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TECHNICAL ASPECTS OF SOUND

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TECHNICAL ASPECTS
of
SOUND

edited by

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VOLUME I

SONIC RANGE AND AIRBORNE SOUND



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PREFACE

Fifty years ago he who wished to study acoustics seriously had no choice of text-book, it was RAYLEIGH or nothing. To RAYLEIGH's famous two volumes—still a standard work which remains surprisingly fresh on the fundamental aspects of sound—have since been added quite a number of general works on the subject together with a few on special applications. There remains however a lack of a 'Handbook' (at least, in the English language) to cover all the technical aspects of the subject and this hiatus it is the object of the present text to fill, so that both the research worker, industrialist and advanced student shall be able to survey, at a fairly advanced level, the whole of the applications which this rapidly advancing branch of physics has reveiced in the past few decades.

The aim has been to find experts, each in his special field, to write the several parts into which the subject matter has been divided. To this, I have added an introduction to cover the basic theory and to introduce the symbols, which are also listed (pp. XVII and XVIII). The publishers and I are indeed grateful to those who have collaborated in this venture by making written contributions.

It was intended originally to present this text in a single volume, but as it grew, it became apparent that two volumes would be necessary. This, the first, deals with the sonic range and, in the main, with airborne sound. The second, which will be issued in 1954, will cover the ultrasonic range and applications thereof, including underwater propagation.

Newcastle-on-Tyne, April 1953 E. G. RICHARDSON

ERRATA

p. 354 in the heading of § 4, for 'Electrical' read 'Acoustical'.
p. 383 line 9 from bottom, for 3% read 0.3%.
p. 384 line 2 from bottom, for *a* read ξ.
line 1 from bottom, the dot on the second ξ should be deleted.
p. 385 line 6, for 'magnetic' read 'electromagnetic'.
p. 390 line 6 from bottom, for 100 read 200.
p. 391 the upper half of Fig. 210 should be deleted.
p. 400 the sentence on lines 2 and 3 should appear on p. 388 after line 2.
p. 402 in Fig. 222, for 1% read 1°
p. 403 line 7 from bottom, for 'distortion meter' read 'volume indicator'.
p. 404 line 9 from bottom, for 'on the light source' read 'the film on the slit'.
line 1 from bottom, for 'in recording' read 'for densiometry'.
p. 407 line 9, for 16 read 18.3.
p. 408 the legend to Fig. 228 should be completed by:
(a) both slits in rest; *(b)* sound slit works; *(c)* both slits work.
p. 419 line 7, for kc read c.
p. 424 line 14, for KONES read KRONES.
line 20, for SEIBARTH read SEIBERTH.
line 6 from bottom, for SCHEDDER read SCHODDER.

Although the names 'gramophone' and 'magnetophone' are registered trade names they have been used here in a general sense.

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LIST OF SYMBOLS*

(Brackets indicate occasional use)

a	absorption coefficient, (radius)
A	absorption units, (arbitrary constant)
b	breadth, (Newtonian velocity of sound)
B	bending moment stiffness, (arbitrary constant)
c	velocity of sound, (concentration)
C	specific heat, capacitance
d (D)	diameter, thickness
D	transmission loss
e	base of natural logs, (elasticity)
E	Young's modulus, (energy density)
$f = c/s$	frequency
F	force
g	acceleration of gravity, general coefficient
G	rigidity
h	height
i	ordinal number
I	intensity
j	ordinal number
k	thermal conductivity
K	stiffness coefficient, bulk modulus
l	length
L	inertance
m, M	mass, (mutual reactance)
n	a/λ or $2\pi a/\lambda$
O	interval
p	pressure, (modulus of complex quantity)
P	power, (porosity)
q	argument of complex quantity
r	reflection coefficient, spherical co-ordinate, specific air resistance
R	radial distance, resistance
s	condensation, stiffness coefficient (space factor)
S	area, surface
t	time, transmission coefficient
T	time period, kinetic energy

* The list comprises mechanical usage only and does not include the usual electrical symbols, as used specially in Division Five.

u, v, w velocity components of fluid or surface
v volume
V potential energy
w weight
W energy, (characteristic impedance)
x, y, z co-ordinates
X volume displacement, reactance
$\mathfrak{z}$ specific impedance
Z acoustic impedance

α attenuation constant, (coefficient of expansion)
β phase constant
γ propagation constant
Γ $\gamma/i\omega$
δ phase angle
Δ log dec.
ε loss factor (factor in conical pipes)
ζ $\sqrt{\frac{\omega}{\nu}} \cdot R$
η viscosity coefficient, (fractional change of temperature)
θ temperature, spherical co-ordinate, angle of incidence
ι $\sqrt{-1}$
κ acoustic conductivity, (radius of gyration)
μ damping factor, (micron), (exponent of horn), (absorption per wave-length)
ν kinematic viscosity
ν' thermal diffusivity
ξ particle displacement
π circular constant, (pressure ratio)
ρ density
$\rho c = W$ characteristic impedance
σ coupling coefficient, Poisson's ratio
τ time of reverberation (of relaxation)
φ velocity potential
ψ stream function } (factors of a complex quantity), velocity ratio
χ ratio of specific heats of gas
ω, Ω pulsatance, angular velocity

CHAPTER 1

INTRODUCTION

E. G. RICHARDSON
Physics Department, King's College, University of Durham, Newcastle-on-Tyne (England)

The various divisions of sound treated in this book have a common background of theoretical analysis. It is the object of this introduction to expound this acoustical syntax in terms of the symbols which will be used through the book.

§ 1. Vibrations of a Particle

As the simplest case consider a particle of mass m and stiffness s executing small vibrations without sustained forcing under the condition that the sum of its potential energy (T) and kinetic energy (V) is constant, or, if ξ represents the displacement;

$$\tfrac{1}{2}m\dot{\xi}^2 + \tfrac{1}{2}s\xi^2 = \text{constant} \tag{1}$$

of which the solution is an S.H.M. at the *natural* time-period $2\pi\sqrt{m/s}$ or pulsatance $\omega = \sqrt{s/m}$.

Incidentally the kinetic energy, taken to represent the intensity of the sound, will be proportional to $\dot{\xi}^2$ or to $\xi^2\,\omega^2$.

If we introduce damping, we have to add a term proportional to $\dot{\xi}$ (still supposing ξ—and so $\dot{\xi}$—small)

Then, from (1),

$$m\ddot{\xi} + \mu\dot{\xi} + s\xi = 0 \tag{2}$$

of which the solution is

$$\xi = \xi_0 \exp.-(\alpha \pm \iota\omega)t \tag{3}$$

with $\alpha = \mu/2m$; $\omega^2 = s/m - \mu^2/4m^2$.

§ 2. Forced Vibrations; Resonance

The above treatment covers the free or natural vibrations. Suppose now an inexorable force $Fe^{\iota\omega t}$ acts on the particle. We write;–

$$m\ddot{\xi} + \mu\dot{\xi} + s\xi = Fe^{\iota\omega t} \tag{4}$$

and try a solution;

$$\xi = \xi_0 \sin(\omega t - \delta)$$

for the forced vibration.
Substituting in (4) we derive for the response:

$$\xi_0/F = [\mu^2\omega^2 + (s - m\omega^2)^2]^{-\frac{1}{2}} \tag{5}$$

and

$$\tan\delta = \frac{\mu\omega}{s - m\omega^2}$$

$$\text{or } \cos\delta = (s - m\omega^2)/[\mu^2\omega^2 + (s - m\omega^2)^2]^{\frac{1}{2}} \tag{6}$$

If the effect of viscosity is small, the response is a maximum $(\mu\sqrt{s/m})^{-1}$ when $\omega^2 = s/m = \omega_0^2$, say, and cos δ (the power factor) is 1.

At the same time the phase of the particle, represented by δ, will lag $\pi/2$ behind the forcing vibration.

When a system is capable of executing a number of simple harmonic resonant vibrations (the fundamental and partial vibrations, *vide infra*) and the damping coefficient α remains constant, the response represented by (5) will change inversely with ω.

The 'sharpness of the resonance' is inversely as μ. This factor called quality factor (Q) is defined:

$$Q = \omega_0/(\omega_2 - \omega_1) \tag{7}$$

where ω_2 and ω_1 are the pulsatances above and below resonance at which the power has dropped to one half.
Since the average power is the mean value of $\xi_0^2 \sin\omega t \cos(\omega t - \delta)$ which from (5) and (6) is $F^2\mu/2[\mu^2\omega^2 + (s - m\omega^2)^2]$ this average will be half the maximum when $\omega m - s/\omega = \pm\mu$ or when the roots of the equation satisfy $\omega_2 - \omega_1 = \mu/m$ and $Q = \sqrt{sm}/\mu$.

The solutions (5) and (6) correspond to the steady state forcing. Before this is established, 'transients' corresponding to (3) will subsist.

§ 3. Coupled Vibrations

So far we have considered what RAYLEIGH called an inexorable forcing system, but in practice the driven will react on the driver. Together, they form a coupled system. When the two components have comparable mass the roles of driver and driven may be periodically interchanged as kinetic energy is bandied between them. The coupling coefficient σ may be regarded as determining the rate of this interchange or of the magnitude of the 'feed-back'.

The effect of one system on the other is that of an added force due to the mass-acceleration of the other. The equations of the two systems (neglecting damping which considerably complicates the analysis) are

$$\left.\begin{aligned} m_1\ddot{\xi}_1 + M\ddot{\xi}_2 + s_1\xi_1 &= 0 \\ m_2\ddot{\xi}_2 + M\ddot{\xi}_1 + s_2\xi_2 &= 0 \end{aligned}\right\} \tag{8}$$

where the second terms represent the mutual reaction. Differentiating twice;

$$m_1\ddddot{\xi}_1 + M\ddddot{\xi}_2 + s_1\ddot{\xi}_1 = 0 \tag{9a}$$

$$m_2\ddddot{\xi}_2 + M\ddddot{\xi}_1 + s_2\ddot{\xi}_2 = 0 \tag{9b}$$

and substituting in (9a) the value of $\ddddot{\xi}_2$ obtained from (9b) we get;

$$m_1\ddddot{\xi}_1 - \frac{M^2}{m_2}\ddddot{\xi}_1 - \frac{s_2 M}{m_2}\ddot{\xi}_2 + s_1\ddot{\xi}_1 = 0$$

again substituting in this equation the value of $\ddot{\xi}_2$ obtained from (8) and dividing out by m_1 we get;

$$(1 - \frac{M^2}{m_1 m_2})\ddddot{\xi}_1 + (\frac{s_1}{m_1} + \frac{s_2}{m_2})\ddot{\xi}_1 + \frac{s_1 s_2}{m_1 m_2}\xi_1 = 0 \tag{10}$$

with the same equation for ξ_2. Now $s_1/m_1 = \omega_1^2$ and $s_2/m_2 = \omega_2^2$ where ω_1 and ω_2 are the natural pulsatances of the respective systems removed from mutual influence. Further, if the combined systems follow a S.H.M. of pulsatance Ω

$$-\ddot{\xi}_1 = \Omega^2\xi_1 \; ; \; \ddddot{\xi}_1 = \Omega^4\xi_1$$

Putting finally $\frac{M^2}{m_1 m_2} = \sigma^2$, (10) becomes, as an equation for Ω;

$$(1 - \sigma^2)\Omega^4 - (\omega_1^2 + \omega_2^2)\Omega^2 + \omega_1^2\omega_1^2 = 0 \tag{11}$$

σ is called the 'coefficient of coupling'.

This equation has in general two roots, showing that the complex system has two natural frequencies. In the special case of $\sigma = 0$, the roots of Ω are ω_1 and ω_2; otherwise the effect of coupling is to separate the roots so that, assuming ω_1 greater than ω_2, the roots of Ω become respectively greater than ω_1 and less than ω_2 as the coupling gets closer. If $\omega_1 = \omega_2 = \omega$

$$\Omega = \frac{\omega}{\sqrt{(1 \pm \sigma)}}$$

This property is utilized in what is called 'multiple resonance', by coupling two resonators in order to get a system having two resonance peaks, which are in general, respectively above and below the upper and lower resonance peaks of the separate systems.

§ 4. Double Forcing; Combination Tones

If two systems of pulsatance ω_1 and ω_2 respectively act on a particle then—in the absence of damping— the latter will execute a motion in which the difference, $\omega_1 - \omega_2$, is prominent. If the particle has a linear response, such as that characteristic of the particle we have so far discussed, this difference tone and the forcing primaries are the only ones found in the response, but if restoring forces proportional to higher powers of the displacement supervene, other combination tones will be found. The condition of linearity is, of course, most likely to be broken if the amplitude is large, that is, if the drivers are powerful. For instance, we could write

$$m\ddot{\xi}^2 + s'\xi + s''\xi^2 = \xi'_0 \exp.\, i\omega_1 t + \xi''_0 \exp.\, (i\omega_2 t + \delta) \tag{12}$$

and then we should arrive at a solution containing terms in $j\omega_1 \pm j'\omega_2$ where j and j' are integers. This includes the summation tone $\omega_1 + \omega_2$. The response falls off with the order of j and j' so that only the lower orders are usually important.

Note that any terms involving even powers of ξ—such as the third term in (12)—introduce rectifying action since they do not change sign with the displacement.

§ 5. Fourier Theorem

The theorem turns on the possibility of representing a complex periodic function as a series of sines and cosines of continually increasing order, thus;-

$$\begin{aligned} f(x) = \tfrac{1}{2}a_0 + a_1 \sin x + a_2 \sin 2x + a_3 \sin 3x \ldots . \\ + b_1 \cos x + b_2 \cos 2x + b_3 \cos 3x \ldots . \end{aligned} \tag{13}$$

If ωt is put in place of x these terms will be recognised as simple harmonic motions, so that the theorem states that the motion of a particle can be represented as a superposed series of S.H.M.'s whose frequencies cover the series of natural numbers. Such a series is called a 'harmonic series' and the partial motions represented by the terms of (13) are called 'harmonics' of the first or 'fundamental'. The usefulness of such a separation into partial vibrations lies in the possibility of determining the values of the amplitudes of the components making up a given periodic function. The general coefficients as derived by Fourier are given by the series;

$$a_j = \frac{1}{\pi}\int_{-\pi}^{+\pi} f(x) \sin jx \, dx; \qquad b_j = \frac{1}{\pi}\int_{-\pi}^{+\pi} f(x) \cos jx \, dx$$

It usually is practicable to determine a_j and b_j by approximate integration.

The FOURIER series can also be written as a series of sines alone, representing components in different phases, thus

$$f(x) = \tfrac{1}{2}\, a_0 + a_1 \sin (x + \delta_1) + a_2 \sin (2x + \delta_2) \ldots .$$

representing a series of harmonics, all having different phase-angles of lead.

To obviate the tedious labour of calculating the coefficients required to complete the FOURIER analysis of a given curve, a number of instruments have been devised to perform the operation (*cf.* Chap. 17, § 4).

§ 6. Relaxation Oscillations; Sub-harmonics

If the driver is not to be represented by a periodic function but rather by an occasional impulse lasting a short time, the amplitude of the particle will decay between impulses. These are called relaxation

oscillations. Naturally the impulses must be so timed that the boost is given in the correct phase to increase, temporarily, the oscillations. If the maintaining impulse occurs at frequencies somewhat above or below that of the driven system, the amplitude falls on one side of resonance and rises on the other so that there is no peak as the frequency of the driver is lowered through resonance. The simple theory of forced vibrations fails here, so that terms involving ξ^2 must be introduced into (2) and (4).

Now suppose that the damping μ in (4) is a negative quantity so that the solution involves a factor $e^{+\alpha t}$ and suppose that the amplitude is limited by an additional damping factor which grows with it. VAN DER POL chooses an equation of the type;

$$\ddot{\xi} = 2\alpha(1 - \xi^2)\,\dot{\xi} - \omega^2\xi$$

in which the net damping is negative until $\xi^2 = 1$. He shows that if α is not too small, the square of the frequency is determined by the quantity $\omega^2/2\alpha = s/\mu$. The period $2\pi\sqrt{\mu/s}$ is known as the 'time of relaxation' of the system since it determines the rate at which the stress disappears as it is taken out in the resulting shear.

§ 7. Wave-Motion

Confining ourselves to S.H.M., a progressive wave along the axis of x will be represented by

$$\dot{\xi} = \dot{\xi}_0 \text{ exp. } (\alpha - i\beta)\,x. \text{ exp. } i\omega t = \dot{\xi}_0 \text{ exp. } (\gamma x + i\omega t) \qquad (15)$$

in which α is the attenuation and β the phase constant. The wavelength λ is $2\pi/\beta$ and γ is called the propagation constant. The velocity c of the wave will then be ω/β. A retrogressive wave will be represented by the same formula save that $+\beta$ will replace $-\beta$.

A stationary wave can be represented by the sum of two unattenuated progressive waves of equal amplitude and frequency, threading the system in opposite directions, thus;

$$\dot{\xi} = \dot{\xi}_1 + \dot{\xi}_2 = \dot{\xi}_0 \text{ (exp. } -i\beta x + \text{exp. } i\beta x) \text{ exp. } i\omega t = 2\dot{\xi}_0 \cos \beta x \text{ exp. } i\omega t \qquad (16)$$

with characteristic nodes (places of no motion) at $x = 2(j+1)\,\lambda/4$ and antinodes (maximum amplitude) at $x = j\,\lambda/2$, j being integral.
Since the natural modes of vibration of an extended system are often

characterised by 'end conditions' which locate a node or antinode at or near each termination the possible natural wave-lengths are in turn determined by these possible modes together with the appropriate velocity of sound in the system at this wave-length.

Sometimes, a set of pseudo-stationary waves may be set up, under the alternative conditions;

(a) that the component waves are attenuated. If at $x = l$, $\dot{\xi}_0 = 0$, then from (15);

$$\dot{\xi} = 2\dot{\xi}_0 \exp.\,(\alpha + i\beta)l.\,\sinh\{(\alpha + i\beta)(x - l)\}\exp.\,i\omega t$$

whence, putting $A = \dot{\xi}_0 \exp.\,\alpha l$ and omitting the time factor,

$$|\dot{\xi}|^2 = 2A^2[\cosh\{2\alpha(x - l)\} - \cos\{2\beta(x - l)\}] \tag{17}$$

(b) that the reflection coefficient (of amplitude) at one end of the system is not unity but the fraction r. Then the two components represented by $\dot{\xi}_1$ and $\dot{\xi}_2$ in (16) are to be replaced by

$$\left.\begin{aligned} \dot{\xi}_1 &= \frac{\exp.\{-(\alpha + i\beta)x\}}{1 - r.\exp.\{-(\alpha + i\beta)2l\}} \cdot \exp.\,i\omega t \\ \dot{\xi}_2 &= \frac{(1 - r)\exp.\{-(\alpha + i\beta)(2l - x)\}}{1 - r.\exp.\{-(\alpha + i\beta)2l\}} \exp.\,i\omega t \end{aligned}\right\} \tag{18}$$

In other circumstances combinations of conditions (a) and (b) must be considered.

§ 8. Longitudinal Plane Waves

This type of motion arises when a piston source, many wave-lengths wide, radiates into a fluid medium. If K is the appropriate elasticity modulus, ρ the density and the increase of pressure δp produces an increase of density $\delta\rho$, we may write $K = \rho\delta p/\delta\rho$ or $\delta p/s$ where s is called the condensation. In a cylinder of unit area δx long with its axis along the direction of propagation x, the passage of a progressive wave causes an extension of length $\delta\xi$, if the gradient of displacement with position is $\delta\xi/\delta x$; this gradient is also equal to the condensation.

$$\frac{\partial(\delta p)}{\partial x} = -K\frac{\partial s}{\partial x} = -K\frac{\partial^2\xi}{\partial x^2} = \rho\ddot{\xi} \tag{19}$$

for δp is the force on the unit area at one end of the cylinder and the

first term represents the difference of the forces on the two ends or the mass $\times$ acceleration, as we have set it in the last term. $\sqrt{K/\rho}$ has the dimensions of velocity and can be shown to be identical with what one regards as the velocity of a wave, (c). In a condensation wave travelling along a column of gas, the adiabatic relationship between pressure and density is usually taken, making $c^2 = \chi p/\rho$, χ being the ratio of specific heats.

We can also, as in hydrodynamics, introduce the idea of a velocity potential φ, such that $\partial\varphi/\partial x = \dot{\xi}$, the particle velocity. From (19)

$$-\dot{\varphi} = c^2 s \tag{20}$$

Spherical Waves

In this case, taking r as the radial distance, for a diverging wave $\partial\varphi/\partial r$ represents the particle velocity. To satisfy this condition as well as (19), let

$$rs = \exp.\{i\beta(ct - r)\}$$

$$\varphi = -c^2 \int^{\cdot} s\,dt = -\frac{c}{i\beta r} \exp.\{i\beta(ct - r)\} \tag{21}$$

$$\dot{\xi} = \frac{\partial\varphi}{\partial r} = \frac{c}{i\beta r}\left(\frac{1}{r} - i\beta\right)(rs) \tag{22}$$

Also
$$\delta p = \rho c^2 s = \frac{\rho c^2}{r}.\,(rs). \tag{23}$$

We shall now obtain the equations of motion of a number of simple vibrating systems.

§ 9. Stretched String

(i) *In Transverse Vibration*

In the absence of stiffness of string or yielding of the bridges over which the string is stretched, the components of tension F tending to restore a displaced segment δx long of the string are at one end $F\dfrac{\partial y}{\partial x}$, at the other $F\left(\dfrac{\partial y}{\partial x} + \dfrac{\partial^2 y}{\partial x^2}\cdot\delta x\right)$, y being the displacement. If the

string has a mass m per unit length, the equation for the mass $\times$ acceleration of the element is

$$F \frac{\partial^2 y}{\partial x^2} \delta x = my\delta x \tag{24}$$

giving a transverse wave velocity of $\sqrt{F/m}$.

(ii) *In Longitudinal Oscillation*

The development is similar to that for the aerial wave (19) with YOUNG's modulus E standing for K in a velocity $c = \sqrt{E/\rho}$.

If, as is usual, the string is gripped at the ends these become nodes and the full harmonic series of partial tones, based on a fundamental wave-length twice the length of the string, is possible.

§ 10. Bar in Transverse Vibration

The bending moment at any point x on the bar can be represented as $B = (Fx)$.

$$\text{or } \frac{\partial^2 B}{\partial x^2} = \frac{\partial F}{\partial x}$$

We now require an expression for B in terms of the configuration of an element δx long. Let it be bent into an arc of a circle of large radius r, so that we may take $\partial^2 y/\partial x^2$ to equal the curvature. Owing to the bending, the original slice at a distance δr from the 'neutral surface' (which remains unchanged in length) becomes stretched by the fraction $\delta l/\delta x = \delta r/r$. This is the strain on the slice; the stress is found by multiplying this by YOUNG's modulus E. Then the total force on this slice of cross section δS, becomes $E \frac{\delta l}{\delta x} \cdot \delta S = E\, Sr.\, \delta S \cdot \frac{\partial^2 y}{\partial x^2}$. Multiplying this by δr to obtain the bending moment about the neutral surface and integrating over the whole area of the element;

$$B = E \Sigma (\delta r^2 \delta S) \frac{\partial y^2}{\partial x^2} = E.\, \kappa^2.\, S. \frac{\partial^2 y}{\partial x^2} \tag{25}$$

where κ is written for the 'spin radius' of the section about the neutral surface. The restoring force on the element is

$$-\frac{\partial F.}{\partial x}\delta x = -\frac{\partial^2 B}{\partial x^2}\delta x = -E\kappa^2 S.\frac{\partial^4 y}{\partial x^4}.\delta x$$

Equating this to the mass $\times$ acceleration of the element we obtain finally;-

$$\frac{\partial^4 y}{\partial x^4} = -\frac{\rho \ddot{y}}{\kappa^2 E} = -\frac{\ddot{y}}{\kappa^2 V^2} \tag{26}$$

where V is the speed of longitudinal waves in the bar.

On the assumption of simple harmonic and undamped motion; $\frac{\partial^4 y}{\partial x^4} = \frac{\rho\omega^2}{\kappa^2 E}\cdot y = k^4 y$ (say); of which a solution is

$y = (A \cos kx + B \sin kx + C \cosh kx + D \sinh kx) \exp. i\omega t$

(i) *Bar Clamped at One End*

Taking the fixed end as the origin of x and the free end at $x = l$ we find the multiplier of exp. $\iota\omega t$ in the last equation to be given by values of kl which satisfy

$$\sec kl = -\cosh kl \tag{27}$$

This equation may be solved by plotting graphs of $y = -\cosh kl$ and of $y = \sec kl$, the required solutions being given by the values of the intersections of the two curves. The corresponding relative frequencies are proportional to k^2.

TABLE OF kl

1.88	4.69	7.85	11.00	14.14	17.28 etc.

The fundamental of such a bar is therefore given by

$$\sqrt{\frac{\rho}{E}}\cdot\frac{\omega l^2}{\kappa} = (1.88)^2$$

(ii) *Bar Supported at Both Ends*

In this case we find $\sin kl = 0$. Therefore kl is a multiple of π. The nodes of the partials are now equidistant, l has successive values in

the ratio, 1, 2, 3 etc., as for vibrating string, but the partials have frequencies proportional to l^2 and therefore in the ratio 1, 4, 9, etc.

§ 11. Circular Membrane

If the membrane has radius a, no stiffness and is subject to a pressure over its surface under tension F, so that $P = F\xi_0/a^2$;

$$\xi = \frac{Pa^2}{4F}\left(1 - \frac{r^2}{a^2}\right) = \xi_0\left(1 - \frac{r^2}{a^2}\right)$$

showing that the variation of displacement ξ with radius r is paraboloidal. If ρ is the mass per unit area, the total kinetic energy

$$T = \int \tfrac{1}{2}\dot{\xi}_0^2\left(1 - \frac{r^2}{a^2}\right)^2 2\pi\rho r\delta r = \frac{1}{6}\pi\rho a^2\dot{\xi}_0^2$$

and the potential energy

$$V = \int_0^{\xi_0}\int_0^a 2\pi r\delta r\, P\delta\xi = \pi F\xi_0^2$$

The natural pulsatance in the absence of damping is then

$$\omega_0 = \sqrt{\frac{6F}{\rho a^2}} = \frac{2.44c}{a} \tag{28}$$

c being the velocity of the waves ($\sqrt{F/\rho}$). (This may be compared with the ideal piston source (p. 16) for which we derive the mass coefficient m to be $\frac{1}{3}$ of the total mass M and the stiffness coefficient s to equal $2\pi F$.)

The equation for the vibrations, in polarcoordinates, is

$$\ddot{\xi} = c^2\left(\frac{\partial^2\xi}{\partial r^2} + \frac{1}{r}\frac{\partial\xi}{\partial r} + \frac{1}{r^2}\frac{\partial^2\xi}{\partial\theta^2}\right) \tag{29}$$

The solution for simple harmonic vibrations is given in terms of BESSEL functions as;

$$\xi = A J_n(\beta r)\cos j(\theta + \alpha)\cos\omega t \tag{30}$$

Here θ is the angle which a radius vector r makes with a fixed polar line, A and α are constants, j is the integral order of the mode. If as

usual, the membrane is fixed round its circumference, we have the boundary condition $J_n(\beta a) = 0$ which fixes the possible values of β for the various modes.

§ 12. Acoustical Impedance

As we shall derive the theory of vibrating fluid columns in terms of acoustical impedance, this concept will first be introduced in relation to gaseous vibrations.

Taking the ideal case in which the displacement ξ of the gas is uniform over any one section (area S) and viscous resistance may be neglected, if the applied pressure is $P \sin \omega t$, the resulting velocity produced can be written in the form $\dot{X}/S$ where X is of the dimensions L^3 and is a species of 'volume displacement'. Then the impedance (Z) is defined by the relation

$$\dot{X} = \frac{P \sin \omega t}{Z} \tag{31}$$

Suppose for example we have a cylindrical volume of gas, area of cross-section S, mass m, elasticity s, free from viscous forces, and subjected to an applied force $F \sin \omega t$. The equation of motion will be

$$m\ddot{\xi} + s\xi = F \sin \omega t \tag{32}$$

As the instantaneous displacement is uniform over any one section of the gas, though varying from section to section, we can put $\xi S = X$, $F = PS$, where P is the maximum value of the applied pressure. Then;

$$\frac{m}{S^2}\ddot{X} + \frac{s}{S}X = P \sin \omega t$$

m/S^2 is called the inertance (L) and S^2/s the capacitance (C) of the body of gas. Then

$$\dot{X} = \frac{P \cos \omega t}{\frac{s}{\omega S^2} - \frac{\omega m}{S^2}} = \frac{P \cos \omega t}{\frac{1}{\omega C} - \omega L}$$

§ 13. Helmholtz Resonator as Inertance and Capacitance in Series

The resonator consists of a cylinder of area S_2, volume v communicating via a neck of area S_1 and length l with the atmosphere. If we write for the mass of air in neck $\rho l S_1$, its inertance $L = \rho l / S_1$ where l is to include the end-correction. The capacitance of air in the cylindrical body is given by s/S_2^2 where s is the restoring force for unit volume-displacement,

Therefore
$$\frac{1}{C} = \frac{\partial p}{\partial v} = \frac{\chi p}{v} = \frac{c^2 \rho}{v}$$

and the natural pulsatance

$$\omega = \sqrt{\left(\frac{S_1}{\rho l} \cdot \frac{c^2 \rho}{v}\right)} = c \sqrt{\frac{\kappa}{v}} \tag{33}$$

in which κ is called the conductivity of the orifice.

§ 14. Impedance of Pipes

Up to the present we have considered the motion in the system to be wholly in phase. If we have to deal with conduits or sections of filters of length comparable to the wave-length of the transmitted sound we must, of course, use the wave equations.

The particle velocity at any section along the pipe at a distance x from the end then takes the form

$$\dot{\xi} = (A \sin \frac{\omega}{c} x + B \cos \frac{\omega}{c} x) \exp. i\omega t$$

Whence
$$\xi = \frac{1}{i\omega} (A \sin \frac{\omega}{c} x + B \cos \frac{\omega}{c} x) \exp. i\omega t$$

But
$$\frac{\partial}{\partial x} (\delta p) = - \chi p \frac{\partial^2 \xi}{\partial x^2}$$

Whence
$$\delta p = - \rho c^2 \frac{\partial \xi}{\partial x} = i\rho c (A \cos \frac{\omega}{c} x - B \sin \frac{\omega}{c} x) \exp. i\omega t$$

$$Z_e = \frac{\delta p_e}{\dot{X}_e} = \frac{i\rho c}{S} \left[\frac{A \cos \frac{\omega}{c} l - B \sin \frac{\omega}{c} l}{A \sin \frac{\omega}{c} l + B \cos \frac{\omega}{c} l} \right] \tag{34}$$

§ 15. Specific Impedance

Consider the case of plane waves incident normally on a boundary between two media. Let the equations of the incident wave moving from left to right be

$$\xi_1 = \exp.\, i(\omega t - \beta x);\ \dot{\xi}_1 = i\omega\xi_1$$

Then

$$cs_1 = -c\,\frac{\partial \xi_1}{\partial x} = i\beta c\xi_1 = \dot{\xi}_1$$

where s_1 is the condensation. For the reflected wave

$$\xi_2 = \exp i\,(\omega t + \beta x);\ -cs_2 = \dot{\xi}_2$$

while for the transmitted wave,

$$\xi' = \exp.\, i\,(\omega t - \beta' x);\ c's' = \dot{\xi}'$$

Since the net particle velocity normal to the boundary must be the same on each side

$$\dot{\xi}_1 + \dot{\xi}_2 = \dot{\xi}'$$

or

$$cs_1 - cs_2 = c's'$$

Since the normal pressure at the boundary must be the same in each medium.

$$Es_1 + Es_2 = E's'$$

writing E and E' for the elasticities. From these two equations, we obtain

$$\frac{s_2}{s_1} = \frac{E'c - Ec'}{E'c + Ec'} = \frac{\rho'c' - \rho c}{\rho'c' + \rho c} \tag{35}$$

This ratio expresses the relative pressure amplitudes in the reflected and incident sound waves. There will in fact be no reflection if $\rho'c' = \rho c$. This product of the density and velocity of sound is a characteristic of the medium and the type of wave and is called 'characteristic (or wave) impedance'. In many experiments, the first medium will be air and the second a solid of impedance $\varkappa$. The amplitude reflection coefficient can then be written $\dfrac{\varkappa - \rho c}{\varkappa + \rho c}$ in accordance with (35). In

general a complex impedance may be written $\zeta = |\zeta|e^{i\varphi}$ where $|\zeta|$ is the modulus (real part) and φ the argument of ζ .When unit area perpendicular to the direction of propagation is considered, the ratio $\delta p/\dot{\xi}$ is called 'specific impedance'.

In a plane wave these two impedances are synonymous for

$$\zeta = \frac{\delta p}{\dot{\xi}} = \frac{Es}{cs} = \rho c \tag{36}$$

This impedance, being real and independent of frequency, is a pure resistance.

In spherical waves, however, the impedance is a function of frequency. Dividing equations (23) and (22) we obtain the characteristic impedance for this type of radiation, *viz;*

$$\zeta = \rho c \left(\frac{i\beta}{r^{-1} + i\beta} \right) = \rho c \left(\frac{\beta^2 r^2 + i\beta r}{1 + \beta^2 r^2} \right) \tag{37}$$

the imaginary term representing a reactance. When βr is small, this impedance is $\rho c \beta^2 r$,[2] the so-called 'radiation resistance'.

In giving the impedance theory of pipes we should properly have included this radiation impedance as a consequence of the change-over from plane to spherical waves at the mouth. Its influence on the natural frequencies of the system is small however, and the effect is bound up with the end-correction of the tube. A more important aspect of this matter appears in the calculation of the ratio of the transmitted to the reflected energy. Not only will the considerations leading to equation (35) apply to a material change in the medium, but in addition whenever there is a sudden change in the total acoustic impedance there will be reflection. Such change may occur at a constriction in virtue of a change in S, or at the end of a pipe in virtue of the change from plane to spherical waves. Energy will be radiated most efficiently, *i.e.* there will be least reflection, if the impedances of the tube and of the open air at the end are matched; this cannot be done in a cylindrical pipe, but is more efficiently done with a horn of gradual flare, especially one whose section increases exponentially. Such a system gradually transforms plane waves introduced at the throat into spherical waves at the bell, at all but the lowest frequencies, apart from the viscous damping of the air in the tube.

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$$\zeta = \rho c \left(\frac{i\beta}{r^{-1} + i\beta} \right) = \rho c \left(\frac{\beta^2 r^2 + i\beta r}{1 + \beta^2 r^2} \right) \tag{37}$$

the imaginary term representing a reactance. When βr is small, this impedance is $\rho c \beta^2 r^2$, the so-called 'radiation resistance'.

In giving the impedance theory of pipes we should properly have included this radiation impedance as a consequence of the change-over from plane to spherical waves at the mouth. Its influence on the natural frequencies of the system is small however, and the effect is bound up with the end-correction of the tube. A more important aspect of this matter appears in the calculation of the ratio of the transmitted to the reflected energy. Not only will the considerations leading to equation (35) apply to a material change in the medium, but in addition whenever there is a sudden change in the total acoustic impedance there will be reflection. Such change may occur at a constriction in virtue of a change in S, or at the end of a pipe in virtue of the change from plane to spherical waves. Energy will be radiated most efficiently, *i.e.* there will be least reflection, if the impedances of the tube and of the open air at the end are matched; this cannot be done in a cylindrical pipe, but is more efficiently done with a horn of gradual flare, especially one whose section increases exponentially. Such a system gradually transforms plane waves introduced at the throat into spherical waves at the bell, at all but the lowest frequencies, apart from the viscous damping of the air in the tube.

§ 16. The Pulsating Sphere as Source

The potential satisfying (21) and giving a velocity $\dot{\xi}_0 \exp. i\omega t$ at $r = a$ (the surface) is

$$\varphi = \frac{A}{r} \exp. i(\omega t - \beta r - a)$$

so that

$$-\frac{\partial \varphi}{\partial r}_{r=a} = A \frac{1 + i\beta a}{a^2} \exp. i\omega t = \dot{\xi}_0 \exp. i\omega t$$

Whence

$$\varphi = \frac{a^2 (1 - i\beta a)}{r (1 + \beta^2 a^2)} \dot{\xi}_0 \exp. i(\omega t - \beta r - a)$$

The pressure at the surface of the sphere is then

$$p_a = r\dot{\varphi}_{r=a} = \rho\omega a^2 \frac{(\beta a + i)}{(1 + \beta^2 a^2)} \dot{\xi}_0 \exp. i\omega t$$

Whence the radiation impedance is

$$Z = \rho c. \frac{\beta^2 a^2 + i\beta a}{1 + \beta^2 a^2} \tag{38}$$

of which the first term is a resistance and the second a reactance. At high frequencies (small wave-length compared to a) the radiation resistance approaches ρc whereas the reactance diminishes to nil.

§ 17. The Oscillating Piston as Source

Let n represent the ratio of the radius a to the wave-length λ. When n is small the effect at a distance is as of a point source, *i.e.*, the front is spherical. But in the neighbourhood of a vibrating piston when $n \gg 1$, the effect of this piston on the surroundings can be considered as made up of a set of elementary sources each performing S.H.M. distributed over its area S and radiating forward over a solid angle π. The velocity potential at a distance r from the elementary source δS is then

$$\delta\varphi = \frac{\xi_0 c i\omega t}{2\pi r} \exp. (-i\beta r)\, \delta S$$

and that due to the complete piston is

$$\varphi = \frac{\xi_0 e^{i\omega t}}{2\pi} \iint \frac{\exp.(-i\beta r)}{r} \delta S$$

From the aspect of a point at a distance x along the axis, the potential is;

$$\frac{\dot{\xi}_0 e^{i\omega t}}{2r} \int_0^a \frac{2\pi y\, dy}{\sqrt{(x^2+y^2)}} \exp.(-i\beta\sqrt{x^2+y^2}) = \frac{i\dot{\xi}_0 e^{i\omega t}}{\beta} \Big[\exp.(-i\beta\sqrt{x^2+y^2}) \Big]_0^a$$

y being the radius of an annulus on the disc, distant from the point $r = \sqrt{x^2+y^2}$.

Writing the particle velocity at the piston, $\dot{\xi}_0 \cos \omega t$ and $n\lambda$ for a we derive

$$\varphi = \frac{-\xi_0}{\beta} [\sin(\omega t - \beta\sqrt{x^2+n^2\lambda^2}) - \sin(\omega t - \beta x)] =$$

$$= \frac{2\xi_0}{\beta} \sin\left(\frac{n^2\lambda^2\beta}{4x}\right) \sin\left(\omega t - \beta x - \frac{n^2\lambda^2\beta}{4x}\right)$$

provided $r \gg n\lambda$. The pressure amplitude p is then ρ times the maximum value of $\dot{\varphi}$ in the period, *i.e.*

$$p = \frac{2\rho\dot{\xi}_0\omega}{\beta} \sin\left(\frac{n^2\lambda^2}{4x}\beta\right) = 2\rho c\dot{\xi}_0 \sin\left(\frac{\pi n^2\lambda}{2x}\right) \tag{39}$$

Up to the point $x_0 = n^2\lambda$ along the axis, the beam does not diverge but the amplitude passes through a series of maxima and minima, the last occurring at x_0. From this point the beam begins to diverge at an angle which is roughly that of the angle subtended by the disc seen from x_0.

§ 18. Absorption of Sound in a Fluid

From the NAVIER-STOKES equations of the dynamics of a viscous fluid, together with an equation of continuity*, we can derive the equation for the propagation of a plane wave in terms of the particle velocity u, the pressure gradient $\partial p/\partial x$ and the kinematic viscosity υ;-

$$\frac{\partial u}{\partial t} = -\frac{1}{\rho}\frac{\partial p}{\partial x} + \frac{4}{3}\upsilon\frac{\partial^2 u}{\partial x^2} \tag{40}$$

* See for example, LAMB's *Hydrodynamics*, p. 328.

This together with the usual expression relating the excess pressure and the condensation to the velocity of sound transforms to

$$\frac{\partial^2 u}{\partial t^2} = c^2 \frac{\partial^2 u}{\partial x^2} + \frac{4}{3}\upsilon \frac{\partial^3 u}{\partial x^2 \partial t} \tag{41}$$

Assuming a solution of the type

$$u = u_0 \exp.\, i\,(\omega t - \beta x) \cdot \exp.\, \alpha x$$

we derive

$$\alpha^2 (c^2 + \frac{4}{3} i\upsilon\omega) + \omega^2 = 0$$

or, neglecting the square of $\upsilon\omega/c^2$

$$\alpha = -\frac{i\omega}{c} - \frac{2}{3}\frac{\upsilon\omega^2}{c^3} \tag{42}$$

Taking the real part, the absorption coefficient is then

$$\frac{2}{3}\frac{\upsilon\omega^2}{c^3} \text{ or } \frac{8}{3}\frac{\pi^2\upsilon}{\lambda^2}.$$

When thermal conductivity has to be allowed for the equation for the temperature θ of a volume v of gas takes the form

$$\frac{\partial\theta}{\partial t} + (\chi - 1)\frac{\theta}{v}\cdot\frac{\partial v}{\partial t} = \upsilon'.\frac{\partial^2\theta}{\partial x^2}, \tag{43}$$

υ' being the thermal diffusivity.
Introducing the condensation s $(= \delta\rho/\rho)$ and an analogous quantity $\eta(= -\delta\theta/\theta)$ for temperature changes

$$\frac{\partial\eta}{\partial t} - (\chi - 1)\frac{\partial s}{\partial t} = \upsilon'\frac{\partial^2\eta}{\partial x^2}$$

(40) becomes

$$\frac{\partial u}{\partial t} = -b^2\left(\frac{\partial s}{\partial x} + \frac{\partial\eta}{\partial x}\right) + \frac{4}{3}\upsilon\frac{\partial^2 u}{\partial x^2} \tag{44}$$

b^2 being p/ρ, the square of the isothermal velocity of sound. With the usual relation $\dot{s} = -\dfrac{\partial u}{\partial x}$

$$\left.\begin{aligned} &\frac{\partial^2 u}{\partial t} = b^2\left(\frac{\partial^2 u}{\partial x^2} - \frac{\partial^2 \eta}{\partial x \partial t}\right) + \frac{4}{3}\upsilon\frac{\partial^3 u}{\partial x^2 \partial t} \\ \text{and} \quad &\frac{\partial \eta}{\partial t} + (\chi - 1)\frac{\partial u}{\partial x} = \upsilon'\frac{\partial^2 \eta}{\partial x^2} \end{aligned}\right\} \tag{45}$$

Our equations for α now become on substituting in these two simultaneous equations

$$\left.\begin{aligned} &\omega^2 + \alpha^2 b^2 + \frac{4}{3} i\upsilon\omega\alpha^2/u - i\omega\alpha b^2\eta = 0 \\ &(\chi - 1)\alpha u + (i\omega - \upsilon'\alpha^2)\eta = 0 \end{aligned}\right\} \tag{46}$$

whence

$$\omega^3 + \left\{c^2\omega + \left(\frac{4}{3}\upsilon + \upsilon'\right) i\omega^2\right\}\alpha^2 + \upsilon'\left(ib^2 - \frac{4}{3}\upsilon\omega\right)\alpha^4 = 0 \tag{47}$$

In most sonic waves $\upsilon\omega/c^2$ is small,

then the roots of (47) are given by

$$\alpha_1^2 = -\omega^2/c^2 \; ; \; \alpha_2^2 = i\omega c^2/\upsilon' b^2 = \frac{i\omega\chi}{\upsilon'}$$

otherwise

$$\alpha^2 = -\frac{\omega^2}{c^2}\left[1 - \left\{\frac{4}{3}\upsilon + \left(1 - \frac{b^2}{c^2}\right)\upsilon'\right\}\frac{i\omega}{c^2}\right] \tag{48}$$

α_1 and α_2 may be added to give the total absorption due to viscous and thermal causes.

§ 19. Viscous Damping in Narrow Tubes

If P is the pressure amplitude of a simple harmonic force acting along a tube of radius r_0, the equation of motion of an annular ring at radius r:

$$\mathrm{P} = \left[i\omega\rho - \frac{\eta}{r}\frac{\partial}{\partial r}\left(r\frac{\partial}{\partial r}\right)\right]\dot{\xi}$$

where η now stands for viscosity and of which the solution is

$$\dot{\xi} = -\frac{P}{\eta\Psi^2} + AJ_0(\Psi r)$$

where $\Psi^2 = -i\omega/\upsilon$ and J_0 is the BESSEL function of zero order, υ standing for the kinematic viscosity coefficient, η/ρ.

Integrating for the mean velocity over the tube and making $\dot{\xi} = 0$ at $r = r_0$;

$$\bar{\dot{\xi}} = -\frac{P}{\eta\Psi^2}\left[1 - \frac{2}{\Psi r_0}\cdot\frac{J_1(\Psi r_0)}{J_0(\Psi r_0)}\right] \tag{49}$$

When $\sqrt{\frac{i\omega}{\upsilon}}\cdot r_0 < 1$ the whole tube is a laminar boundary layer in the sense of POISSEUILLE's law and the resistance $(P/\dot{\xi})$ is $8\,\eta/r_0^2$. When $\sqrt{\frac{i\omega}{\upsilon}}\cdot r_0 > 10$, the commoner case in ultrasonics, the resistance becomes

$$P/\dot{\xi} = i\omega\rho + (1+i)\,2\,\frac{\sqrt{\eta\rho}}{r} \tag{50}$$

Taking this with (49) and the sound equation

$$P = \rho c^2 \frac{\partial^2 \xi}{\partial x^2}$$

we get $$\left(\rho + \frac{1}{\upsilon}\frac{\sqrt{2\rho\eta}}{\sqrt{r}}\right)\ddot{\xi} + \frac{1}{r}\sqrt{2\rho\eta\omega}\cdot\dot{\xi} = \rho c^2 \frac{\partial^2 \xi}{\partial x^2} \tag{51}$$

Assuming a S.H.M. solution, corresponding to a damped wave along the axis of x, we obtain

$$\alpha = \frac{1}{rc}\sqrt{\frac{\omega\upsilon}{2}}\;;\; c' = c\left(1 - \frac{1}{r}\sqrt{\frac{\upsilon}{2\omega}}\right)$$

for the absorption and velocity in the tube, c being the free-space velocity.

