowels a, e, i, o, u, recorded at a speed of 16 spectra per second. The frequency scale is the same as that of fig. 7.

While it is generally known that the consonants are characterized for the most part only by certain introductory and transitional vibrations (most of them cannot be "held"), it is clearly apparent from the sections of film reproduced that vowels also fail to represent a completely periodic vibration. Each vowel is in fact characterized by a series of harmonic components which lie in certain formative regions, relatively independent of the pitch of the fundamental tone of the sound.

The fundamental tone, which may be quite different for male and female voices, in the case of the subject of the spectrograms in fig. 10 is about 250 c/s (peak on extreme left, band filter No. 10) for the e, i, o and u. In the case of the a during its pronunciation the fundamental tone is seen to rise slightly from 175 to 225 c/s, and in the case of oo it falls slightly from 250 to 175 c/s. Such variations in the fundamental tone are of little importance for the separate sounds of speech, but they do not...
play an important part in the capacity of the language for expression. This is an important point for investigation in phonetics. If the position and intensity of the other peaks in the spectrum are measured, the following data (table I) are found for instance for the a (third spectrum from the top) (applying the calibration curve of fig. 6).

Table I

<table>
<thead>
<tr>
<th>Filter No.</th>
<th>Frequency c/s</th>
<th>Voltage mV</th>
</tr>
</thead>
<tbody>
<tr>
<td>16</td>
<td>210</td>
<td>8</td>
</tr>
<tr>
<td>28</td>
<td>421</td>
<td>14</td>
</tr>
<tr>
<td>35</td>
<td>630</td>
<td>19.5</td>
</tr>
<tr>
<td>44</td>
<td>1060</td>
<td>75</td>
</tr>
<tr>
<td>56</td>
<td>2120</td>
<td>40</td>
</tr>
<tr>
<td>64</td>
<td>3360</td>
<td>30.5</td>
</tr>
</tbody>
</table>

As may be expected theoretically 4), these frequencies are multiples of the fundamental tone (210 c/s). The strongest overtone is found at 1060 c/s, in which region lies the most important formative of the a. The weaker maxima usually occurring as well, which in this case for instance occur at 2120 and 3360 c/s, are less essential to the character of the vowel.

Upon investigation of a series of different voices we found for the most important formative of each of the vowels mentioned the frequency regions given in table II, column A. For the sake of comparison the values given by

Table II

Most important formative regions of six vowels, according to measurements with the apparatus described. For the main formatives our own measurements (column A) are to be compared with values given in the literature 1) (column B).

<table>
<thead>
<tr>
<th>Vowel</th>
<th>Subordinate formative c/s</th>
<th>Main formative c/s</th>
<th>A</th>
<th>B</th>
</tr>
</thead>
<tbody>
<tr>
<td>oe</td>
<td>150-300</td>
<td>400-500</td>
<td>366-548</td>
<td></td>
</tr>
<tr>
<td>o</td>
<td></td>
<td>950-1498</td>
<td>548-1504</td>
<td></td>
</tr>
<tr>
<td>a</td>
<td>150-300</td>
<td>1888-2980</td>
<td>1543-2926</td>
<td></td>
</tr>
<tr>
<td>e</td>
<td>300-600</td>
<td>2378-3786</td>
<td>2569-2762</td>
<td></td>
</tr>
<tr>
<td>i</td>
<td>150-300</td>
<td>2523-3786</td>
<td>2192-3687</td>
<td></td>
</tr>
</tbody>
</table>

5) The spectra of the vowels are formed due to the fact that certain harmonics of the vocal chord vibration, which contains the fundamental tone with a large number of harmonics, are amplified by resonance of the cavity of mouth, throat and nose. See also J. de B o e r and K. de B o e r, The Laryngophone, Philips Techn. Rev. 5, 6, 1940.