DIVISION ONE

ACOUSTIC MEASUREMENTS AND MATERIALS

CHAPTER 2

STANDARD SOURCES OF FREQUENCY AND THEIR CALIBRATION

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§ 1. Introduction

Ability to measure and to produce frequencies of high accuracy is of ever-increasing importance in many fields of science. In acoustics, accurate measurement of frequency is necessary in the calibration and adjustment of electroacoustic devices and electrical networks such as filters and equalizers. Carefully calibrated frequency sources must be employed in many investigations in psychoacoustics, mechanical vibration, ultrasonics and other fields.

In recent years, primary standard frequency sources of extremely high stability have been developed. Accurate and convenient techniques are now available for calibrating these sources and for calibrating secondary and laboratory standard frequency sources from these primary standards.

§ 2. Primary Frequency Standards

The standard unit of time is the *second*, defined as 1/86, 400 of a mean solar day. The unit can be measured very accurately from astronomical observations, although variations are known to exist in astronomical time.

Reference standards of frequency are located at national laboratories and other laboratories in England [1], the United States of America [2], and other countries. At present, the most accurate primary frequency

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standards are vacuum-tube oscillators with frequency controlled by carefully prepared quartz crystals [3]. In a typical installation, the oscillator operates at 100,000 c/s, and multiples and submultiples of this frequency are produced through frequency conversion circuits. A submultiple frequency of 1000 c/s derived from the master oscillator is used to drive a synchronous clock. Indications of this synchronous clock can be compared with astronomical time over long intervals, to obtain a very precise check of the oscillator frequency. The rates of the best crystal oscillators may be determined in this way, with an accuracy better than 1 part in 10^8.

In the United States, broadcast services are maintained to provide standard frequencies and time signals derived from crystal oscillators [4]. These services can be utilized by laboratories in basic scientific studies where precise time measurement is essential, and in the calibration of secondary standards of frequency throughout a wide range of audio and radio frequencies.

Recently discovered resonance phenomena in molecules and atoms may soon have application in the stabilization of standard frequency sources [5]. Oscillators are being developed in which these phenomena are utilized to control the frequency of oscillation, and hence, through frequency conversion circuits, to control frequencies low enough to operate clocks and other mechanisms. The precision of these oscillators would be even greater than that of crystal oscillators, since many resonance phenomena are invariant; *i.e.* the frequency is not a function of temperature, pressure, and dimensions. The possibility exists that these sources will provide a new standard of time even more stable than astronomical time itself.

§ 3. Other Standard Frequency Sources

Frequency standards commonly employed in laboratories may be classified in terms of their degree of stability. The precise laboratory standard is usually a constant frequency oscillator controlled by a quartz crystal or by a precision tuning fork. When less accuracy is desired, commercial audio oscillators are available in which the frequency is determined by LC or RC feedback circuits.

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(i) *Quartz Crystal Oscillator and Clock*

The quartz crystal oscillator and clock used as a laboratory frequency standard is modelled after the primary frequency standards discussed previously [3]. The installation consists of a constant-frequency oscillator, suitable frequency dividers, and a synchronous clock mechanism. Maximum frequency stability is obtained by careful temperature control of the master oscillator and crystal unit. The oscillator frequency can be checked against time signals originating from primary standards, and it will usually hold constant to less than one part in 10^7.

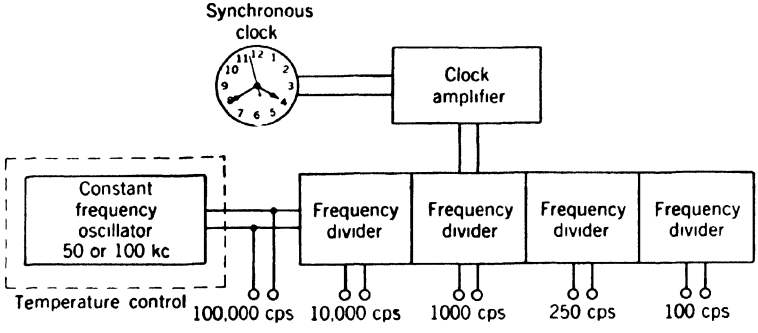


Fig. 1. Block diagram of laboratory frequency standard with unfiltered outputs at 100, 250, 1000, 10,000 and 100,000 c/s. (From BERANEK, *Acoustic Measurements*).

A block diagram of a suitable laboratory standard of frequency of this type is given in Fig. 1. It consists of a constant-frequency oscillator with a frequency of 50 or 100 kc/s and four multivibrators or frequency dividers for producing frequencies of 10,000 c/s, 1,000 c/s, 250 c/s, and 100 c/s. The output of one of the frequency dividers is amplified by an audio amplifier and used to drive a synchronous clock with an easily readable sweep second hand.

The vacuum-tube circuit used for maintaining the quartz resonator in oscillation should provide maximum stability in both frequency and amplitude. The circuit that at present most nearly approaches the ideal was developed by L. A. MEACHAM, and is known as the bridge-stabilized oscillator [6]. This oscillator, consisting of an amplifier and a WHEATSTONE bridge, is shown in Fig. 2. The frequency-controlling resonant element is used as one arm of the bridge, and resistance

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elements are placed in the other three arms. The output of the amplifier is connected across one of the diagonals of the bridge, and the unbalance potential appearing across the conjugate diagonal is applied to the amplifier input terminals. Amplitude stabilization is provided automatically by a thermally controlled resistance in one of the arms.

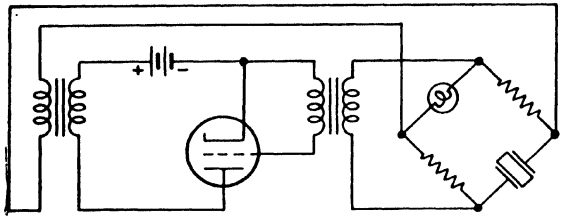


Fig. 2. Simplified circuit for bridge-stabilized oscillator of MEACHEM [6].

The circuit for a common frequency-divider of the multivibrator type is shown in Fig. 3. If conventional multivibrator circuits are used, their outputs will be rich in harmonics. Suitable band-pass filters and amplifiers will be necessary to obtain sine waves of any harmonic frequency.

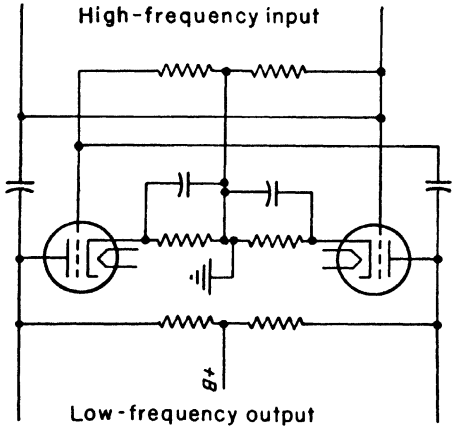


Fig. 3. Frequency-divider circuit of the multivibrator type.

(ii) *Precision Fork* [7, 8, 9]

Tuning forks were among the first and most commonly used frequency standards. Although less accurate than quartz crystal units, they have the advantage of direct low frequency operation, precluding the use of frequency dividers. A tuning fork can be easily excited and maintained in vibration by means of an electrical feedback circuit, and a synchronous clock mechanism can be driven directly by the output of the oscillator amplifier. If the material of the fork is selected to have low temperature coefficient of modulus of elasticity, the output frequency will remain remarkably constant at normal ambient room temperatures.

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(iii) *Variable-Frequency Oscillators*

In many applications in acoustics it is necessary to generate oscillations whose frequency can be set quickly at any arbitrary value with a high degree of accuracy. The stability of this type of oscillator is necessarily considerably lower than that of constant frequency standard signal generators. Several methods can be employed to obtain variable frequency control with electronic circuits.

In one type of oscillator, an RC network in the feedback circuit is used to control the frequency. This circuit may take the form of a simple phase-shifting network [10], a bridge network [11], or a frequency-selective twin-T network [12]. The continuously variable elements are usually resistances. It is possible to obtain stable, linear circuit operation by the use of a thermal element for amplitude control. With this circuit, the entire audible frequency range cannot usually be covered by one rotation of the oscillator dial, and hence range switching is necessary.

Another useful oscillator is the beat frequency type [13], which covers the entire audible frequency range with a single rotation of the tuning dial. The audio frequency produced is equal to the difference between the frequencies of two radio frequency oscillators, and is obtained from a detector and filter circuit at the output of the oscillators. One of the oscillators is usually fixed, and the other is variable by tuning an LC resonant circuit. Very careful design must be used to provide a stable beat frequency oscillator, since if either of the two oscillators drifts by a small percentage, the difference frequency will change by a large percentage. If careful temperature control is exercised, and if oscillating circuits of like thermal coefficients are selected, the output will remain quite stable.

§ 4. Calibration of Frequency Sources

The accuracy of laboratory frequency standards can be checked by calibration against reference primary standards. These laboratory frequency standards may in turn be used to check the calibration of other laboratory frequency sources. Many methods are available for performing these calibrations, depending on the frequency range and degree of precision desired [14]. These may be divided into two general

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classes. In the first, direct comparison is made with a standard frequency source by measurement of the difference between standard and unknown frequencies or multiples or submultiples of these. A high degree of precision can be obtained by this procedure, especially when the difference frequency is small. In the second class, the frequency selective properties of electric circuits or vibrating devices are employed to determine the unknown frequency directly.

(i) *Beat Methods* [15]

When the frequency of the source to be calibrated is stable and nearly equal to the standard frequency, a very accurate determination can be made by counting beats. The two signals are combined in a modulator, and the resulting difference frequency is measured either by counting, if the difference frequency is less than a few cycles per second, or by a simple frequency meter (several types are described below) if the difference frequency is greater. When the input frequencies are in the ratio $m : n$, where m and n are small integers, the beat method may also be employed, using higher order modulation products at the modulator output.

When two audio frequencies are to be compared, audible methods may often be used in detecting and counting beats. The presence of beats is detected as periodic changes in the loudness of the combined sounds. When the two periodic sound sources are rich in harmonics, beats between the harmonic frequencies can often be detected.

(ii) *Cathode Ray Oscilloscope*

The cathode ray oscilloscope is often employed in the calibration of frequency sources, especially in the range of audio frequencies. In its simplest form, the procedure is to connect the unknown frequency source to one pair of deflection plates and the standard frequency source to the other pair. When the frequency ratio approaches a value that can be expressed by simple integers, a pattern known as a LISSAJOUS figure [16] appears on the screen. The pattern can be interpreted more easily if the two sources are sinusoidal.

When the ratio of frequencies is exactly equal to, say $m : n$, where m and n are small integers, the pattern is stationary. The frequency

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ratio is equal to the number of times the side of the figure is tangent to a horizontal line to the number of times its end is tangent to a vertical line (See Fig. 4). If the ratio differs slightly from this value, the pattern appears to move about on the screen, since the relative phase of the two waves is continuously changing. When the ratio of the two frequencies is far from being simple, the pattern becomes a blurred rectangle.

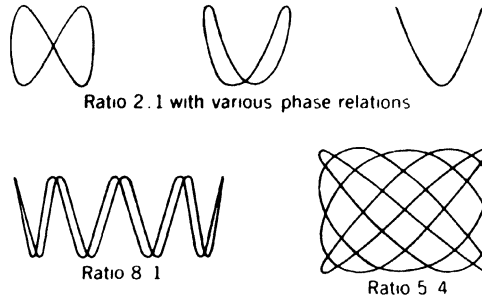


Fig. 4. Typical LISSAJOUS figures with the ratios indicated.

Figure 5 shows a more satisfactory arrangement for performing the frequency comparison for higher order frequency ratios.[13] If the high frequency is inserted in series with the grid at point *A* of (a), then the spot-wheel pattern (b) is obtained. The low frequency produces a circular or elliptical path, since it is applied across the resistance-

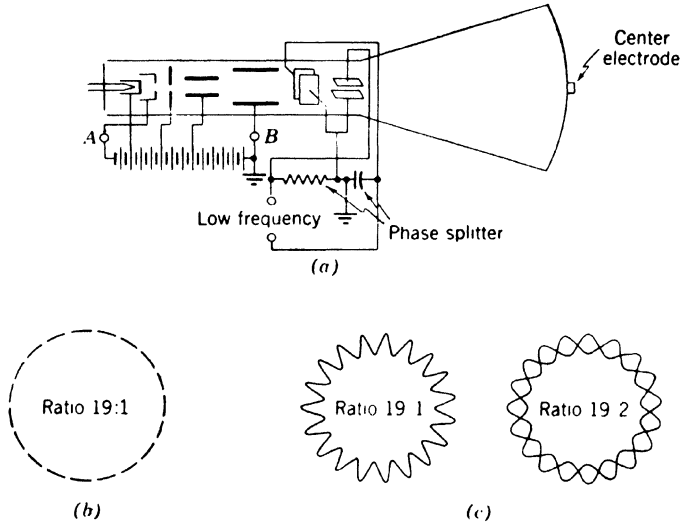


Fig. 5. (a) Circuit for producing spot-wheel or gear-wheel patterns; (b) spot-wheel pattern; (c) gear-wheel patterns (after TERMAN [13]).

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capacitance phase-splitting network connected to the deflection plates. If the frequency ratio is exactly integral, the spot wheel will be stationary; otherwise it will rotate. The gear-wheel patterns of (c) are easier to read and interpret. These may be produced either by inserting the high frequency into the circuit at point B of (a), or by using the tube with centre electrode, and impressing the high frequency between this electrode and ground.

(iii) *Interpolation Oscillator*

An oscillator of special design, called an interpolation oscillator, may be used to perform accurate comparison between a standard frequency and an unknown [15]. The interpolation oscillator should have an output nearly equal to that of the standard oscillator, and its frequency should be variable over a range of at least $\pm$ 10 percent. It must have a frequency which is constant for short intervals of time, and incremental changes in its frequency should be proportional to incremental changes of capacitance of its standard variable condenser.

A procedure that might be followed in determining an unknown frequency using the interpolation oscillator follows. Assume that the unknown frequency is 1043 c/s, and that a 1000 c/s component is obtainable from a frequency divider operated from the standard frequency source. Then connect one pair of plates of a cathode ray tube to the 1000 c/s output and the other pair of plates to the output of the interpolation (variable frequency) oscillator. Adjust the standard condenser in the interpolation oscillator until the LISSAJOUS figure on the screen is stationary, and observe the condenser reading. Finally, replace the 1000 c/s connection by the source of unknown frequency and readjust the standard condenser. The shift of frequency will be given by $\delta f/f = -\delta C/2C$. The frequency difference δf can be determined in this way with fairly high precision, provided the capacitance change δC is small compared to C. If more complex frequency ratios are used in making the reference dial settings, greater precision may be attained in determination of the unknown frequency. For example, to determine a frequency of 1043 c/s, find the dial settings for 1040 and 1050 c/s from cathode ray tube patterns with a standard frequency of 100 c/s and ratios of 52 : 5 and 21 : 2, and determine the unknown frequency by linear interpolation.

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(iv) *Stroboscope*

The stroboscope may be used to measure frequency by comparison, usually in the audio frequency range. In principle, the unknown source of frequency is caused to flash a light which reflects from or shines through a rotating disk marked with a certain number of equal segments of contrasting colour. The speed of rotation is controlled by a standard frequency source. When the flashing rate of the light either is equal to or is a submultiple of the rate of segment passage, the segments appear to stand still, and the frequency under test is known either to be equal to or to be a subharmonic of the standard frequency. When the segment picture appears to stand still but contains two, three, four, or more times as many segment images, the frequency under test is known to be the corresponding multiple of the standard frequency.

A 'chromatic stroboscope' has been developed by YOUNG and LOOMIS [17] to measure frequencies over a continuous range. Twelve disks are rotated at speeds corresponding to the frequencies of the twelve equi-tempered semitones within an octave. On this scale, the interval corresponding to all semitones is equal to $\sqrt[12]{2}$. A typical disk is shown in Fig. 6. The device described by YOUNG and LOOMIS is especially valuable when it is used in conjunction with a laboratory standard of frequency that can be adjusted over a range of one semitone, such as a tuning fork [9]. Unknown frequencies which fall between a semitone are determined by shifting the standard frequency a known amount, which in turn changes the rotational rate of the disks so that one of the rotating patterns on one of the disks is brought to a standstill. The unknown frequency can thus be determined in terms of octaves and semitones, and then computed in cycles per second.

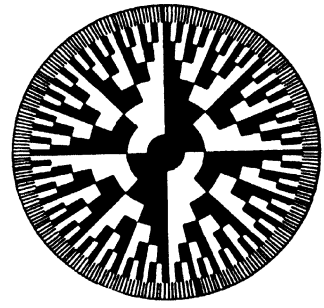


Fig. 6. Typical stroboscope disk.

(v) *Electronic Meters*

Many types of experiments require the measurement of frequency to

within only a few percent. For these cases an electronic meter that operates on condenser discharge current is satisfactory. The fundamental circuit of such a meter [18] is shown in Fig. 7. The periodic wave whose frequency f is to be determined is first converted to a square wave by limiting, as indicated by the source at the left with internal impedance R. If the time constant RC is small, condenser C charges to the maximum voltage V and charge CV, f times per second. The condenser can discharge through meter M, and the direct current indicated by M will be equal to CVf. The meter scale can thus be calibrated linearly in terms of frequency.

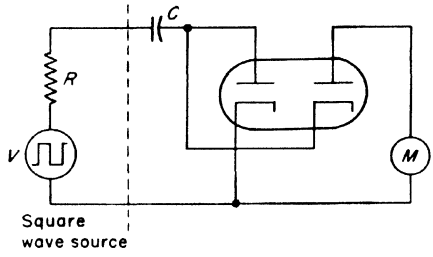


Fig. 7. Simplified circuit for electronic frequency meter.

(vi) *Bridge Methods*

Several bridge methods may be used for measuring frequency, especially in the audio frequency range. One such circuit, known as the WIEN bridge, is shown in Fig. 8(a). A three-terminal equivalent network eliminating the use of the transformer is shown in Fig. 8(b)[12]. The two resistances R_1 and R_2 are made variable over a suitable range, but are also kept equal.

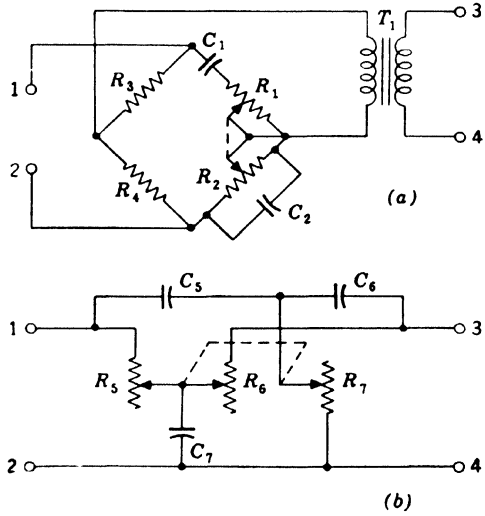


Fig. 8. Schematic diagram of two forms of WIEN bridge.

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If $C_1 = C_2$ and $R_3 = 2R_4$, the condition for balance reduces to

$$f = \frac{1}{2\pi R_1 C_1}. \qquad \text{(I, 1a)}$$

For the twin—T circuit the condition for zero output is

$$f = \frac{1}{2\pi R_5 C_5} \qquad \text{(I, 1b)}$$

provided these additional relations are satisfied:-

$$C_5 = C_6 = C_7/2$$
$$R_5 = R_6 = 2R_7.$$

In both these cases, the variable ganged resistances can be calibrated directly in terms of frequency.

(vii) *Vibrating Reed*

For the measurement of low frequencies, clamped steel bars excited directly by a vibrating machine or electromagnetically have been employed for many years [19]. HARTMANN-KEMPF developed an electromagnetically excited series of such bars, the resonant frequency of each being separated from its neighbour by 1 c/s. The amplitude of vibration of the bars was observed visually, and the device was quite useful in the approximate control of power frequencies. A single reed of variable length can also be used; the length is varied until the reed is brought into resonance as observed visually. The fundamental frequency f for such a bar clamped at one end is given by (*cf.* Chap. 1, § 10).

$$f = \frac{.56\kappa}{l^2}\sqrt{\frac{E}{\rho}} \text{ (approx.)} \qquad \text{(I, 2)}$$

where

l = length of the bar
E = Young's elastic modulus
ρ = density
κ = radius of gyration (note that $\kappa^2 = a^2/4$ for a bar of circular section and radius a, and $\kappa^2 = t^2/12$ if the bar is rectangular and of thickness t in the plane of vibration)

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CHAPTER 3

MEASUREMENT OF PRESSURE AMPLITUDE AND VELOCITY AMPLITUDE

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§ 1. Introduction

In any laboratory where acoustic measurements are made, there is need for an accurately calibrated standard microphone for the measurement of the strength of a sound field. This standard may itself be used to yield accurate measurement of acoustic quantities in a given experiment, or it may be used for the calibration of other microphones. For the case of a plane progressive sound wave, the strength of the sound field may be expressed in terms of one of several quantities—sound pressure, particle velocity, particle displacement, intensity, etc. Most measuring devices respond to sound pressure or particle velocity, but others have been devised.

The recently developed reciprocity technique is at present the most widely used method for performing the absolute calibration of such standard microphones. Other techniques may be divided into two general classes. In one class, the pressure amplitude or velocity amplitude existing at a point in space is measured by direct observation of the physical effect (mechanical, thermal, etc.) on some measuring device placed in the sound wave. The RAYLEIGH disk and the smoke particle technique are examples of this 'wave measurement' method. Of historical interest are the methods of measuring absolute pressure described by ALTBERG and by RAPS [1]. In the other class, use is made of a sound generator whose characteristics can be computed theoretically. Examples are the thermophone, the pistonphone and the electrostatic actuator.

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§ 2. Reciprocity Technique

The reciprocity technique for the absolute measurement of sound has been developed comparatively recently [2-7]. By this method, the calibration of electromechanical transducers can be performed by the observation of electrical quantities only, except for the simple measurement of distance or volume. This procedure is at present considered to be the most accurate absolute calibration technique, and is used almost exclusively by standards laboratories in the calibration of standard microphones.

The reciprocity theorem, as applied to lumped-constant electrical networks, may be stated as follows. In a linear, passive two terminal-pair network, the two transfer impedances are equal. This principle may be extended to electroacoustical systems that involve distributed rather than lumped parameters. This extension will be assumed here, although a rigorous proof has been presented by FOLDY and PRIMAKOFF [8]. Analysis shows that the principle of reciprocity applies to most linear, passive electromechanical or electroacoustical systems, although certain special types of transducers or combinations of transducers are exceptions [9]. These exceptions are normally of little practical importance.

(i) *Free Field Method*

Consider a transducer in a medium of characteristic impedance ρc, and a point P at distance d from the transducer, d being large compared to the dimensions of the transducer (Fig. 9). To examine the application of the principle of reciprocity to this situation, a simple hypothetical experiment is performed. The transducer is operated first as a source, with sound pressure measurements performed at point P, and then as a microphone, with a vanishingly small spherical source of radius a at point P. For operation as a source, let i be the transducer driving current and p the sound pressure at P. For operation as a microphone let e be the open circuit voltage at the transducer terminals, and u the

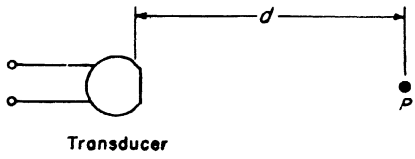


Fig. 9. Demonstration of reciprocity for a transducer in a free field.

volume velocity of the small spherical source at P. By the principle of reciprocity, the transfer impedances are equal, *i.e.*,

$$\frac{p}{i} = \frac{e}{u}. \qquad \text{(I, 3)}$$

For application in the free-field reciprocity calibration of a microphone or speaker, this result may be placed in a more convenient form. Define the following quantities:-

p_{ff} = free-field sound pressure before insertion of the microphone in the field.

$M = \frac{e}{p_{ff}}$ = free-field sensitivity of the transducer as a microphone.

$S = \frac{p}{i}$ = free-field space calibration as a speaker at a distance d.

Z_r = radiation impedance of vanishingly small spherical source of radius a. The value of Z_r is $i\rho c/2a\lambda$, where λ is the wavelength *.

The pressure at the surface of the sphere is uZ_r, hence

$$p_{ff} = uZ_r \frac{a}{d} \exp. - i(2\pi d/\lambda)$$

$$= u \frac{\rho c}{2d\lambda} \exp. - i\,[(-\pi/2) + (2\pi d/\lambda)].$$

Hence $$M = \frac{e}{p_{ff}} = \frac{e}{u} \cdot \frac{2d\lambda}{\rho c} \exp. i\,[(-\pi/2) + (2\pi d/\lambda)].$$

Also, $$S = \frac{p}{i} = \frac{e}{u} \text{ from (I, 3).}$$

Therefore $$\frac{M}{S} = \frac{2d\lambda}{\rho c} \exp. i\,[(-\pi/2) + (2\pi d/\lambda)]. \qquad \text{(I, 4)}$$

This equation, showing that M/S is a constant independent of the transducer, is the form in which the reciprocity principle will be applied. The ratio M/S is sometimes called the reciprocity parameter.

The basic experimental procedure employed in the free-field calibration of a microphone can now be described. The transducer U whose free-field calibration M is to be determined need not be rever-

* The unit of impedance here is the acoustic ohm per sq.cm., not that of the Introduction (*cf.* eqn. 35).

sible. An auxiliary reversible transducer U' is required, with an assumed free-field sensitivity M', and a free space calibration S'. There are two steps in the experimental procedure, illustrated in Fig. 10. First, with

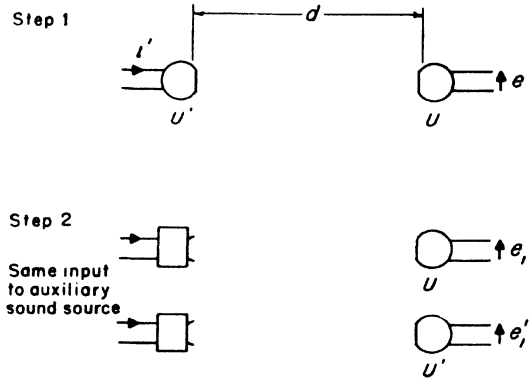


Fig. 10. The two steps in the free field reciprocity calibration of a transducer.

U' acting as a source with driving current i', and U as a microphone at distance d, the microphone open-circuit voltage e is measured. From the relations developed above,

$$e = Mp_{ff} = Mi'S' = Mi' \frac{\rho c}{2d\lambda} M' \exp. - i[(-\pi/2) + (2\pi d/\lambda)]$$

and hence $$MM' = \frac{2d\lambda}{\rho c} \left(\frac{e}{i'}\right) \exp. i[(-\pi/2) + (2\pi d/\lambda)] \qquad \text{(I, 5)}$$

A second experiment is now performed to obtain a comparison calibration of U and U'. An auxiliary sound source of constant output is used, and U and U' are alternately placed at the same point in the sound field of this source. The separation between this point and the source should be sufficiently large to provide an essentially plane wave at the point. The open-circuit voltages e_1 and e'_1 are measured, yielding the ratio

$$\frac{M}{M'} = \frac{e_1}{e'_1} \qquad \text{(I, 6)}$$

Combination of (I, 5) and (I, 6) yields the calibration of U:-

$$M = \left|\left(\frac{e}{i'}\right)\left(\frac{e_1}{e'_1}\right)\frac{2d\lambda}{\rho c}\right|^{\frac{1}{2}} \exp. i[(-\pi/4) + (\pi d/\lambda)] \qquad \text{(I, 7)}$$

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Thus with the measurement of the ratios $\frac{e}{i'}$ and $\frac{e_1}{e'_1}$ and of the distance d, the microphone calibration is obtained exactly. It is apparent that the equations also given the values of M' and S'. All quantities should, of course, be expressed in a consistent system of units such as the MKS system.

Certain practical details must be observed in the calibration procedure. The desired orientations of U and U' must be selected and maintained properly in both experiments. For greater accuracy, the distance d between U and U' in the first experiment should be as large as possible consistent with level requirements. There may be some uncertainty in the location of the centre of the transducer from which distance measurements are made. For precise calibration, the acoustic centres of the transducers should be located by auxiliary experiments [10, 11].

A variation of the above procedure, called 'self-reciprocity', has been described by CARSTENSEN [12]. This method permits the calibration of a single reversible transducer without the aid of auxiliary transducers. The transducer is placed at a distance $d/2$ in front of a large perfectly-reflecting wall, as in Fig. 11. A short pulse of sinusoidal current i' is applied to the transducer, and the resulting pulse of acoustical energy propagates toward the wall. The reflected pulse develops an open-circuit voltage e at the transducer terminals. The microphone calibration M may be obtained from equation (I, 5) replacing M' by M:

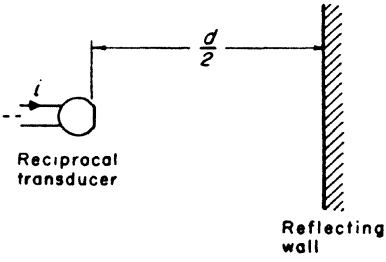


Fig. 11. Self-reciprocity calibration of a transducer.

$$M = \left| \left(\frac{e}{i'} \right) \frac{2d\lambda}{\rho c} \right|^{\frac{1}{2}} \exp. i\,[(-\pi/4) + (\pi d/\lambda)] \qquad \text{(I, 8)}$$

The distance d must be equal to several wavelengths if a pulse length of several cycles is employed. Also, the width of the reflecting wall must be of the order of several wave-lengths. In practice, these limitations restrict the use of the self-reciprocity procedure to the high-

frequency range. The method has been applied successfully in the calibration of underwater sound transducers [12].

(ii) *Closed Chamber Method*

The reciprocity technique may also be applied to the calibration of microphones that can be enclosed in a small cavity or 'coupler' [11]. Such a calibration is in terms of the pressure at the diaphragm rather than the free-field sound pressure, and is commonly called a *pressure calibration*. The difference between the pressure calibration and the free-field calibration is caused by the effects of diffraction and cavity resonances, which are especially important at high frequencies.

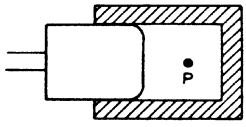


Fig. 12. Demonstration of reciprocity for a transducer in a cavity.

Consider a reversible transducer coupled to a cavity having an acoustic impedance Z_c (Fig. 12). To investigate the application of the reciprocity principle in this case, we examine the results of a simple hypothetical experiment. For the moment, the cavity is assumed to be small, with uniform sound pressure throughout its volume, and the transducer diaphragm impedance is assumed large compared with Z_c. The same symbols as in the free-field case are used, with the following exceptions: p now represents the (uniform) sound pressure in the cavity when the transducer acts as a source, and u represents the volume velocity of a hypothetical point source P placed within the cavity. The transducer is operated, as before, as a source and as a microphone, and by the principle of reciprocity we obtain the relation I, 3.

For operation as a microphone, the pressure calibration M is now equal to e/p_d, where p_d is the pressure in the cavity and at the diaphragm. p_d is related to the source volume velocity u by the equation

$$p_d = u Z_c \tag{I, 9}$$

Hence $M = \dfrac{e}{uZ_c}$, and $S = \dfrac{p}{i} = \dfrac{e}{u}$, yielding the ratio

$$\frac{M}{S} = \frac{1}{Z_c}. \tag{I, 10}$$

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This result shows that M/S depends only on the properties of the cavity, and not on the particular transducer.

The basic experimental procedure for the closed chamber method is the same as for the free-field method. First the transducer U to be calibrated and the auxiliary transducer U' are both coupled into a cavity as in Fig. 13, and the microphone open-circuit voltage e is measured with U' acting as a source with driving current i'. This voltage is given by

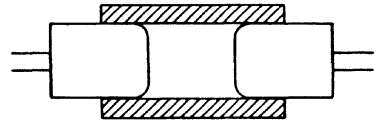


Fig. 13. Cross-sectional sketch of a cavity for calibrating two transducers.

$$e = Mp_d = M i' S' = Mi' \frac{M'}{Z_c}$$

Hence

$$MM' = Z_c \frac{e}{i'}. \qquad \text{(I, 11)}$$

To obtain a comparison calibration of U and U', an auxiliary sound source with constant output is coupled into a cavity, together with U and U' in turn. The open-circuit voltages e_1 and e'_1 are measured, to yield the ratio

$$\frac{M}{M'} = \frac{e_1}{e'_1} \qquad \text{(I, 12)}$$

Finally, we obtain

$$M = \left| \left(\frac{e}{i'} \right) \left(\frac{e_1}{e'_1} \right) Z_c \right|^{\frac{1}{2}} \qquad \text{(I, 13)}$$

which is the desired pressure calibration of transducer U. The cavity impedance can be computed from the dimensions of the cavity, using the relation (*cf.* Chap. 1, § 13)

$$Z_c = \frac{-i\rho c^2}{\omega v} \qquad \text{(I, 14)}$$

where v is the volume of the cavity, ω is the pulsatance, ρ is the density, and c is the velocity of sound in the medium.

In the practical calibration procedure, several corrections must be applied to this result.[11, 13] At high frequencies, the cavity may no longer act as a pure compliance, with uniform pressure throughout its volume.

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Corrections must be applied to account for wave motion in the cavity. To reduce the size of these corrections and to extend the frequency range, the cavity is made small, and hydrogen may be introduced into the volume. A further correction must be applied if the diaphragm impedance is not large compared to Z_c. The effect of finite diaphragm impedance may be accounted for by a change in the effective volume of the cavity. These and other errors arising in the closed coupler reciprocity calibration procedure are discussed in some detail by BERANEK [11].

§ 3. Wave Measurements

(i) *Rayleigh Disk*

The RAYLEIGH disk has long been used as a primary means for measuring the magnitude of the particle velocity in a plane wave sound field. Lord RAYLEIGH [14] first observed that under some circumstances a torque is exerted on a thin disk suspended by a fine fibre in a stream of air, and he recognized the phenomenon as a means for measuring the intensity of sound.

The action of the RAYLEIGH disk may be inferred qualitatively from the general character of the lines of flow around it (Fig. 14). The pressure at various points on the surface depends on the square of the velocity of flow at these points, by BERNOULLI's principle. For the case of steady flow, the velocity at points A and B would be greater than at points C and D. This would produce smaller pressure at points A and B, and the disk would tend to align itself so as to face the stream. For the case of alternating flow, as in a sound wave, there would be a similar torque, computed now from the mean square tangential velocities at the disk surface.

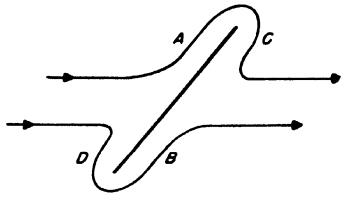


Fig. 14. Lines of flow around a RAYLEIGH disk.

The following relationship between the torque and the stream velocity has been developed by KOENIG [15]:-

$$L = \frac{4}{3}\rho_0 a^3 \overline{\dot{\xi}^2} \sin 2\varphi, \qquad \text{(I, 15)}$$

where L = time average of the torque tending to rotate the disk about its vertical axis, this vertical axis being normal to the stream velocity.
ρ_0 = density of medium
a = radius of disk
$\overline{\dot{\xi}^2}$ = mean square particle velocity in the wave
φ = angle between the direction of the stream velocity and the direction of a line drawn normal to the surface of the disk.

This equation is approximate only, since the following assumptions are made in its derivation:- (a) the fluid is incompressible, (b) the disk is an infinitely thin ellipsoid, (c) the disk is rigidly held, (d) the disk itself is rigid, (e) the scattered sound can be neglected, (f) the fluid is non-viscous and (g) there is no vortex motion around the disk.

KING [16] has performed a theoretical analysis of the RAYLEIGH disk, assuming the disk to be freely suspended instead of rigidly held, and including diffraction effects. His corrected formula for the case of a plane progressive wave, assuming the wave-length is large compared to the disk diameter, reduces to

$$L = \frac{4}{3}\rho_0 a^3 \overline{\dot{\xi}^2} \sin 2\varphi \left(\frac{m_1}{m_1 + m_0}\right) \qquad \text{(I, 16)}$$

Where m_1 = mass of disk
m_0 = virtual or added mass due to the displacement of fluid by disk
$= \frac{8}{3}\rho_0 a^3$

KING's original formula also includes terms of the order $(\beta a)^2$, where $\beta = \frac{\omega}{c}$. The finite thickness of the disk may be accounted for by replacing m_1 by $m_1 - m_w$ where m_w is the mass of the medium displaced by the disk. This correction is small for calibrations in air, but may be appreciable in water. The assumption of an incompressible fluid should remain valid, provided $\beta a \gg 1$.

An experimental investigation by MERRINGTON and OATLEY [17] has led to an evaluation of the effects of viscosity and circulation around the disk. They deduced that the effects of viscosity were small, but that the presence of vortex motion around the disk could produce a 10% discrepancy in the KING formula. These effects were studied by SCOTT [18], who presented an empirical correction curve to be applied to the torque calculated by the KING formula.

Mechanical resonances in the disk itself have been found to produce errors in the determination of the particle velocity at the frequencies in the vicinity of resonance [19]. Measurements at these frequencies should be avoided.

In summary, the absolute measurement of particle velocity can be performed with reasonable accuracy using the RAYLEIGH disk, and applying the corrected KING formula. BERANEK [11] presents a discussion of the practical details in the design and calibration of RAYLEIGH disks. Consideration is given to disk dimensions and materials, suspension fibres, and torque measuring methods.

(ii) *Smoke Particle Method*

Direct determination of the particle *amplitude* in a sound wave may be accomplished by measurement of the length of the traces of smoke particles suspended in the sound wave. This technique was first advanced by CARRIÈRE [20], and later used by ANDRADE [21] and by ANDRADE and PARKER [22]. If the smoke particles are assumed to be spherical and of known radius, it is possible to compute their motion in the sound field. The ratio of the amplitude of motion of the spherical particle to that of the surrounding medium can be determined from a formula due to STOKES. For smoke particles with an effective radius of 0.3 μ or less, such as cigarette smoke particles, this ratio is fairly close to unity, even up to a frequency of 5000 c/s.

Measurement of the amplitude of motion of the smoke particles is accomplished by observation through a microscope having a calibrated eyepiece scale. The particles should be well illuminated by means of an arc lamp and lens system. Suitable apparatus of this type is described by SCOTT [18], who used the smoke particle method for comparison with RAYLEIGH disk measurements.

(iii) *Hot Wire Method*

By the cooling of an electrically heated wire (of Pt or Ni, about 0.001″ diameter) the velocity amplitude in a resonator or column of air can be measured [23, 24]. The calibration is done by oscillating a similar wire on a tuning-fork in still air (*cf.* Chap. 13, § 30).

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§ 4. Primary Sources of Sound

(i) *Thermophone*

When an alternating current is passed through a thin strip of metal (foil) or circular wire, heating effects are produced, causing the temperature of the conductor to increase and decrease with respect to the surrounding medium. This gives rise to an acoustic wave whose amplitude is proportional to the variation in temperature near the foil [25]. If the foil or wire is mounted in a cavity, it is possible to compute the sound pressure developed in the cavity in terms of the current in the foil. A microphone coupled to the cavity can therefore be calibrated in terms of the known sound pressure in the cavity.

In practice, an alternating current I_1 is superimposed on a steady current I_0 in the conductor, with $I_0^2 \gg I_1^2$. This produces pressure variations at the fundamental frequency of I_1, with negligible second harmonic distortion.

A theoretical treatment of the thermophone was originally developed by ARNOLD and CRANDALL [26], and refinements of this theory were offered by WENTE [27], SIVIAN [28], BALLANTINE [29], GEFFCKEN and KEIBS [30], and COOK [6]. BERANEK [11] presents a summary of these theoretical considerations, together with a discussion of the practical details in the calibration procedure.

The accuracy of the thermophone method has been evaluated by comparison with reciprocity calibrations. Using the formulas of GEFFCKEN and KEIBS [30], agreement within 1.0 db can be expected.

(ii) *Pistonphone* [27, 31, 32, 33]

The pistonphone consists of a rigid-walled chamber, terminated on one side by the microphone to be calibrated and connected to a small cylinder on the opposite side. A piston in this cylinder is driven in and out with a sinusoidal motion, producing variations in pressure within the chamber. A microscope is provided to observe the amplitude of motion of the piston. The r.m.s. sound pressure produced in the cavity can be computed from the formula

$$p = \frac{\xi_0 S \chi p_0}{v} \qquad \text{(I, 17)}$$

Where v = effective volume of the enclosure with the piston in mean position. (The effective volume includes the volume of the cavity plus an equivalent volume due to the finite microphone diaphragm impedance.)

p_0 = atmospheric pressure

χ = ratio of specific heats for the gas

S = area of piston

ξ_0 = r.m.s. amplitude of motion of the piston, where a positive sign indicates an inward movement of the piston.

(iii) *Electrostatic Actuator*

At very high frequencies, closed coupler pressure calibrations of microphones are not feasible because of wave motion in the cavity. Recently, several laboratories have used an earlier apparatus, the electrostatic actuator [29], for obtaining pressure calibrations of miniature condenser microphones. A fixed slotted plate is mounted very near the diaphragm of the microphone (Fig. 15), and an alternating

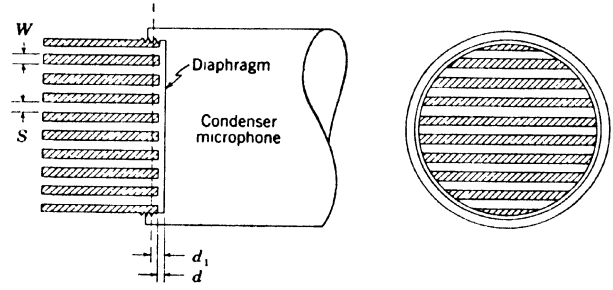


Fig. 15. Cross-sectional sketch of an electrostatic actuator mounted in front of the diaphragm of a condenser microphone. (From BERANEK, *Acoustic Measurements*).

plus a steady potential are applied between the slotted plate and the diaphragm. The alternating potential should be small compared with the steady polarizing voltage.

The effective alternating pressure on the microphone diaphragm may be computed from the formula

$$p = \frac{E_0 e}{4\pi d_1^2} \qquad \text{(I, 18)}$$

where p = rms pressure

E_0 = direct polarizing potential

e = r.m.s. alternating potential

d_1 = effective spacing between grill and microphone diaphragm.

In applying this formula, a consistent system of units must be used, such as the MKS system. The effective spacing d_1 will be slightly greater than the actual spacing d because of the presence of the slots. The necessary corrections in terms of the slot width S and the width of the grill sections W have been presented by BALLANTINE [29] and BERANEK [11].

An advantage of this method is that a uniform pressure is obtained over the surface of the diaphragm, independent of the frequency and the characteristics of the medium. The method can be used to extend condenser microphone calibrations to ultrasonic frequencies.

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CHAPTER 4

BEHAVIOUR OF ABSORBING MATERIALS

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§ 1. Introduction

Absorbing materials like wood fibre plates, textiles, resonant panels etc. are wanted in practice mainly in the following cases:

(a) to shorten the reverberation time of auditoria (see Chap. 6, § 3),
(b) to eliminate echoes in auditoria (see Chap. 6, § 5),
(c) to improve acoustic conditions in noisy offices, factories and the like (see Chap. 8, § 4),
(d) to reduce the sound level in air ducts with the aid of an adequate lining (see Chap. 8, § 4).

In this chapter we shall not enter into the mathematical treatment of these problems themselves, but only give the basic knowledge of the properties of the materials used for these purposes. The experimental technique for the measurement of these properties will only be given in outlines.

Sound absorption is practically always a question of conversion of sound energy into heat with the aim of *reducing the reflection of sound* back into the room or enclosure where the sound is present.

The total energy impinging on a wall will partly be reflected, partly absorbed in the wall itself and partly transmitted to the space behind the wall. Labelling these energy fractions:

the energy reflection coefficient	$\|r\|^2$
the absorption coefficient	a
the energy transmission coefficient	$\|t\|^2$

respectively, we obviously have

$$|r|^2 + a + |t|^2 = 1. \qquad \text{(I, 19)}$$

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The complicated writing of the symbols for the energy reflection and transmission coefficients originates in the wish to reserve the symbols r and t for analogous complex quantities, *viz.* the amplitude reflection and transmission coefficients, to be introduced further on.

An excellent absorbing material absorbs, say, 90%. It is often argued by laymen that such materials should be very effective for sound insulation purpose, *i.e.* for the prevention of sound transmission from one room to another, since 'practically all' sound energy disappears. This reasoning is almost completely wrong. If the remaining 10% should be transmitted entirely the transmission loss would be only 10 decibels (db), a rather poor result indeed. A wall having a good transmission loss of, say, 50 or 60 db only transmits 10^{-5} or 10^{-6} of the incident energy and the absorption coefficient should be as high as 99.99(9)%, if the absorption is to be the only reason of the transmission loss. This being practically impossible it is clear that absorbing layers alone are almost useless for insulation purposes.

Most practical constructions with a high transmission loss ($|t|^2$ very small) are almost completely reflecting ($|r|^2 \approx 1$) so one might say that a low transmission is obtained by greatly increasing the reflection, whereas reflection is reduced as a rule by increasing the absorption.

§ 2. Oblique and Normal Incidence

In the majority of cases where absorbing materials are employed they have to absorb sound impinging from practically all possible directions, *e.g.* in auditoria and offices. In the case of an echo that has to be counteracted by absorption a more or less well defined direction of incidence can be given for which the absorption shall be high. From this it is clear that the absorption coefficient should be known as a function of the angle of incidence of the sound wave.

Now the architect who has to find the appropriate material for a given case has to make a choice from a great variety of materials differing in price, absorption coefficient as a function of frequency, thickness, ease of application, aesthetical respect etc. If the absorption of the material should be given furthermore as a function of the angle of incidence the mutual comparison of the materials would hardly be possible any longer.

One possible way to get rid of the dependence upon the angle of

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incidence is to give the absorption coefficient averaged over all possible directions (§ 13). The advantage of doing this is that this average coefficient can be used directly in many practical problems. Moreover it can be measured directly with the reverberation method (Chap. 5) that in its main features is an imitation on a small scale of practice. A reverberant test room of some hundred cubic metres volume or more is used. The time of reverberation of the room decreases when the sample of *ca.* 10 m^2 is put into it. From the times of reverberation before and after the application of the sample its coefficient of absorption for random incidence can be calculated. The main disadvantages from a practical point of view are the well-known difficulties as to the accuracy and reliability of the results. The absorption coefficient found for a given sample varies with the laboratory carrying out the test and even in the same laboratory with the mode of application of the sample in the reverberation room and the number of m^2 of the sample used. These discrepancies are so great [1] that intercomparison of results from different laboratories is hardly possible, whereas the way in which the measurements are carried out in one and the same laboratory even should be standardized. Nevertheless the reverberation method for measuring sound absorption is indispensable in many cases.

It has turned out feasible to calculate the absorption behaviour of not too complicated materials and constructions from basic properties like air resistance, thickness, porosity etc. These theoretical results can therefore be used to understand the absorption mechanism of a material involved and to indicate the way to further improvement of the material. A very clear example of this is found in the class of porous absorbers behind perforated panels.

Since theory gives in the first place the absorption behaviour at normal incidence a method for measuring at normal incidence was very desirable. Many of these have been developed to give reliable results which are independent of the testing laboratory. Usually a standing wave is set up and the instruments so used are generally called interferometers (see § 12). They consist of a tube with a loud-speaker and the sample at either end. The sound pattern in the tube, generated by the loud-speaker, is examined in a suitable way and the absorptive behaviour of the sample at normal incidence is calculated from the measurements. On account of the great scientific merits of

this method, results of interferometer measurements will be given. Practice requires also however information of the absorption coefficients at oblique incidence or at least an average value for random incidence. This information can be given from a simple interferometer measurement provided one condition is fulfilled *i.e.* the material involved should be 'locally reacting' (see § 13). Often this is not the case and then only approximate figures for oblique incidence can be computed from the known behaviour at normal incidence.

Summarizing we can say that the reverberation method would be of very great value for practice if the results were more accurate and reliable, whereas interferometer results at normal incidence have the drawback of being inaccurate for oblique incidence. Therefore both methods have to be used, the former for gathering practical results, the latter *also* for research and development purposes.

§ 3. The Various Absorption Mechanisms

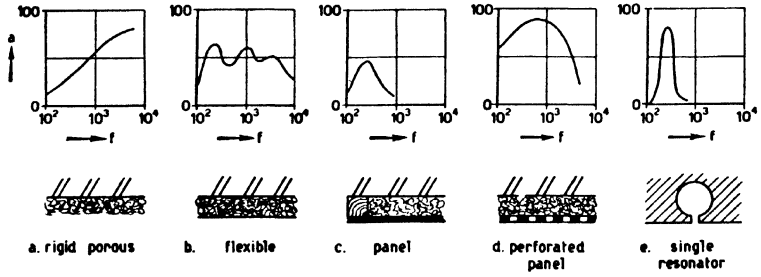


Fig. 16. Five fundamental absorption mechanisms and their typical absorption characteristics.

In Fig. 16 the most important ways of absorption are given in a schematic manner. Fig. 16a symbolizes the group of non-flexible porous materials like acoustic plasters. They absorb, thanks to the viscosity of air. The impinging sound causes the air inside the pores to vibrate. The relative motion of air and solid skeleton entails dissipation of sound energy into heat by air friction. Moreover exchange of heat between the heated compressed air (or cooled rarified air) and the solid skeleton contributes to sound energy losses. The principal quantities that govern the behaviour are the thickness of the layer, the frequency, the air resistance, the porosity and the

structure factor, the latter being a measure of the irregularities in the form of the pores and their directional distribution. The layers need not be homogeneous throughout. Moreover such porous rigid layers may be used at some distance in front of a rigid wall, thus providing a layer of air between the material and the wall. The result is higher absorption values in the low frequency range. As a rule porous layers of rigid structure directly fixed to a wall absorb the higher frequencies (say above 500 c/s) more effectively than the lower ones (see Fig. 22, 24, 26, 27, 28, 29).

A prerequisite for absorption by layers of this type is that the pores are accessible to the sound waves, so the surface too should be porous, not only the interior. Dust and dirt easily adhere to the surface. Cleaning is not very easy and painting is only allowed if special care is taken not to fill up the pores. Special water paints may be used to this end.

A class of materials that has not the latter disadvantages is that of the compliant layers (Fig. 16b) .Their surface may be non-porous but then the surface coating should preferably be lighter than 0.1 kg/m^2 in order to avoid absorption in a too narrow frequency band. With a view to the necessarily rather high flexibility the interior should almost certainly be porous. The absorption is due to internal friction in the compliant skeleton, air friction and heat exchange. Quantities that are of importance for their behaviour are (besides those mentioned for the porous materials) the surface density of the surface coating, the stiffness and the density of the skeleton.Properly designed materials of this class begin already at low frequencies, say 200 c/s, to absorb reasonably. At the higher frequencies the absorption coefficient tends to decrease owing to the mass of the surface coating. Painting is not too dangerous if not too much mass is added to the surface. Pure representatives of this class are not used in practice. It is believed however that many of the modern soft fibrous layers of open texture absorb partly in virtue of their flexibility. Some of them have such a high air resistance that the sound waves certainly compress more or less the whole material.

Light panels mounted at some distance in front of a rigid wall absorb also, thanks to their capability of giving way to the sound waves. At normal incidence they can be treated as a mass (the panel) mounted on a spring (the layer of air). This system has a resonance frequency

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(p. 74) at which vibration is easily set up by the impinging sound. Due to internal friction in the panel and energy losses near the clamping of the framed panels absorption occurs. Practical constructions normally have their resonance frequency rather low and therefore absorb mainly in the low frequency range (say 50-700 c/s). They may therefore be used advantageously in auditoria as a welcome addition to the porous surfaces (*e.g.* carpets). Their behaviour is mainly governed by the mass per square metre of the panel, the thickness of the layer of air and the amount of frictional damping.

Perforated panels (Fig. 16d) form a class of absorbing constructions that are in wide spread use nowadays. The incoming sound wave leaves the perforated heavy panel practically motionless, but forces the air inside the perforations to vibrate to and fro. The air inside the porous material appears to act as a spring on which the little masses of air within the holes are mounted. The system has a resonance frequency the magnitude of which can be controlled by varying the depth of the layer of air and the perforation. The porous filling material makes the construction absorb in a satisfactorily broad frequency range. Moreover this filler is wanted to ensure reasonable absorption at oblique incidence. These materials combine the advantages of panels as to cleaning and painting with a high absorption coefficient in an important frequency range. They absorb namely the low frequencies better than normal porous absorbers. Their behaviour can be treated mathematically as that of the resonators (Fig. 16e). The main variables governing the behaviour of perforated panels are the perforation, the thickness of the layer of air and the damping due to the filler. The mathematical treatment is moreover so simple that the development of suitable perforated panels for any purpose is no longer a difficulty.

§ 4. The Acoustical Impedance, Reflection and Absorption

Suppose a plane travelling harmonic wave to impinge in a *normal direction* upon a surface, *e.g.* a wall covered with some sort of absorbing material. A plane travelling wave will be reflected with diminished intensity due to the absorption of the wall. If the complex sound pressure of the incoming and reflected wave be labelled p_i and p_r

respectively, and the complex reflection coefficient r at a point is defined as usual to be the ratio

$$r = \frac{p_r}{p_i} \tag{I, 20}$$

at that point, it can be shown (*cf.* Chap. 1, § 15) that

$$r = \frac{Z - \rho c}{Z + \rho c}, \tag{I, 21}$$

where ρc is the characteristic or wave impedance in air, ρ is the density of air (in kg/m³ in MKS-units), c the sound velocity in air ($\pm$ 340 m/sec) and

$$Z = \frac{p}{u} = \frac{p_i + p_r}{u_i + u_r} \tag{I, 22}$$

is the specific acoustic impedance at that point. u_i and u_r are the particle velocities in the direction of the reflecting surface due to the incoming and reflected wave, respectively. The validity of (I, 21) can be shown immediately by substituting

$$p_i = \rho c u_i, \quad p_r = -\rho c u_r \tag{I, 23}$$

in (I, 22) and using (I, 20).

Eq. (I, 21) holds at any point in the air. If one chooses a point just in front of the reflecting surface, Z is the impedance of the surface and r the reflection coefficient just before the surface. Since the intensities of the two waves are proportional to $|p_i|^2$ and $|p_r|^2$, the *energy* reflection coefficient is

$$|r|^2 = \left| \frac{Z - \rho c}{Z + \rho c} \right|^2 \tag{I, 24}$$

(*cf.* eqn. (35) of Chapter 1) and if the non-reflected energy fraction is absorbed or assumed to be so, the energy absorption coefficient obviously is

$$a = 1 - |r|^2 = 1 - \left| \frac{Z - \rho c}{Z + \rho c} \right|^2. \tag{I, 25}$$

So the reflection and absorption are completely governed by the impedance of the surface, apart from the constant characteristic impedance of the air.

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Although the absorption coefficient a gives sufficient information for most practical applications, the impedance, which determines also the absorption coefficient, is a much more attractive quantity to characterize the surface.

The complex impedance Z may be written either as

$$Z = (p + iq)\,\rho c,$$

or as

$$Z = |Z| \exp i\varphi,$$

where $|Z|$ is obviously the modulus of Z, and φ its argument.

The complex reflection coefficient will be written

$$r = |r| \exp i\Delta.$$

According to (I, 21) a certain value of r belongs to every given value

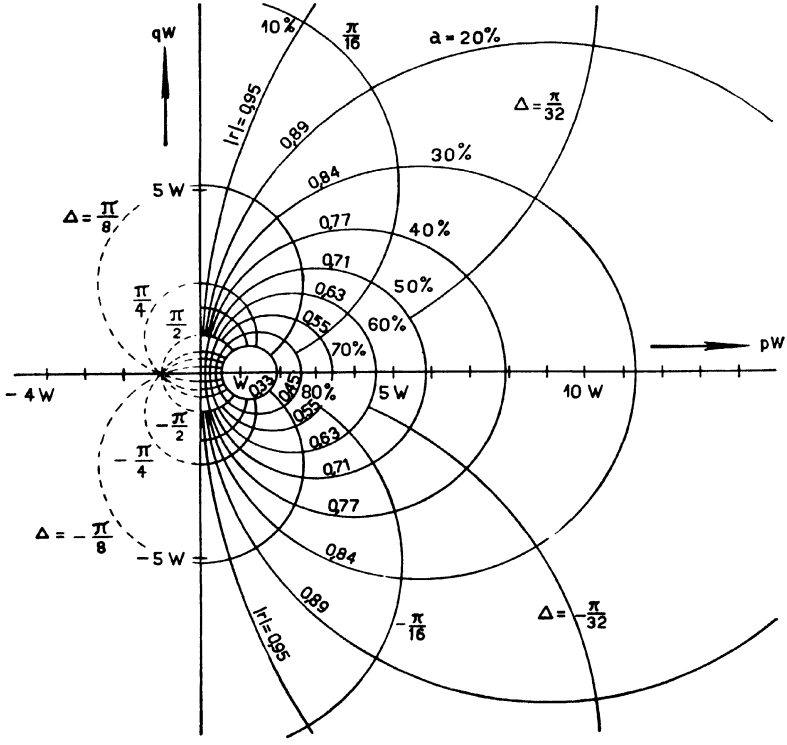


Fig. 17. Representation of the equation $r = (Z - W)/(Z + W)$ in the complex Z-plane. $Z = (p + iq)W$ and $W = \rho c$ = wave impedance of medium in front of the reflecting surface.

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of Z and vice versa. Much more lucidly than by eq. (I, 21) the interrelation of r and Z can be shown by a graph, either by Fig. 17, 18 or 19. In Fig. 17 the complex r-plane is represented in the complex Z-plane, *i.e.* for every couple of values $|r|, \Delta$ the corresponding impedance Z is computed and the couple $|r|$, Δ is noted at the corresponding point in the Z-plane. Subsequently the points of equal $|r|$ and Δ in the Z-plane have been connected. The family of curves $|r| =$ constant turn out to be a set of circles, as well as those of $\Delta =$ constant. They intersect orthogonally. The values of $|r|$ and Δ belonging to the various circles are indicated along them. The circles $|r| =$ constant are at the same time circles of constant absorption coefficient. The corresponding values are also given in the figure.

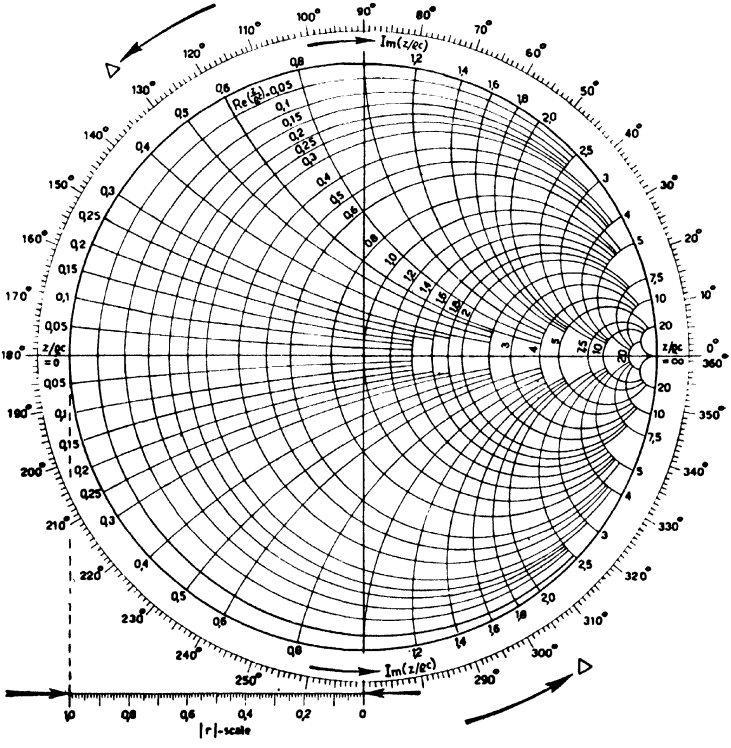


Fig. 18. Representation of $Z/\rho c$ in rectangular coordinates in the complex r-plane ($r = |r| \exp i\,\Delta$).

In Fig. 18 the reverse procedure is followed. The $Z/\rho c$-plane has been represented in the r-plane here, and, since $|r|<1$ for all passive surfaces, this figure is limited within a circle of $|r| = 1$. The value of

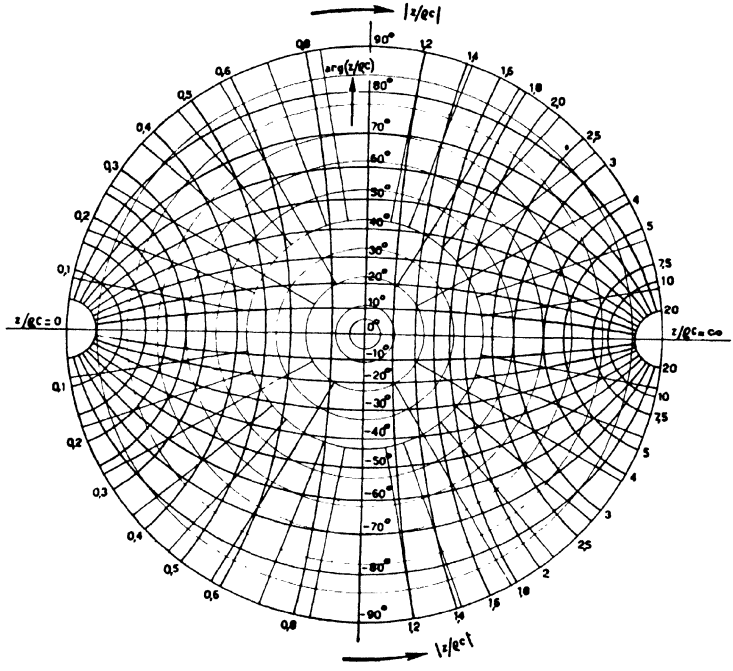


Fig. 19. As Fig. 18, but $Z/\rho c$ in polar coordinates ($Z/\rho c = |Z/\rho c| \exp i\varphi$).

$Z/\rho c$ corresponding to a given value of r is indicated through the intermediary of circles of constant p and q, the real and imaginary part of $Z/\rho c$ respectively. For some applications polar coordinates for $Z/\rho c$ are to be preferred. This leads to Fig. 19, where the value of $Z/\rho c$ is represented with the aid of two sets of circles of constant $|Z/\rho c|$ and φ respectively.

§ 5. Outlines of the Computation of Impedances

In many sound absorbing materials damped progressive waves are of primary importance. A well-known example is that of porous

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absorbents like acoustic plasters and the like. Another simple example is that of homogeneous layers of a flexible material. The motion of plane waves in such material is governed by

$$-\frac{\partial p}{\partial x} = \rho \frac{\partial u}{\partial t} \tag{I, 26}$$

$$-\frac{\partial p}{\partial t} = K \frac{\partial u}{\partial x} \tag{I, 27}$$

in exactly the same way as in air or another gas. p is again the sound pressure, u the particle velocity, ρ the density and K a specific stiffness. Depending upon the material under consideration the quantities will be defined more precisely later on. *E.g.* the concept 'particle velocity' needs definition in the case of a porous plaster since under close examination the material is not at all homogeneous.

Properly speaking (I, 26) and (I, 27) are only valid for simple harmonic vibrations. In fact either ρ or K, or even both of them, can be complex, thus causing the waves in the material to be damped.

$$\rho = \rho_r (1 - i \tan \delta_\rho) \tag{I, 28}$$

$$K = K_r (1 + i \tan \delta_K) \tag{I, 29}$$

ρ_r, K_r are the real part of ρ, K respectively, δ_ρ, δ_K are called the angle of loss of ρ, K respectively.

It is easily shown by substitution that (I, 26) and (I, 27) are satisfied by

$$p = \hat{p} \exp (-i\omega \sqrt{\frac{\rho}{K}}\, x), \tag{I, 30}$$

$$u = \frac{\hat{p}}{\sqrt{K\rho}} \exp (-i\omega \sqrt{\frac{\rho}{K}}\, x), \tag{I, 31}$$

if $\hat{p}$ is supposed to vary as $\exp i\omega t$. (I, 30) and (I, 31) represent the sound pressure and the particle velocity in a progressive wave.

If we put

$$c = \sqrt{\frac{K}{\rho}}, \tag{I, 32}$$

p and u vary in space and time as

$$\exp i\omega (t - \frac{x}{c}),$$

the well-known expression for a progressive wave with sound velocity

c. If K and/or ρ are complex, the sound velocity is complex too, which is another way of saying that the travelling wave is damped.

Frequently the propagation exponent γ or its components $\alpha + i\beta$ (p. 6) are used.

In a progressive plane wave a fixed complex ratio p/u exists, independent of the time and place. This is seen from (I, 30) and (I, 31) to be

$$W = \frac{p}{u} = \sqrt{K\rho} = \rho c. \qquad \text{(I, 33)}$$

This ratio of sound pressure and particle velocity is generally called the characteristic or wave impedance, even when ρ and/or c are complex. For plane waves a material is completely described by its propagation constant γ and wave impedance W, both as a function of frequency.

As an application we shall describe the way in which the impedance of a two-layer-system or a multiple layer system can be computed (see Fig. 20). A plane wave is supposed to impinge normally on the

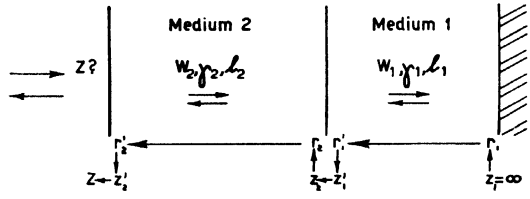


Fig. 20. The computation of the impedance of a multilayer system.

system. In each layer an incoming and a reflected wave can be distinguished. The computation starts at the rigid wall and progresses gradually to the front side of the system. The various steps are indicated symbolically by quantities and arrows at the bottom of Fig. 20; they run in formulae:

$$\begin{aligned}
Z_1 &= \infty \\
r_1 &= (Z_1 - W_1)/(Z_1 + W_1) \text{ (see eqn. (I, 21))} \\
r'_1 &= r_1 \exp(-2\gamma_1 l_1), \text{ since } p_r \text{ [} p \text{] varies as } \exp(-\gamma l) \text{ [} \exp(+\gamma l) \text{]}. \\
Z'_1 &= W_1 (1 + r'_1)/(1 - r'_1) \text{ (see eqn. (I, 21))} \\
Z_2 &= Z'_1 \text{ since } p \text{ and } u \text{ are continuous at the boundary.} \\
r_2 &= (Z_2 - W_2)/(Z_2 + W_2) \text{ (see eqn. (I, 21))} \\
r'_2 &= r_2 \exp(-2\gamma_2 l_2) \qquad \text{(see } r'_1\text{)} \\
Z'_2 &= W_2 (1 + r'_2)/(1 - r'_2) \text{ (see eqn. (I, 21))} \\
Z &= Z'_2 \qquad \text{(continuity)}
\end{aligned}$$

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The whole procedure is based on eq. (I, 21), the relation between r and x [$r = r_0 \exp(-2\gamma x)$] and the continuity of Z at a boundary. It may be replaced by one single rather complicated recurrence formula for the impedance, well-known in the theory of four-terminal networks:

$$Z_j/W_j = \frac{Z_i \cosh \gamma_j l_j + W_j \sinh \gamma_j l_j}{Z_i \sinh \gamma_j l_j + W_j \cosh \gamma_j l_j}. \qquad \text{(I, 34)}$$

We see from (34) that the impedance of a layer, rigidly backed, is $Z = W \coth \gamma l$, if l is its thickness.

§ 6. Theory of Absorption by Rigid Porous Layers

Many commercial materials belong to this important class, either approximately or actually, *e.g.* all porous materials with cement as binder, fibreglass, rock wool, wood fibre plates etc. In exceptional cases the flexibility of the material probably plays a part in the absorption process (see § 8).

The study of the behaviour even of the rigid materials is very much complicated by the unknown structural peculiarities. To simplify things we shall confine ourselves in the beginning to the propagation of waves in cylindrical pores of radius R. Since motion of the air is hindered by viscosity we expect a resistive term in the equation of motion, something like eqn. (40) of the Introduction, which may be written in the form

$$-\frac{\partial p}{\partial x} = \rho^* \frac{\partial u}{\partial t}, \qquad \text{(I, 35)}$$

if now ρ^* is considered as a complex quantity †

$$\rho^* = \rho - i\frac{r}{\omega}. \qquad \text{(I, 36)}$$

It goes without saying that the last two equations only have a physical meaning if a simple harmonic wave is considered.

Theoretical and experimental studies have taught that r is by no means a constant. It depends upon frequency and is 'slightly' complex

† The symbol r will be used in this chapter henceforward in resistive terms as a measure for the energy losses due to all sorts of friction. In the case of air friction r is called the specific air resistance. The symbol r will be used only very exceptionally for the complex reflection coefficient (*e.g.* in § 12).

itself. We borrow at once expressions for ρ^* for low and high frequencies from RAYLEIGH [2] and a general expression for all frequencies from CRANDALL. [3] In order to avoid confusion this special value of ρ^* for cylinders shall be indicated by $\rho^*_{cyl.}$

$$\left.\begin{aligned} \rho^*_{cyl.} &= \frac{4}{3}\rho\left(1 - i\,\frac{6}{\zeta^2}\right) && \left(\begin{matrix}\text{low fr.}\\ \text{or } \zeta<1\end{matrix}\right)\\ \rho^*_{cyl.} &= \rho\left(1 - 2\,\frac{\sqrt{-i}}{\zeta}\right) && \left(\begin{matrix}\text{high fr.}\\ \text{or } \zeta> 10\end{matrix}\right)\\ \rho^*_{cyl.} &= \frac{\rho}{1 - \dfrac{2}{\zeta\sqrt{-i}}\,\dfrac{J_1(\zeta\sqrt{-i})}{J_0(\zeta\sqrt{-i})}} && \text{(all fr.)} \end{aligned}\right\} \qquad \text{(I, 37)}$$

in which ζ is a dimensionless quantity

$$\zeta = \sqrt{\frac{\omega}{\upsilon}}\cdot R \qquad \text{(I, 38)}$$

υ is the kinematic coefficient of viscosity for air, and J_0, J_1 are BESSEL functions of the order zero and one respectively. In Fig. 21 the quantity $\rho^*_{cyl.}/\rho$ is shown in the complex plane as a function of its variable ζ.

At *low* frequencies the real part of $\rho^*_{cyl.}$ exceeds the density of free air by 33%. This is due to the immobilization effect of the viscosity. The air in close proximity to the walls of the pore is hindered considerably in its motion, in much the same way as it would be by an increase in density. The imaginary part of the complex density is inversely proportional to ω, thus pointing to an air resistance that is independent of frequency (compare eq. (I, 36)), and equal to its steady flow value

$$r = \frac{8\eta}{R^2}$$

as was to be expected from POISSEUILLE's law.

At *high* frequencies the density is practically normal, the resistance now being proportional to $\sqrt{\omega}$. This result was first obtained by HELMHOLTZ. When a steady flow is started in a pore there is a small time lag between this moment and the time at which the velocity distribution in the pore has attained its steady form (POISSEUILLE distribution). When a low frequency sound wave progresses in the pore the change as a function of time of the flow at a certain point

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is so slow that the velocity distribution is always quasi-steady, giving rise to the normal POISSEUILLE value of air resistance. If the frequency is high there is no longer time for building up the steady distribution. The air in the centre of the pores has already changed its direction of motion before the walls have notified it of their presence. The effect of viscosity remains confined to a thin layer of air near the walls, the interior of the air content vibrating to and fro en bloc (*cf.* § 19 of the Introduction and the 'annular effect', § 25 of Chapter 18)

When practical materials are dealt with, several concepts remain vague without a further definition. This applies, *e.g.*, to the concept 'air velocity', since the velocity varies from point to point in the same plane $x =$ constant. We define u in this case as the mean value of the component of the air velocity perpendicular to the plane $x =$ constant under consideration, it being taken zero where solid material is encountered, so that u is identical with the volume current per unit surface (velocity of volume displacement).

We will next show that, if u is defined according to the definition given, we should write for practical materials

$$\rho^* = \frac{s}{P}\ \rho^*_{cyl.}$$

(s and P to be defined next); moreover it can be shown [4] that now ζ should be reckoned from

$$\zeta^2 = \frac{8\omega s\rho}{Pr},$$

provided the pores are essentially cylindrical in shape and of equal diameter. In this expression r stands for the steady flow value of the air resistance. P is the *porosity* and is defined as the ratio of the free air content in a material and its overall volume. The lower the porosity P the higher the pressure gradient should be in order to obtain the same volume flow, thus explaining why P occurs in the denominator of ρ^*.

The factor s is called the *structure factor* [5]. It is essentially greater than unity, is frequently between 2 and 10, but values as high as 25 have already been found. Its definition can only be: 'the factor by which $\rho^*_{cyl.}$ should be multiplied, apart from the influence of P, in order to meet experimental results'. In a qualitative way it is easy to explain that an additional factor $s > 1$ should be introduced. In steady flow only part of the free air content is involved. The remaining

fraction is practically or actually motionless because it is contained in 'culs-de-sac', *i.e.* in cavities with one entrance only, or in pores that have an unfavourable direction with respect to the direction of propagation. All these ineffective cavities and pores might have been solid material without changing the equation of motion, *i.e.*, P might have been chosen accordingly smaller. The introduction of s in the numerator has the same effect. It goes without saying that the distinction between motionless and flowing air is too fine. Actually it is impossible to determine the limit. It can be shown [6] that a random distribution of the directions of cylindrical pores yields a factor $s = 3$, even disregarding possible 'culs de sac'.

We shall now give, referring for the derivation again to the literature, an expression that might be the most suitable one for the equation of continuity. Whereas in the equation of motion the problem was to find the complex density ρ^* we are now in search of the complex bulk modulus K^*, needed in the equation

$$-\frac{\partial p}{\partial t} = K^* \frac{\partial u}{\partial x}.$$

We start again with the behaviour of a progressive wave in a cylindrical pore. Disregarding the viscosity from the beginning the problem was studied by KOSTEN [7] along the lines of KIRCHHOFF. This procedure is analogous to that of CRANDALL [3] who gave the expression for the complex density disregarding thermal conductivity at the outset. Labelling this value of K^* for cylinders $K^*_{cyl.}$ we arrive at

$$\left.\begin{aligned} K^*_{cyl.} &= p_0\left[1 + \frac{1}{8} i \frac{\chi - 1}{\chi}(\mathfrak{P}_\mathfrak{r}\zeta)^2\right] && \begin{pmatrix}\text{low fr.}\\ \text{or}\,\zeta<1\end{pmatrix} \\ K^*_{cyl.} &= \chi p_0\left[1 + 2\sqrt{-i}\,(\chi-1)/(\mathfrak{P}_\mathfrak{r}\zeta)\right] && \begin{pmatrix}\text{high fr.}\\ \text{or}\,\zeta>10\end{pmatrix} \\ K^*_{cyl.} &= \frac{\chi p_0}{1 + \dfrac{2(\chi-1)}{\mathfrak{P}_\mathfrak{r}\zeta\sqrt{-i}}\dfrac{J_1(\mathfrak{P}_\mathfrak{r}\zeta\sqrt{-i})}{J_0(\mathfrak{P}_\mathfrak{r}\zeta\sqrt{-i})}} && (\text{all fr.}) \end{aligned}\right\} \quad (\text{I, 39})$$

in which

p_0 = the barometric pressure ($\approx 10^5$ N/m^2),

$\chi = C_p/C_v$ = ratio of specific heats (≈ 1.4 for air),

$\mathfrak{P}_\mathfrak{r}$ = Prandtl Number* (≈ 0.86 for air),

$\zeta = \sqrt{\frac{\omega}{\upsilon}}\, R$ as before.

* Prandtl Number is the ratio of the product of viscosity and specific heat to the thermal conductivity of a fluid.

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We see from (I, 39) that at extremely low (high) frequencies the bulk modulus has its isothermal (adiabatic) value p_0 (χp_0), as was to be expected. The dependence of K^*/K_{ad} upon frequency is depicted in Fig. 21, ζ being the measure for frequency. Due account has generally been paid to the complex nature of K^* at high frequencies, not so, however, at low frequencies.

In order to take structural details of practical materials into account the expressions for $K^*_{cyl.}$ of eqn. (I, 39) should obviously be divided by the porosity P since the air in a unit volume of material is apparently stiffer the lower the porosity is.

Summarizing, we get that rigid porous materials are governed by

$$\gamma = i\omega\sqrt{\rho^*/K^*} \qquad K^* = K^*_{cyl.}/P \qquad K^*_{cyl.} \text{ see Fig. 21}$$
$$W = \sqrt{K^*\rho^*} \qquad \rho^* = s\rho^*_{cyl.}/P \qquad \rho^*_{cyl.} \text{ see Fig. 21}$$

$$\zeta^2 = \frac{8\,\omega\, s\,\rho}{Pr}.$$

For the computation of the impedance Z and absorption coefficient a of a layer or composite layers of such materials see § 5. Part of the tiresome computational work can be avoided with Fig. 21, where $\sqrt{K^*\rho^*}$ and $\sqrt{\rho^*/K^*}$ for cylinders are given directly as a function of the variable ζ.

In Fig. 22 the absorption characteristics $a\,(f)$ of layers of various resistances r, structure factors s and thickness l are collected (curves drawn in full). The layers are assumed to be rigidly backed, *e.g.* by fixing them to a brick wall. The porosity P is taken to be 80%. Its influence on the absorption coefficient is not very interesting and rather small. For small values of a, proportionality between a and P may be assumed. The influence of P on high values of a is even less. As a rule, therefore, a high value of P is advantageous, 70% being a reasonable minimum. The thickness l is indicated along the various curves, that of 1 cm thickness being drawn thicker than the others for rapid orientation. The very important frequency range 200–2000 c/s is indicated by vertical lines. The conclusions from Fig. 22 are:

(a) for layers of (infinitely) large thickness the resistance should be as low as possible; the structure factor also.

(b) this is no longer true for layers of small thickness. For $r = 10^4$

and $l = 0.01$ or 0.0316 m the absorption is poor at low frequencies; taking $r = 10^5$ gives a considerable improvement at low frequencies, whereas the loss in absorption at the higher frequencies is not great.

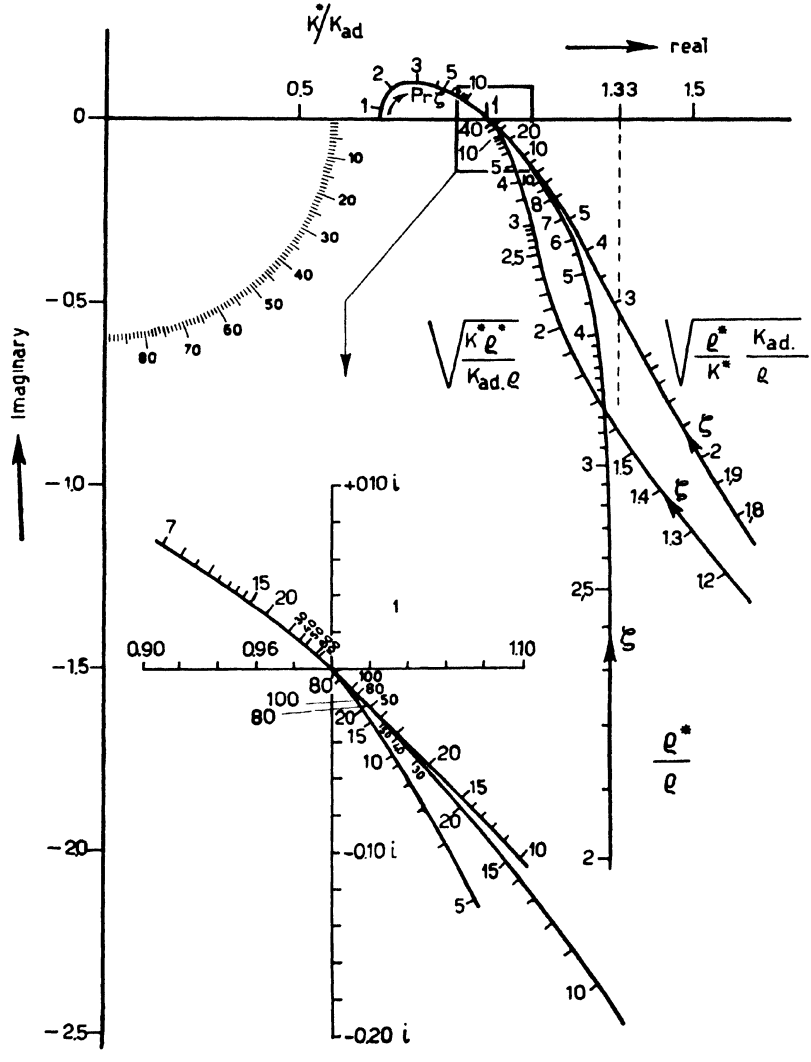


Fig. 21. The complex density, bulk modulus, sound velocity and characteristic impedance in cylinders as a function of $\zeta = \sqrt{\omega/\upsilon}.R$.

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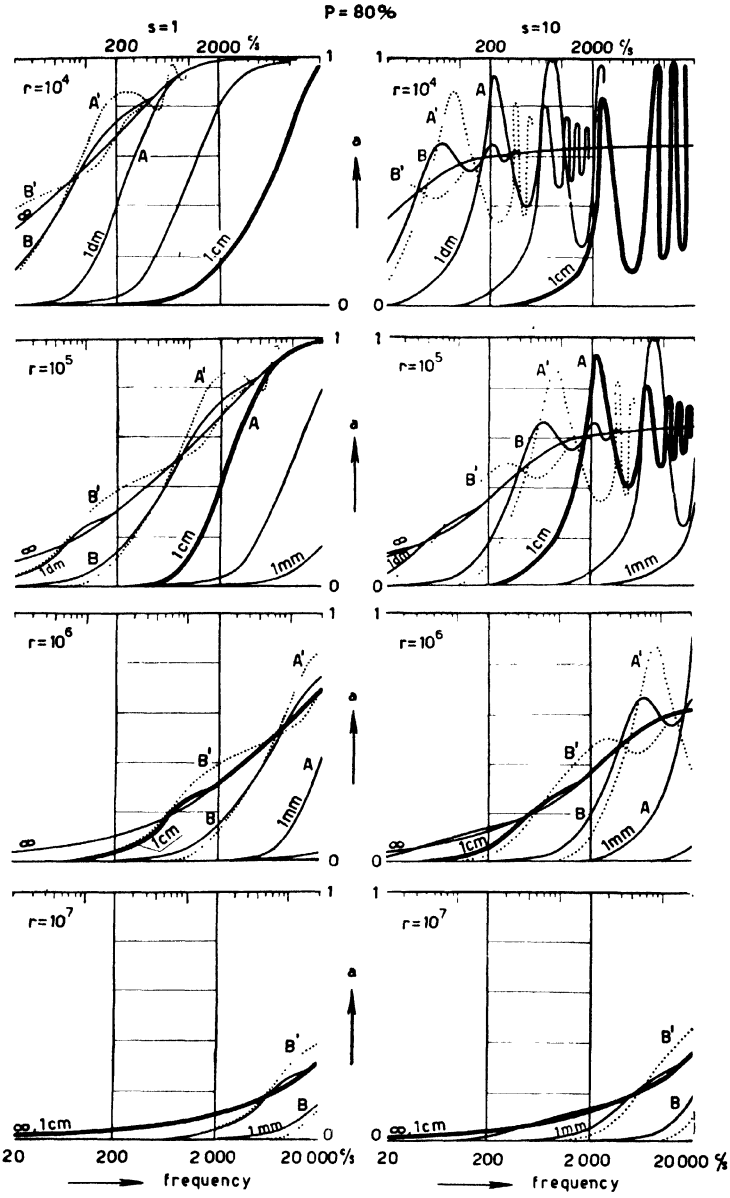


Fig. 22. The absorption coefficient of porous rigid layers, fixed directly to the wall, as a function of frequency, air resistance r, structure factor s and thickness (indicated along the curves; the thickness increases by a factor $3.16 = \sqrt{10}$ from curve to curve from right to left). The porosity P is taken 0.8. The dotted curves A′ and B′ relate to the layers of curves A and B (drawn in full) backed by a layer of air 2.16 times thicker than the layer itself (total thickness of A′ and B′ therefore 3.16 times that of A and B, resp.).

A still higher resistance makes the absorption still more uniform, but the average value (*e.g.* between 200 and 2000 c/s) is greatly impaired. An optimum value of $r = 10^5$ might be suggested. Indeed a value of this order of magnitude (10^5 MKS = 100 CGS) is often encountered in practical materials.

(c) the advantageous influence of s being greater than unity is obvious from the figures for which $r = 10^4$ and $r = 10^5$ for both $s = 1$ and 10. At low frequencies, however, s is of no importance. This is physically clear from the fact that at low frequencies the inertia of the air plays no part, the stiffness of the air being preponderant. So an apparent increase in density is of no importance either.

Although it is certain that several other effects must cause discrepancies between practical and theoretical behaviour measurements on practical materials are well in agreement with the theory presented (see § 7).

Although Fig. 22 provides us with much fundamental knowledge many questions, theoretical and practical, remain, *e.g.*

(1) how does the behaviour change, if the layer of porous material is fastened at some distance in front of the rigid wall?
(2) what is the influence of dirt, cleaning, painting?
(3) how is the behaviour of perforated porous rigid materials to be explained?

Query 1. The change in behaviour due to a layer of air behind a porous layer can roughly be described as an apparent increase of the thickness l of the layer, the apparent increase even being somewhat greater than the thickness d of the layer of air, *viz.* d/P. So, if porous layers of low porosity are considered, a thin layer of air d may have a great influence. Since thick layers generally absorb better than thinner ones a layer of air is usually advantageous. This does not apply to layers of very high resistance r (say $r = 10^6$ or 10^7 MKS). Thin layers of such materials are already acoustically speaking infinitely thick, so that a further apparent increase in thickness has no effect (see however the panel effect § 9). Theoretical examples have been added in Fig. 22 (broken curves A′ and B′). They relate to porous layers of a thickness equal to that of the curves A and B, backed by a layer of air so as to make the combined layer $\sqrt{10}$ times (3 . 16 times) thicker. The simple result given above (apparent increase in thickness d/P) predicts that the dotted curves should coincide approximately with the curves drawn in full for a 3.16 times greater thickness (*i.e.* a shift to the left

by 'one curve'.) This holds indeed as a rough approximation. In comparison with the layer of porous material with the same overall thickness the layer backed by air lacks part of the viscous resistance. This manifests itself clearly in a less 'damped' characteristic (see especially the curves for $s = 10$) and somewhat higher resonance frequencies (a shift by somewhat less than 'one curve' instead of by somewhat more). The advantageous effect of the layer of air is obvious, especially, however, at low air resistance (top of figure).

Query 2. Porous surfaces are liable to collect dust in their pores and to grow dirty, especially where the circulation of air is strong, *e.g.* above radiators of central heating. It is therefore reasonable to require that they should allow of cleaning and/or of washing and painting. It is difficult to draw up practical directives in this respect. As long as rigid materials are concerned a condition of primary importance is that the surface treatment may not hamper the passage of air from the outside inwards, since this is a prerequisite for absorption. So paints containing a minimum of solid material should preferably be used. Oil painting is very dangerous. The detrimental influence of a dense coating is easily understood in a quantitative way from impedance considerations. The coating may be treated as a very thin layer of relatively great air resistance backed by the impedance of the unpainted layer. So the impedance curve of the treated layer, giving the impedance in its complex plane as a function of frequency is essentially the same as that of the untreated layer but for a shift in the direction of the real axis over a distance governed by the air resistance of the coating. In the low frequency range this means an improvement. In the medium and high frequency range, where absorption coefficients above 40% may be found, the absorption is reduced.

A theoretical example is given in Fig. 23. The layer is characterised by the quantities $r = 3.16 \times 10^4, P = 0.8, s = 10, l = 0.0316$ (3.16 cm). The thick curve is the absorption characteristic of the untreated surface, whereas the other curves depict the behaviour of the same layer treated with coatings of air resistance $1 \times 420; 2 \times 420; 4 \times 420; 8 \times 420$, resp. (420 is the characteristic impedance of air). The absorption characteristic of the material is flattened, but on the whole a resistive surface layer is unfavourable.

Query 3. Perforated porous rigid materials of the Acousti-Celotex type are known for their high absorption at frequencies much lower

than for the non-perforated material. Their behaviour might be understood in the following way. If the porous material had been non-porous but for the macroscopic holes (of, say, 6 mm diameter) that are distributed in a regular pattern all over the material, the layer would have

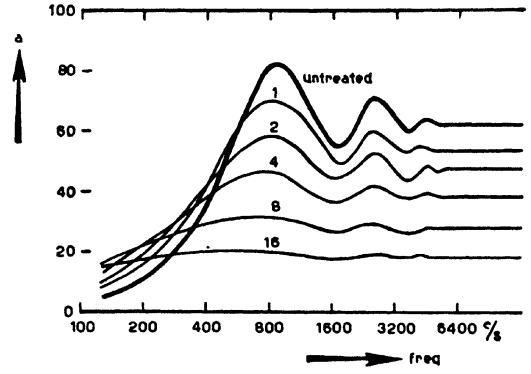
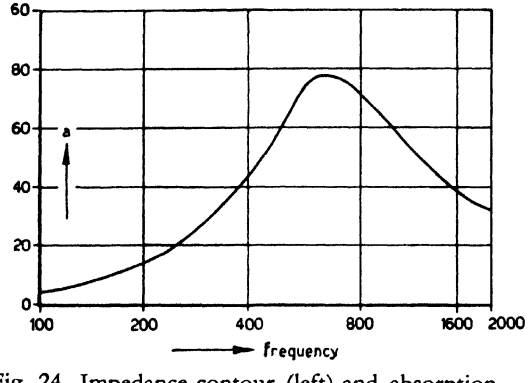


Fig. 23. The effect of painting a porous layer.

had a resonance frequency when $d = \frac{1}{4}\lambda$, with d the depth of the holes and λ the wave-length. The material is however porous thus introducing the possibility for the air that is going to be compressed to escape sideways in the material. The stiffness in the main pores is apparently decreased and since the velocity of sound is $\sqrt{K/\rho}$, a decreased stiffness results in a decreased velocity of sound, *i.e.* in a lowered resonance frequency. Since the air volume in the porous material generally is some 25 times the volume in the main pores, K might



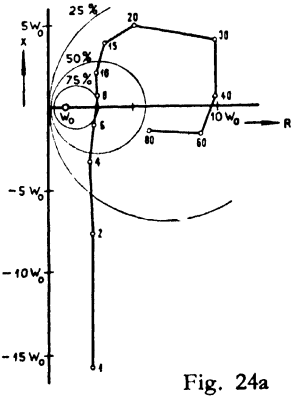


Fig. 24. Impedance contour (left) and absorption characteristic (right) of Celotex C-4 after BERANEK [8].

Fig. 24a

be apparently 25 times reduced, *i.e.* a 5-times reduced resonance frequency is not unlikely. For a thickness of 2.5 cm this yields a resonance frequency of 700 c/s which indeed is often found approximately. An example is given in Fig. 24, a result of BERANEK with Celotex C-4, measured at normal incidence of sound.

The electric analogue of air as a medium for sound waves is well-known to be a long line with distributed self-inductance (series) and capacity (parallel), the density being comparable with the inductance, the compliance (the inverse of the stiffness) with the capacity. In the case of the perforated materials under discussion the compliance of the air in the main pores is increased by the compliance of the air in the porous substance. This additional volume of air can only be reached, however, via a viscous resistance. The electrical analogue of an infinitesimally small part of our acoustic long line is obviously that of Fig. 25, C_1 representing the air in the main holes, C_2 that in the porous substance, the resistance being the translation of the viscous air resistance of the porous substance. The ratio C_2/C_1 is obviously governed by the ratio of the quantities of air in the porous material and in the main holes respectively, so that the resistance is the only parameter that can be chosen to suit experiment. As compared with non-perforated materials an essential difference is that whereas for non-perforated materials ρ was 'very' complex and K not, the reverse is true for the perforated materials.

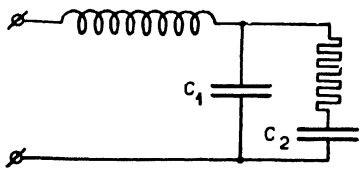


Fig. 25. Electrical analogue to Celotex C-4.

Without following this picture further we may mention that the same explanation might be given in other words by assigning to the material a structure factor s of about 25 that decreases gradually for higher frequencies. The main holes act as 'pores', the porous substance forms the 'cul-de-sac'.

A question of great practical importance is how porous materials may be protected against mechanical damage. Especially the soft materials (glass silk, rock wool, etc.) are in need of such a protection. Perforated panels of metal, hard board and the like are used for this purpose. The influence of such a protection on the absorption behaviour will be considered in § 10; it can be mentioned here, however, that

a sheet, perforated by more than, say, 30% by surface, does not give a noticeably different behaviour. [9] The tendency, especially of sheets that are perforated less, is to shift the absorption characteristic to the lower frequencies, which as a rule is considered to be advantageous.

§ 7. Typical Absorption Characteristics of Porous Layers

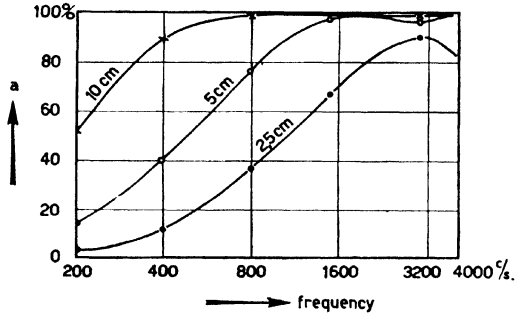


Fig. 26. SCOTT's results on glass wool of three thicknesses.

In Appendix I (at end of book) a survey is given of results for practical materials. The fact that a certain material does appear in the list does not imply that the material in question belongs to the very best

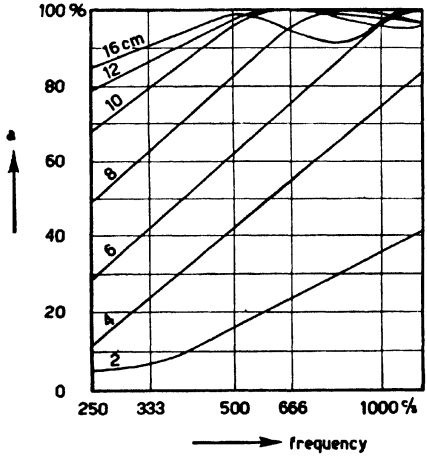


Fig. 27. Results of FERRERO and SACERDOTE on glass wool of various thickness.

materials, nor does not appearing of any material mean that it has proved unsatisfactory.

In this section we confine ourselves to the description of a few illustrative results of measurements at normal incidence of sound.

In Fig. 26 SCOTT's results [10] at normal incidence of sound on rock wool of density 80 kg/m³, porosity 97%, steady flow air resistance 8600 MKS-units and three different thicknesses are shown. They corroborate the theory (compare Fig. 22, for $s = 1$, $r = 10^4$). SCOTT derived from these results the structure factor s to be between 1 and 2. Similar results of FERRERO and SACERDOTE [11] are shown in Fig. 27 relating to nearly the same material with density 90 kg/m³ (compare again Fig. 22 for $s = 1$, $r = 10^4$).

These results are typical indeed of all low density fibrous layers. Their absorption characteristic is smooth; their absorption at high frequencies is already high for thin layers; at low frequencies too for layers of, say, 10 cm thick.

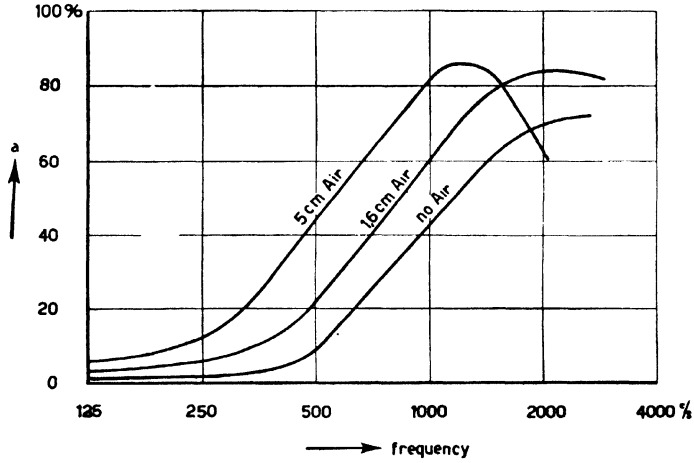


Fig. 28. MARIENS' results on glass wool layers backed by a layer of air.

In Fig. 28 MARIENS' [12] results are given relating to a layer of glass silk of density 80 kg/m³ backed by a layer of air. All curves comply well with theory (compare again Fig. 22 for $s = 1$, $r = 10^4$).

Fig. 29 depicts the behaviour of an acoustic plaster, 21 mm thick, porosity 72%, static air resistance 31000 MKS-units, backed by a layer of air of various thicknesses, indicated along the curves [13]. The

curve without air backing (0 cm) shows up essentially the behaviour of Fig. 22 for $s = 10$. Indeed plasters have as a rule a rather high structure factor. The advantageous influence of the layer of air is obvious. Very thick layers of air shift the absorption peak to the very

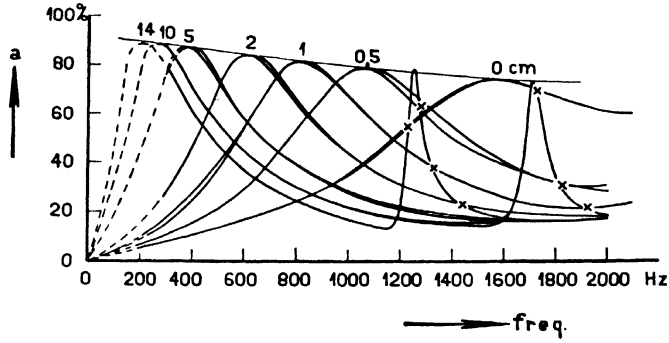


Fig. 29. Absorption of a layer of acoustic plaster, 21 mm thick, backed by various layers of air (indicated along the curves).

low frequencies. As a consequence of the low air resistance the absorption at the high frequencies is then poor. The sharp second resonances of the thick layers are also in compliance with theory.

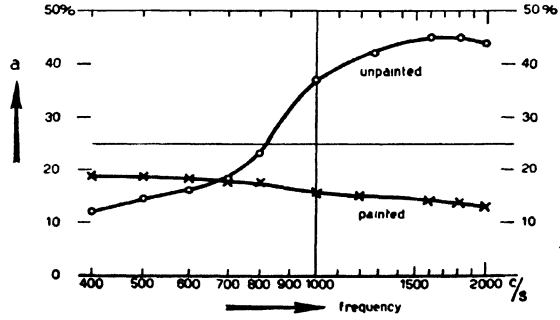


Fig. 30. Measured effect of inadequately painting a porous layer.

Finally Fig. 30 shows the detrimental effect of inadequate painting in agreement with what theory predicts (compare Fig. 23). The sample under test was 21 mm thick and of the sawdust type with cement binder.

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§ 8. Theory of Absorption by Flexible Materials; Some Results

Many materials used as absorbents, *e.g.* rock wool and hair felt, are so flexible that it is rather likely that their solid skeleton will be set into vibration to some extent by the impinging sound waves. As a matter of fact one easily feels with the finger tips the mechanical vibration of such materials when acted upon by a not too weak sound wave. We shall consider here some of the theoretical and practical aspects of these vibrations. The theory is rather complicated, even when simplifying assumptions are made, and since the practical applications are as yet rather scanty we shall only give the outlines of the simplest case here, *i.e.* we shall confine ourselves to materials that are so porous that their porosity can, by approximation, be put equal to unity. The theoretical treatment for the case of smaller porosity is given elsewhere in detail [14].

As with rigid porous materials the fundamental problem is the derivation of expressions for characteristic impedance and propagation constant. The computation of impedances of layers or composite layers of given thickness, then, is tiresome, but rather elementary. The situation is essentially complicated, however, since there turn out to be two propagation constants γ' and γ'', and even four characteristic (wave) impedances W_1', W_1'', W_2' and W_2''. This results from the possible existence of two wave types that may travel through such a medium, each with its own damping and its own velocity of propagation, *i.e.* with its own γ. Let us take *e.g.* a glass tube filled with air. In the air inside the tube sound travels at approx. 340 m/sec; solid-borne sound waves in the tube itself travel, however, with the sound velocity in glass. If we take into due account the coupling of glass and air, the 'glass wave' will also be found to some extent in the air, *i.e.* also in the air a wave type with 'glass velocity' will be found and conversely, in the solid material a wave travelling with approx. 340 m/sec is present in principle. These phenomena originate in the coupling of the two media. They are of no importance in the example given, of course. In the case of a very flexible, light, porous medium it is likely, however, that the damping, sound velocity and absorption behaviour are considerably influenced by these coupling effects. We should still expect two wave types (two γ's), each being present in both media, *i.e.* each having two characteristic impedances.

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The mathematical basis is given by two equations of motion

$$-\frac{\partial p_1}{\partial x} = \rho_1 \frac{\partial u_1}{\partial t} + g(u_1 - u_2) \qquad \text{(I, 40)}$$

$$-\frac{\partial p_2}{\partial x} = \rho_2 \frac{\partial u_2}{\partial t} + g(u_2 - u_1) \qquad \text{(I, 41)}$$

and the two equations of continuity

$$-\frac{\partial p_1}{\partial t} = K_1 \frac{\partial u_1}{\partial x} \qquad \text{(I, 42)}$$

$$-\frac{\partial p_2}{\partial t} = K_2 \frac{\partial u_2}{\partial x} \qquad \text{(I, 43)}$$

where

p_1, p_2 = force per unit area of material in frame and air resp.,
ρ_1, ρ_2 = mass per unit volume of frame and air resp.,
u_1, u_2 = (mean) velocity of frame and air resp.
K_1, K_2 = specific stiffness of frame and air resp. That of air, K_2 may be supposed to be equal to the isothermal bulk modulus, *i.e.* the barometric pressure, since coupling only occurs with rather high air resistance, in which case isothermal conditions prevail.
g = a coupling constant of the dimension of an air resistance.

If the vibration of the medium could be stopped, *e.g.* by increasing ρ_1 and/or making K_1 tend to infinity, eqn. (I, 41) would become identical with

$$-\frac{\partial p_2}{\partial x} = s\rho_2 \frac{\partial u_2}{\partial t} + r u_2,$$

from which it is clear that

$$g = i\omega\rho_2(s - 1) + r. \qquad \text{(I, 44)}$$

We seek a progressive wave solution, *i.e.* one in which all quantities vary as $\exp(i\omega t - \gamma x)$, to satisfy the eqns. (I, 40) to (I, 43). We, therefore, put $-\gamma$ for $\partial/\partial x$ and $i\omega$ for $\partial/\partial t$ in these equations. We then obtain four homogeneous equations in p_1, p_2, u_1 and u_2 that only allow for a solution if the determinant of coefficient vanishes. This yields an equation in γ, from which this quantity can be computed, namely

$$\Gamma^4 - \Gamma^2\left[\frac{\rho_1 + \sigma}{K_1} + \frac{\rho_2 + \sigma}{K_2}\right] + \frac{\rho_1\rho_2 + (\rho_1 + \rho_2)\sigma}{K_1 K_2} = 0 \qquad \text{(I, 45)}$$

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in which Γ stands for $\gamma/i\omega$, the reciprocal quantity of the complex sound velocity, and σ for $g/i\omega$. Since only even powers of Γ occur we find two pairs of solutions that will be labelled $\pm\ \Gamma'$ and $\pm\ \Gamma''$. The minus signs are related to sound waves in the negative direction of x and may be dropped for the time being. We are now in a position to compute all the necessary ratios, namely

$$\left.\begin{aligned} W_1 &= p_1/u_1 = K_1\Gamma \\ W_2 &= p_2/u_2 = K_2\Gamma \\ \frac{u_2}{u_1} &= \frac{-K_1\Gamma^2+\rho_1+\sigma}{\sigma} = \frac{\sigma}{-K_2\Gamma^2+\rho_2+\sigma} = -\frac{K_1\Gamma^2-\rho_1}{K_2\Gamma^2-\rho_2} \\ \frac{p_2}{p_1} &= \frac{K_2\,u_2}{K_1\,u_1} = \frac{K_2\,u_2}{K_1\,u_1}. \end{aligned}\right\} \quad \text{(I, 46)}$$

In these expressions the roots of Γ from eqn. (I, 45) enter and since there are two of them (disregarding the negative ones), there are two values of each of these quantities, *e.g.* two characteristic impedances W_1' and W_1''.

We now proceed to the computation of impedances. The most important example from a practical point of view is perhaps a layer of thickness l of flexible material fixed to a rigid wall. We then superimpose four waves, two waves in the positive direction and two in the negative direction. The boundary conditions at $x = 0$ and $x = l$ provide us with the necessary equations for computing the ratio of their amplitudes and we finally obtain for the impedance

$$\frac{1}{Z} = \frac{u_2'/u_1'}{W_1'(p_2'/p_1' - p_2''/p_1'')}\tanh\gamma' l + \frac{u_2''/u_1''}{W_1''(p_2''/p_1'' - p_2'/p_1')}\tanh\gamma'' l \quad \text{(I, 47)}$$

This suggests the presence of a shunt connection, the admittance of the air and frame waves being added. Physically this means that if the air wave enters with difficulty into the pores its impedance may be short-circuited more or less by the flexibility of the frame.

Another extreme case may be considered, *viz.* the case of the porous, flexible layer coated by a thin impervious surface layer of negligible thickness and mass m per unit surface. We, then, may put the impedance of the layer as

$$Z_{tot} = i\omega m + Z,$$

where Z is the specific impedance of the layer itself. The boundary

condition at $x = 0$ (front) is changed in comparison with the previous case. Along similar lines to those indicated above we find

$$Z_{tot} = i\omega m + W_1'(1+\pi')\frac{1-\varphi''}{\varphi'-\varphi''}\coth\gamma' l + W_1''(1+\pi'')\frac{1-\varphi'}{\varphi''-\varphi'}\coth\gamma'' l. \quad \text{(I, 48)}$$

in which π denotes p_2/p_1 and φ u_2/u_1.

Experiments on flexible porous layers have been carried out by KOSTEN [14]. The layers were of soft sponge rubber of low density (Laticel 120 kg/m³, Dunlopillo 200 kg/m³). In Fig. 31 a few characteristics are reproduced of layers with normal porous surface; they

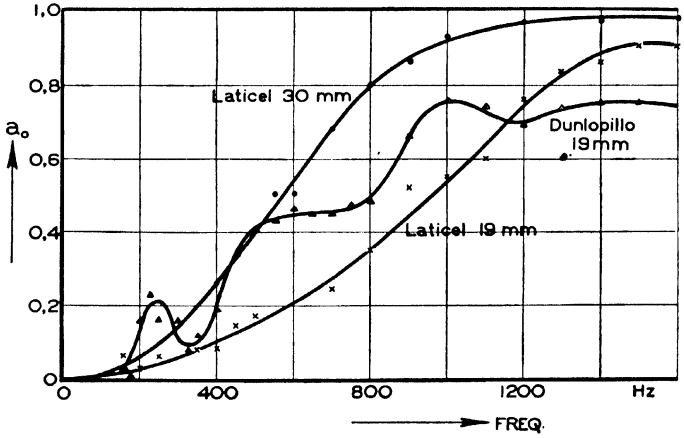


Fig. 31. Absorption by sponge rubber layers with porous surface.

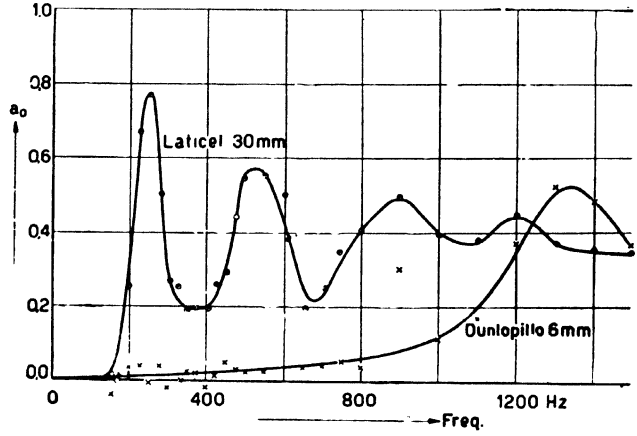


Fig. 32. Absorption by sponge rubber layers with impervious light surface coating.

show up the behaviour as of rigid porous layers, but that for the Dunlopillo-curve indicates some resonances in the elastic frame. Fig. 32 gives characteristics of similar layers coated with a thin, almost massless, non-porous surface layer. Comparison with Fig. 31 reveals the great change in behaviour due to the coating. Imperviously coated rigid layers do not absorb at all, stressing the fact that in the case of Fig. 32 the absorption exists thanks to the flexibility of the layers.

From a practical point of view the theoretical results for materials with a porous surface (eqn. (I, 47)) are more important than those of coated layers (eqn. (I, 48)) In many light fibrous materials the following inequalities hold

$$\underset{\text{(heavy skeleton)}}{\rho_1 >> \rho_2} \qquad \underset{\text{(very flexible skeleton)}}{K_1 << K_2}.$$

It can be shown that then (I, 47) simplifies considerably, *viz.*

$$Z = \sqrt{K_2\rho_2}\,\sqrt{\frac{1+\sigma/\rho_2}{1+\sigma/\rho_1}}\,\coth i\omega l\sqrt{\frac{\rho_2}{K_2}}\sqrt{\frac{1+\sigma/\rho_2}{1+\sigma/\rho_1}} \qquad \text{(I, 49)}$$

If the material had been rigid, the term σ/ρ_1 would have been absent, which makes it clear that some sort of short-circuiting takes place. Whereas waves of very low frequency in *rigid* porous materials are extremely damped and have a wave impedance that tends to infinity for decreasing frequency, the same low frequency wave in *flexible* porous materials is less damped and has an impedance that remains finite. Measurements of SCOTT [10], WÜST [15] and KOSTEN, KASTELEYN, VAN DEN EIJK [16] on such materials do not show up this behaviour. They found a more or less 'classical' behaviour that can be explained with the theory of rigid materials, although the materials investigated were certainly flexible. This is probably due to the fact that such measurements are carried out in tubes that hamper the vibration of the skeleton. Recent measurements in the laboratory of the author [17], carried out with special precautions not to impede vibration of the fibres, and to some extent also the measurements of FERRERO and SACERDOTE [11] on glass wool and those of MARIENS [12] on glass wool of 40 kg/m^3, reveal the predicted behaviour clearly.

It is therefore believed that the absorption behaviour in practice of flexible materials is influenced advantageously by flexibility. Fig. 33 shows an example of what theory predicts in a special case. At small

thickness of the layer its flexibility is of no importance (see *e.g.* the curves for $l = 1$ cm). Above $l = 3$ cm a considerable gain in the low frequency range is obtained.

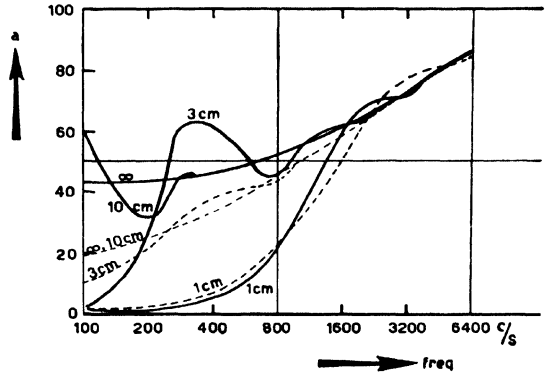


Fig. 33. Improvement of absorptive behaviour due to flexibility according to theory. Assumptions: $r = 2 \times 10^5$, $\rho = 100$ kg/m^3, $s = 3$, eqn. (I, 49).

§ 9. Theory and Results of Panel Absorbents

An example of a panel absorbent is an impervious lining fixed at some distance in front of a rigid wall (Fig. 34). Wooden panel constructions of this type are frequently used in room acoustics for aesthetical reasons or with a view to increasing absorption in the low frequency range. The essential quantities that govern their behaviour are:

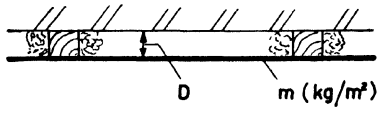


Fig. 34. Panel absorbent.

(a) the mass per unit surface m,
(b) the thickness (width) of the air gap D,
(c) the amount of damping (see further on).

The air in the gap acts as a spring on which the panel is mounted as a mass. Let us first consider the behaviour at normal incidence of

sound. If p be the sound pressure before the panel, the equation of motion is

$$p = (i\omega m + r + \frac{\chi p_0}{i\omega D}) u, \qquad \text{(I, 50)}$$

where as previously
χp_0 = the adiabatic bulk modulus of air,
r = the resistance of the system per unit surface,
p = sound pressure in front of the panel,
u = velocity of the panel surface.

The mass-term does not need explanation, the resistance term is inserted in its simplest form (ru) as frequently done. That the stiffness-term is indeed $\chi p_0 u/i\omega D$ is proved by firstly replacing $u/i\omega$ by the displacement y of the panel, bearing in mind then that y/D is the relative compression of the air and, finally, that the bulk modulus is indeed defined as equal to the ratio of the applied pressure to the fractional compression. It will be seen from (I, 50) that the panel is supposed to vibrate as a piston, remaining flat and moving to and fro as a whole. Practical panels, even at normal incidence of sound, are unable to perform this motion, owing to the boundary conditions (clamping or simple mounting). It might be thought that this would upset the whole reasoning since these boundary conditions would give rise to a stiffness-term in (I, 50) due to the stiffness of the panel itself. With practical constructions, however, the proper stiffness of the panel is negligible in comparison with that of the layer of air (see example further on).

We derive immediately from (I, 50) that the specific impedance of the panel is

$$Z = i\omega m + r + \frac{\chi p_0}{i\omega D}. \qquad \text{(I, 51)}$$

For 100% absorption r should be made equal to ρc, *i.e.* the characteristic impedance of air, whereas the imaginary parts should compensate each other, *i.e.* the system should be in resonance. This occurs at

$$\omega_{res.} = \sqrt{\frac{\chi p_0}{mD}} \qquad \text{(I, 52)}$$

Example: 5 mm plywood (density 600 kg/m³), $D = 0.03$ m, $\chi p_0 = 1.4 \times 10^5$, $f_{res.} = 200$ c/s. With panels of practical size the lowest resonance frequency as

governed by its mass and own stiffness is of the order of 20 c/s, indicating that the proper stiffness is some 100 times smaller than that of the layer of air.

Panel absorbents are most suitable at low frequencies. If rigid materials like wood or hard board are used, m is bound to a rather high practical minimum. The only means for getting a rather high resonance frequency of, say, 500 to 1000 c/s is, then, to use a very thin air gap (say a few millimeters). Resonance is then obtained by the mutual compensation of a large inertia force and a large stiffness force in opposition of phase. For frequencies differing slightly from resonance the matching of these forces will be completely upset, *i.e.* the impedance just below and just above resonance has already a considerable imaginary part, decreasing the absorption accordingly. In other words, although a high resonance frequency with rigid panels is obtainable no satisfactory constructions are feasible in this frequency range, since they will absorb only in a very narrow frequency band; they are said to be too selective. There are two ways to provide constructions that are not too selective, both being used in practice.

(1) m can be reduced considerably by merely taking an impervious light textile layer, as *e.g.* oil-cloth. E. MEYER [18] investigated construc-

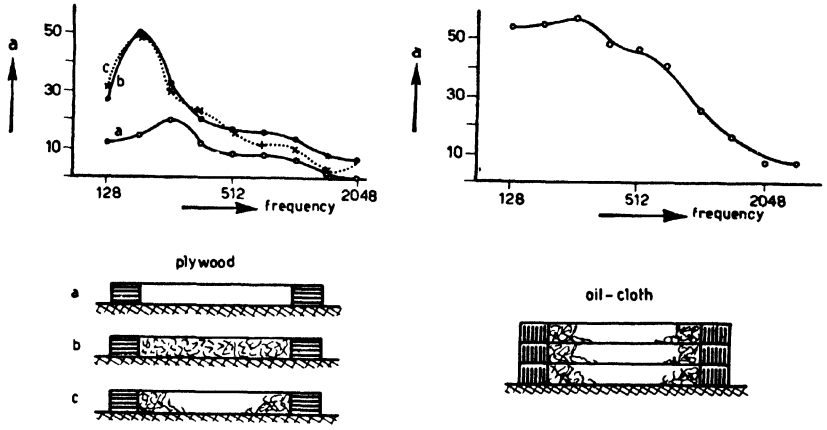


Fig. 35. MEYER's results [18] on panel absorption.
Left: a. Plywood panel with unfilled gap.
b. Ditto, gap completely filled.
c. „ , gap filled to a small extent.
Right: Absorption by a multiple panel absorbent of oil-cloth.

tions consisting of several of these layers, placed one before the other (multiple panel construction) and obtained 50% absorption in a satisfactory frequency range (Fig. 35).

(2) D can be increased, thus reducing the resonance frequency. This is done with board and wooden panels (see example above and [18-19]).

For the design of suitable panels the theory of perforated panel absorbents may be used with advantage. The basic knowledge will be given later (§§ 10, 11) and used here without derivation. We choose the absorption coefficient at resonance $a_{res.}$ and the number of octaves O within which the absorption shall exceed $a_{res.}/2$. We enter Fig. 38 with these quantities and read from the graph $1/\kappa D$. We then enter Fig. 39 with the desired resonance frequency $f_{res.}$ and the value of $1/\kappa D$, just found, and read from the graph D, the width of the air gap. The value of m can then be computed by replacing $1/\kappa D$ by its expression $m/\rho D$ that should be used for the panel absorbents of this section in stead of $1/\kappa D$ (compare eqn. (I, 50) and (I, 53)), bearing in mind that the numerical value of $m/\rho D$, as well as ρ (= 1.2) and D, are known.

Example:

$a_{res.} = 33\%$, $O = 2.1$; read from Fig. 38 $1/\kappa D = m/\rho D = 50$

$1/\kappa D = 50$, $f_{res.} = 77$c/s ; read from Fig. 39 $D = 0.1$ m

$m/\rho D = 50$, $D = 0.1$; computed $m = 6$ kg/m².

At random incidence of sound a higher absorption is found than at normal incidence. Enter Fig. 46 with $a_0 = 33\%$ and read the mean value at normal incidence to be about 50%.

In Fig. 36 recent results of PARKIN and PURKIS are reproduced. The example given above is chosen so as to verify these results. Indeed the discrepancies are within the errors of measurement, the panels being of 1 cm plywood. The resonance frequencies, actually found, are somewhat higher than those computed. This is due to the proper stiffness of the panels that was even increased artificially by braces.

The remaining difficulty in the design is to adjust the resistance to the appropriate value. The resistance r originates to some extent in internal friction in the panel itself, furthermore in energy losses near

the clamping and is frequently increased above this involuntary level by filling the air gap by suitable flexible, cheap, fibrous substances. MEYER [18] showed that it is not necessary to fill the whole gap; the absorbing material in the centre may be omitted if suitable material is provided near the boundaries.

Interesting theoretical considerations are given by L. CREMER [20]. He showed, *e.g.* that also at oblique incidence reasonable absorption is to be expected and this at a somewhat *higher* frequency, contradictory to the first impression.

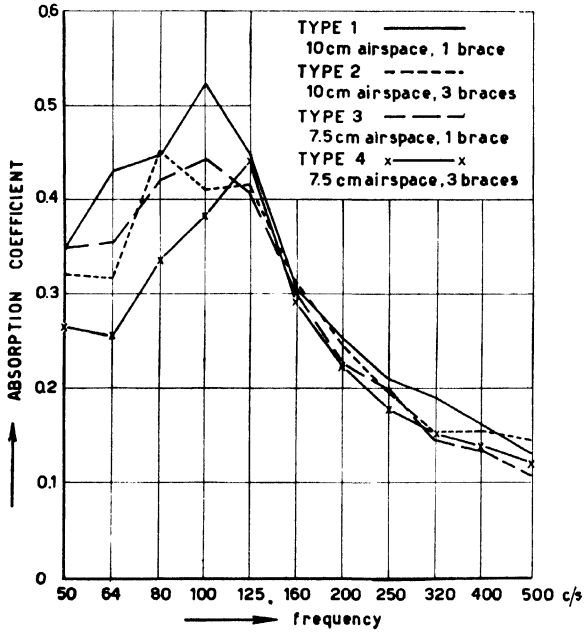


Fig. 36. Absorption by panel absorbents at low frequencies according to design and measurement of PARKIN and PURKIS [19].

§ 10. Single Resonators and Perforated Panels

In the Introduction § 13, the theory of the HELMHOLTZ resonator is given. In such a device the air in the neck provides the mass of the system, that in the reservoir the spring. Various values have been ascribed to the 'conductivity', as RAYLEIGH [22] called it, of the orifice or neck, according to the geometry of the system. The conductivity κ determines in fact the extent of the air which vibrates. Formulae [23]

which have been proposed for κ are set out in Chapter 18, § 24.

A resonator, just like other three-dimensional systems, has $\infty 3$ resonance frequencies. The reason that we can speak about *the* resonance frequency is that the lowest resonance frequency is much smaller than the other ones. *E.g.* a resonator of 1 dm^3 volume with a circular hole of 1 cm diameter in the thin wall has its first resonance at about 170 c/s. At the higher resonance frequencies the wave-length is smaller than about double the linear size of the volume, say smaller than 0.2 m, *i.e.* they exceed 1700 c/s approx. If a resonator is built into a wall such that the neck communicates with the room, it will be a weak spot in the wall for sound in the neighbourhood of its resonance frequency. It comes into action, if a wave impinges upon the wall. Now if the vibration in the resonator does not imply conversion of vibrational energy into heat – no viscous or thermal losses – the action consists of storing some sound energy in its own vibration and diffusing as a point source part of the incident sound. If the impinging wave ceases, the resonator reverberates for some time, restoring in this way its own energy to the room. Absorption can be provided for by adequately damping the vibration of the resonator. This can be done by a suitable filling material in the resonator, *e.g.* a fibrous light substance. It is most effective as damping agent in or near the neck since there the velocities are maximum. Whether the resonator is more a diffusor or an absorber depends upon the ratio k of the viscous resistance of the resonator, *e.g.* due to the filling material, and the radiation resistance of the mouth, $k = 1$ being a rough limiting value between both activities.

It can be shown that the effective 'open window' surface afforded at resonance by the resonator equals

$$A_{res.} = \frac{\lambda^2_{res.}}{2\pi} \frac{4k}{(1+k)^2} \qquad \text{(I, 53)}$$

Single resonators absorb rather selectively unless k is great. This is measured by the number of octaves O within which the absorption exceeds 50% of the absorption at resonance. One finds [24]

$$O = 1.442\,(1+k)\frac{\kappa}{\lambda_{res.}} \qquad \text{(I, 54)}$$

Single resonators have been in use for a long time. If only selective

absorption is wanted they are very helpful. If, however, absorption in a wide frequency range is required they are uneconomical, and perforated panels backed by a layer of porous material, to which we now will proceed, are more suitable. They are the more practical construction in which the same idea is incorporated.

A perforated panel construction can be treated along the same lines as single resonators. Suppose a plane wave impinges upon the panel. The sound pressure just in front of the panel sets the air in the construction into vibration. Again the kinetic energy, and therefore the mass of the system, is to be sought in and near the holes whereas the air behind the panel essentially behaves as a spring (potential energy). We may put at once

$$Z = i\omega\frac{\rho}{\kappa} + r + \frac{\chi p_0}{i\omega D} \qquad \text{(I, 55)}$$

where

κ = the conductivity of all the holes together per unit surface of the panel; if the number of holes per m² is n, the separate conductivities must be added.

r = the real part of the specific acoustic impedance due to involuntary causes, or introduced on purpose to combat a too great selectivity in the construction.

D = the volume per m² of the panel, *i.e.* the width of the air gap (see Fig. 37).

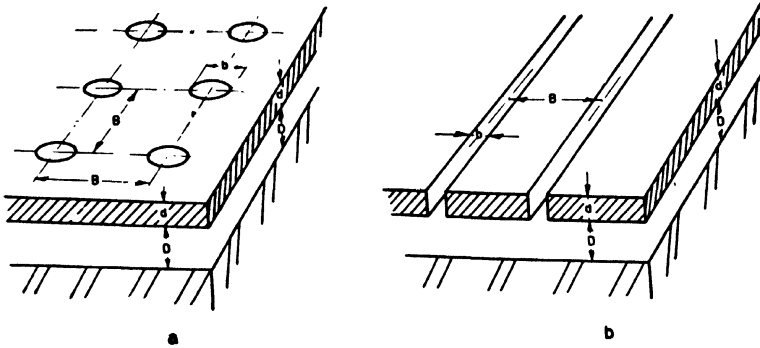


Fig. 37. Dimensions of panel absorbents:
a. circular perforations b. slit perforation

It is clear that κ, r and D govern completely the absorptive behaviour, resonance frequency, absorption at resonance, selectivity, etc.

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Instead of r the ratio k of r and the characteristic impedance ϱc of air will also be used.

The formulae describing the behaviour turn out to be [25]

$$\left.\begin{aligned} f_{res.} &= \frac{c}{2\pi}\sqrt{\frac{\kappa}{D}} \\ a_{res.} &= \frac{4k}{(1+k)^2} \\ O &= 1.442\,\sqrt{\kappa D}(1+k) \end{aligned}\right\} \qquad \text{(I, 56)}$$

where $c = 342$ m/sec and O is the number of octaves within which the absorption exceeds $a_{res}/2$.

In order to get practical constructions the air gap D is often filled with some sort of porous material. It is suggested in the theoretical considerations given above that of the three quantities κ, D and r governing the behaviour, κ and D only depend upon macroscopical geometrical data of the resonator so that the introduction of a filling material might be supposed to alter neither κ nor D, *i.e.* not $f_{res.}$ This is only true in a first rather rough approximation. D is apparently decreased since part of the air volume is taken up by the incompressible solid material. If this were 10%, a 5% increase in resonance frequency should be expected. κ is however decreased either, and, as a rule relatively to a greater extent than D, so that the resulting effect of the introduction of a filling material is, as a rule, a decrease in resonance frequency. The decrease in κ due to the filling material is obvious from the fact that due to the structure of the porous medium the air inside it is apparently s/P times denser and since the inner end-correction is normally located in the porous substance it is increased by a factor s/P, decreasing κ accordingly. The resulting change in resonance frequency depends upon various features of the construction; it is a maximum when the panel is very thin. Then the resonance frequency decreases roughly by a factor

$$\sqrt{(1 + s/P)/2.}$$

Taking $s = 2.1$ (rather high for these cases) and $P = 0.7$ (rather low), we find a $\sqrt{2}$ times lower resonance frequency ($\frac{1}{2}$ octave lower) as a practical maximum of the frequency shift. This demonstrates clearly the fact that design with a very high accuracy cannot be expected if filling materials are used.

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Something should be said about absorption at oblique incidence. If the gap is completely filled every hole may be treated as the entrance of an independent resonator. The impedance of the panel may then be supposed to be independent of the direction of incidence. If the filling material does not fill the gap completely, coupling of the individual resonators in the gap will influence their behaviour considerably. It is a considerable complication in theoretical respects [26] and is as a rule disadvantageous. The coupling might be counteracted, however, by dividing up the gap in numerous separate squares of relatively small dimensions, say 20×20 cm^2 or 50×50 cm^2, by the insertion of wooden strips.

Up till now perforation with circular holes at relatively large mutual distance has been considered. We, then, can borrow the necessary formulae for the conductivity (*cf.* Chap. 18). New problems arise when the number of circular holes per m^2 is so greatly increased that the interaction of the flow of air through neighbouring holes becomes of importance or when the perforation is essentially different, e.g. infinitely long parallel slits at a regular mutual distance. The basic approximate formulae of eqn.(I, 56)are still valid, but the difficulty is to predict the conductivity per m^2 of such constructions, RAYLEIGH'S formulae being no longer applicable, TABER JONES has, however, discussed the conductivities of slits (*cf.* Chap. 18, Fig. 19).

It has been shown by J. M. A. SMITS and KOSTEN [27] that for slit resonators the product κD can be written as

$$\frac{1}{\kappa D} = \frac{B}{b}\frac{d}{D} + \frac{B}{D}\frac{l(b/B, B/D)}{b} \qquad \text{(I, 57a)}$$

where B, b, D, d are geometrical quantities, defined in Fig. 37b and l is a known, but rather complicated function of its variables b/B and B/D (see Fig. 40). A similar expression has been given in the same paper [27] for resonator panels with circular holes. It is based on the results of NIELSEN, who took into due account the velocity distribution in the holes, and runs

$$\frac{1}{\kappa D} = \frac{4}{\pi}\left[\frac{B^2}{b^2}\frac{d}{D} + \frac{B}{b}\cdot\frac{B}{D}\cdot F\left(\frac{b}{B}\right)\right] \qquad \text{(I, 57b)}$$

where again the reader is referred to Fig. 37a for the meaning of B, b, D, d in this case and F is a known function (see Fig. 40).

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§ 11. The Design of Perforated Panel Absorbents; Results

The design is based primarily on eqn. (57). The requirements may be stated as given numerical values of the resonance frequency $f_{res.}$, the absorption coefficient at resonance $a_{res.}$ and the number of octaves O for which a exceeds $a_{res.}/2$. One can, then, compute κ, D and k from eqn. (I, 56). It is, however, simpler to solve this problem graphically. We enter Fig. 38 with the given values of $a_{res.}$ and O, and read from the graph $1/\kappa D$ and k. We, then, enter Fig. 39 with the value of $1/\kappa D$ just found and $f_{res.}$, and read D from the graph.

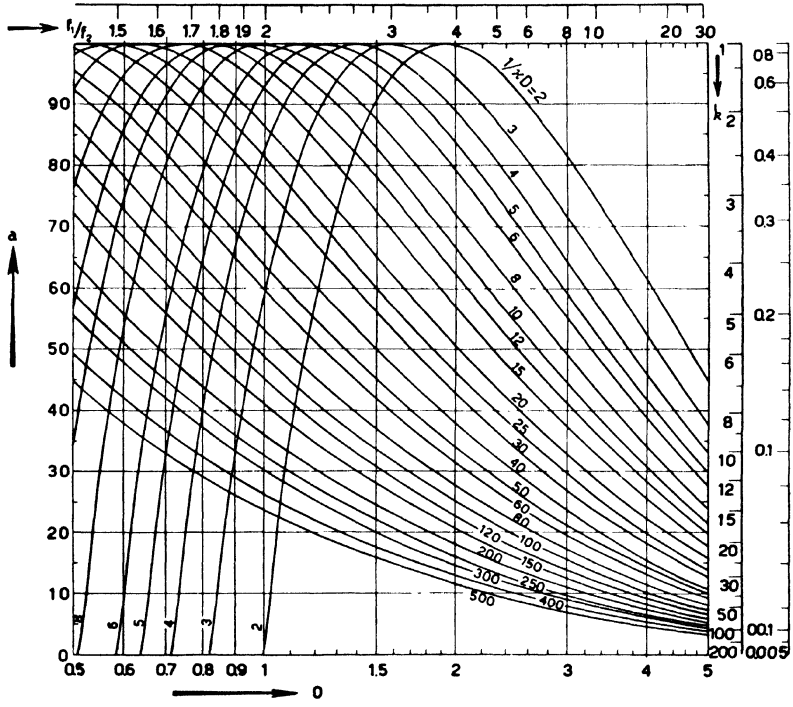


Fig. 38. Design chart of perforated panel absorbents. Requirements: $a_{res.}$ and O (curve width in octaves). Read from the graph: $1/\kappa D$ and k (see also Fig. 39 and 40).

Finally we enter Fig. 40 with the value of $1/\kappa D$, keeping in mind the value for D that was already found, and read from the graph all

data B, b, d that govern the perforation. The use of Fig. 40 is explained with the following example.

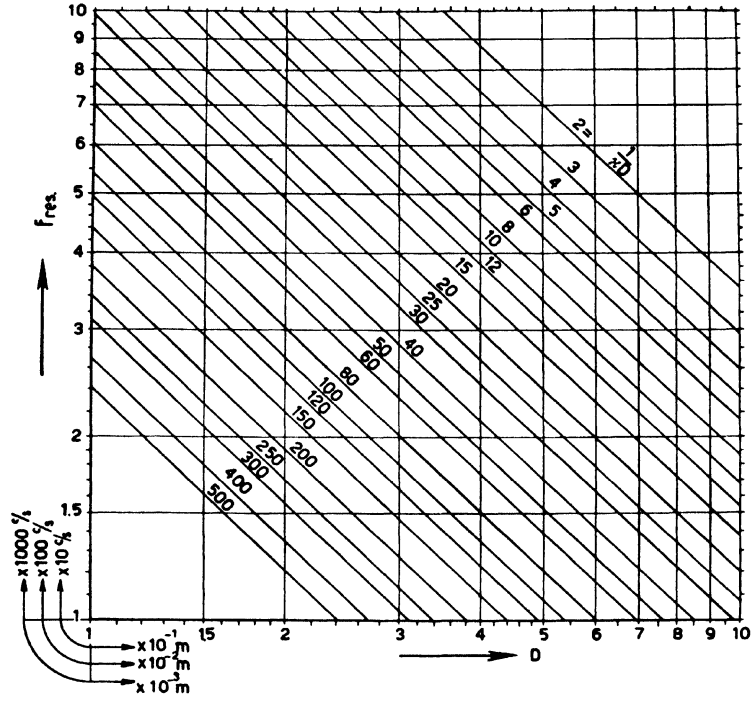


Fig. 39. Design chart of perforated panel absorbents. Entrance: resonance frequency $f_{res.}$ and $1/\kappa D$ (from Fig. 38). Read from the graph the air gap D (see Fig. 37).

Example. Requirements: $a_{res.} = 60\%$, a exceeds 30% in two octaves ($0 = 2$) $f_{res.} = 500$ Hz.

Design: Fig. 38 yields $1/\kappa D = 13$. Fig. 39 yields $D = 3 \times 10^{-2}$m $= 3$ cm. Enter Fig. 40 b with $1/\kappa D = 13$ and $D = 0{\cdot}03$. The value $1/\kappa D = 13$, as read from the ordinate axis of Fig.40 b is marked by two points on a slip of paper. This is moved parallel to itself in horizontal and vertical direction until the two marks indicate a sympathetic design. The lower mark is bound to remain during this translation in the lower half of the figure and indicates d/D, *e.g.*, $d/D = 0.1$, *i.e.* $d = 3$ mm since $D = 3$ cm. The upper mark is bound to remain in the upper half, and indicates B/D. Two systems of curves are presented. Numerical values of B/D with a bar (*e.g.*$\bar{3}$) relate to slit absorbers, the unbarred values to normal circular perforation. In our case choosing $p = 2\%$ (= the percentage perforation; see abscisscae in the middle of the figure) a suitable pair of dimensions

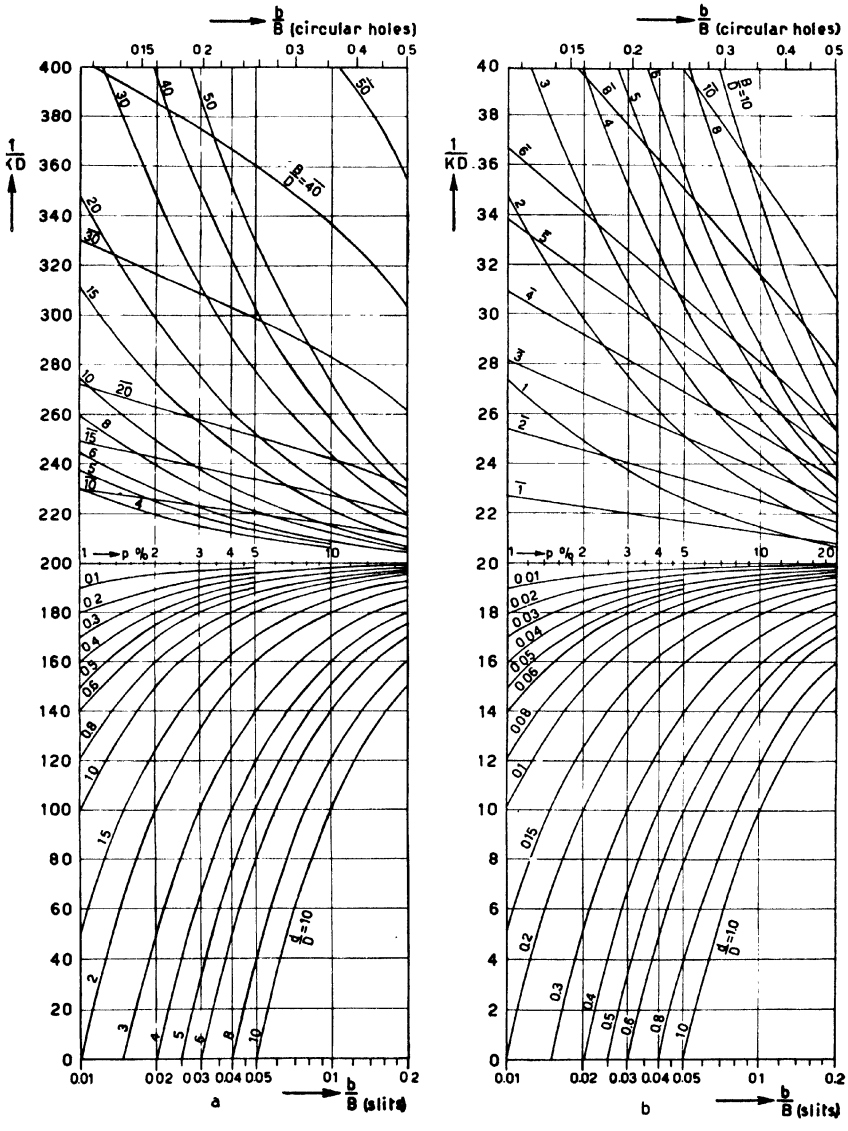


Fig. 40. Design chart of perforated panel absorbents. Entrance: $1/\kappa D$ (from Fig. 38) and D (from Fig. 39). Read from the graph dimensions of the perforation (see Fig. 37 and example p. 90).

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for slit absorbers is: $B/D = \overline{3.5}$, *i.e.*, $B = 10$ cm; $b/B = 0.02$ (see bottom of figure), *i.e.*, $b = 2$ mm. A combination of dimensions for circular holes and the same degree of perforation is found to be: $B/D = 1.6$, *i.e.*, $B = 5$ cm; $b/B = 0.16$, *i.e.*, $b = 8$ mm.

The most difficult problem in design is the question of finding the filling material or other means for adjusting k to the desired value. Without giving here the derivation [28] we give a formula that might be of some help in this choice. If b is much greater than the thickness d of the panel (compare Fig. 37), and if porous material is applied just behind the panel, *e.g.* porous material filling the whole gap,

$$r' = 2r\kappa \qquad \text{(I, 58)}$$

where

r' = specific air resistance of the filling material,
r = $k \times \rho c = k \times 420$ MKS-units,
κ = conductivity per m^2 of the construction.

Taking as reasonable mean values

f_{res} = 500 c/s
a_{res} = 70%
O = 3 octaves

we arrive at $r' = 2 \cdot 10^4$ MKS-units for the air resistance of the filling material, a value of the order encountered in rock wool, glass wool, felt etc. Measurements at normal incidence have turned out to be essentially in agreement with theory [29].

Two graphs with results that cannot be verified with the simple theory presented here are shown in Fig. 41 and 42.

The resonator of Fig. 41 was a thin perforated metal plate placed in the interferometer. The air space behind the perforated plate was 30 mm, the gap was filled with glass wool. Unfortunately no impedances were measured. The perforation consisted of circular holes in a 'cubical lattice', 5.6 mm apart from each other, 4 mm in diameter, the whole plate had about 175 holes. Variations of the resonator were made by closing a great number of these holes. The curves in Fig. 41 from left to right correspond to 1, 2 × 2, 3 × 3, 5 × 5, 7 × 7, 9 × 9 and 11 × 11 holes opened in the middle of the plate. The curves of 1 to 5 × 5 holes show a second resonance at 800 to 900 c/s. This has

nothing to do, however, with the resonator, these irregularities being probably due to a vibration of the metal covering plate.

In practice combinations of resonators, in series (Fig. 42a) or in parallel, are sometimes used. Their advantage is a high absorption

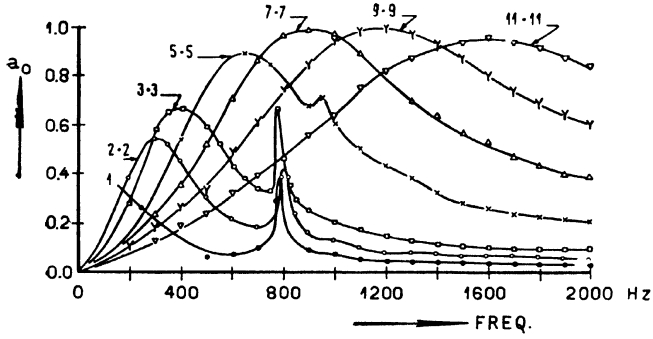


Fig. 41. Absorption characteristics of the same resonator with different perforations.

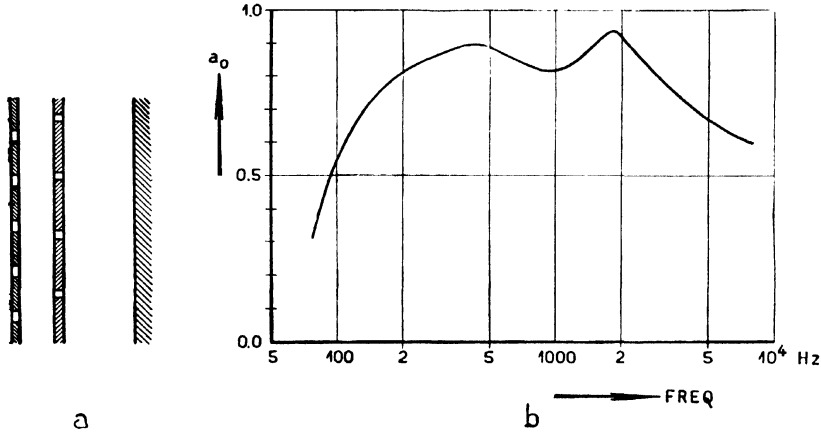


Fig. 42. Double resonator in series (after BRÜEL).

coefficient in a wide frequency range due to the two or more resonance frequencies inherent in such systems. Fig. 42b is an example taken from BRÜEL [30].

§ 12. Measurement of Acoustic Impedance by Interferometer *

Since several critical surveys of the numerous methods for measuring the acoustic impedance at normal incidence have been given elsewhere[31], we confine ourselves here to the description of the principle of the method that seems to be used most frequently, the constant length interferometer. At the end of this section the basic idea of a few other methods will be given in outline. An interferometer can be described as a tube in which a sound source and the sample under test are inserted, at either end, one facing the other. If the sound source operates at such a low frequency that the wave-length is greater than 1.707 times the diameter D of the tube, only plane waves can travel in the tube. In the neighbourhood of the source these plane waves are disturbed if the sound source generates non-plane waves. The same holds for the neighbourhood of the sample. A non-homogeneous sample may reflect the incoming plane wave as a distorted reflected one. At some distance from the sample the reflected wave will be plane again if the wave-length is large enough. Therefore all interferometers using plane waves can only be used up to frequencies less than $c_0/1.7\,D$, c_0 being the velocity of sound in free space. This restriction cannot be neglected if the source provides us with ideally plane waves since all samples are more or less inhomogeneous and will reflect progressive non-plane waves if the frequency chosen is too high. Assuming that this restriction is fulfilled and denoting the impedance of the sample by Z we may describe the sound field in the tube as the superposition of two plane waves, an oncoming and a reflected one. Owing to absorption the reflected one will be of smaller intensity than the oncoming one. We may borrow from equation (I, 21) the relation between the reflection coefficient $r = p_r/p_i$ directly before the sample and the impedance Z

$$r = \frac{Z - \rho c}{Z + \rho c} \quad \text{or} \quad \frac{Z}{\rho c} = \frac{1 + r}{1 - r}. \tag{I, 59}$$

Remembering that the propagation constant γ of a gas is $i\omega/c$ we can write down the pressure at a distance x in front of the sample

$$p(x) = p_i \exp(-i\omega x/c) + r p_i \exp(+i\omega x/c). \tag{I, 60}$$

* Reverberation methods are discussed in Chapter 5.

The exponentials have modulus unity and only change the phase of the progressive waves (damping is neglected!). Writing as previously

$$r = |r| \exp i\Delta,$$

(I, 60) takes the form

$$p(x) = p_i \exp(-i\omega x/c) + |r| p_i \exp i(\omega x/c + \Delta).$$

The pressure $p(x)$ will be a minimum for those values x_j of x for which both terms are in opposite phase, *i.e.*, for

$$(\omega x_j /c + \Delta) - (-\omega x_j /c) = -(2j + 1)\pi,$$

where j is an integer. The latter equation yields

$$\Delta = -(2j + 1)\pi - 4\pi x_j/\lambda. \qquad \text{(I, 61)}$$

This equation provides a means of measuring Δ. It may be computed from the position of the places of minimum sound pressure in the tube. It turns out to be too inaccurate to compute λ from frequency and sound velocity.

More reliable values of λ are obtained by measuring the positions of two minima (*e.g.*, x_0 and x_1) and eliminating λ. One sees at a glance from (I, 60) that the ratio of maximum and minimum pressure is

$$\frac{p_{max}}{p_{min}} = \frac{1 + |r|}{1 - |r|}. \qquad \text{(I, 62)}$$

This ratio can easily be measured, whence $|r|$ is found. So (I, 61) and (I, 62) are a basis for measuring the modulus and argument of r.

Several methods for exploring the sound field have been worked out, differing in the amount in which the sound field is disturbed, *viz.*

(a) A small microphone, movable in the tube; the sound field is especially disturbed at the high frequencies, but at low frequencies this method is used with advantage [32].

(b) The microphone is kept outside the tube, but is acoustically connected to the point inside the tube where the sound pressure is to be measured by an exploring tube or probe tube of a diameter very small in comparison with the interferometer tube itself. The sound field is disturbed less than in (a). The remaining disturbance originates in the geometrical shape of the probe tube and in the fact that its opening has not an infinitely high impedance [33].

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(c) The microphone is built into the tube wall behind a stiff membrane and can be moved together with the whole tube, in which loudspeaker and sample holder fit in an airtight way like pistons (see Fig. 43). Since the acoustic impedance of the membrane may be high the remaining disturbance of the sound field can be made negligible [34].

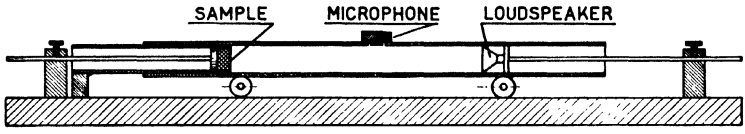


Fig. 43. Constant-length interferometer without exploring tube.

Many other methods can be considered as derivatives of the method just described, *e.g.*

(d) The microphone is fixed near the source of constant strength and the frequency is changed or the sample moved in the tube. The curve width of the pressure-frequency (or -length) curve near resonance is measured as well as resonance frequency (or -length) [35].

(e) An ordinary loudspeaker can act at the same time as a source and as a microphone. The change in impedance of the loudspeaker is measured as frequency or length are varied. This is called the reaction method [36].

(f) A variation of (*d*) has been developped measuring the ratio of maximum and minimum pressure [37]. An analogous variation of (e) is obtained by measuring the maximum change in impedance [38].

(g) The complex ratio of the pressures at two fixed points can be measured.

One method has been developed that is practically acoustic throughout [39]. An acoustic bridge of WHEATSTONE is built, in which the sample under test is compared with an impedance that can be varied. The detector for stating equality of both impedances may be the ear. The difficulty is the manufacturing and calibration of the variable acoustic impedance.

§ 13. Absorption at Oblique Incidence; Locally Reacting Materials

Suppose a plane sound wave to impinge on the surface with impedance Z, in a direction making an angle θ with the normal to the surface (Fig. 44). The reflected wave will be as shown in the figure. Let p_i

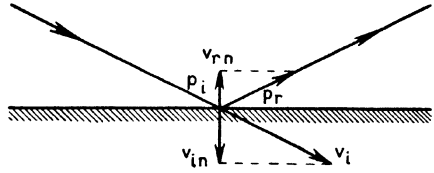


Fig. 44. Oblique incidence.

and p_r be the pressures at the surface in any point, then

$$Z = \frac{p_i + p_r}{v_{in} + v_{rn}}, \tag{I, 63}$$

if v_{in} and v_{rn} are the normal components of the respective particle velocities towards the surface. Now

$$v_{in} \frac{p_i}{\rho c} \cos\theta \text{ and } v_{rn} = -\frac{p_r}{\rho c} \cos\theta,$$

and therefore

$$\frac{Z \cos\theta}{\rho c} = \frac{p_i + p_r}{p_i - p_r}. \tag{I, 64}$$

The complex reflection coefficient p_r/p_i, and, therefore, the absorption coefficient $1 - |p_r/p_i|^2$ bears the same relation to $Z \cos\theta$ as to Z in the case of normal incidence. Therefore, the circle-diagrams Fig. 17, 18 and 19 also apply to this case, provided they are supposed to give absorption coefficient and phase-jump in the complex $Z \cos\theta$ -plane. If the impedance at oblique incidence, defined by (I, 63), is independent of the angle of incidence θ, *i.e.* if it is equal to the impedance at normal incidence, the material is said to be 'locally reacting.' Indeed a possible dependence of Z upon θ originates in lateral interaction of the sound waves in the material. An example of a locally reacting material is a honeycomb structure, all pores being quite independent in their behaviour. A wall covered with single resonators is also rigorously locally reacting. The normal velocity in every pore or resonator is entirely dictated by the sound pressure before it.

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It can be shown that normal absorptive materials may be approximately considered as locally reacting. The reasoning amounts to saying that the sound velocity in such materials is materially smaller than in free air, owing to the structure factor s being greater than unity and the air resistance of the material, so that obliquely impinging waves are refracted towards the normal in the material to such an extent that the waves in the material behave practically as normally incident ones.

For locally reacting material, the following conclusion holds: since the impedance of many materials is larger than ϱc both as to their real and imaginary part, an apparent decrease of the impedance as a consequence of the multiplication by $\cos\theta$ will generally increase the absorption coefficient. Only for very small impedances (near resonance) or for very small $\cos\theta$ (almost grazing incidence) the diminution of Z to $Z\cos\theta$ will be disadvantageous. It, therefore, is plausible that the mean value of the absorption coefficient, if all directions are taken into account, is greater than that for normal incidence.

Although the assumption that practical materials are locally reacting is reasonable, it is worth while considering the other limiting case, in which the lateral interaction is a maximum. This occurs in materials with $s = 1$ and low air resistance. CREMER [40] discussed this case for layers of infinite thickness and showed that Z then varies as $\cos\theta^{-1}$ resulting in an absorption coefficient independent of θ, as was to be expected.

An adequate mean value of the absorption coefficient for locally reacting materials is found if the energy absorbed by a surface element that is exposed to a completely diffuse sound field is considered. The incident energy per unit area comprised in a small solid angle $d\Omega$ in a direction with an angle θ to the normal is proportional to $\cos\theta d\Omega$, since the apparent surface of the surface element under consideration is proportional to $\cos\theta$. Let the fraction a_θ be absorbed. Then the mean value $\bar{a}$ is defined as

$$\bar{a} = \frac{\text{absorbed energy}}{\text{incident energy}} = \frac{\int a_\theta \cos\theta \, d\Omega}{\int \cos\theta \, d\Omega}. \qquad \text{(I, 65)}$$

Now a is only dependent upon θ; therefore take as elementary solid angle $d\Omega$ the small solid angle between a circular cone around the

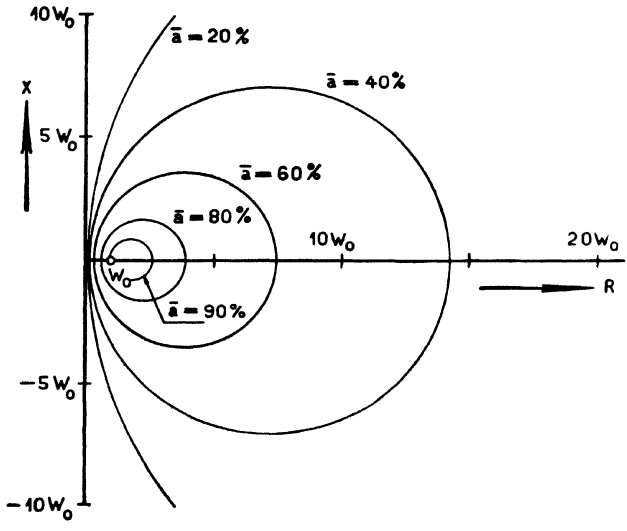


Fig. 45. The mean value of the absorption coefficient of a locally reacting surface as a function of the impedance.

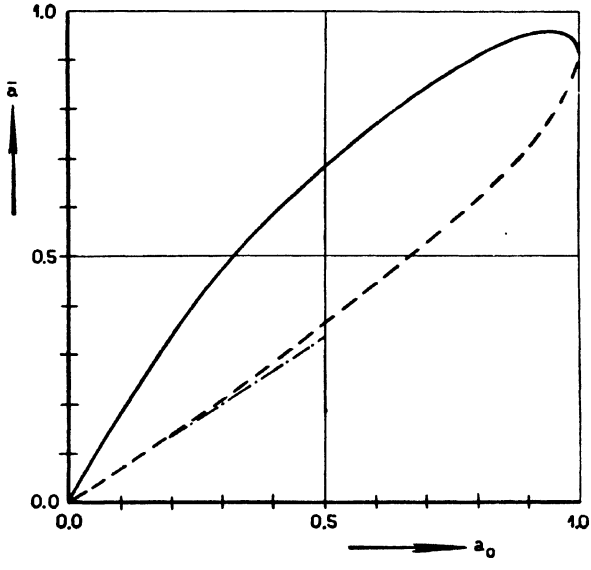


Fig. 46. The comparison between mean and normal value of the absorption coefficient for locally reacting surfaces with real impedance.

normal with vertical angle 2θ and a similar cone with vertical angle $2(\theta + d\theta)$, then $d\Omega = 2\pi \sin\theta \, d\theta$ and

$$\bar{a} = \int_0^{\pi/2} a_\theta \sin 2\theta \, d\theta.$$

For locally reacting surfaces a_θ is known as a function of θ, so the integral can be evaluated. This seems to have been done independently by MORSE *et al* [41]. and KOSTEN and ZWIKKER [42]. If the impedance Z is written as $R + iX$, we obtain

$$\bar{a} = 8\frac{\rho c R}{R^2 + X^2} - 8\left(\frac{\rho c R}{R^2 + X^2}\right)^2 \ln\left(\frac{R^2 + X^2}{\rho^2} + \frac{2R}{\rho c} + 1\right) + \\ + 8\frac{R}{X}\left(\frac{\rho c X}{R^2 + X^2}\right)^2 \left(\frac{R^2}{X^2} - 1\right) \tan^{-1}\frac{X}{R + \rho c}. \qquad \text{(I, 66)}$$

A more adequate representation is, of course to draw contours of constant $\bar{a}$ in the complex Z-plane. This is done in Fig. 45, which is the parallel of Fig. 17 for random incidence. The contours can hardly be distinguished from circles, but actually are not. Another way of representation is given by MORSE [43]. The maximum value of a turns out to be 0.95. In Fig. 46, $\bar{a}$ is compared to a_0 on the supposition that the impedance is real. The dotted part of the curve refers to impedances less than ρc. One sees that for impedances greater than ρc $\bar{a}$ is appreciably larger than a_0.

§ 14. Measurement of Sound Absorption at Oblique Incidence

Although the dependence upon the angle of incidence of the absorption coefficient and even the mean absorption coefficient for random incidence can be estimated with fair accuracy from § 13 something should be said about measuring methods at oblique incidence.

The early attemps of L. CREMER, SCHUSTER and SPANDÖCK may be omitted here for brevity. They used large samples in free space. Diffraction phenomena complicated the experimental problem and reduced the accuracy. New methods have been developed, however, that are comparable in character and accuracy with the interferometer at normal incidence.

With one such method [44] a parallelopiped-shaped enclosure is used

of which one wall is covered with the sample under test. If the dimensions are taken adequately, such an enclosure has characteristic frequencies in the interesting frequency range lying well apart from each other. One such frequency is excited by a sound source. The mode of vibration can be predicted from the dimensions of the enclosure and the frequency. The sample is subjected in this way to plane waves impinging at an angle of incidence that follows from the mode of vibration involved. The curve width of the response as a function of frequency and the decay constant of the damped vibration after stopping the source are dependent on the absorption coefficient of the sample at that angle of incidence. Either of them can be used to suit our purpose. Recently a still simpler method has been developed[45] allowing the use of the well-known technique of normal incidence interferometers. In the latter frequencies below a critical value are used in order to prevent the progression of non-plane waves. The new method is the reverse of the former. The frequency is raised above the critical frequency, enabling a non-plane wave excited by a dipole sound source to be propagated, as indeed it is, whereas the plane wave is suppressed as far as possible. A tube of rectangular cross-section one side rather longer than the other is used for reasons of symmetry. The non-plane wave type can be considered as the superposition of two plane waves at a certain angle with the axis of the interferometer. In Fig. 47 a superposition of two plane waves of the same frequency, progressing at a certain mutual angle, is shown. For reasons of symmetry the velocity component perpendicular to the dotted planes is zero all over the plane, *i.e.* the suggested superposition can exist between two parallel walls. The walls need not be of infinite size. A set up of Fig. 48 is used. If the rectangular tube is closed by the sample a similar wave with reduced intensity will be reflected, giving rise to an interference pattern with the incoming wave just as in normal incidence interferometers. This sound field is explored and yields the absorption coefficient at oblique incidence. The apparent angle of

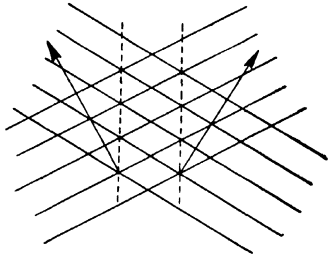


Fig. 47. Superposition of two plane waves.

incidence varies with frequency. Without changing the dimensions of the tube it is not feasible however to choose both the frequency and the angle of incidence.

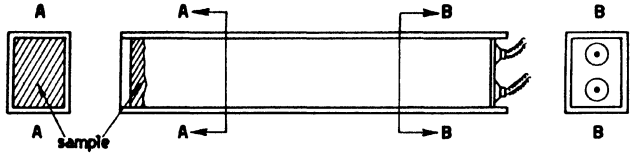


Fig. 48. Interferometer for oblique incidence measurement.

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DIVISION TWO
ACOUSTICS OF BUILDINGS

CHAPTER 5

THEORY OF SOUND TRANSMISSION

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§ 1. Introduction; Mass Law

In this section the theoretical aspects of sound transmission through walls and floors will be treated in outline. A clear distinction must be made in this respect between so-called air-borne sound and structure-borne sound. Sources of air-borne sound are, *e.g.*, the human voice, a violin player, and a more or less elastically supported wireless set, whereas typical sources of structure-borne sound are footsteps on floors, hammering, moving of chairs, a violoncello player etc. Since theory has been of little help up to now for structure-borne sound abatement of floors, little will be said about it here.

The simplest case of sound transmission of air-borne sound is that of a plane progressive harmonic wave impinging normally on a plane single homogeneous wall of thickness d_w and density ρ_w (see Fig. 49). For further simplification the one-dimensional case will be considered first, in which the wall and the wave are supposed to be infinitely extended in both directions [1].

If the specific acoustic impedance Z that the wall, backed by a half-infinite space of air, offers to the impinging wave, were known, the problem could be solved very easily with the aid of the well-known reflection formula (*cf.* Chap. 1, eqn. (35))

$$r = \frac{Z - \rho c}{Z + \rho c} \quad , \tag{II, 1}$$

where r is the complex ratio of the pressures p_r and p_i of the reflected and the incident wave, respectively, and ρc stands for the characteristic

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impedance of air. For, then, the energy reflection coefficient $|r|^2$ is known; from it the energy transmission coefficient $|t|^2 = 1 - |r|^2$ can be deduced, since no absorption occurs, and finally the transmission loss D_o can be computed according to the definition equation

$$D_o = 10\ {}^{10}\log 1/|t|^2.$$

Now one easily sees that

$$Z = i\omega m + \rho c \qquad \text{(II, 2)}$$

where $m = \rho_w d_w$ is the mass per unit area of the wall. This formula is based on the assumption that the wall moves with the same velocity throughout. Its reactance due to its mass is obviously $i\omega m$ per unit area. It is backed by the characteristic impedance ρc of the air. The impedances $i\omega m$ and ρc must be added since they have the same velocity (acoustical series connection, see Fig. 49).

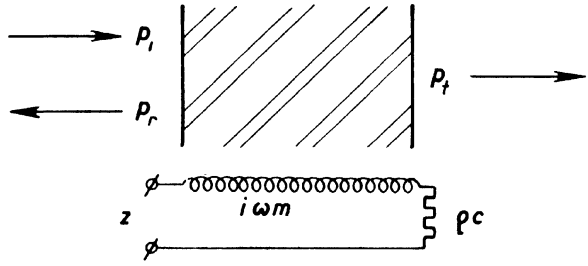


Fig. 49. The homogeneous wall and its electrical analogue.

The assumption of a constant velocity throughout the wall can be shown to be fulfilled accurately in most practical cases up to fairly high frequencies, because

(a) the wave-length at audio frequencies in the wall is very large in comparison with the wall thickness;

(b) the characteristic impedance of the wall material is very large in comparison with that of air (a condition often overlooked because it is fulfilled at all frequencies).

Combining (1) and (2) we get immediately

$$D_o = 10\ {}^{10}\log |t|^{-2} \left(= 10 \log \left| p_i \,/\, p_t \right|^2 \right)$$

$$= 10 \log \left[1 - \left| \frac{i\omega m}{i\omega m + 2\rho c} \right|^2 \right]^{-1}$$

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$$D_o = 10 \log \left| \frac{(2\rho c)^2}{(\omega m)^2 + (2\rho c)^2} \right|^{-1} \qquad \text{(II, 3)}$$

or approximately

$$\boxed{D_o \approx 20 \log \frac{\omega m}{2\rho c}}$$

the index zero in D_o pointing to normal incidence.

Eq. (II, 3) is generally known as the *mass law*. It indicates that the transmission loss D_o is completely governed—as far as the wall itself is concerned—by the mass per unit area of the wall. D_o should increase 6 db for every doubling of m. The same applies to the dependence upon frequency, *i.e.* for every octave increase in frequency D_o should increase by 6 db.

The derivation of (II, 3) looks straightforward and indeed it is. Nevertheless its validity is rather limited owing to assumptions that are not fulfilled in practice, *viz.* the sound does not impinge normally, but is in fact distributed at random over all possible directions in practice. Moreover the walls are not at all of infinite size. Clamping conditions at the edges are likely to be of importance. The third assumption that wave motion in the wall itself may be disregarded can be shown to be a safe one.

§ 2. The Coincidence Effect [2]

Several attempts have been made to develop theories of transmission based on assumptions that do not differ so much from practice as those underlying the mass law. The stiffness can be taken into account [3] by the addition of a term $1/i\omega C$ in eq. (II, 2), thus providing a system of one resonance frequency, in the neighbourhood of which a decreased transmission loss could be explained. Since a wall has, however, an infinite number of resonance frequencies it is not surprising that this approach turns out to be generally inapplicable. Another attempt can be made by treating oblique incidence along the same lines as the absorption of locally reacting materials under oblique incidence (*cf.* p. 100). It is, however, certain that walls are not in the least locally reacting since the stiffness of the wall provides the wall elements of a rather tight sideway coupling. The first scientifically sound approach was

made by SCHOCH [4] who proved that even for walls of finite dimensions the mass law should still hold for normal incidence at sufficiently high frequencies and further that the transmission loss could only be expected to be diminished in the frequency region that comprises a number of the lowest resonance frequencies. His theory is not applicable, however, to oblique incidence.

The most promising approach up to now is that of CREMER [2]. His theory is applicable to oblique and normal incidence on walls of *infinite* size, taking the stiffness of the wall into due account. The restriction that the wall should be of infinite size is thought, however, not to affect the results unduly.

A description in detail of CREMER's theory would surpass the space limit in this book. The reader is referred to the highly interesting original paper or to CREMER's book.

Let a plane single wave of pulsatance ω impinge with an angle of incidence θ upon an infinite wall. The pressure distribution along the surface is easily shown to have a periodicity corresponding to $\exp\left(-\frac{i\omega \sin\theta}{c} x\right)$, where x indicates the distance along the surface in the plane of incidence and c the ordinary sound velocity in air. Apart from possible transients a stationary forced vibration of the wall of the same periodicity is to be expected in the form of a flexural wave travelling along the wall.

Therefore we may put in the differential equation governing the motion of the wall,

$$\partial/\partial x = -i(\omega \sin\theta)/c. \qquad \text{(II, 4)}$$

If the sound pressure at the side of incidence be p_1, at the transmission side p_2, the differential equation is

$$p_1 - p_2 - B\frac{\partial^4 y}{\partial x^4} = m\frac{\partial^2 y}{\partial t^2}. \qquad \text{(II, 5)}$$

B is a measure for the flexural stiffness and is

$$B = \frac{EI}{1-\sigma^2}.$$

where E is the modulus of elasticity, I the moment of inertia per unit width, σ POISSON's ratio and y the displacement of the wall. Now

introducing $i\omega$ for $\partial/\partial t$ and $-(i\omega \sin\theta)/c$ for $\partial/\partial x$ and finally replacing $i\omega y$ by the wall velocity u we get

$$p_1 - p_2 = \left[i\omega m - i \frac{B\omega^3 \sin^4\theta}{c^4} \right] u. \qquad \text{(II, 6)}$$

The quantity between brackets is the specific impedance of the wall. It consists of two parts, the first being the normal inertia part; the second term is a capacitive one owing to the minus sign. It does not vary however as ω^{-1}, but as ω^{+3}. We deduce from (II, 6) that for every angle θ a frequency can be found for which the impedance vanishes. This is not called resonance, but coincidence and this for several reasons. To begin with there is not a single coincidence frequency, but it is readily deduced from (II, 6) that above a critical frequency

$$f_c = \frac{c^2}{2\pi} \sqrt{\frac{m}{B}}, \qquad \text{(II, 7)}$$

where coincidence takes place with $\sin\theta = 1$ or $\theta = \pi/2$, coincidence *always* takes place at a certain angle of incidence θ, θ decreasing with

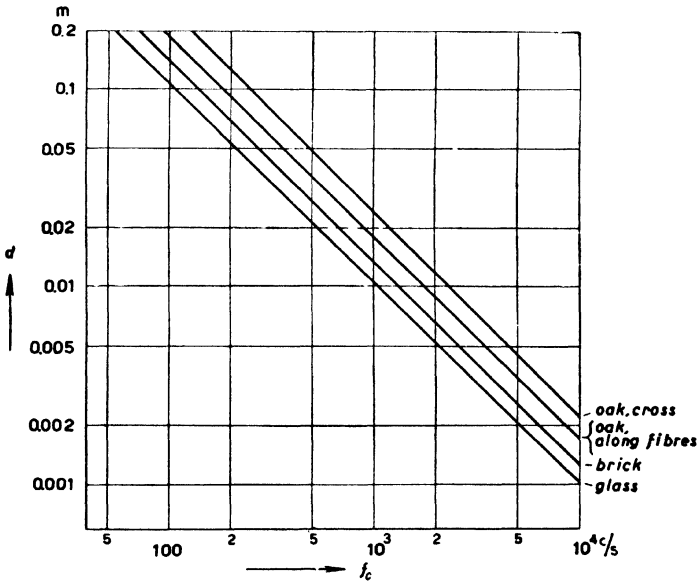


Fig. 50. Critical frequencies according to CREMER for a few materials of various thicknesses.

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increasing frequency. So there is a coincidence *region* with a lower limit given by (II, 7). Moreover this critical frequency has nothing to do with the lowest resonance frequency, clearly demonstrated by the fact that the dimensions of the wall do not enter into (II, 7); finally it increases with decreasing thickness of the wall, whereas the reverse applies to the lower resonance frequencies.

Eq. (II, 7) can be transformed into

$$f_c \approx \frac{c^2}{2c_{long} \cdot d}, \qquad \text{(II, 7a)}$$

where $c_{long} = \sqrt{E_w/\rho_w}$, *i.e.* the sound velocity of longitudinal waves in the wall, and d the thickness of the wall.

If the form between brackets of (II, 6) vanishes, a travelling flexural wave with finite amplitude can exist without driving pressure. This is obviously a so-called free progressive wave, and since the periodicity of the air wave along the wall must be equal to that of the wall the condition of coincidence can also be written

$$\frac{c}{\sin \theta} = c_1, \qquad \text{(II, 7b)}$$

where c_1 is the sound velocity of the free flexural wave along the wall. c_1 increases with frequency and since c_1 should be greater than c to satisfy (II, 7b), this equation shows that a lower frequency limit must be exceeded to make coincidence possible. We conclude from (II, 6)

(a) at very low frequencies the mass law should be valid,
(b) above a critical frequency, given by (II, 7), coincidence occurs at any frequency, but only for sound energy impinging in a specified direction that gradually tends to become perpendicular the higher the frequency is.
(c) the coincidence will probably decrease the transmission loss above the critical frequency, also if random incidence is present.

Along lines similar to those of SCHOCH it can be shown that we may disregard the restriction to infinitely wide walls without much change in the conclusions.

We must be content to borrow without any derivation the formulae for the average transmission loss in the coincidence region from CREMER's original paper. They have been derived by him on the assumption that the energy transmission above the critical frequency is

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entirely due to the energy transmitted in the neighbourhood of the coincidence angle for any frequency. He finds after adequate approximation for frequencies at least twice the critical frequency

$$\bar{D} = \tfrac{1}{2} D_{oc} + 20 \log (f/f_c), \qquad \text{(II, 8)}$$

where D_{oc} is the transmission loss at the critical frequency according to the mass law at normal incidence (eqn. (II, 3)).

This means that well above the critical frequency the transmission loss should be diminished by half its value at the critical frequency, which generally means a considerable decrease.

Internal friction is disregarded however. This can easily be done by assuming the modulus of elasticity that enters in the stiffness constant B of eqn. (II, 6) to be complex according to

$$E^* = E(1 + i\varepsilon).$$

The quantity ε is generally called the tangent of the angle of loss of the material; it is more or less independent of frequency. The numerical value of ε can be chosen so as to fit theory and experiment.

Within $\pm$ 2 db error CREMER then arrives, for frequencies at least twice the critical frequency, at

$$\bar{D} = D_{oc} - 10 \log 1/\varepsilon + 30 \log f/f_c - 3,$$

thus indicating a drop in the transmission loss that is 'adjustable' due to the uncertain value of the internal friction, and a *steeper slope* of the characteristic in the coincidence region owing to the factor 30 instead of 20 in the mass law. This is in keeping with various experimental results [2].

§ 3. The Single Cavity Wall

The equations governing the cavity wall of Fig. 51 in the case of normal incidence are

$$p_1 - p = i\omega m_1 u_1,$$

$$u_1 - u_2 = \frac{i\omega d}{\chi p_0} p,$$

$$p - p_t = i\omega m_2 u_2.$$

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Only the second equation needs some explanation. It is practically equal to the definition equation of the bulk modulus of air, that is known to be equal to χp_0. The quantity $(u_1 - u_2)/i\omega$ is the compression,

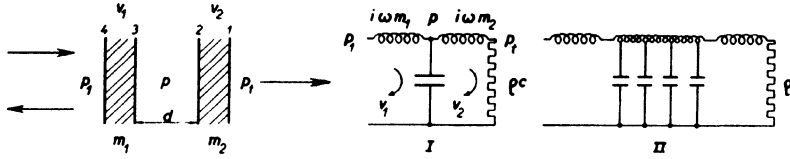


Fig. 51. The single cavity wall, a simplified electrical analogue for low frequencies (I) and a more general electrical analogue (II).

which after division by d gives the relative compression. This must be equal to the ratio of the excesspressure to the bulk modulus by the definition of the latter. The derivation is valid as long as the wavelength in air is very great compared with d, which normally is the case with cavity walls up to rather high frequencies. The electrical analogue I will now be clear. It allows us to compute directly the impedance of the cavity wall

$$Z = i\omega m_1 + \frac{\frac{\chi p_0}{i\omega d}(i\omega m_2 + \rho c)}{\frac{\chi p_0}{i\omega d} + i\omega m_2 + \rho c} \qquad \text{(II, 9)}$$

and from it the transmission loss D_0 at normal incidence along the lines of § 1.

$$r = (Z - \rho c)/(Z + \rho c)$$
$$|t|^2 = 1 - |r|^2$$
$$D_0 = 10 \log |t|^{-2}$$

This method of approach is possible since there is no energy loss in the system, so that the transmission is calculable from the reflection.

From (II, 9) one easily derives that at extremely low frequencies (far below $\omega_{1res.}$, eqn. (II, 11))

$$Z = i\omega(m_1 + m_2) + \rho c, \qquad \text{(II, 10)}$$

i.e. the cavity wall behaves in this frequency range like a single wall

of the same mass. After adequate approximation one finds further that the system has a resonance frequency at about

$$\omega_{1res.} = \sqrt{\frac{(m_1 + m_2)\chi p_0}{m_1 m_2 d}}, \qquad \text{(II, 11)}$$

which reduces to

$$\omega_{1res.} = \sqrt{\frac{2\chi p_0}{md}} \qquad \text{(II, 11a)}$$

for a cavity wall with identical leaves. The two leaves vibrate in this case in opposite phase, dividing the compressibility of the layer of air in between them. To give some idea of the order of magnitude of this first resonance frequency, if

$$m_1 = m_2 = 200 \text{ kg/m}^2 \text{ (ca } \tfrac{1}{2}\text{— brick leaves)}$$
$$d\ (= 5 \text{ cm}) = 0.05 \text{ m.}$$
$$f_{1\,res.} \approx 20 \text{ c/s.}$$

This implies that eqn. (II, 10) is of no practical importance in general.

It can easily be shown that the transmission loss is exactly zero in the first resonance, if the density of the air in the cavity is taken into account (see further and Fig. 52), independent of the stiffness of the wall.

Between the first resonance and the higher resonance frequencies, to be considered next, the following approximate formulae hold:

$$D_o = 20 \log \frac{\omega^3 m_1 m_2 d}{2\rho^2 c^3}$$
$$= D_{o1} + D_{o2} - 20 \log \frac{c}{2\omega d}. \qquad \text{(II, 12)}$$

They can be derived with careful approximation from eqn. (II, 9).

This generally means a considerable gain in transmission loss over that of a single wall of the same mass, and since (II, 12) applies in an extended frequency range it proves the effectiveness of a cavity wall for insulation of airborne sound.

At rather high frequencies (say more than 2000 c/s) wave motion in the cavity may no longer be disregarded and therefore neither the basic equations, nor the electrical analogue I in Fig. 51 are valid. Analogue II in Fig. 51 is the correct one up to frequencies at which

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wave motion in the walls should be regarded and, therefore, also the corresponding inductances must be replaced by suitable 4-terminal networks. We will not enter into discussion of the rather complicated formulae governing the behaviour in this range of high frequencies. It should be stated, however, that if wave motion in the air is taken into account an infinite number of resonance frequencies is encountered, where the transmission loss is approximately zero, and even exactly so if the leaves are of equal mass.

This can be seen without calculation from Fig. 52, where the impedance z of the cavity wall has been constructed for a few frequencies.

The construction can be accomplished in the following way. In point 1 of the

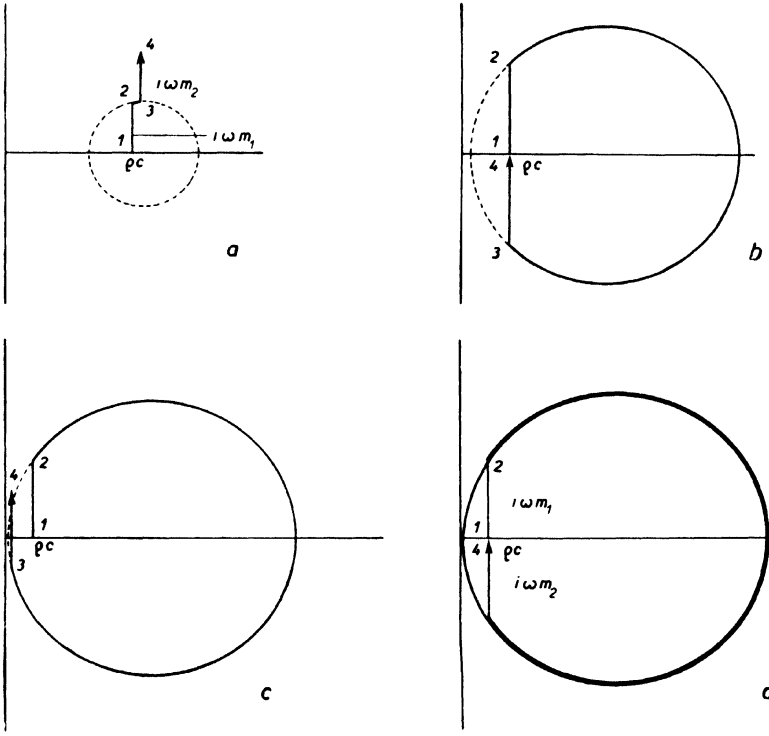


Fig. 52. The construction of the impedance of the cavity wall at
a. extremely low frequencies,
b. first resonance,
c. a few hundred c/s,
d. second resonance.

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cavity wall (see Fig. 51) the impedance to the right of this point is equal to ρc, the characteristic impedance of air. The corresponding point in the impedance planes of Fig. 52 is also labelled 1. If we proceed to the left in Fig. 51 from 1 to 2 the impedance changes accordingly. If the stiffness of the wall is not taken to be infinite the corresponding change (1 to 2) in the impedance planes of Fig. 52 is part of a circle of constant reflection coefficient that is followed in a clockwise direction (see also Chap. 4 § 4 Fig. 17). The circle has a very great diameter as a consequence of the characteristic impedance of the wall material being very much greater than that of air and, since we only go through a small part of it (the wave-length in the wall still being large in comparison with the wall thickness), this part may be approximated by a straight line of length $i\omega m$. This approximation is identical with assuming the wall to be of infinite stiffness. This assumption is generally made (also in analogues I and II), but is superfluous for the graphical construction. Proceeding further to the left from 2 to 3 another part of a circle is gone through, more rapidly this time since the sound velocity in air is much smaller than in the wall. The circle is of a smaller diameter, ρc now being the basic point of the set of circles, out of which one circle is followed. From 3 to 4 part of an enormous circle is again passed through. It can be approximated by a vertical line $i\omega m_2$. The resulting impedance ζ in point 4 depends of course upon the values of m_1, m_2, d and ω. A few conditions that should be fulfilled to have zero transmission loss can be easily read from Fig. 52. For the first resonance (at say 20 c/s, see Fig. 52b) m_1 should obviously be equal to m_2, in order to get a symmetrical figure that ends in ρc. The same applies to the higher resonance frequencies (Fig. 52d). Moreover one reads from this figure (Fig. 52d) that for the second and higher resonance frequencies d should very nearly be equal to a whole number of half wave-lengths in air. Fig. 52d shows the situation for the second resonance. The loop from 2 to 3 comprises a complete circle, corresponding to one half wave-length displacement along the cavity thickness, and an incomplete circle that corresponds, however, to a negligibly small part of a half wave-length (compare the scale of the phase angle Δ along circles of great diameter in Fig. 17 of Chap. 4).

Since the second and higher resonances occur at $d \approx j\lambda/2$ this implies *e.g.*, for a cavity of $d = 0.1$ m, dips in the transmission loss characteristic at about 1700, 3400, 5100 c/s.

In Fig. 53 [5] a theoretical curve is given for a double window, showing a number of dips in the characteristic above 1000 c/s due to this effect. Curve A is the characteristic of a single window of the same mass, curve B that of eqn. (II, 12). The gain at the higher frequencies over the single window is considerable but for the various dips. The picture given is in general agreement with measurements. Cavity walls are sometimes found to be inferior at very low frequencies in comparison with the single wall of the same mass. At higher frequencies, however, the reverse is true.

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It should be borne in mind that attention is paid neither to the finite dimensions of practical constructions nor to the various causes of energy dissipation.

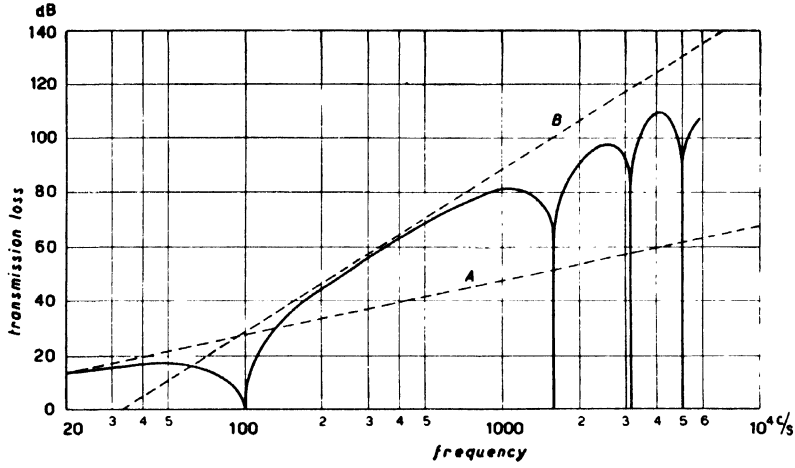


Fig. 53. Theoretical transmission loss [5] in decibels as a function of frequency for a double window ($m_1 = m_2 = 7.5$ kg/cm², $d = 0.1$ m).

§ 4. Multiple Cavity Walls [6]

If a great many identical leaves are mounted at regular intervals the electrical analogue becomes a long LC-filter with a low-pass character.

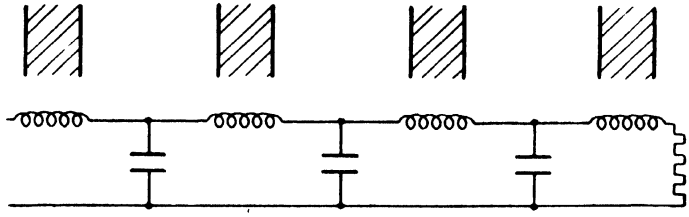


Fig. 54. The multiple cavity wall and its analogue for low frequencies.

All frequencies above the cut-off frequency should be reflected. One computes easily for this frequency

$$\omega = 2\sqrt{\frac{\chi p_0}{md}} \qquad \text{(II, 13)}$$

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The difficulty is to find practical constructions with a low cut-off frequency and an acceptable overall thickness. Nevertheless E. MEYER succeeded in obtaining surprisingly good results (55 db above 90 c/s with a wall of only 50 kg/m², but 45 cm thickness).

§ 5. Some Theoretical Aspects of Structure-Borne Sound

Abatement of structure-borne sound can be divided into two parts. One should first try not to generate it. If it is already present in the solid material the problem is to prevent its propagation. Two fundamental formulae can be given in this respect. If a travelling wave in one solid medium impinges in normal direction upon the boundary with a second medium of infinite extent, reflection takes place. The expression for the reflection coefficient can be borrowed at once from absorption theory to be

$$r = \frac{\rho' c' - \rho c}{\rho' c' + \rho c} \qquad \text{(II, 14)}$$

(*cf.* Chap 1, eqn. 35) where $\rho' c'$ and ρc are the characteristic impedances in the respective media. The expression holds also for flexural and shear waves. From it the transmission loss in decibels may be calculated along the same lines as with air-borne sound:

$$D = 10 \log [1 - |r|^2]^{-1} .$$

It turns out that a transition from metal to brickwork or vice versa gives about 3 db transmission loss, whereas 8 db approximately is found for a transition wood-metal or metal-wood.

Eqn. (II, 14) has a very limited applicability since the second medium is supposed to be of infinite extent.

A second formula of more practical value can be given for the case of an elastic slab enbedded in solid medium (Fig. 55). The electrical analogue of Fig. 55 will be clear; the slab is considered as a spring without mass (electrical analogue a capacity), the medium at the transmission side is supposed to be infinitely long, so its impedance can be put equal to ρc, the characteristic impedance of the solid. Since the impedance of the slab is $1/i\omega C$ (C is the compliance of the slab per unit area) we get immediately for the impedance z just before the slab

$$\frac{1}{Z} = i\omega C + \frac{1}{\rho c}$$

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which, introduced in the reflection formula, gives for the reflection coefficient

$$r = \frac{i\omega C}{i\omega C + 2/\rho c}$$

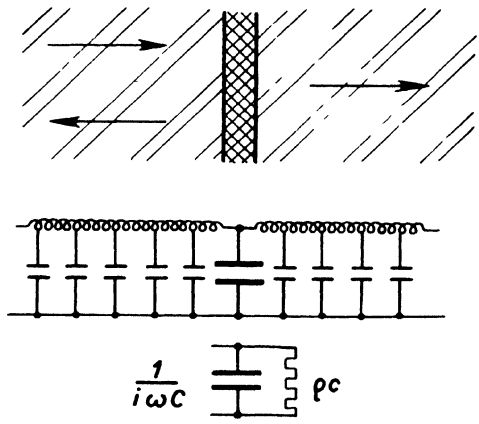


Fig. 55. Reflection of structure-borne sound by an elastic slab.

and from r as previously

$$\text{transmission loss} = 10[1 - |r|^2]^{-1}$$

$$= 10 \log \left[\left(\frac{\omega C \rho c}{2} \right)^2 + 1 \right]. \qquad \text{(II, 15)}$$

This is an expression very similar to the mass law. It turns out that in order to obtain a reasonable transmission loss fairly compliant slabs are necessary. A 3 db-gain is obtained for

$$\omega C \rho c = 2.$$

Taking $\rho c = 10^7$ MKS-units (brickwork) and $\omega = 2000 (f \approx 300$ c/s), C should be in this case 10^{-10}, and since $C \approx d/E$, where $d =$ thickness of the slab and E its YOUNG's modulus, E should be about 10^8 MKS-units for a slab of 1 cm thickness. This seems to be in keeping with practice.

References p. 121

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CHAPTER 6

ROOM ACOUSTICS AND THE DESIGN OF AUDITORIA

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§ 1. Introduction

Investigating the behaviour of sound waves in rooms is essentially different from doing so in free space where sound sources are usually considered to be small and all the energy flows away from them, often under steady state conditions. In a room the bounding surfaces confine the sound energy to the space they enclose and the waves undergo many reflections before they are absorbed completely. The types of sound sources encountered in practice are no longer always small, even compared with the room dimensions; they usually have directional characteristics which are difficult to specify and they emit sounds which are essentially transient and not steady state in character. This gives rise to a very complex distribution of sound in even the simplest of enclosures.

If it were analytically possible to specify completely the variations in sound intensity due to frequency, space and time in any room under conditions of normal use it is debatable whether such an analysis is necessary. It would only be essential if one particular distribution was preferable above all others and it would then be the objective to design rooms which gave the required distribution. Now the final arbiters who decide whether or not a room is acoustically satisfactory are the people who have to use it for the purpose for which it is designed. Their judgment must be based on subjective experience and the answers they give are largely a matter of personal taste, so there is no unique solution but rather the designer must aim to give the greatest satisfaction. There are relatively few cases where the required sound

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distribution does not depend on subjective experience, but these generally arise in the measurement of acoustic properties of materials to be used in rooms needing subjective appraisal.

It is first necessary, then, to determine from common experience what are the positive and negative attributes most desirable for a particular purpose. These qualities must then be related to the physical properties of the room which can be measured by objective methods. The study of the behaviour of sound waves in rooms known to have good and bad qualities provides a means of obtaining this relation as long as a sufficient number of objective techniques are available. At present, there are a limited number of techniques and these are more useful in fault finding than in measuring positive qualities. Thus while it is reasonably possible to design a new room which will have no obvious faults it is by no means certain that it will be a good room acoustically.

§ 2. Desirable Qualities

Rooms whose acoustic qualities are important can, broadly speaking, be divided into two classes:

(a) Those used primarily for speech,

(b) Those used for music.

There are, of course, rooms which have to be used for both and, if necessary, some compromise has to be obtained here.

Now one effect of enclosing the sound source is to concentrate the energy so that more reaches the ears of listeners than would in free space. With speech, this means that the unaided voice is able to reach a greater number of people and the speaker prefers a room in which the energy is not absorbed quickly, giving a higher sound level for a given amount of effort. The primary requirement for the listener is that the room shall not distort the speech in any way, leaving it both intelligible and natural. That is, the room has merely to act as an acoustic transmission line, reproducing faithfully at the ears of the listener the sounds put into it by the speaker.

With music, however, either as the result of long experience in listening in enclosed spaces or as a fundamental fact, it is found desirable for the sounds emitted to be modified by the acoustic behaviour of the room to give a more pleasant effect in the ears of the listener. Some rooms have the qualities necessary to give this enhancement, others

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have not and it is here that the subjective opinions differ. Those rooms which are most pleasing appear to have some or all of the following qualities:

(a) Resonance or fullness of tone, described also by various other terms, all of which imply that the sounds have a certain amount of body and the performers do not find it difficult to get an adequate amount of sound to the listeners.
(b) Good definition so that a succession of quickly played notes are all heard clearly and distinctly.
(c) Adequate balance so that all departments of the orchestra are heard everywhere in the correct proportions.
(d) Sufficient blending that a number of instruments playing together sound as an ensemble rather than a juxtaposition of individual sounds.
(e) Freedom from troublesome echoes.
(f) A low level of intruding noise.

Consideration of the way in which sound waves behave in rooms is necessary to see how these properties can be obtained.

§ 3. Sabine Analysis

With a steady sound source radiating in a room there is an energy balance reached when the rate of radiation from the source is equal to the rate of absorption. Now sound energy is absorbed to a small extent at high frequencies in the air itself, but principally at any surface exposed to the sound waves. The absorption coefficient, which is the absorption per unit area of surface, is defined as the ratio of absorbed to incident energy. The steady state sound level produced by a source of constant power varies inversely with the total amount of absorption in the room. When the source is suddenly stopped, however, the energy density falls to zero at a rate which is again determined by the total absorption present. Since most of the sounds commonly met with are essentially transient in character this rate of decay of sound is a very important quantity in room acoustics.

If, as SABINE did, it is assumed that in the steady state condition the energy distribution is uniform all over the room, that no particular direction is preferred and that energy is being absorbed continuously

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at the bounding surfaces, air absorption being neglected, then the energy density E at time after the source starts is given by [1]:-

$$E = (4\bar{E}/Ac)\ [1 - e^{-Act/4V}] \qquad \text{(II, 16)}$$

where $\bar{E}$ is the rate of emission of energy from the source,
A the total absorption in the enclosure,
V the volume of the enclosure,
c the velocity of sound in air.

The maximum, or steady state value of E is $(4E/Ac)$ and the rate of decay determined by the term $e^{-Act/4V}$. The form of this relation was first determined experimentally by W. C. SABINE who defined the reverberation time as that taken for the energy density to fall through 60 decibels (db). The energy equation gives this time to be:

$$\tau = \frac{24 \log_e 10\, V}{Ac}$$

For measurements in English units the constant is equal to 0.049.

i.e. $$\tau = 0.049 V/A \qquad \text{(II, 17)}$$

The value of the constant when metres are used is 0.16.

The range of 60 db chosen by SABINE is quite arbitrary but was convenient experimentally in his case.

If truly diffuse conditions do hold, the decay curve is a straight line on a logarithmic scale whose slope can be used to determine the reverberation time. In practice it is often not possible to obtain a full decay of 60 db and the value is taken from the slope of the decay obtainable.

Now the total absorption in the room is the sum of all the individual amounts of absorption, each of which is the product of a surface area S and an appropriate absorption coefficient a. Since the total surface area is finite and the maximum value of a is unity when all the energy is absorbed the value of A remains finite and hence from eqn. (17) so does τ. Obviously the free space value of $\tau = 0$ is to be expected in this case and the SABINE equation thus gives the wrong value when the surfaces are highly absorbing.

In practice the decay curves obtained are often not straight lines showing that the conditions for the simple geometrical analysis are not obeyed. With non-uniform conditions the behaviour of the sound has to be investigated by wave analysis which is a complex procedure

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except for rooms of the simplest shape. Because of this, and since also the geometrical analysis gives results which are of value in a large number of practical cases, there have been various attempts to modify the SABINE equation. These mainly take into account the fact that, apart from that in the air itself, absorption does not take place continuously, but only at discrete intervals when the sound waves strike the walls. EYRING [2], by considering the source in the room together with a set of image sources corresponding to single, double and multiple reflections in the walls each with a strength proportional to the number of reflections, derived the formula

$$\tau = \frac{0{\cdot}049\ V}{S[-2{\cdot}3\ \log_{10}(1-\bar{a})]} \qquad (11, 18)$$

where $\bar{a}$ is the average absorption coefficient of all the room surfaces and S the total surface area.

At high frequencies and in rooms of large volume the effect of air absorption is no longer negligible and the formula is further modified to:

$$\tau = \frac{0{\cdot}049\ V}{-2{\cdot}3\ S\ \log_{10}(1-\bar{a}) + 4mV}$$

where the value of m, the air absorption coefficient which varies with humidity and frequency, is given by the curves of Fig. 56 [3].

§ 4. Measurement of Reverberation Time

There are various methods of measuring reverberation time. Obviously the original method of listening by ear in conjunction with a stopwatch, although remarkably accurate in the case of a few observers long practised in the art, is only really effective when the period is long. The present technique is to use high quality audio-frequency amplifiers fed from a non-directional microphone situated at the position in the room for which the reverberation time is wanted. Band-pass filters are incorporated into the amplifiers and the output could be displayed on a cathode ray tube from which the reverberation time could be read directly or from photographs of the trace. Some logarithmic device incorporated into the circuit is necessary to give a reasonable proportion of the 60 db decay on the screen. A more convenient method is to use a logarithmic level recorder which converts

the linear output from the amplifier into the logarithmic displacement of a stylus across a waxed paper travelling underneath at a pre-selected

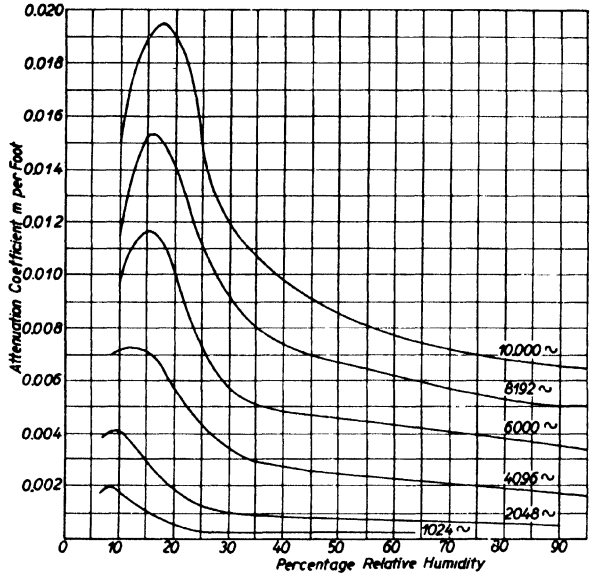


Fig. 56. Variation of Attenuation of Sound in Air with Humidity (KNUDSEN).

speed. Since its highest writing speed is between 500 and 1000 db/second it can follow any rapid fluctuations in the decay if required, or, set at lower writing speeds, the curve can be smoothed to any extent desired. Fig. 57 shows a typical reverberation decay curve at two

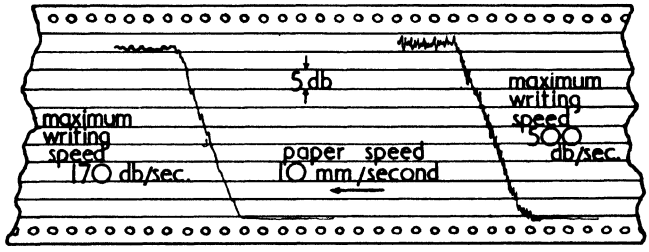


Fig. 57. Typical Decay Traces obtained with 1 kc/s ± 100 c/s @ 5 c/s Warble Tone in an unfurnished Living-room having a Reverberation Time of 2.2 sec.

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different writing speeds of the recorder. The reverberation time can be read off directly from the slope with the aid of a special protractor calibrated for the particular total range across the paper in db and for the speed of traverse underneath.

In the case of auditoria of various kinds the reverberation time will vary according to the total number of people present since they provide a considerable proportion of the total absorption. It is therefore important to know the reverberation time under these conditions and with the same type of sound source as will be heard in practice, *i.e.* the orchestra. Three types of noise sources are suitable for reverberation measurements: warble tones, white noise and orchestra. Pure tones give inconsistent results, the curves are often critical with respect to frequency and microphone position and are also rarely straight lines, hence frequency modulated tones are used. The amount of deviation from the mean frequency and the rate of warbling determine the spectrum obtained. With a frequency deviation $\pm \Delta f$ and warble frequency α there are approximately $2\Delta f/\alpha$ pure tone components in the warble band [4]. To get as great a number of those components as possible with amplitudes not differing too greatly, the ratio $\Delta f/\alpha$ should be as large as possible. But when the absorption characteristics, and hence the decay time, vary with frequency Δf should be small and α sufficiently rapid for the band to be swept through in a short time compared with the decay. A reasonable compromise is obtained with a deviation of $\pm$ 10 per cent of the mean frequency at a rate of 5 c/s.

White noise, *i.e.* that which contains all frequencies, when used as a source can either be generated electrically or by use of a pistol since the noise produced by this is sufficiently white for practical purposes. The particular band of frequencies for which the reverberation time is required is then filtered electrically from the complete spectrum. Generally bands an octave wide are sufficient but when the decay rate varies with frequency the method gives an answer corresponding to the longest decay time in the band, which may not be that of the midpoint. Hence in some cases $\frac{1}{3}$rd octave bands are desirable.

With an audience present the orchestra is usually the only sound source available for reverberation time measurements and in practice the values so obtained agree closely with those using a pistol as source.

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§ 5. Reverberation Time as an Index of Room Performance

Although the reverberation time is by no means the only physical quantity which can be measured objectively it is still by far the most important factor in linking up with subjective experience. The optimum reverberation time for a room, generally specified at a frequency of 500 c/s, depends on its volume and the use to which it is to be put.

The best listening conditions for speech are obtained with a short reverberation time. This is because the character of speech is such that the energy is concentrated into discrete syllables separated from each other by small but finite lengths of time. The effect of too long a reverberation time is to fill in these gaps and mask the following syllables, thus reducing intelligibility. The requirement for a long reverberation time to give a high sound level for the unaided voice is not really important since a well-designed speech reinforcement system can provide an adequate sound level with a high degree of naturalness and intelligibility in an auditorium of practically any size. In general a reverberation time of about one second is the longest that can be tolerated for satisfactory speech listening conditions.

With music, however, the optimum reverberation time is longest for choral and organ music, perhaps as the result of a long tradition of listening to such music in very reverberant churches. For orchestral music the period is a little less and is shortest for chamber music. The reverberation formula shows that, other things being equal, τ increases according to the linear dimensions, *i.e.* is proportional to the cube root of the volume. Fig 58 shows the recommended optimum reverberation time for orchestral music as found by BAGENAL and WOOD [5] from a study of British and European concert halls, and as given by KNUDSEN and HARRIS [6] from a similar study in America. The figure also shows the measured reverberation times with audience of a number of existing British concert halls [7].

The reverberation characteristic as a function of frequency is also important. It should show no marked peaks or dips and it is probably best for it to be flat over the whole frequency range. It is generally found to fall off at high frequencies due to air absorption while at low frequencies it may be allowed to rise because of the crowding of the contours of equal loudness at this end of the scale. At 100 c/s the value may increase up to $1\frac{1}{2}$ times the figure at 500 c/s. [8]

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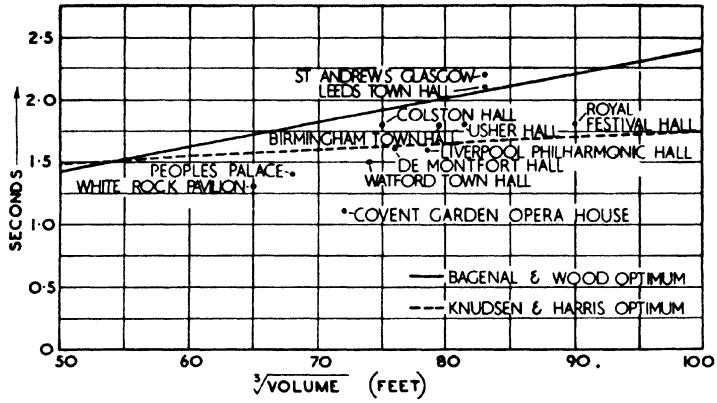


Fig. 58. Reverberation Times of (full) Halls at 500 c/s.
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§ 6. Wave Analysis of Room Behaviour

In introducing a talk on the modern principles of room acoustics BOLT [9] compared the SABINE and associated geometrical analysis of the acoustic behaviour of room to listening to an orchestra from one of the furthest seats. The gross outline of the music is apparent enough but the detail of the fast moving lines is lost and it is difficult to follow the weaker passages without effort, so a broad generalised effect of the whole is obtained. In an endeavour to fill in some of this missing detail the wave theory of room acoustics was developed. By starting at the other end of the scale and studying the behaviour of individual waves in relation to the boundary conditions we get the component parts of the whole picture.

The materials which form the bounding surfaces of a room are of great importance in governing the way in which the room behaves. Their acoustic properties can be uniquely specified by the ratio of pressure to particle velocity at the surface, which is the acoustic impedance and is generally complex. Since also all the other field variables can be evaluated from these two quantities they are the most convenient choice.

Considering first the limiting case of a rectangular room with perfectly hard walls the solution of the wave equation

$$\nabla^2 p = 1/c^2 \, \partial^2 p / \partial t^2 \qquad \text{(II, 19)}$$

is of the form: $p = \begin{matrix}\cos \\ \sin\end{matrix}\left(\omega_x x/c\right)\begin{matrix}\cos \\ \sin\end{matrix}\left(\omega_y y/c\right)\begin{matrix}\cos \\ \sin\end{matrix}\left(\omega_z z/c\right)\exp(-2\pi i f t)$ (II, 20)

where $$f = (\tfrac{1}{2}\pi)\,(\omega_x^2 + \omega_y^2 + \omega_z^2)^{\frac{1}{2}}$$ (II, 21)

Choice of either the sine or cosine gives a symmetrical pressure wave about an origin in the centre.

Since pressure and velocity are related by:-

$$u = \left(1/i\omega\rho\right)\left(\partial p/\partial x\right) \text{ etc.,}$$ (II, 22)

we get for the velocity u in the x direction

$$u = \left(\omega_x x/2\pi i f \rho c\right)\begin{matrix}-\sin \\ \cos\end{matrix}\left(\omega_x x/c\right)\begin{matrix}\cos \\ \sin\end{matrix}\left(\omega_y y/c\right)\begin{matrix}\cos \\ \sin\end{matrix}\left(\omega_z z/c\right) \exp.\,(-2\pi i f t)$$

The condition that the normal velocity at the wall face ($x = \pm l_x/2$) is zero for a perfectly hard surface means that

$$\omega_x = \left(c/l_x\right) n_x \pi \quad (n_x = 0, 1, 2, 3 \text{ etc.})$$ (II, 23)

where even values of n_x arise from choice of the cosine function in the expression for p and odd values from the sine. The resultant natural frequencies or normal modes of vibration of the room are:-

$$f = \omega/2\pi = \left(c/2\right)\left[\left(n_x/l_x\right)^2 + \left(n_y/l_y\right)^2 + \left(n_z/l_z\right)^2\right]^{\frac{1}{2}}$$ (II, 24)

Thus there is a normal mode for each combination of integral values of n_x, n_y and n_z caused by a plane wave starting along a line whose direction cosines are given by the corresponding values of ω_x/ω, ω_y/ω and ω_z/ω.

The form of equation (24) suggests that the various allowed values of the frequency, f, can be plotted in three-dimensional cartesian co-ordinates with components $\omega_x/2\pi = cn_x/2l_x$ etc. Since n_x, n_y and n_z are integral numbers, all the different frequencies permitted by the equation will lie on a lattice in this frequency space at points corresponding to integral multiples of $c/2l_x$ in the f_x direction, $c/2l_y$ in the f_y direction and so on as shown in Fig. 59. Thus each point is associated with an element of volume $(c/2l_x)(c/2l_y)(c/2l_z)$ in frequency space and the radius vector from the origin to any point gives the frequency of

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that particular normal mode by its length and the direction of the standing wave by its direction.

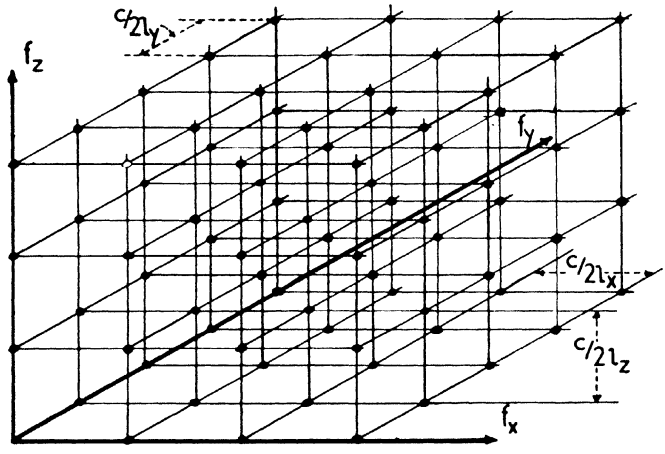


Fig. 59 Distribution in Frequency Space of Normal Modes for a Rectangular Room.

When the frequency is high enough the total number of normal modes, N, less than this frequency is given approximately by the volume of the positive octant of the sphere, radius f, divided by the element of volume occupied by each point.

$$i.e.\ N \approx (\pi f^3/6)\,/\,(c^3/8l_x l_y l_z) = (\pi f^3/6)(8V/c^3) \qquad \text{(II, 25)}$$

where $V = l_x l_y l_z$ is the room volume.

Now those lattice points lying on the axes themselves have three-quarters of their associated elements of volume lying outside the positive octant. They represent those normal modes for which two of the three integers n_x, n_y and n_z in equation (24) are zero and the standing waves are purely in directions perpendicular to the wall faces, *i.e.* along the axes of the room and are known as axial waves. The number of these in the x direction is f divided by the lattice spacing *i.e.* $2fl_x/c$ and the total number of axial waves in all three directions, N_{axial}, is $(2f/c)(l_x + l_y + l_z)$. But for the rectangular room the total edge length, $L = 4(l_x + l_y + l_z)$.

$$N_{\text{axial}} = fL/2c \qquad \text{(II, 26)}$$

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The other special class of normal modes is that for which only one n in equation (24) is zero and are due to waves travelling parallel to any of the wall surfaces in the room. All the remaining points lying in the $f_x f_y$, $f_y f_z$ and $f_z f_x$ planes of frequency space, except those on the axes themselves represent such modes. The total number having a frequency less than f is given by the area of those parts of the disks, radius f, lying in the planes bounding the positive octant, divided by the appropriate areas occupied by each point in the planes, less $(fL/2c)$ for the axial waves.

$$i.e.\ N_{\text{tangential}} = \left(\pi f^2/4\right)\left[\left(2l_x/c\right)\left(2l_y/c\right) + \left(2l_y/c\right)\left(2l_z/c\right) + \left(2l_z/c\right)\left(2l_x/c\right)\right] - \left(fL/2c\right) \qquad \text{(II, 27)}$$

But the total surface area, S, of the room is $2(l_x l_y + l_y l_z + l_z l_x)$

$$\therefore N_{\text{tangential}} = (\pi f^2 S/2c^2) - (fL/2c) \qquad \text{(II, 28)}$$

The rest of the points in frequency space represent those waves for which none of the n's are zero, *i.e.* oblique waves, and the number of these by a similar analysis is found to be:

$$N_{\text{oblique}} = (4\pi f^3 V/3c^3) - (\pi f^2 S/4c^2) + (fL/8c) \qquad \text{(II, 29)}$$

Thus the corrected value for the total number of normal modes of all types is

$$N = (4\pi f^3 V/3c^3) + (\pi f^2 S/4c^2) + (fL/8c) \qquad \text{(II, 30)}$$

By differentiation of any of these expressions for N with respect to f the number of such modes in a bandwidth df is obtained. The symmetry of a rectangular room is such that different modes of vibration can have the same resonant frequency. This degeneracy is important at low frequencies where the total number of modes in a given band is small and in the region of a many-fold degeneracy all the energy is concentrated at that particular frequency giving a less uniform response for the room.

The expression for the total number of modes in a room of given volume has been shown [10] to be independent of room shape. The term involving L becomes indeterminate in some cases when the room is not rectangular. It is therefore impossible to change the total number

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of modes merely by choosing a particular shape for the room.

So far no account has been taken of the loss of energy at the walls. The absorption coefficient introduced previously has only a precise meaning when the sound field is completely diffuse, and, according to the present theory, this is only true at high frequencies when there are enough normal modes in a given frequency band to assume that all directions of the corresponding waves in the room are equally probable. Very often this is not true and it is more convenient to consider the energy lost in terms of the acoustic impedance of the surface, defined as the complex ratio of sound pressure to the normal air velocity at the surface. The real part of this quantity governs the energy loss and the imaginary part the phase of the reflected wave. The effect of the absorbing material is to modify completely the sound distribution in the room, since each standing wave is reacted upon differently and it is not generally sufficiently accurate to compute the result of introducing absorbent walls merely as a change of boundary conditions to the case with rigid walls already considered. When the walls of a rectangular room are not perfectly hard they affect the damping of the various standing waves in different ways. Waves travelling parallel to a wall are damped approximately half as much as those striking it obliquely. This means that the distribution of normal modes present under steady state conditions changes as the sound is allowed to decay; the oblique waves being absorbed most strongly, the tangential waves less so and the axial waves least of all. Decay curves have double slopes when such conditions are found in practice.

Wave analysis of the behaviour of sound in rooms of simple geometrical shape with uniform absorbents on the walls is complex; for rooms of normal shape having an irregular distribution of absorption it is infinitely more so. The real value of wave acoustics at present does not lie in attempting exact calculations for auditoria of complicated shape but in investigating more basic problems, particularly the behaviour of sound absorbing materials.

§ 7. Other Techniques and Indices

(i) *Pulse Measurements*

Apart from the straightforward measurement of the reverberation time as the slope of the decay curve it is also useful to examine in

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detail the fluctuations in this curve, particularly during the first few hundred milliseconds. If a pulse of sound such as a pistol shot is used as a source, any obvious echoes should show up as peaks in the decay, but some sort of criterion is necessary to decide whether or not any particular peak constitutes an echo. The work of HAAS [11] on the influence of single echoes on the intelligibility of speech is useful in this respect.

HAAS found that if two different sources were emitting the same sound reaching the listener's ear at the same time in equal intensities then the apparent direction of the sound is somewhere between the two sources. If, however, one of them is delayed so that the sounds from it reach the listener between 5 and 35 milliseconds after the other then everything appears to come from the undelayed source and it is not obvious that the delayed source is operating at all. For delays between 35 and 50 milliseconds the delayed source is recognised as being present but the direction is still located with the undelayed sound. It is only with delay times greater than 50 milliseconds that the secondary source is heard as a distinct echo. The effect is independent of the position of the delayed source and the increase in loudness is proportional to the increase in power, *i.e.* an increase of 3 db on a single source. It is also found that the secondary source has to be 10 db louder than the primary, when delayed 5 to 35 milliseconds, for the two to be heard with equal loudness.

Beyond 50 milliseconds the delayed sound begins to impair the

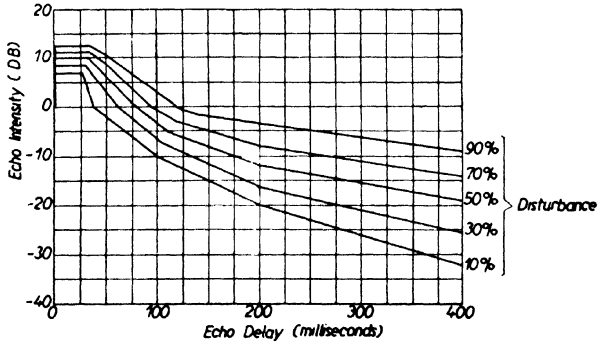


Fig. 60. Tentative Criterion for the transient Response of Rooms (DOAK and BOLT).

naturalness and intelligibility of speech, the percentage disturbance being a function of the relative intensity and delay time of the echo. DOAK and BOLT [12] have extended the original work with the aid of other results and replotted the percentage disturbance curves of HAAS in the form shown in Fig. 60 as a tentative criterion for the transient acoustic response of rooms.

The HAAS results only strictly apply for single echoes and were obtained with speech, whereas in large rooms the direct sound is followed by a multiplicity of reflections. It is known for instance that a long delayed echo is less disturbing than the curves of HAAS would suggest when other echoes are present in the intervening time. The criterion of Fig. 59 takes some account of this for speech and it may be possible to develop a similar criterion for music. In any case the fine structure of the decay curve is important in assessing the transient response of rooms.

The inertia of the level recorder is too great for the initial part of the trace to be examined for transient response. A long afterglow cathode ray tube with a comparatively slow speed scan is the most useful apparatus and the trace can be photographed if a permanent record is wanted. If the sound causes a linear displacement of the trace only the first 20 db approximately of the decay can be examined but this is probably the most important range. Incorporation of some sort of logarithmic device in the signal amplifiers enables this range to be extended with some reduction in the possible accuracy.

(ii) *Transmission Irregularity*

The performance of a room can also be measured under steady state conditions. If a loudspeaker emitting a pure tone is placed in one part of the room the sound pressure at any other point varies considerably with the frequency of the source. The room acts as a frequency sensitive transmission line between the source and the microphone measuring pressure.

To measure this variation of response with frequency the microphone output, after amplification, is fed to a logarithmic level recorder and the frequency of the source is changed sufficiently slowly to give effectively steady state conditions at any instant. The trace obtained varies with the rate of change of frequency, the writing speed of the

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recorder and the frequency bandwidth; a typical trace is shown in Fig. 61.

The transmission irregularity is defined by WENTE [13] as the sum of

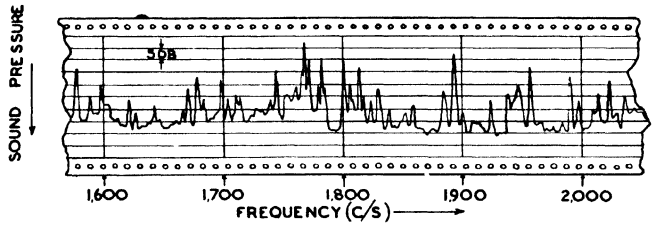


Fig. 61. Typical Transmission Irregularity Trace.

all sound pressure levels at maxima of the curve (in decibels) minus the sum of all the sound pressure levels at minima of the curve (in decibels).

A more significant figure, first suggested by ROOPE and BOLT [14], is the frequency irregularity, F.I., which is the transmission irregularity per cycle and is obtained by dividing throughout by the frequency interval, *i.e.*

$$\text{F.I.} = \frac{\Sigma_{fa}^{fb} \text{ db peak} - \Sigma_{fa}^{fb} \text{ trough}}{fb - fa} \qquad \text{db/cycle/sec.}$$

The greatest rate of change of frequency, R, permissible without losing steady state conditions is given by

$$R < (2/\pi\sqrt{3})\,(7/\tau)^2$$

where τ is the reverberation time of the room being measured.

This means that a single measurement from 50 to 10000 c/s takes about three hours. The frequency irregularity is calculated from the trace by dividing it into suitable frequency bands, probably those $\frac{1}{3}$rd octave wide are most convenient, and plotting the results as a function of the frequency at the centre of each band.

The average values of F.I. for different rooms seem to vary between 1 and about 5. It has been shown experimentally [14, 15] that measures which increase the amount of diffusion in a room, polycylindrical surfaces for instance, tend to reduce the frequency irregularity but to

establish a positive criterion for diffusion in terms of F.I. requires many more results. A very similar trace is obtained if the frequency is kept constant and the microphone moved in space. This could give a figure for the space irregularity, defined in a similar way, and also be a measure of the diffusion.

§ 8. Steady State Sound Pressure Levels

As well as the fine variations in pressure just mentioned the average sound pressure produced by a point source often varies considerably over the seating area of a large auditorium. Warble tones may be used for this measurement but white noise is probably better. Apart from indicating seating areas where the sound is quieter there is no obvious correlation at present with other acoustical qualities of the room.

§ 9. Distribution of Reverberation Time

Since the SABINE conditions of complete diffusion of sound in a room are very rarely fulfilled there is generally some small variation of reverberation time with position. If a large number of positions are measured, the curve showing the distribution of the values about their mean can be plotted. According to FURRER [16] this is a normal error or Gaussian curve for rooms which are acoustically good.

In a large concert hall where upwards of a hundred positions are chosen it is a great help to record on magnetic tape the full frequency band decay at each position using a pistol shot or other white noise as a source. The frequency analysis can be obtained by replaying the tape through filters. For any of the frequency bands the results have to be analysed statistically to get the mean value, the standard deviation which is a measure of the scatter about the mean, and the skewness which should be zero for the symmetrical Gaussian distribution.

§ 10. Articulation Tests

In auditoria used for speech the primary requirement is for the intelligibility to be high. When a sound reinforcement system is used it is also desirable for the speech to sound natural and for as little

distracting reverberant sound as possible to be heard, but neither of these is as important as the intelligibility.

The most convenient way to measure the intelligibility is to use articulation test methods. In such tests a speaker reads from a carefully selected list of monosyllabic words and the percentage of these which are correctly understood by a team of listeners seated successively at different points in the auditorium gives the relative articulation. Such tests are discussed in Division IV, Chap. 12.

The percentage articulation obtained at different seating positions is a good measure of the relative intelligibility. Complete word and sentence intelligibility is always higher than that for monosyllabic words [16] since the sense can often be obtained by the context and inflections. If words of more than one syllable or a sentence were used for testing, however, the figures would be too high for the differences between various seat positions to be significant. The following figures give a scale relating percentage articulation and intelligibility;

P.A. = 96 ; Perfect hearing.
85-96 ; Highly satisfactory.
75-85 ; Satisfactory.
65-75 ; Speech understandable but with difficulty.
below 65 ; Unsatisfactory.

§ 11. The General Principles of Acoustic Design

Although the science of acoustics has progressed considerably since the days of W. C. SABINE the acoustic design of rooms, particularly those intended to hold a great many people, is still as much an art as an exact science. To a certain extent this will always be so since there is no one unique way in which the sound waves should be made to behave, the criteria of perfection are too subjective for this, so the problem of design resolves itself into the use of general principles, applicable to all types of rooms, together with the experience gained by studying the behaviour of good rooms already existing.

One of the primary requirements, although it is a negative one in some respects, is for a low level of intruding noise. Whether the room

is used for music or speech, it is essential to have a low ambient noise level if the sounds made in it are to be heard to their best advantage. Choice of a quiet site is not always possible, traffic on adjacent roads and railways in any large town generates sufficient noise for particular care to be taken in effective planning and provision of adequate sound insulation. Ancillary buildings, foyers for instance in the case of concert halls, can give a certain protection against external noise, but sounds coming from inside the building, particularly corridors, are also likely to cause trouble. The ventilation plant is also a possible source of noise in itself besides being a channel by which the external noise can be transmitted into the very heart of a building.

Coming now to the question of proper design, it is the large halls intended to seat audiences of two or three thousand people which are the greatest problem. For a given seating capacity the variables available are the shape in plan, in section and the volume per seat. Of the various possible plans, those involving large concave surfaces are best avoided because of focussing effects which give an uneven distribution of sound over the seating area. If curved surfaces are necessary the radius of curvature should either be twice as long as, or very short compared with the room dimension in the direction of the radius. It is desirable for as many of the audience as possible to be close to the sound source as long as they are in front rather than to the side of it. This is because human voices in particular have directional characteristics which throw most of the high frequency sound forward. In addition, the path difference between the sound reaching any listener directly and that arriving as a strong first reflection from a hard surface should nowhere exceed 60 ft. or the two will not reinforce each other but be heard independently. These conditions are best satisfied either with rectangular halls of not too great a width or with slightly fan-shaped halls. An arrangement where the main body of the hall is rectangular and the walls at the source end are splayed so as to throw as much sound as possible down the length probably gives a more even distribution of sound than any other.

The audience is a highly absorbing area which attenuates sounds travelling at grazing incidence, so to get enough direct sound the seating is usually raked sufficiently to give good sight lines. Very often this is not possible when the hall has to be used for other purposes needing a flat floor, in which case a raised platform is essential. This too should

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be raked as far as practicable or the sight lines will only be good to the front edge and not for the whole of the platform.

Balconies which project too far into the hall always reduce the amount of sound reaching those seats beneath them, particularly when they extend forward a greater distance than the height above the floor at the front edge. Those which also extend round the sides of the hall and are more than two or three rows deep give rise to an appreciable deadening effect over a large part of the ground floor. The poor listening conditions, generally found beneath deep balconies, can be partly cured by increasing the amount of direct and first reflected sound reaching these areas with the aid of suitably placed reflecting surfaces at the source end.

Those listeners in the balcony itself are helped considerably by reflections from the ceiling and as long as the seating is adequately raked good listening conditions are generally obtained here, even when the back rows are considerably further from the source than those on the ground floor. The height and shape of the ceiling are very important. Its position governs the total volume of the auditorium and hence, for a given seating capacity, it fixes the volume per seat. Halls which are acknowledged to have good musical tone generally have values lying between 180 and 240 cubic feet for the volume per seat although where unaided speech is the primary consideration lower values are usual. The ceiling itself is a most useful reflector and is the most important single surface in the auditorium. It is dangerous to leave such a large area completely plane, particularly when it is parallel to the floor, for fear of strong low frequency resonances and flutter echoes which are liable to develop between any parallel surfaces.

In halls of classical design the ornamentation, such as coffering in ceilings, pillars, plaster panelling and recessed windows in side walls, gives enough diffusion to the reflected sound for such faults to be avoided. The more severe modern architectural style avoids such ornamentation so some measures for diffusion must be introduced deliberately when parallel surfaces are necessary. With side walls this may take the form of serrations formed by panels placed at an angle to the line of the wall. When such panels are backed by a suitable air space they also give low frequency absorption previously provided by the plaster on lath and wood panelling in the classical design.

Reflections from the rear wall can arrive back at the source as echoes

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with appreciable time delay. When the seating is steeply raked the area of rear wall is small but it is safest either to make it as absorbent as possible or to arrange for it to be tilted forward so that the reflections reinforce the sound to the rear seats. Other possible sources of echoes are the corners formed by the junction of ceiling and side walls. If these are right angles any sound approaching the corner line normally is reflected back along the same path, probably with a path difference exceeding 60 feet and so is heard as a distinct echo. Such corners are either made highly absorbing or are curved to avoid this effect.

Reflections from nearby surfaces are always a great help to performers in enabling them to hear themselves and others with them, giving better ensemble and more chance of good blending of the sound. There are some types of music, Wagner and some other romantics for instance, for which this blending is more essential than others. In the true Wagner theatre the orchestra is housed in a well which is almost completely enclosed, resulting in a thorough blending before the sound issues forth to reach the audience. Most auditoria, however, have to cater for various types of music and speech, and the design cannot be subjugated to one particular type as with the Bayreuth Theatre. Splay surfaces on either side, if not completely plane, and a ramped platform provide a certain amount of inter-reflection, but much of the sound goes upwards or is reflected up from the floor. The ceiling design is therefore very important at the source end in giving reinforcement to those seating areas furthest away, under deep balconies for instance, and in giving further reflections back to the performers. Since the path difference for these should not greatly exceed 60 ft. or the sounds will be heard as distinct echoes, this limits the height of the ceiling above the stage. The alternative, which has been employed in some post-war British concert halls, is to have a substantial reflector suspended at a suitable height above the platform, some parts being horizontal and others inclined at the correct angle for directing sound out into the auditorium.

The audience is a highly absorbing area, particularly when the seating is raked. If sound is directed particularly to this area before it is allowed to become reverberant, that is if a lot of the energy arrives at the listener within 50 milliseconds of the direct sound, then this gives good clarity and definition. Such conditions are excellent for speech since the reverberant sound reduces rather than aids intelligibi-

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lity; for music, however, a balance has to be maintained between definition and fullness of tone which is closely connected with the reverberant sound.

§ 12. Calculation of Reverberation Time

At some stage in the design of a large hall it is desirable to calculate the reverberation time to be expected at various frequencies to see whether more or less absorption is necessary. This entails calculating the area of all the different types of surface and assigning to each type its appropriate absorption coefficient. For most materials this is a simple though tedious procedure, except perhaps at low frequencies where absorption coefficients cannot be measured very accurately. There are some surfaces to which it is difficult to give the correct absorption and this particularly applies to the audience. Fig. 62 shows

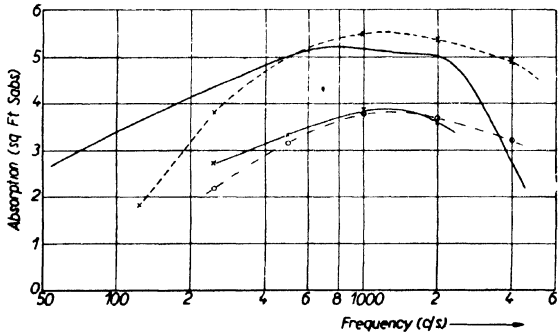


Fig. 62. Absorption per seated Person; (a) ———— Increase due to Person sitting on hard chair, (b) - - - - seated Person, (c) x ———— x seated in church Pews, (d) – – – – seated on Floor.

[(a) is calculated from reverberation times given by PARKIN and TAYLOR [18], (b) is from INGERSLEV; 'Lille Akustik', (c) is from KNUDSEN and HARRIS; 'Acoustical Designing in Architecture', (d) is from Building Research Station data.]

the absorption per seated person as a function of frequency according to various authorities and for different kinds of seats. At mid and high frequencies the audience often provides more than half of the total absorption. Those parts of the floor which are screened from direct view by the audience should be credited with only about 90% at 125 c/s, 80% at 500 c/s and 20% at 2000 c/s of the amount calculated.

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Other surfaces to which it is difficult to assign suitable values are comparatively thin partitions such as suspended ceilings backed by large air spaces.

§ 13. Speech Reinforcement Systems

Most large auditoria have to be used at times for speech and, although in a well-designed hall with reasonably low ambient noise level it should be possible to hear a trained voice unaided by loudspeakers, it is often found desirable to install a speech reinforcement system. Recent developments in such systems have shown that the intelligibility and naturalness of the amplified voice can be greatly improved when the design takes into account the disturbing effect of echoes discussed previously (p. 129)

With any system the direct sound of the voice travels out to the listener; if it is arranged that this is reinforced by the loudspeakers at least 5 and not more than 35 milliseconds after the direct sound reaches him, then the level of reinforced sound can be up to 7 db greater than the direct without the presence of the loudspeakers being noticed. All the sound appears to come from the true source. The appropriate time delay can be obtained by arranging the single group of loud speakers above and a little behind the direct source, but it may not be possible to get adequate reinforcement for those seats furthest away without destroying the realism for some of the nearer seats due to too high a level from the loudspeakers.

An alternative is to have a modified high level system with a few groups of loudspeakers reinforcing the sound to those seats particularly needing it. The signal to these loudspeakers has to be delayed by an amount corresponding to the path difference from the sound source plus 5 to 35 milliseconds. It has also to be arranged that the sound from the loudspeakers travels only outwards in the same direction as the direct sound, for any going back towards the source reaches those listeners with a longer time delay and hence is heard as an undesirable echo. The appropriate time delays for each set of loudspeakers can be conveniently obtained by recording the direct sound on magnetic tape and replaying it at fixed intervals later.

In very reverberant auditoria such as churches, the reverberant sound coming as it does up to several seconds behind the direct sound

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seriously impairs the intelligibility of speech. Any normal loudspeaker system increases the level of the direct and reverberant sound by the same amount with a possible improvement in intelligibility but with listening conditions still difficult. Recently the use of highly directional loudspeaker arrays has been found to give greatly improved listening conditions in such circumstances [18]. The delayed sound from the loudspeakers is beamed to fall only on the listeners which constitute a highly absorbing area and very little sound is reflected to become reverberant.

§ 14. Measurement of Sabine Absorption Coefficient and its Relation to Normal Impedance Measurements

The absorption coefficient of a material is defined as the ratio of non-reflected to incident energy on the sample and the value obtained depends on the particular distribution of sound in its immediate vicinity. Thus the absorption provided by a material is not a unique characteristic value but depends on its size and shape, its position in a particular room, as well as on its physical properties. If, however, the SABINE conditions are fulfilled and the sound field is completely diffuse, all the constants of the distribution disappear apart from the energy. Hence as long as there are no non-linear effects there is a definite value of absorption coefficient depending only on the characteristics of the material. Since in large rooms and at high frequencies the SABINE conditions are fulfilled, it is this value which is most often required and for which the term absorption coefficient has come generally to be known.

Reverberation chamber method

In the reverberation chamber method of measuring absorption, precautions are taken as far as possible to ensure that reverberant conditions are obtained, but this is not always possible at low frequencies where an excessively large chamber is necessary. Also the use of SABINE absorption coefficients to determine the reverberation time of small rooms and at low frequencies gives erroneous results since again the conditions are not truly reverberant. Nevertheless the reverberation chamber method is particularly useful for comparative measurements

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of absorption characteristics, either of the same material at different frequencies, or of different materials. It would be even more useful if different laboratories when measuring absorption could use precisely the same technique and conditions so that the results obtained could be more readily compared. Since in practice it is the size and shape of chamber which differ most from one laboratory to another, it is this which is likely to prove the greatest stumbling block to standardisation of method even when the same technique of measurement is used.

Existing reverberation chambers vary a lot in shape and size but there is probably little advantage in using a chamber much greater than about 100 m^3 in volume. More diffuse sound fields are obtained if some of the walls are set obliquely although convex surfaces and pillars have been used as diffusers. Any one of the linear dimensions should not differ too widely from the others. The reverberation time of the empty chamber must be as long as possible so that the change due to introducing the sample is large; smooth hard plaster on a solid backing gives the lowest absorption for bare walls but at high frequencies dust on the surfaces and absorption in the air may limit the empty reverberation time. In any case this should be at least 5 seconds at 400 c/s and below and it should not fall below 2 seconds at 6400 c/s.

The material under test is best divided into three parts and approximately equal areas put on each of three mutually perpendicular surfaces. The apparent value of absorption coefficient obtained varies a little with the amount of absorption put into the chamber, equal increments in area giving steadily decreasing increments in absorption. A total area of about 10 square metres should be used.

A choice of sound source from either white noise or warble tones is possible, both being generated electrically, the use of a pistol being ruled out in this instance since it is inconvenient and generates excessive sound pressure in such a confined space. If the full frequency band of white noise is used, when the reverberation time is changing rapidly with frequency due to the absorption characteristics of the sample being measured, two effects may occur. Firstly, the reverberation time obtained is the longest value in any particular $\frac{1}{3}$rd octave band and is not necessarily that of the centre frequency. Secondly the energy from the adjacent band where the reverberation time is longer may cause erroneous values to be measured if the cut-off of the electrical filters is not sufficiently steep. This could be overcome by a further

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set of filters in the loudspeaker circuit which would also have the advantage of giving a greater dynamic range without fear of overloading the amplifiers feeding the loudspeakers.

Warble tones are generally more convenient to use as a source, however, with a warble frequency of 5 to 10 c/s and a deviation of $\pm$ 10 per cent of the centre frequency. The loudspeakers should be placed in corners and directed diagonally outwards into the chamber.

The technique, as with measurements of reverberation time in auditoria, is to use a non-directional microphone coupled to the necessary amplifier through band-pass filters, preferably $\frac{1}{3}$rd of an octave wide and certainly not wider than an octave. The amplifier is followed by a logarithmic level recorder which traces the decay and gives a slope from which the reverberation time can be measured. These traces should extend over a range of at least 30 db and those which show multiple slopes should be excluded from the evaluation.

The reverberation time, τ_1, of the empty chamber is measured first to get a_1 the average absorption coefficient of the bare surfaces. Readings are taken at several microphone positions and the average value used to determine the absorption; below 200 c/s six positions, from 200 to 400 c/s four and above 400 c/s three positions should be sufficient. If a warble tone of gradually changing frequency is used, a single microphone position is sufficient when the tone is switched off correspondingly more often during the passage through a $\frac{1}{3}$rd octave. The measurements are repeated with the sample in position on three surfaces giving τ_2, the new reverberation time. As long as this is not too short the simple SABINE equation is adequate to determine the absorption coefficient a_2 of the sample. The first measurement gives

$$\tau_1 = \frac{0{\cdot}049\,V}{S_t\,a_1}$$

where V is the volume of the chamber in cubic feet and S_t the total surface area, from which a_1 can be calculated. The second measurement then gives

$$\tau_2 = \frac{0{\cdot}049\,V}{(S_t - S_s)\,a_1 + S_s\,a_2}$$

where S_s is the total area of the sample. At high frequencies the difference between τ_1 and τ_2 is very small since the absorption in

the empty chamber is already quite high, thus the reverberation time has to be measured more accurately to get significant results, and if necessary the correction made for absorption in the air with the aid of the curves of Fig. 56.

It is customary to state the results at $\frac{1}{3}$rd octave intervals using the frequencies 100, 125, 160; 200; 250, 320, 400, 500, 640; 800; 1000 c/s and so on up to about 5000 c/s. Above this frequency the results have little significance and are of little use in practice.

It is essential for the mounting conditions to be specified as accurately as possible, particularly when samples are not mounted directly on to the rigid wall of the chamber. This is because there is often resonant absorption at low frequencies when thin panels of material are fixed over an airspace and changes in the method of mounting can give appreciable difference in the low frequency absorption.

§ 15. Impedance Measurements

With materials which do derive some or all of their absorbing properties from resonance phenomena the reverberation chamber method is usually the only possible one to use since full scale samples are essential. The absorption coefficients of most porous absorbents, however, depend mainly on the properties of the materials themselves which can be measured using quite small samples. Of these properties the propagation constant and the characteristic impedance are the ones of interest and these can be determined quite accurately. From their value the behaviour of the material in a given sound distribution can be computed.

Now the normal acoustic impedance of a material is defined as the complex ratio of the sound pressure p to the normal particle velocity u at the surface, *i.e.*, $Z = p/u$. The relation between this quantity and the absorption suffered by a wave striking the boundary between two media is discussed in Chap. 4.

The absorption coefficient at random incidence may be taken very approximately 1.5 times that at normal incidence.

The impedance is very little more fundamental as a property of the material than the absorption coefficient and although at present the calculation of absorption coefficients from impedance values does not yield results which are much more reliable for the practical design of

auditoria than those measured in the reverberation chamber, at least the impedance can be determined more accurately and can be derived from the more fundamental properties of the material. When the actual distribution of sound in an enclosure can be found more accurately and not just assumed to be truly diffuse as has to be done in most practical cases, then the effect of the bounding surfaces can be best found in terms of the impedance.

Methods of measuring the impedance of materials are described in Chapter 4, § 12. These methods mostly involve a determination of the 'standing wave ratio' (eqn. I, 60).

Besides these interferometer methods it is possible to use a resonance method, useful for samples with low absorption coefficients. In this method the microphone position is fixed close to the source whose frequency is variable. With a perfectly reflecting plate at the sample end the initial wave suffers multiple reflections between this and the source which acts as a closed end. Resonance occurs when

$$2l = n\lambda \text{ } i.e. \text{ when } f = nc/2l.$$

When the tube is closed by the sample in question, however, the phase change of the reflected wave is such that the new resonance condition is given by;

$$2\pi\,(2l/\lambda) - \delta = 2\pi n, \text{ or the resonant frequency is;}$$

$$f' = (c/2l)\,(n + \frac{\delta}{2\pi})$$

The resultant amplitude of the reflected waves is;

$$1 + r + r^2 + \text{ - - } = (1 - r)^{-1}$$

The steepness of the resonance curve provides a more exact criterion. If $2\Delta f$ is the width measured at half maximum energy this is related to the absorption by; [19]

$$2\Delta f = (c/4\pi l)\,(a + a^2/_2)$$

The relation between the SABINE absorption coefficient $a_{sab.}$ and the magnitude and phase angle, φ, of the impedance is developed in detail by MORSE and BOLT [20] giving

$$a_{sab} \rightarrow (8 \cos \varphi/3) \left[1 - \frac{2 \cos \varphi}{\eta} \ln \eta + \frac{\varphi}{\eta} \frac{\cos 2\varphi}{\sin \varphi} \right] \text{ as } \eta \rightarrow \infty$$

and

$$a_{sab} \rightarrow \frac{8\eta}{3} \cos \varphi \left[1 - \frac{3}{2} \eta \cos \varphi \right] \text{ as } \eta \rightarrow 0,$$

where η in this case is $(|Z|/\rho c)$.

They also give a set of curves from which the value of a_{sab} can be obtained when η and φ are known.

These relations are only true for completely diffuse sound fields. In reverberation chambers, however, it is possible that truly diffuse conditions are not obtained below about 2000 c/s and the values of a so obtained are larger than those computed from impedance measurements. In large irregularly shaped auditoria the SABINE conditions do hold and so it is the value a_{sab} which then characterises the behaviour of the material.

§ 16. Acoustical Control Surfaces

Since the most important single quantity governing the acoustical performance of a room is the reverberation time and this in turn depends directly on the amount of absorption in the room, it is most necessary to see how this absorption is obtained. Apart from the audience, which in the case of concert halls provides as much as half of the total amount at mid frequencies, and the air absorption which often limits the reverberation time at high frequencies in large halls, it is the bounding surfaces that give the rest of the absorption. So much of the art of acoustic design relies on the correct selection and placing of the materials forming these surfaces that it is essential to know how they behave.

There are two distinct classes of absorbers;

(a) Porous materials, which can either be fibrous or granular in character.

(b) Resonant absorbers in which the associated air cavity may or may not contain porous absorbents and may be connected to the main volume of the room.

Broadly speaking the first kind, which absorb over a wide frequency band, are more useful at mid and high frequencies while the resonant

absorbers are more frequency-selective and are generally used at the lower frequencies. There is, however, no rigid line of demarcation between the two classes and some materials may behave in either way depending on their physical properties and on the way in which they are mounted. The behaviour of the two types is described with the relevant theory in Chap. 4, §§ 8–11.

The best examples of purely porous materials which have high absorption coefficients are rock wool, glass silk, felts and soft fibre-boards, for all of which the absorption increases with thickness for layers of practical size. It is generally undesirable to use such materials as direct facings on walls because the loose surface, which is essential to this type of absorber, collects dirt and is easily damaged. The effect of painting such surfaces is to reduce their absorbing properties appreciably at medium and high frequencies, particularly when heavy oil bound paints are used.

An air space behind a layer of rigid porous absorbent improves the low frequency absorption; the action of the air increases the apparent thickness of material. When the layer of absorbent is semi-rigid, however, it can act as a panel resonator, generally at low frequencies, with the material itself providing its own damping either internally or by absorption of energy from the air cavity by the rear surface. This again gives greater low frequency absorption but by a different mechanism. It is here that we are coming into the second category mentioned above, that of resonant absorbers.

Such resonators being highly tuned only absorb energy over a comparatively narrow frequency band, therefore it is impracticable to use such resonators spaced sufficiently far apart to act individually as a general absorbent for controlling reverberation time. However, they have been useful for damping out bad low frequency resonances which are liable to occur even in the most carefully designed rooms. The ceiling of the Royal Festival Hall, London, for instance, contains over 1000 holes behind which Helmholtz resonators could have been placed if it had been found necessary.

It is possible, however, to use such resonators as a practical absorbent by building them together in panels. In this case it is not even necessary for the individual cavities to be separated; a common enclosed air space can serve a considerable number of necks but it is essential for the internal damping to be high. INGERSLEV [21] gives the following

details of a typical resonant panel absorber particularly useful for noise reduction.

A perforated facing $^3/_{16}''$–$\frac{1}{4}''$ thick spaced 1″–2″ away from a rigid wall or ceiling by means of battens. Between 8 per cent. and 12 per cent. of the area is perforated by $^3/_{16}''$–$\frac{1}{4}''$ holes spaced $\frac{1}{2}''$ apart. A blanket of porous material having the right flow resistance (50 to 100 gr. $cm^{-3}sec^{-1}$) is placed in the airspace. It need not completely fill the cavity, about 1″ thickness is sufficient, but it is important for it to be adjacent to the facing material and not by the rigid wall.

Such an absorbent having a $3\frac{1}{2}''$ deep air cavity containing a 1″ rock-wool blanket gives an absorption coefficient greater than 0.6 in the frequency band 150–2000 c/s. It is equally possible to have the perforation in the form of short slits instead of round holes when slits are more acceptable architecturally. Obviously such perforated facings are useful for protecting the soft faced, purely porous, type of absorbers. The frequency f_{max} below which the facing does not cause any appreciable reduction in the absorbing properties of the porous material is given by

$$d^2/a < {}^{7000}/f_{max}$$

for perforations consisting of a square lattice of circular holes radius a cm, spaced a distance d cm apart. When the perforations are in the form of slits whose length is greater than $\frac{1}{3}$rd. of the wave-length at the resonant frequency the absorber becomes a slit resonator. Fig. 63

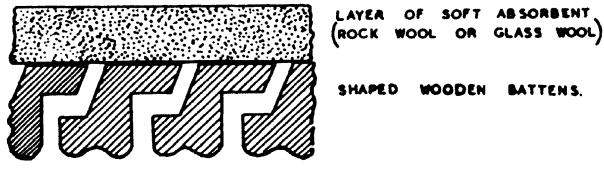


Fig. 63. Section of Knucklebone Type Slit-Resonator Absorbent.

shows a cross section of the battens of the so-called knucklebone absorbent in which it can be seen that the slits are specially shaped to avoid exposing the soft porous material behind them.

The other class of resonant absorbers are the non-perforated panels. Here the resonant frequency depends on the mass and stiffness of the

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panel itself as well as on the airspace behind. The boundary conditions, particularly the method of fixing such panels, are very important in determining the effective total stiffness and it is these conditions which are liable to differ in practical constructions. With plates of thickness 3–5 mm, larger than 50 × 50 cm and backed by an airspace d, less than 5 cm the resonant frequency is given by:

$$f_0 = \frac{188}{\sqrt{md}}$$

where m is the mass per square cm.

In concert halls and other large rooms of classical design quite an amount of low frequency absorption is provided fortuitously by wood and plaster panelling included for decorative purposes. The simpler lines demanded by modern architecture together with the more solid types of construction needed for good sound insulation using reinforced

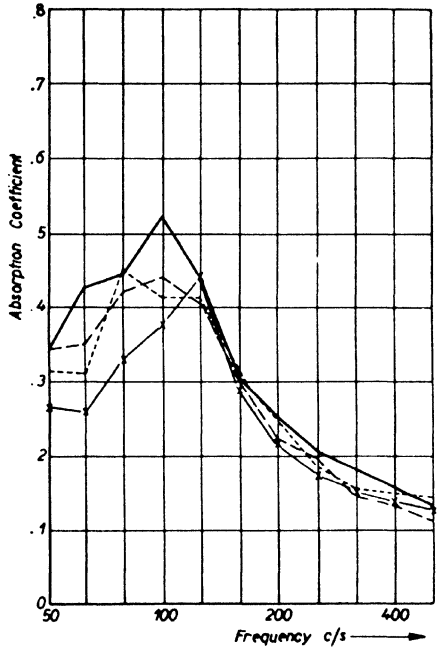


Fig. 64. Absorption Coefficients of Wood Panels for the Royal Festival Hall. —— Type 1, 10 cm. air space 1 brace; - - - - - Type 2, 10 cm. air spac, 3 braces; — — Type 3, 7.5 cm. air space, 1 brace; x —— x Type 4, 7.5 air space 3 braces. *(Crown copyright reserved)*

concrete make it necessary for functional low frequency absorbents to be used to control the reverberation time, particularly as the absorption per head of audience decreases considerably towards low frequencies. When panel type resonators are used for this purpose a certain degree of randomization of the absorption is required so that particular frequencies are not absorbed too strongly. Fig. 64 shows the measured [23] absorption coefficients of plywood panels 1 cm thick made up in units of two sizes, 70×70 cm^2 and 200×70 cm^2, equal numbers of each size being measured. The randomization was obtained by stiffening the back of the panels with cross braces and varying the depth of airspace, which contained rock wool. Including such a blanket of soft porous absorbent immediately behind the panel in the airspace appreciably increases the peak absorption obtainable with resonant panels.

§ 17. Distribution of Absorbing Materials

When the material for any of the inside surfaces of an enclosure is chosen particularly for its absorbing properties it is for one of three reasons;-

(a) to reduce noise;
(b) to prevent echoes;
(c) to control reverberation time.

Considering (a) first; the places where noise reduction is desirable are legion but it is most necessary in foyers and corridors adjacent to halls, debating chambers and conference rooms needing extreme quiet. Treating the ceilings and upper parts of the walls with highly absorbing material and, what is most important, carpeting the floor, not only reduces the noise level in such places but appears to give an atmosphere in which less noise is likely to be made. The materials which have the necessary high absorption coefficients over a broad frequency band are the soft porous absorbents such as rock wool, wood wool, glass wool, the soft fibreboards, felts and asbestos spray. Most of these are not suitable to be directly exposed to view, but quilted if necessary and covered with a perforated facing of harder material they are quite acceptable.

The same broad band absorbents are needed to prevent echoes in large auditoria, but it is best to restrict the use of the highly absorbing materials to those parts which are likely to cause particularly bad

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echoes. Excessive treatment not only decreases the reverberation time at mid and high frequencies but the uneven distribution of absorption can cause bad listening conditions in seats adjacent to the treated areas. When large surfaces such as domes in reverberant buildings need to be made absorbing, acoustic plasters or similar granular materials are more suitable. They are not such powerful absorbents as those mentioned previously but are adequate when their action is not impaired by painting the surface.

Since at mid and high frequencies the audience absorption is by far the greatest factor in controlling the reverberation time, in halls designed for music any further absorption at these frequencies must often be kept to a minimum. Apart from upholstered seating, which is desirable to maintain the full and empty conditions as near the same as possible, the use of soft furnishings such as carpets and curtains has to be avoided. Wood is a very suitable material with which to line many of the remaining surfaces and a number of halls finished in this way have good musical tone. Joist floors may give increased absorption at low frequencies, either by transmission or due to resonance effects with the airspace behind. It is almost essential for the orchestral platform to be of wood and not too solid a construction so that the bass string players produce their best results.

The reflecting surfaces about the platform which are intended to give inter-reflections and reinforcement to particular groups of seats may also be of wood. Here the mounting should be more solid to cut down any tendency towards low frequency absorption and highly polished hardwoods give greater reflection at high frequencies. When the platform is low and the area between it and the front of the audience is not screened from the performer's view this surface may also be made highly reflecting. Marble or other polished stone has the right properties but care has to be taken to see that the ceiling is not parallel to the floor at this point or strong flutter echoes and room resonances may result.

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DIVISION THREE

NOISE

CHAPTER 7

NOISE MEASUREMENT

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§ 1. Definition

Noise is defined[1,2] as 'undesired sound'. This definition brings out the two aspects of noise: the physical aspect, denoted by 'sound', and the subjective aspect denoted by the reaction of the hearer that the sound is 'undesired'. The first of these two is susceptible of definite physical statement in absolute units, but the second is much less definite, being a matter of opinion so that a given sound may be music to one person and noise to another, or even to the one in different circumstances, *e.g.* even an ardent musician hearing good music in the middle of the night when he is tired and wants to go to sleep may consider the music 'noise'.

A noise may consist of a single pure tone or it may contain two or more component tones. If it contains a very large number of component tones evenly distributed over a certain frequency band the tones may be so close together as to be virtually unresolvable. Such a band is called a 'continuous spectrum noise,' the frequency limits being stated if known or described in general terms such as 'high', 'medium', 'low' or 'all' audio frequencies.

The term 'noise' is also used sometimes to denote any unwanted components, such as electrical interference, in an electrical signal even though these electric currents may not be converted into corresponding sound waves so as to be audible. In such cases the term 'electrical noise' is preferable.

§ 2. Measurement

The measurement of noise is made difficult by our inability to measure

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directly the nervous response of the human auditory system or to imitate its response instrumentally to noises of all types. It is true that it is possible to measure [3] the action potentials generated in the auditory nerves of animals but this is not the same thing as measuring their nervous response and, in any case, is hardly practicable with human beings. It is also possible, as will be seen later, to make instruments which respond to certain noises, *e.g.* pure tones, in a similar manner to the human ear but not to noises of all types.

§ 3. Phon Technique

In view of these difficulties, it was decided at an international conference at Paris in 1936 to measure all noises by referring them to a common scale, the basis of reference being equality of loudness between the noise to be measured and a reference sound. For simplicity of specification, and because it is about the geometric mean of the audio range of frequencies, the reference sound was made a pure tone of 1,000 c/s. Natural conditions of listening alternately to the noise and reference tone with both ears while facing the sources of the sounds were specified, the comparisons being made by a number of persons of normal hearing in a uniform progressive sound field. The 'Equivalent Loudness' [1] or 'Loudness Level' [2] of the noise in phons is given by the intensity level in decibels (db) above the reference pressure of 0.0002 dyne/sq. cm r.m.s. of the 1,000 c/s reference tone.

If P dyne/sq. cm r.m.s. is the acoustic pressure of the 1,000 c/s tone which appears to the average observer to be as loud as the noise, the intensity level is $20 \log (\frac{P}{0.0002})$ db. This number of db is, therefore, also the equivalent loudness or loudness level of the noise in phons. A laboratory with sound absorbent walls, called an anechoic chamber, for carrying out fundamental measurements of loudness on reproduced noises is shown in Fig. 65. An observer is shown seated in a skeletal chair in front of and facing a large horn type loudspeaker. He hears the noise and reference tone alternately, each for about 2 seconds, and registers his judgement of which is louder by pressing a push button once or twice. After each judgement the magnitude of the reference tone is changed for another presentation, care being taken to avoid any continuous sequence in magnitude or order of presentation. Results of tests [14] on 10 observers are shown in Fig. 66.

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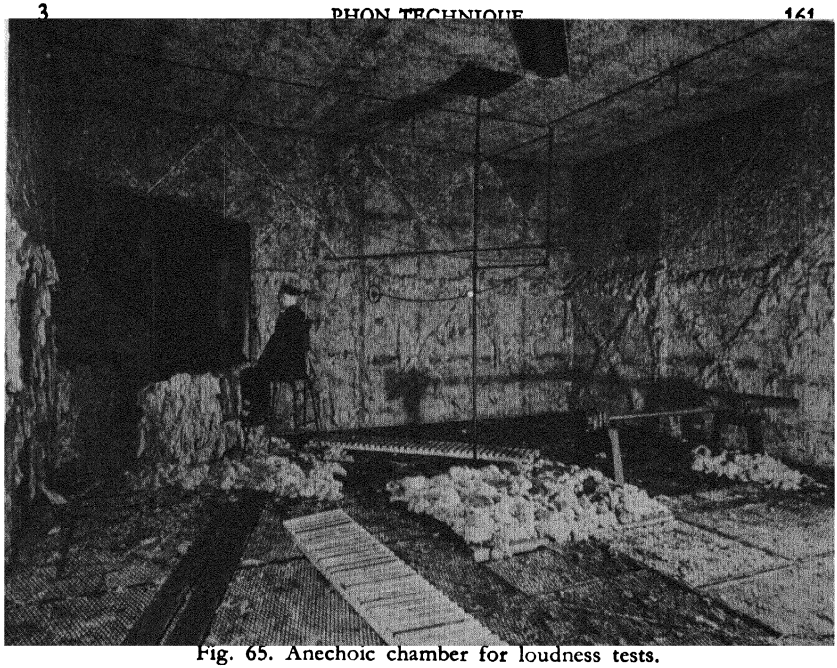


Fig. 65. Anechoic chamber for loudness tests.

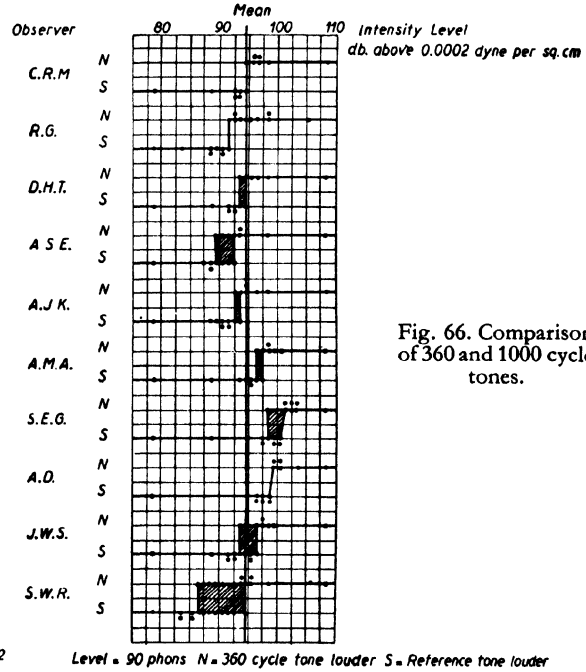


Fig. 66. Comparison of 360 and 1000 cycle tones.

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Level = 90 phons N = 360 cycle tone louder S = Reference tone louder

It is seen that some observers can discriminate very sharply in relative loudness while others have a region of uncertainty. The r.m.s. deviation of the judgements from their mean value is called the standard deviation, S, of the judgements and in the case of Fig. 66, $S = 3$. This is fairly representative, values as high as 4.5 being reached only in a minority of cases. If n is the number of observers, the probable error, Y, of the mean relative to the true value is given, for a normal distribution, by

$$Y = 0.67 \frac{S}{\sqrt{n-1}}$$

The probability of the errors being $2Y$ is 1 in 5 and Y is 1 in 22. When $S = 4.5$ and $n = 10$, $Y = 1$. For the more usual value of $S = 3.5$, 10 observers are sufficient to give a probable error of less than 1 phon. This is adequate for most practical purposes but if any particular result is in doubt a repeat test with another group of observers affords the best safeguard.

§ 4. Equal Loudness Relations of Pure Tones

One most important application of the phon technique is to the determination of the rate of exchange on a loudness basis between tones of different frequencies. This information, as will be seen later, forms the basis for objective meters for measuring sounds. The information is conveniently given in Fig. 67 which shows the results of two independent determinations [4,5]. The curves are called 'equal loudness contours' or 'isophons', since each curve joins points which correspond on the frequency and intensity scales to pure tones which appear, to the average observer, to be equally loud. The considerable differences which are apparent between the two determinations, especially at low frequencies and high intensities, have not been satisfactorily explained. They may be due, in part, to the additional stimulus or condition associated with the use in one case [5] of calibrated headphones compared with a direct reference to the unaided ear and natural conditions of listening in the other case [4]. Further determinations are being carried out by the National Physical Laboratory near London so in due course it should be possible to deduce a single set of curves which will be truly representative of the average listener.

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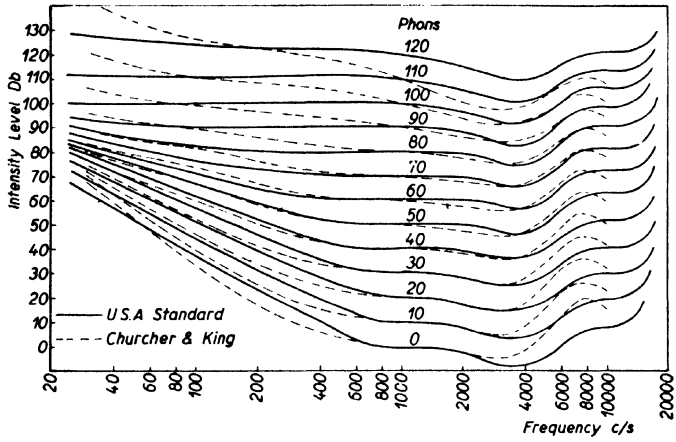


Fig. 67. Equal loudness contours.

§ 5. Secondary Noise Meters

Although the technique of the phon, with its reference to the human ear as the final arbiter in all questions of loudness measurement, is eminently suitable as an ultimate standard it is, like most other ultimate standards, quite unsuitable for routine measurements. The requirements of free space listening conditions and the ability to switch the noise to be measured on and off instantaneously are seldom encountered in practice where it is usually necessary to measure a noise in existing surroundings and it is not possible to start and stop the noise quickly at will. As in many other cases, secondary meters have had to be developed capable of measuring under practical conditions and of being referred to the primary standard. These secondary noise meters are of two main types, 'subjective' and 'objective' depending on whether or not the operator is called upon to exercise critical hearing faculties during the measurement. Both types are made in portable form so that they can be taken to the noise instead of vice versa.

§ 6. Subjective Noise Meters

For convenience in use, particularly for portability, most subjective noise meters have made use of earphones rather than loudspeakers for reproducing test tones.

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(A) *Masking*

Early attempts to develop a subjective noise meter made use of the principle of masking [6] of one sound by another. An earphone was spaced a short distance from one of the operator's ears and connected through an attenuator to a valve oscillator or buzzer. A preliminary test in a perfectly quiet room determined the attenuator setting for a just audible sound in the earphone. A similar test in the presence of the noise in question required a higher setting on the attenuator and the difference between the two settings, or raising of the threshold, was taken as a measure of the noise. It would need further justification before it could be said that the loudness of the noise was being measured since the criterion used was masking, not loudness. This view is supported by the wide variations experienced in the masking readings as a tone of constant loudness is varied in frequency, it being apparent that masking is critically dependent on harmonic and near-harmonic relationship between the noise and the tone in the earphone. In spite of its simplicity, therefore, the masking method has not been continued.

(B) *Equality*

There are three simple ways in which an observer can listen to a reference sound in earphones and compare it in loudness with a noise.

(i) *Spaced Headphones*

If two spaced earphones, such as those used for masking tests, are worn by the observer and supplied with a reference tone, it is possible with quiet noises for the observer to hear the noise and tone simultaneously and for him to adjust the latter until it sounds as loud as the former. With loud noises it is usually not possible to obtain a loud enough tone from the earphones and overloading and non-linearity may occur. Apart from these limitations, there are two fundamental objections to the method. Firstly the earphones have a frequency-discriminatory shielding effect on the noise and, secondly, the noise and reference tone are heard simultaneously. Both these features are departures from the phon technique.

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(ii) *Single Earphone*

By covering one ear with a well-fitting earphone the observer can listen to the noise with one ear normally and to the reference tone with the other. The method has been investigated [4] fully and found to be moderately successful when compared with the fundamental standard. Small differences between ears can be allowed for by interchanging the functions of the two ears and taking the mean of the two results. However, the method is suspect since the observer does not follow the phon technique of listening with both ears first to the noise and then to the reference tone. In practice it is found that some observers tend to drift towards a false balance, a tendency which is not noticed when both ears are similarly occupied.

(iii) *Double Earphone*

The condition of alternate listening with both ears can be achieved [7] by using two headphones with well-fitting soft rubber ear caps for listening to the reference tone. The noise is heard alone by listening naturally with both ears while facing the source and then the reference tone is heard, also alone and in both ears, by quickly placing the earphones over the ears. The effectiveness of soft rubber ear caps in excluding the noise when listening to the reference tone has been checked and attenuations of the order of 20-30 db at medium frequencies have been observed [4]. The reference tone can be a pure tone of any desired frequency but for ease in interpreting results in phons, Fig. 67 shows that frequencies of 800 to 1,000 c/s are preferable. Fig. 68 shows a commercial instrument in use. This model uses a pure reference tone of 800 c/s, it being claimed that loudness balancing of common noises is easier with this tone than with one of 1,000 c/s. A full investigation of the performance of this meter has shown [4] that with several observers it is capable of giving results in phons with a probable error of 1 to 2 phons for a noise of 40 to 106 phons.

It is possible that some observers would find the process of loudness balancing easier if a complex noise, such as is produced by a buzzer or by thermal agitation were used instead of a pure tone. However, experience shows that the process of loudness balancing, even of dissimilar sounds like a pure tone and a complex sound, is easier than

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one might imagine, very little practice yielding good results in spite of a feeling on the part of some observers that the results will have little meaning.

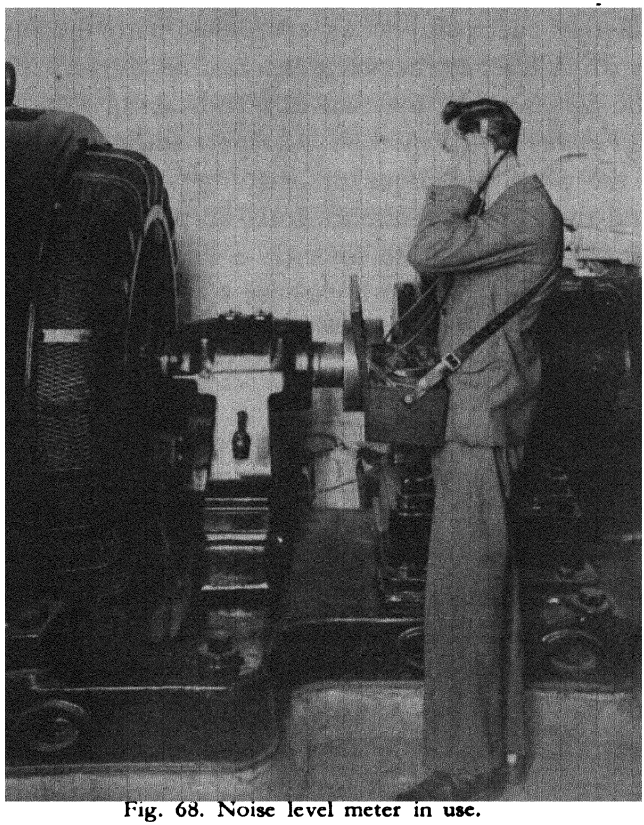


Fig. 68. Noise level meter in use.

§ 7. Objective Noise Meters

When microphones and amplifiers of stable performance became available it was an obvious step to add a measuring instrument such as an oscillograph [8] or thermal ammeter and to notice a rough correlation between the loudness of a sound and the magnitude of the resultant instrument deflection. However, anomalies were soon noticed, sounds of different pitch causing deflections which did not agree with

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mental estimates of relative loudness. An overall response similar to that of the average listener was required.

(i) *Sound Level Meter*

By including suitable filter networks in the amplifier circuit it is possible to make the overall response of a simple noise meter agree for pure tones with any of the pure tone equal loudness contours of the average listener given in Fig. 67. Such an instrument has been standardised in U.S.A. by the American Standards Association [9] and is known as a Sound Level Meter. It has three responses corresponding to the U.S.A. Equal Loudness Contours (Fig. 67) for '40' and '70' phons and 'Flat', the latter being used for all levels of 80 phons and above. It is also claimed that the time constant of the instrument is such as to make the response to sounds of short duration approximate to that of the human ear.

(ii) *N.P.L. Meter*

This meter was developed by DAVIS [10] at the National Physical Laboratory 'for the measurement of moderate and loud, steady and impulsive noises'. In addition to a circuit similar to the conventional sound level meter, the instrument incorporates a 'nearly-peak' meter which it is claimed makes the response correct for impulsive noises. For simplicity, only a half-wave rectifier is used in the 'nearly-peak' circuit, so, to take care of unsymmetrical waves, it is necessary to reverse the microphone leads and take the larger of the direct and reverse readings. A full-wave rectifier would remove this necessity but would complicate the circuit. DAVIS has shown [10] by many tests that the readings of the meter are in accord with subjective estimates for a wide range of noises of 80 phons and over, especially the impulsive noises associated with motor vehicles.

(iii) *Metro Vick Objective Noise Meter*

As a basis for the development of this meter, the law of summation of the human ear was determined. Equivalent loudness measurements were made [4] on many sounds of known composition and from the

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results the law of summation was deduced. The final curve [11] is given in Fig. 69. It had been observed [4] that the readings of a sound level meter on complex noises were low compared with the subjectively

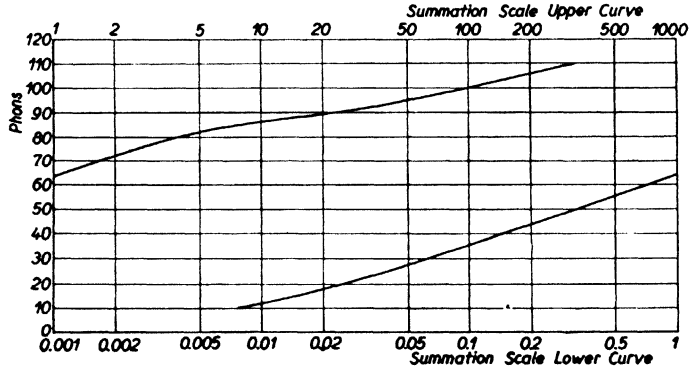


Fig. 69. The summation of tones of known equivalent loudness.

determined value in phons by amounts which increased with the number of components in the noises. This is illustrated in Fig. 70, which shows errors of up to 20 or more for noises having many components. It followed, therefore, that if the number of effective components in

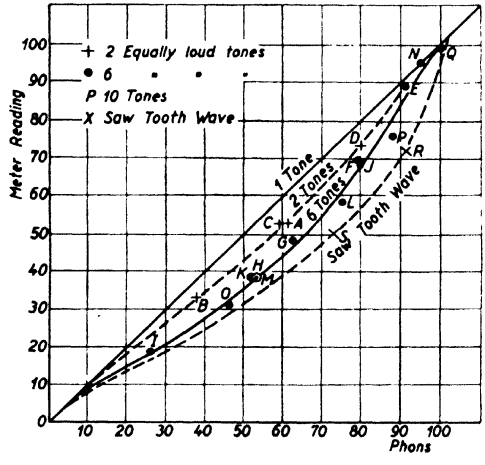


Fig. 70. Performance of sound level meter with noises of different complexity.

a noise could be determined, the correction to be added to the sound level meter reading could be estimated from Fig. 70 or, better still, computed from Fig. 69. An indication of the number of equally loud components can be derived on a statistical basis [11] from the ratio of peak /r.m.s. of the output of the sound level meter. A unique relationship is not possible owing to the critical dependence of the peak value on the relative phases of the components. Fig. 71 gives the average

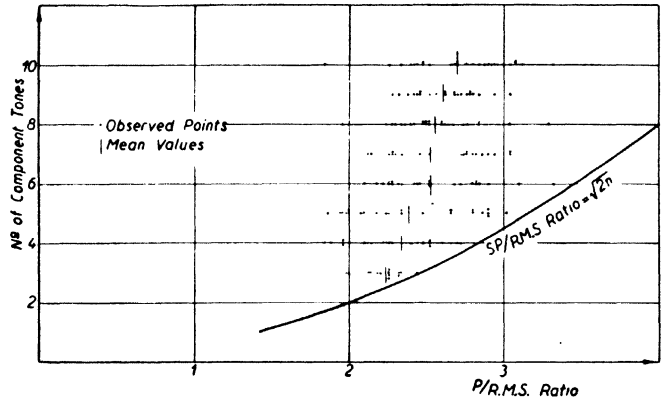


Fig. 71. Relation between peak/r.m.s. ratio and number of component tones.

relation for equally loud components and also the limiting values when all the components are in phase together.

It is seen that the uncertainty due to phase relationships is considerable. From Figs. 69 and 71 the corrections of Fig. 72 are computed. It will be noticed that the corrections are given for the component levels, not the sound level meter readings. The amounts to be subtracted from the latter to give the former are in most cases small. This is a refinement which can generally be omitted in practice.

In operation, the meter has two instruments, a rectifier type reading sound level in db and a full-wave peak valve voltmeter. The attenuators controlling sensitivity are adjusted until the first instrument reads a marked value when the second instrument reads directly in peak/r.m.s. ratio. The sound level in db is then read from the attenuators and the correction to be added to give equivalent loudness or loudness level in phons is read from Fig. 72.

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An analysis of the performance [11] of the meter, given in Table I, has shown that the errors are substantially less than those of the sound level meter or peak meter. The reason for the increased errors of the

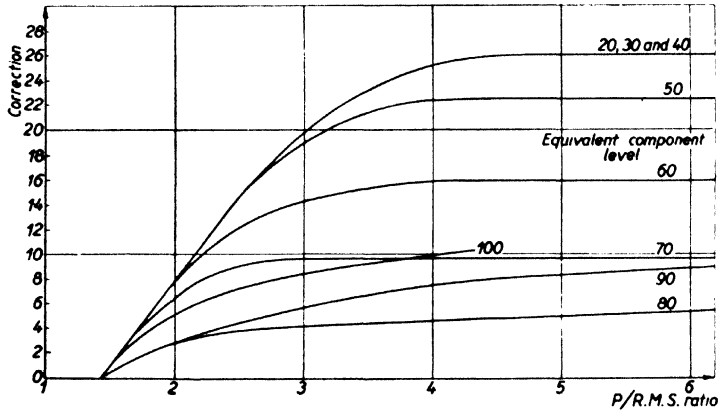


Fig. 72. Corrections to be added to sound level meter readings in db. to give equivalent loudness in phons.

sound level and peak meters at levels below 80 phons has been shown [11] to be due to the markedly different slopes of the summation curves used in the three meters below this level.

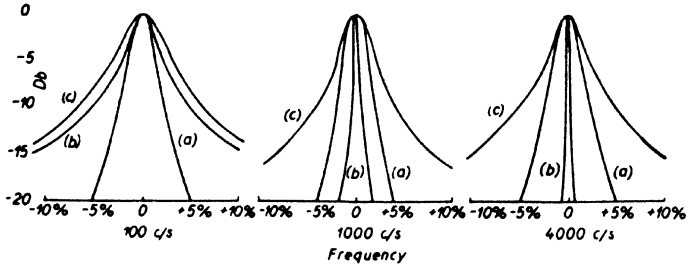


Fig. 73. Selectivity of analysers.
(a) Twin loosely-coupled tuned-circuit type.
(b) Heterodyne type, magnetostrictive rod filter.
(c) Feed-back type.

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TABLE 1 SUMMARY OF RESULTS

Particulars of Noises			Objective Noise Meter					Primary Standard					
Type of Noise	Rate of Repetition per sec.	S.E. db	Peak/r.m.s. Ratio		Equivalent Component Level db	Correction db	Phons	E.L. 10 observers		Reading as Peak Meter	Differences from Primary Standard		
			Max.	Min.				Phons	P.E.		O.N.M.	Peak	S.E.
Thermionic	—	40.3	2.95	2.95	29.4	19.2	59.5	63.3	1.4	46.7	−3.8	−16.6	−23.0
,,	—	61.4	2.95	2.95	50.5	18.5	79.9	81.8	0.8	67.8	−1.9	−14.0	−20.4
,,	—	82.4	2.95	2.95	71.5	8.8	91.2	98,9	0.9	88.8	−7.7	−10.1	−16.5
Sharp Impulsive	50	28.3	7.7	6.6	13.1	26.0	54.3	57.3	1.3	43.0	−3.0	−14.3	−29.0
,, ,,	50	47.4	8.2	7.5	32.2	26.0	73.4	77.3	1.0	62.7	−3.9	−14.6	−29.9
,, ,,	50	75.3	7.8	7.0	60.1	15.8	91.1	95.3	1.0	90.1	−4.2	− 5.2	−20.0
,, ,,	12.5	38.6	12.5	12.0	23.4	26.0	64.6	61.2	1.0	47.6	+2.5	−14.5	−23.5
Impulsive	50	34.0	4.6	3.4	20.0	25.8	59.8	61.2	0.8	44.3	−.14	−16.9	−27.2
,,	50	55.0	4.9	3.4	40.5	25.9	80.9	81.1	1.0	65.8	−0.2	−15.3	−26.1
,,	50	71.9	4.9	3.3	57.5	17.5	89.4	90.5	0.9	82.7	−1.1	− 7.8	−18.6
,,	26	54.9	5.0	3.6	40.2	26.0	80.9	81.8	0.8	65.9	−0.9	−15.9	−26.9
,,	12.5	35.0	5.1	4.6	20.3	26.0	61.0	53.3	1.7	46.2	+7.7	− 7.1	−18.3
,,	12.5	54.9	5.0	3.9	40.2	26.0	80.9	81.7	1.2	65.9	−0.8	−15.8	−26.8
,,	12.5	75.0	4.9	3.8	60.4	15.7	90.7	91.0	1.0	85.8	−0.3	− 5.2	−16.0
Transformer	—	84.3	2.01	1.57	79.5	3.0	87.3	91.8	0.9	87.4	−4.5	− 4.4	− 7.5
Transformer (TG)	—	51.8	2.67	2.06	42.2	16.4	68.2	72.6	1.0	57.4	−4.4	−15.2	−20.8
,, (TG)	—	32.6	2.29	1.89	25.5	11.8	44.4	46.4	0.9	36.8	−2.0	− 9.6	−13.8
D.C. Motor	—	72.0	2.79	2.79	61.7	13.1	85.1	85.1	1.2	77.9	0	− 7.2	−13.1
,,	—	32.3	2.79	2.79	22.0	18.0	50.3	52.1	1.0	38.2	−1.8	−13.9	−19.8
,,	—	86.4	3.07	3.07	75.1	6.9	93.3	99.7	1.3	93.2	−6.0	− 6.5	−13.3
A.C. Motor	—	59.3	2.98	2.53	48.3	19.1	78.4	77.0	1.5	65.8	+1.4	−11.2	−17.7
,,	—	30.9	2.99	2.46	20.3	18.7	49.6	48.8	2.1	37.4	+0.8	−11.4	−17.9
10/80 Phons	—	90.7	2.52	2.33	81.8	4.0	94.7	94.1	0.7	95.7	+0.6	+ 1.6	− 3.4
3/20 ,,	—	24.4	2.26	2.00	17.4	11.7	36.1	32.6	1.0	28.5	+3.5	− 4.1	− 8.2
								Mean without sign			2.7	10.8	19.0
								Mean with sign			−1.3	−10.6	−19.0

(iv) *Calculation of Noise from Analyses*

If the composition of a complex sound is known or can be obtained by analysis giving the intensity level and frequency of the component tones it is possible to calculate the equivalent loudness, or loudness level, of the whole sound in phons.

The first method of carrying out this calculation was developed by FLETCHER and MUNSON [5] (1933) and has been described in detail. Allowance is made for the masking effect of the components on each other by introducing masking factors. This necessitates a considerable amount of computation. A simpler method [4,12] which, nevertheless, has been shown to give results which are in many cases closer to the subjectively determined values, makes use of the summation curve of Fig. 69. The method is illustrated by Table II (p. 171) which shows the calculation for the noise of a turbo-generator. The components are first turned from intensity levels in db to equivalent loudness in phons by Fig. 67, using the dotted curves. The corresponding values on the loudness summation scale, Fig. 69, are then summed and the total converted back into phons, so giving the equivalent loudness of the whole noise.

When the noise has a continuous spectrum or contains one or more bands of components as opposed to discrete tones, the results of a sound analysis will depend on the selectivity of the analyser used. The heterodyne analyser, having an acceptance band of constant width in c/s, is very selective at high frequencies and unselective at low frequencies. Other analysers, notably the degenerative type [13] and variable inductance tuned-circuit type, have a constant percentage selectivity over their whole frequency range. Some selectivity curves are given in Fig. 73. It has been found that, with an analyser having a constant acceptance band with a width at mean ordinate of 6% in frequency, it is possible to compute the loudness of a continuous spectrum noise. The continuous spectrum is replaced by discrete components 6% apart and of a magnitude corresponding to the analyser readings. When these are summed as above the result has been found to agree well with the subjectively determined value. However, when filters an octave wide have been used, the results have been found to be too high. This summation is entirely empirical and justified only when the results are found to agree with the subjectively determined values.

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A further contribution to this problem of summation of sound spectra has been made by BERANEK [14] (1951).

§ 8. Loudness Scale

It is a commonly heard criticism of the phon scale of equivalent loudness that the numbers of phons attributed to two different noises create in the mind of the average person an impression of relative loudness which is very much less than that he would form intuitively on listening to the two sounds in quick succession. This view was found to be so prevalent that an investigation was made [15] to determine what meaning, if any, could be attributed to such comparisons as 'twice as loud'. Not only was it found that the various multiple and sub-multiple estimates of relative loudness of one individual showed a measure of consistency but also that there was substantial agreement between the estimates of different individuals. From these estimates it is possible, by starting from an arbitrary point and working up and down, to construct a phon-loudness curve.

A mean curve derived from the fractional loudness estimates of 34 persons is given in Fig. 74, the starting point being 100 phons = 100 loudness. Additional support for this curve is provided by the differen-

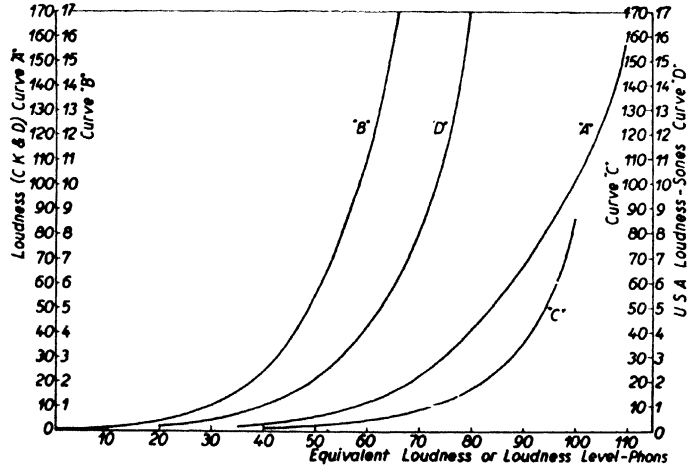


Fig. 74. Relation between equivalent loudness or loudness level and loudness.

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ces in aural perception between one and two ear listening. Assuming that a sound heard with two ears appears to be twice as loud as when heard with one ear, a loudness curve can also be derived from data on one and two ear listening [16]. When this is done, the resulting curve agrees well with that derived from fractional loudness estimates. Similarly good agreement was obtained by DAVIES (1934) [17] with a loudness curve derived from the differences in incremental sensibility between one and two ear listening. The expression (phons)$^5 \cdot 10^{-8}$ has been shown by CHURCHER [16] (1935) to give a good approximation to loudness over the range 40 to 100 phons.

Fig. 74 also gives the corresponding relation between loudness level in phons and loudness in sones determined in U.S.A. and adopted as standard there. A loudness level of 40 phons has been made to have a loudness of 1 sone. The two curves are similar in shape and would be closer numerically if they were adjusted on the arbitrary scales to have a common point. Approximately, log. sones = (phons–40)/30.

§ 9. Structure-borne Sound or Vibration

In addition to the sound waves radiated directly to the surrounding air by a noisy machine installed in a building, there is in many cases a vibration transmitted via the feet and foundations of the machine to the structure of the building. The resulting vibration of walls, floors and ceilings in more or less distant rooms, to which the vibration is conducted, causes sound waves to be radiated there, and so good is the conduction of the vibration in solid structures that the structure-borne noise, as it is called, is often much louder than the original air-borne noise of the machine which penetrates the intervening partitions.

The magnitude of these vibrations transmitted from the machine may be extremely small and yet they may cause an audible noise, *e.g.* in quiet surroundings a person of average hearing can detect the sound radiated by a surface vibrating at 1,000 c/s with an amplitude of 10^{-9} in. If the amplitude were increased to 10^{-6} in. the sound near the surface would be moderately loud and with 10^{-4} in. it would be uncomfortably loud. The detection and measurement of these small amplitude vibrations is fortunately a straightforward matter as detectors of adequate sensitivity are available. These vibration detectors are of various types and they can be substituted for the microphone

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of a noise meter to measure a vibration in amplitude, velocity or acceleration, depending on the type. If suitable differentiating or integrating circuits are included the response of the meter may be changed from that characteristic of the detector to one of the other bases, and if an analyser is included a spectrum of the components of the vibration can be obtained. As in the case of air-borne sounds, the spectrum is very helpful for diagnostic purposes as discussed later.

§ 10. Vibration Detectors

Although it is possible to make vibration detectors of many types [18] only two types are in common use.

(i) *Moving Coil*

The voltage generated by a coil moving so as to cut a magnetic field is proportional to the relative velocity of the coil and field. If one of the two is stationary, the voltage generated is a measure of the vibration of the other in the direction of cutting the flux. Usually the magnetic field is radial in the annular gap of a cylindrical pot magnet and the coil can vibrate axially in this gap as in a moving coil loudspeaker. In some models the pot magnet is held in the hands so that the coil touches the vibrating surface. It is necessary to press the coil on to the surface with sufficient force to make it follow the vibration, a light spring being usually sufficient to drive the small mass of the coil. This type is illustrated in Fig. 75. If no fixed point is available an artificial one is created in the seismic type. In the latter type the pot magnet is fastened to the vibrating surface and the coil is suspended very

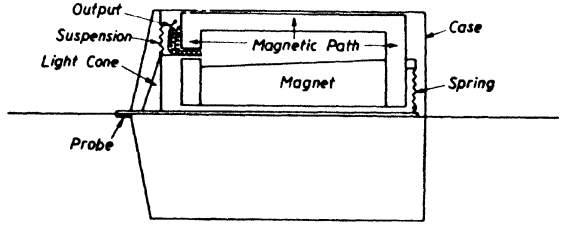


Fig. 75. Moving-coil vibration detector.

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flexibly in the gap so that its axial natural frequency is below the lowest frequency present in the vibration. The coil will then tend to remain stationary and the voltage generated will be a measure of the vibration velocity of the magnet.

(ii) *Piezo-Electric*

When a force is applied to some materials, *e.g.* Rochelle salt, quartz, barium titanate and ammonium di-hydrogen phosphate, the material develops an electric charge so that a voltage appears between electrodes on opposite faces (*cf.* Chap. 20). Since these materials are relatively stiff, a piece of one of them can be used to support a small mass with a natural frequency which is higher than that of most common vibrations. The combination therefore becomes [18] an accelerometer, the voltage generated being proportional to the acceleration multiplied by the mass and divided by the capacitance. Fig. 76 (b)

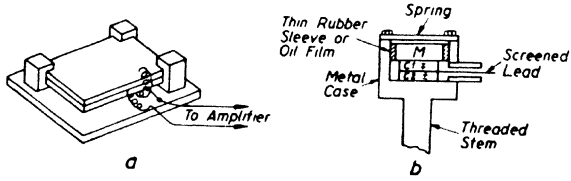


Fig. 76. Inertia piezo-electric accelerometers.
(a) Double-plate Rochelle salt type
(b) Quartz type

shows a quartz crystal accelerometer which is very suitable for precision measurements as it is very robust and has a stable response which is directional and linear in frequency and amplitude up to 10 kc/s. The film of oil indicated between the inertia mass and the case permits axial movement with some desirable viscous damping but prevents lateral movement, thus giving the directional properties. A thin-walled tube of rubber can be used in place of the oil film and is more predictable in performance. However, the sensitivity of this type, even with a pile of crystals to reduce the effect of added capacitance in cables and amplifiers, is only of the order of 10 mV per g.

A much more sensitive type giving about 1 V per g is illustrated

in Fig. 76 (a). The increased sensitivity is obtained partly by using plates of rochelle salt instead of quartz but also by making use of two plates cemented together with electrodes on the outer faces and between the plates and with their piezoelectric axes at right angles. Three corners of the pair are fixed and the fourth is left free as shown in Fig. 76 (a). Such a combination acts somewhat like a bimetal strip to bend the plates and so cause an increased movement of the free corner when a voltage is applied to the electrodes. Conversely, a small force applied to the free corner normal to the plane of the plates causes increased stresses in the plates and increased voltage between the electrodes compared with the same force applied directly to the plates in simple compression. If the base and the three corners attached are vibrated vertically, the free corner will lag due to its inertia and a voltage will be generated proportional to the acceleration. It is claimed by manufacturers that, in addition to giving greater sensitivity, this construction reduces undesirable variation due to temperature. Of the other piezoelectric materials mentioned above, barium titanate has to be electrically polarised before use. Being a ceramic it can be moulded to any desired shape, and soft soldered.

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CHAPTER 8

NOISE REDUCTION

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§ 1. Introduction

When it is necessary to consider possible ways of reducing a noise the alternative courses are (a) to reduce the noise of the source, (b) to reduce the transmission of the noise from the source to where it is undesired. In either case, modifications of one kind or another become necessary and it is usually a problem for the acoustical engineer to decide which alternative, or combination of the two, will effect the desired reduction in the cheapest and most generally acceptable way. The latter qualification is necessary because aesthetic consideration sometimes becomes important and may dictate a solution which is not the economic optimum. It is, therefore, necessary to consider all the implications of the various alternative methods before making the final choice.

In order to determine how much reduction is necessary, the first step is to measure the noise at a time when it is considered too loud. This is often in the middle of the night as it is then that other noises are usually reduced and the listener becomes more critical. The second step is to measure the permissible noise level. If it is not possible to stop the noise in question it may be possible to take readings under similar but tolerable conditions elsewhere. The difference between these two measurements gives the amount of the attenuation which it is necessary to introduce.

§ 2. Reduction at the Source

If the noise comes from a small machine it may be possible to take it to

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an acoustics laboratory where it can be studied without the restrictions usually associated with work on site. Fig. 77 shows an electric motor on test in a lined laboratory. It is bolted to a large concrete block, isolated from the building by an air gap all round the sides, and is loaded

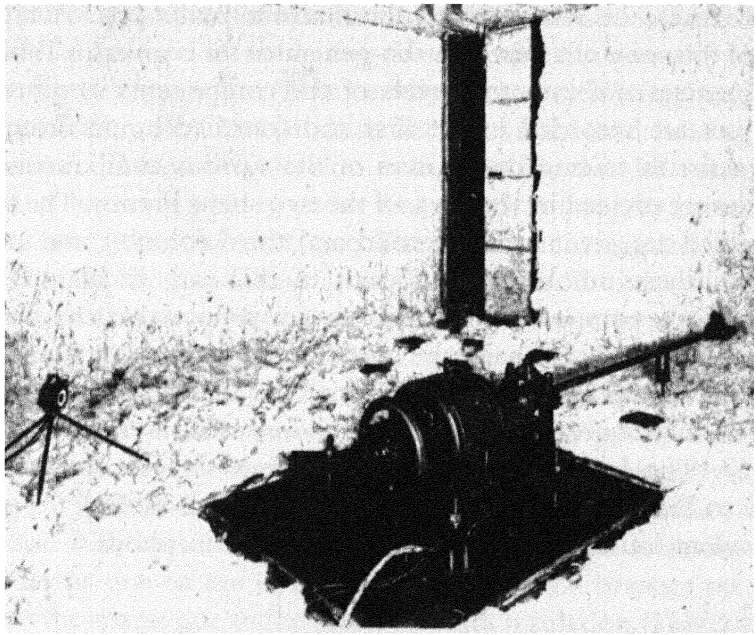


Fig. 77. Noise investigation on electric motor in anechoic chamber.

through flexible couplings and a long shaft by another machine in an adjacent room. A lining of cotton waste 6 in. thick with a 6 in. air space behind covers the walls, ceiling and floor of the room. Tests have shown [4] that the attenuation with distance from a point source of 100 to 10,000 c/s in such a room is within 1 db of the theoretical 6 db for each doubling of the distance to be expected from spherical radiation in free space. A microphone standing near the machine is connected to a noise meter and analyser in an adjoining room where the operator can make the measurements without disturbing the sound field.

If the measurements have to be made on site or in a works the conditions are often largely determined by other considerations. It may not be possible to have a clear space of more than a few yards round a machine and other sources of sound may be contributing to

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the noise. In such cases it is usual to choose the most open side of the machine to stand the microphone and to arrange, by shutting down as many other sources as possible or choosing a suitable time for the tests, that the background noise is at least 10 phons less than the noise to be measured. In the case of analyses, a test of the background noise alone will serve to rule out any components which might otherwise be attributed to the machine under test. A typical analysis of the noise of a geared turbo-generator set is given in Table II. The frequencies and intensity levels of the components as given by the analyser are recorded in the first and fourth columns from the left. To assist in tracing the sources of the various components the frequencies are divided by the r.p.s. of the two shafts in turn. The numbers obtained are given in the second and third columns and an inspection of these numbers is sufficient in this case to identify the sources of all the components. In many cases it is not so easy to identify all the components but usually the loudest components can be traced. The loudest components are determined by transferring intensity levels in db into equivalent loudness in phons, column five, by means of Fig. 67. The loudness numbers of column six are obtained by reference to Fig. 69 and their sum, transferred back by Fig. 69, gives the equivalent loudness of the noise as a whole in phons.

TABLE 2

TYPICAL SOUND ANALYSIS OF GEARED TURBO-GENERATOR

Turbine 7470 R.P.M. = 125 R.P.S.
Gear Pinion 30 teeth. Wheel 224 teeth. Cutting table 360 teeth.
Generator 1000 R.P.M. = 16.7 R.P.S. Slots 100. Commutator bars 200. Fan blades 12.

Frequency c/s	Frequency/Rps. Pinion	Wheel	Intensity Level Db	Equivalent Loudness Phons	Loudness Numbers	Probable Source
125	1	7.5	70	57	0.6	Turbine Unbalance
199	1.6	12	75	69	1.5	Generator Fan
1670	13.4	100	74	75	2.6	Generator Slots
3340	26.8	200	61	65	1.1	Generator Commutator
3740	30	224	75	80	4.0	Gear Contracts
6000	48	360	71	65	1.1	Gear Phantom
			Sum =	87	10.9	

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Considering the sources listed in the last column the first, turbine unbalance, recognised by one impulse per revolution of the pinion in column two, obviously calls for better balancing. This process used to be one of trial and error but now electrical devices give the amount and phase of the unbalance at each end of a rotor. The generator fan note, recognised because of the twelve blades and the same number of impulses per revolution of the wheel in column three, can usually be reduced by increasing the clearance between the blades and the nearest fixed edge. This usually means sacrificing aerodynamic performance, and a compromise between noise and heating may be necessary. The generator slot note, due to the 100 slots in the rotating armature passing under the poles, is radiated by corresponding vibrations of the yoke and frame. The pulsations in magnetic force due to the slots entering and leaving the magnetic field under the poles can be reduced by skewing the slots or the pole tips by a slot pitch, by semiclosing the slots and by grading the air gap from a small value under the centre of the pole to a value several times as large at the pole tips. These three measures, each of which can give 10 db attenuation, are additive and all three are sometimes necessary to reduce the slot vibration in very important cases. The commutator bars, in this case 200, cause a vibration of the brushes and sometimes of the brush arms due to the change in friction of the brushes on the bars and on the spaces due to the undercut mica insulation. This component would be reduced if the insulation, without undercutting, wore away at the same rate as the bars and had the same friction force on the brush as the bars. Such a material has not yet been found, although ruby mica is a near solution, so pulsating friction forces are inevitable. Some reduction, if only temporary, can usually be obtained by skimming up the commutator or by waxing it. The gear contact note is identified by the numbers of teeth on both pinion and wheel since the tooth contacts per second are common to both. Ideally, the profile of the teeth should be such that if the pinion rotates with a constant angular velocity so also does the wheel. Incorrect profiles will superimpose an oscillatory component on the mean angular velocity, so involving accelerating forces at the tooth contacts and journals. These pulsating forces at the journals are transmitted through the thin oil films there to the gear case from which the contact note is radiated. Hobs for cutting precision gears are carefully checked by

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the N.P.L. but small errors, of the order of 0.1 to $1.0 \cdot 10^{-3}$ in., are almost inevitable and can cause loud noises, as we have seen, at frequencies of the order of 10^3 to 10^4 c/s. Careful hand scraping of high spots on the teeth after a running-in period can give reductions of 10 db or more. The other gear note, called the 'phantom' gear note, has a characteristic number, in this case 360, corresponding to the number of teeth on the worm drive on the cutting table on which the gear was cut. This note is due to imperfect mating of the worm drive and the resultant non-uniform rotation of the cutting table which has the effect of superimposing a 'phantom' gear on that being cut having the same number of teeth as the cutting table. Careful correction of the errors in the worm drive can reduce this phantom note to a negligible level. Alternatively a cutting table with a 'creep drive', due to Sir CHARLES PARSONS, can be used to distribute the worm drive error round the gear. This usually leads to an increase in distributed or 'unpitched' noise instead of the outstanding phantom gear note.

The exhausts of piston internal combustion engines and compressors can be fitted with silencers to reduce their noise. These silencers usually consist of a chamber and a long tail pipe such as are installed on motor vehicles. By making the chamber large enough and the tail pipe long and small in cross section, the pulsations in the exhaust are smoothed so that the gas issuing from the tail pipe is in a steady stream. If f c/s is the frequency of the pulsations, v the volume of the chamber in cu. cm, l the length in cm and S the area of cross-section of the pipe in sq. cm, then to smooth out the pulses, the mass reactance of the gas of density ρ g/cc in the pipe must be at least several times greater than the stiffness reactance of that in the chamber,

$$i.e. \quad \frac{2\pi f \rho l}{S} \gg \frac{\rho c^2}{2\pi f v} \quad \text{or} \quad \frac{lv}{S} \gg \frac{c^2}{4\pi^2 f^2}$$

where c is velocity of sound in the gas in cm/sec.

If the attenuation of N times is > 3, $\frac{lv}{S} = \frac{N}{f^2} \cdot 3 \cdot 10^7$ approximately.

It is important to make the receiver and pipe out of sheet metal heavy enough not to dilate to the gas pressure pulsations. The harmonics are attenuated more than the fundamental up to the point

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where the dimensions of the chamber and pipe are comparable with a wavelength.

Electrical transformers are very difficult to quieten at the source. The noise radiated is mostly from core vibrations due to magnetostrictive elongation of the laminations and to flux distortions associated with joints. Magnetostrictive elongations, shown in Fig. 78 for typical

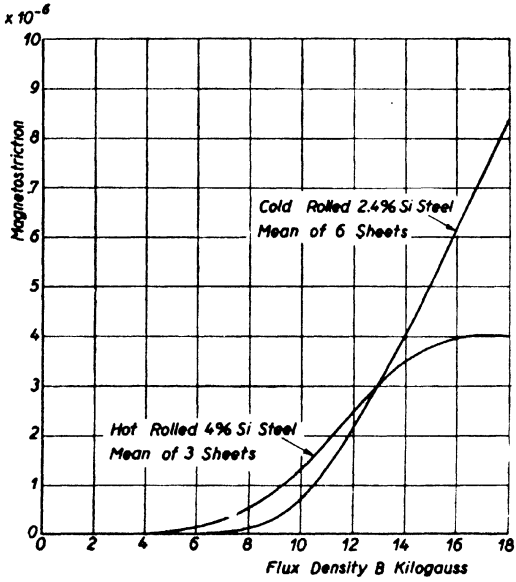


Fig. 78. Magnetostriction of hot and cold rolled transformer steel.

core materials, occur twice per cycle of magnetisation and so give rise to a fundamental tone of twice the supply frequency together with harmonics. Most power transformers are oil-immersed, the core vibrations being transmitted through the oil to the tank sides which radiate the noise to the air. In the past, core materials have been mostly hot rolled silicon steel sheets operating at flux densities up to 13,500 lines/sq. cm. The recently developed material with grain orientation by cold rolling can be operated at densities up to 16,000 gauss or more, but Fig. 78 shows that the magnetostrictive elongation is considerably increased compared with the older material at the lower density. This increase is confirmed by reports from U.S.A. of in-

creased noise. It, therefore, becomes more important to consider ways of reducing transformer noise. One way which has been proposed [19] is to surround the core in the oil with resilient absorbers. These absorbers, consisting of air-filled plastic containers 2.5 to 5 cm thick and extending nearly the whole height of the tank, yield to the pressure pulsations radiated in the oil from the core and pass on to the outer oil and tank side pressure pulsations of reduced amplitude. Attenuations of 25 or more phons are claimed. If m g is the mass per unit area of the oil and tank side looking out from the absorber of thickness t cm, the attenuation N, greater than 3, at frequency f c/s, is given by $N = \frac{4\pi^2 f^2 m t}{g r p}$ where $g = 981$ cm/sec.^{2}, $r = 1.4$ and $p =$ 1,000 g/sq. cm (approximately).

§ 3. Noise Reduction during Transmission

When it is impracticable or uneconomic to reduce a noise at its source there is the alternative of reducing it on its way to where it is considered to be too loud. It is then important to know how the transmission is taking place, *i.e.* whether the noise is air-borne or structure-borne. In some cases it is obvious which type of transmission is involved but in others it may be necessary to make a few simple tests and it may be found that both types are present.

§ 4. Reduction of Air-Borne Noise

The erection of a barrier between the source and listening point is often proposed to create a 'sound shadow' and so reduce the noise in the 'shadow'. The attenuation to be expected in such a shadow has been investigated theoretically by REDFEARN (1940) [20] and the analysis shows that the attenuation is dependent on the height of the barrier in wave-lengths above the line joining source and observer, and also on the angular distance of the observer measured from the top of the barrier into the sound shadow. Attenuations of 10 db may be obtained at ground level near a barrier 10 ft. high but if the observer is at an upstairs window, as often happens in cases of complaint at night, a barrier of 15 to 20 ft. may be required. In such cases complete enclosure of the source of noise is usually more satisfactory. Similarly,

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inside a building, if it is desired to prevent the noise of a certain source from spreading and causing annoyance, enclosure of the source is most effective. Walls of the usual building materials behave as masses. The attenuation provided by such partitions is given [21] in energy units at pulsatance ω by:-

$$\text{Reduction factor} = 1 + \left(\frac{\omega m}{2\rho c}\right)^2 = \left(\frac{\omega m}{2\rho c}\right)^2 \text{ (approx.) where } \omega = 2\pi f$$

m = mass/unit area of the partition
and ρc = characteristic impedance of air.
If m is 1b/sq.ft.,

Attenuation = $20(\log m + \log f) - 29$ db (approximately)

Porous materials such as rock wool are often proposed for sound attenuating partitions. The attenuation in db of walls of such materials is proportional to the thickness, not its logarithm, and also to the density of packing. Fig. 79 permits a comparison of porous and solid

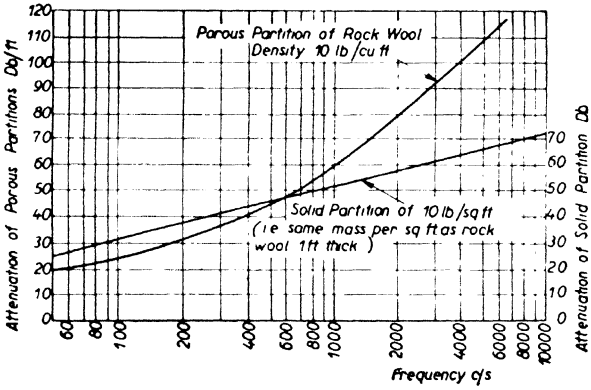


Fig. 79. Sound attenuation of partitions.

walls of similar mass. It is apparent that walls of rock wool become attractive for thicknesses greater than 1 ft. and at high frequencies, otherwise solid walls are better in attenuation.

However, it is important to remember two effects when a source of sound is enclosed. Both the sound intensity and the temperature in the enclosure may be increased. The first is due to 'build-up effect' [19],

the intensity I_{max} to which the sound builds up asymptotically being $I_{max} = \frac{4E}{cA^2}$ where E is the rate of emission of sound energy, c is the velocity of sound in air and A is the total absorption in the enclosure (*cf.* Chap. 6, § 3).

It follows that, for a given source, the final intensity in the enclosure is inversely proportional to the total absorption inside. If bare walls are used, an increase in intensity, compared with free space, of 10 db is often encountered and the effective attenuation of the enclosure is reduced by this amount. It is usually an easy matter to introduce sufficient sound absorption to double that in the bare room and so secure a reduction of 3 db in intensity. However, successive doublings become rapidly more expensive and less justified by their 3 db reduction. The thermal problem created by enclosure may call for special cooling by air or a liquid medium. If air ventilating ducts are used, it may be necessary to treat them with sound absorbent material to attenuate the escaping sound. The attenuation of sound in lined air ducts has been treated theoretically by several investigators but the treatments are not entirely satisfactory because when all the factors, including transmission in the lining material, are taken into account the equations giving the attenuation are virtually unsolvable. For this reason some results of an experimental investigation [22] are given in Fig. 80(a) and (b). The (a) curves refer to 66% space factor, *i.e.* 33% of the cross-section taken up with absorbent material, and the (b) curves to 33%. It is apparent from these curves that for good attenuation the thickness of lining must be not less than one twelfth of a wave-length and the width of air passage not greater than two wave-lengths. To meet these rules it may be necessary to subdivide the ducts by 'splitters' of absorbent material of twice the thickness of linings. These splitters behave, as far as attenuation along a duct is concerned, like two linings back-to-back, so the curves of Fig. 80 still apply for the appropriate space factor. The introduction of bends into a duct with the intention of improving the attenuation at high frequencies is of very little use [22] except at particular frequencies and may even be harmful in reducing attenuation at other frequencies.

When it is necessary to quieten a large room containing many operatives each controlling a source of noise such as a typewriter, it is often advantageous to have a low ceiling and cover it with absorbent

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material such as acoustic tiles. The latter consist of a layer of absorbent material, usually 1 in. or 2 in. thick, covered by a thin layer of perforated hard material, usually plaster or mill-board. The lower the ceiling, the greater the sound attenuation laterally and the less disturbed each operative is by the noise of machines not under his or her control. Experience indicates that this is a point of some psychological importance.

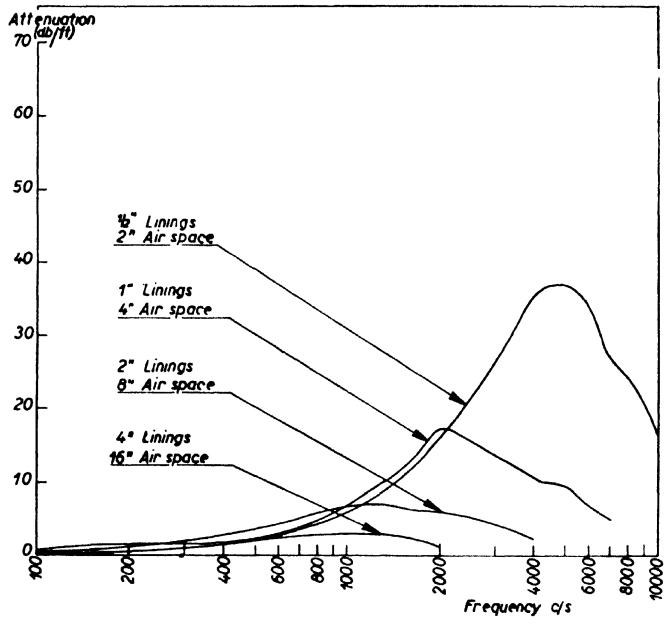


Fig. 80a. Variation of attenuation with thickness of porous linings for ducts in which air space is $66^2/_3\%$ of total duct width. Rock wool linings (13.7 lb/ft^3).

§ 5. Reduction of Structure-Borne Noise

When it is necessary to reduce the structure-borne noise or vibration radiated from a source it is usually possible to effect substantial reductions by interposing resilient mountings between the source and its foundations. An essential condition is that the impedance of the mountings must be small compared with that of the foundations

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at the frequency of the vibration. The reduction factor, N, of vertical vibrations is then given approximately by [18] $N = (\frac{f}{f_0})^2$ where f is the frequency of the vibration and f_0 is the vertical natural frequency of the source on the mountings. If the mountings are perfectly elastic and so obey HOOKE'S Law in vibration, $f_0 = \sqrt{\frac{10}{h}}$ approximately where h = vertical compression of mountings in inches.

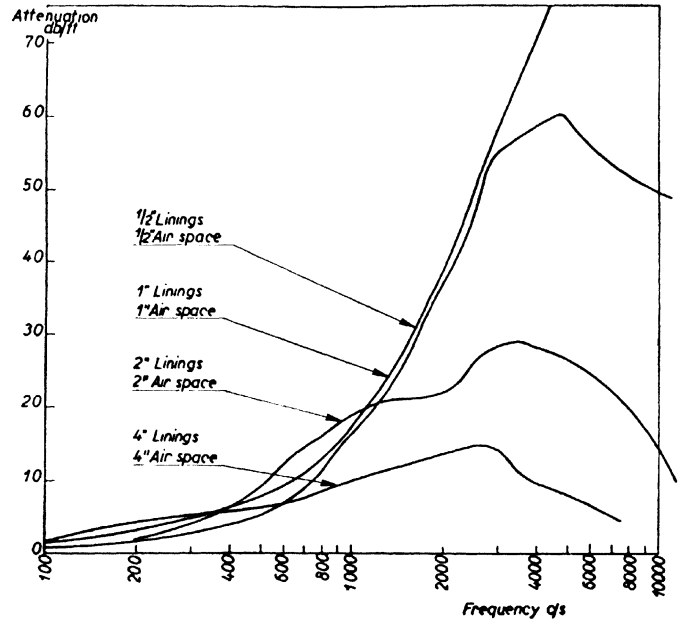


Fig. 80b. Variation of attenuation with thickness of porous linings for ducts in which air space is $33\frac{1}{3}\%$ of total duct width. Rock wool linings (13.7 lb/ft^3).

Most materials are not perfectly elastic so, if the dynamic stiffness [18] is $k \times$ static stiffness, $f_0 = \sqrt{\frac{10k}{h}}$

$$\therefore \text{ Attenuation} = 20 \log \frac{f^2 h}{10k} \text{db.}$$

The relation between attenuation, frequency and compression for $k = 1$ is illustrated by the curves of Fig. 81. If a certain compression

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is indicated by Fig. 81 as being necessary for a given case then this compression must be multiplied by the k of the material used. The ratio k is unity* for steel springs, 1.5 to 2.5 for rubber [18] and higher for cork, felt and certain other proprietary mounting materials. The value increases with frequency for most materials [23] over the range 1 cycle

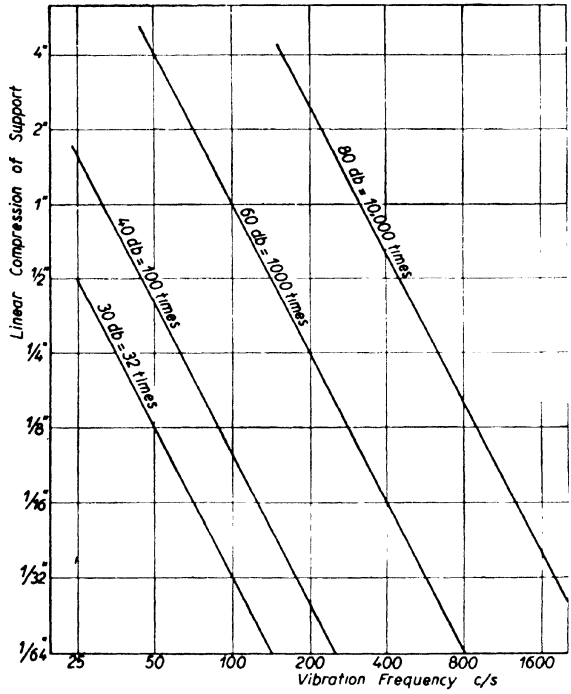


Fig. 81. Variation of vibration attenuation with frequency and linear compression of perfectly elastic support. $k = 1$.

in 10 minutes to 100 c/s. Typical curves for a sample of rubber showing the variation in stiffness and Q with frequency over this range are given in Fig. 82a and b. If facilities are not available to determine the dynamic stiffness of a sample of resilient material, the 'incremental stiffness' or 'zero-frequency dynamic stiffness' under appropriate static load is a better guide to its performance as a mounting than its static stiffness. An increase and decrease of say 5% in load can easily be

* The author's work to date indicates a value between 1.03 and 1.06.

arranged at intervals in a simple static compression life test and the incremental stiffness data obtained in this way are most informative. Some materials deteriorate rapidly under load, the incremental stiffness rising until it is clear that the materials have become virtually solid.

It is usually sufficient to design mountings to give the required attenuation of vertical vibration at the lowest frequency present.

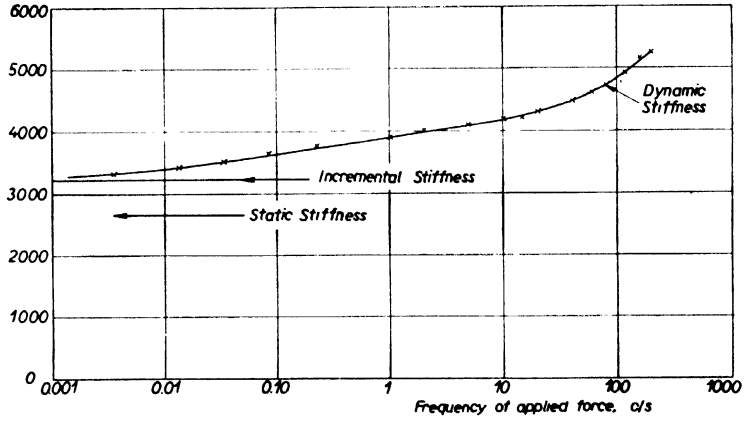


Fig. 82a. Bonded rubber mounting: stiffness, r.m.s. displacement = 0.1 — 0.5 $\times 10^{-3}$ in. Static load = 200 lb.

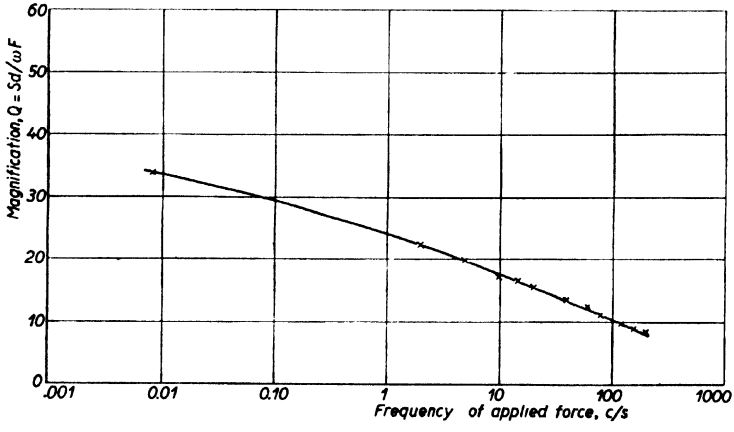


Fig. 82b. Bonded rubber mounting: magnification, $Q = Sd/\omega F$. R.m.s. displacement = 0.1 — 0.5 $\times 10^{-3}$ in. Static load = 200 lb.

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However, there are six possible modes of vibration of a supported mass and some of these are coupled if the centre of gravity of the mass is not in the plane of the supports. If the mountings are not constrained laterally the attenuations of other modes of vibration are usually greater than that of the vertical mode.

In some cases, it is advantageous to mount a machine on a large concrete block which rests on resilient mountings. This has the dual advantages of reducing the vibration of the machine by increasing the mass and of increasing the stability by lowering the centre of gravity. The latter point is important in the case of tall machines like vertical engines. There is also the necessity in the cases of printing presses and stamps to provide a large mass to act as a foundation block. When such large blocks are mounted on resilient mountings it is convenient to arrange for the mounting to be accessible to facilitate inspection and, if necessary, replacement. One way of arranging this is illustrated in Fig. 83.

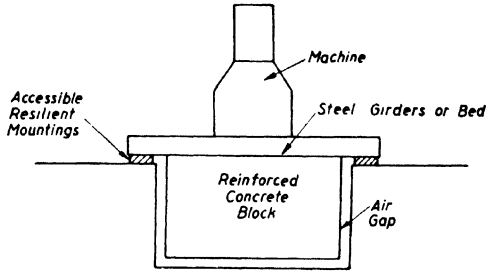


Fig. 83. Tall reciprocating machine on resiliently mounted block.

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DIVISION FOUR
SPEECH AND HEARING

The material in this division was prepared by members of the Transmission Research Department of Bell Telephone Laboratories, Inc., Murray Hill, New Jersey, U.S.A., under the supervision of R. K. POTTER, Director of Transmission Research, and J. C. STEINBERG and W. E. KOCK in charge, respectively, of Research in Speech and Hearing and Research in Physical Acoustics. Contributing members of the Technical Staff were H. L. BARNEY, B. P. BOGERT, H. K. DUNN, N. R. FRENCH, M. B. GARDNER, F. K. HARVEY, W. A. MUNSON, G. E. PETERSON and F. M. WIENER.

CHAPTER 9

SPEECH AND HEARING IN RELATION TO COMMUNICATION SYSTEMS: INTRODUCTION

R. K. POTTER and J. C. STEINBERG

The past few decades have seen significant advances in our knowledge of the acoustic and subjective characteristics of speech and hearing. These advances have been stimulated in part by the increasing importance and wider use of wire and radio speech communication systems. Both involve problems of picking speech up at one point, transmitting it over great distances and reproducing it at another point. In such systems, the speech mechanism and the ear are the terminal elements. To design and engineer the systems properly, information on the acoustic and subjective characteristics of speech and hearing are necessary. The purpose of this division is to bring together the information in this field that has been obtained to date.

The essential elements in speech communication are indicated in Fig. 84. They comprise a message sequence, individuals *A* and *B* and a transmission system. The individuals are represented by three elements; the speech mechanism, the ear and the brain. The speech mechanism generates the sound waves or message, but it acts under the control of the ear and the brain as indicated by the feed-back

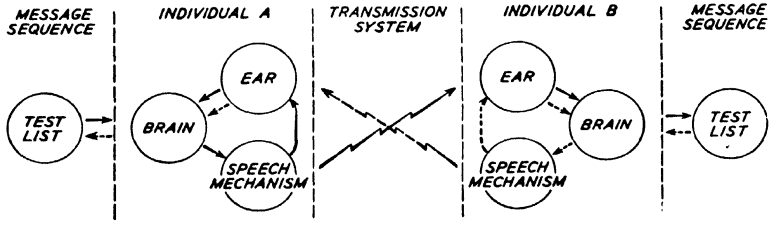


Fig. 84. Elements in communication by speech and hearing.

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paths shown. For convenience, the message sequence is shown separately since it may be a written message such as a test list of speech sounds. It is, of course, closely linked with the three elements representing the individual. For our purposes, speech may be regarded as a habit acquired through the language experience of the individual. A message begins as an idea expressible in the language. The idea starts a sequence of neural and muscular operations of the elements representing the individual. The sequence is carried on from habit and results in a vocal gesture or flow of speech.

The emphasis in the earlier work on speech and hearing was directed more toward the part indicated as 'transmission system' in Fig. 84. Conceptually, the transmission system was thought of as a microphone, a transmitting medium and an earphone. The problem was one of designing the system to fit the message. Consequently, the objectives of speech and hearing studies were concerned principally with transmission requirements for speech. Information was needed on such questions as required frequency and voltage ranges—time distributions of peak voltages—superposition of peak voltages in multiplex transmission—speech power requirements—effects of distortion on speech intelligibility.

To obtain the information, measurements were made on the physical characteristics of speech at the microphone position. The measurements included such quantities as instantaneous, peak, and average levels in the whole speech spectrum and in frequency bands. Also studies were made of the spectrum of individual sounds and their frequencies of occurrence in telephonic speech. The sensitivity of the ear, the intensity and frequency range of audible sound, loudness and masking were investigated. Effects of distortion were studied by means of articulation and intelligibility tests. Test lists of words and sentences were called into the microphone and listeners wrote down the words as heard from the earphones. Empirical relations between the characteristics of the system and the percentage of called sounds correctly received were obtained.

Recent developments in communication theory [1] have tended to shift the emphasis in speech and hearing studies to problems of transmitting the information content of speech, in contrast with transmitting speech wave-forms as such. From this viewpoint, a speech message is regarded as a point in a space of n dimensions. The amount of

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information conveyed by the message is related to the number of dimensions. A similar space can be associated with the transmission system. The capacity or rate at which information can be transmitted by the system is related to the number of dimensions. The problem in speech communication is one of using the system at its maximum capacity. It is generally recognized now that this involves two kinds of operations. One has to do with representing the message by a minimum number of independent dimensions. The other has to do with encoding the message into a signal that best utilizes the capacity of the transmission system. Both operations are in the area of designing the speech messages and signals to fit the transmission system.

One of the problems from this viewpoint is that of determining the information-bearing elements or dimensions in speech. If several persons speak a given word and listeners agree that the same word was spoken, what are the invariants in the several utterances that enable the sounds to be identified as one of a number of possible words? The attack on this problem again involves speech and hearing studies. The emphasis, however, is more on the parts of Fig. 84 designated as 'message sequence' and as 'individual *A* or *B*'.

In hearing, this interest has led to investigations of the type of transmission provided by the ear, especially as regards the information that can be transmitted. This includes studies of the nerve impulse and excitation patterns produced by sound at the several levels in the auditory pathways that extend to the cortex. Speech studies are concerned with the information-bearing elements of speech and their specification from acoustic measurements on speech waves. For the most part, the work has been carried on with detached speech sounds and involves relating their phonetic characteristics to measurable acoustic properties. Studies of the vocal tract as a generator, especially with regard to the constraints it imposes, are in progress. The effects, on information, of the systematic structure in language are also of interest. Finally, the functioning of the overall communication system including sensory and motor terminals and the brain is receiving increased attention [2].

As most of this work is still in progress, extensive data are not available for inclusion here. A good survey of preliminary data and methods may be obtained from the several papers presented at a Speech Communication Conference held in June 1950 at Massachusetts

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Institute of Technology. They appear in the November 1950 issue of the Journal of the Acoustical Society of America.

The first part of the present speech and hearing division deals with the characteristics of speech. This includes information on the speech machanism, the production of speech, and on the physical characteristics of speech sound waves. The second part deals with hearing. Information is given on the properties of the ear, the subjective aspects of perception, and on auditory patterns or form. The third part deals with speech transmission rating. Various types of tests are described and results given for a number of different transmission systems. Since much of the work played a role in microphone development, a discussion of types and characteristics of microphones has been included as a fourth part of the division.

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CHAPTER 10

SPEECH

For authorship, see p. 193

As indicated above, the material in this section is concerned primarily with the physical properties of speech and the physiological activities associated with speech production. This is a part of phonetics. It is recognized that the meaning and structure of language is assuming increasing importance to those interested in the applications of mathematical theories of communication. For information in this area, the reader is referred to sources dealing with phonemics and linguistics [1,2,3].

(A) THE SPEECH PROCESS

§ 1. The Speech Mechanism

Speech is a form of motor behaviour which must be learned and controlled by means of the sense organs. Hearing and kinesthesis, the sense of muscle movement, are the dominant senses in the development and maintenance of speech. The information from these senses is organized and coordinated by the central nervous system in the direction of the speech musculature [4, 5].

(i) *The Speech Musculature*

Speech is produced by various operations upon the breath stream as it passes through the cavities of the nose, mouth, and throat. The air is exhaled from the lungs in the production of most speech sounds. In rare instances, basic speech sounds are produced during inhalation; these sounds are known as clicks.

The speech mechanism is shown schematically in the median section of the head and chest of Fig. 85. To a large extent ,the motor mechanism employed in the production of speech is organized from systems

affording more basic life functions. In particular, speech employs organs from the respiratory system and from those used for chewing

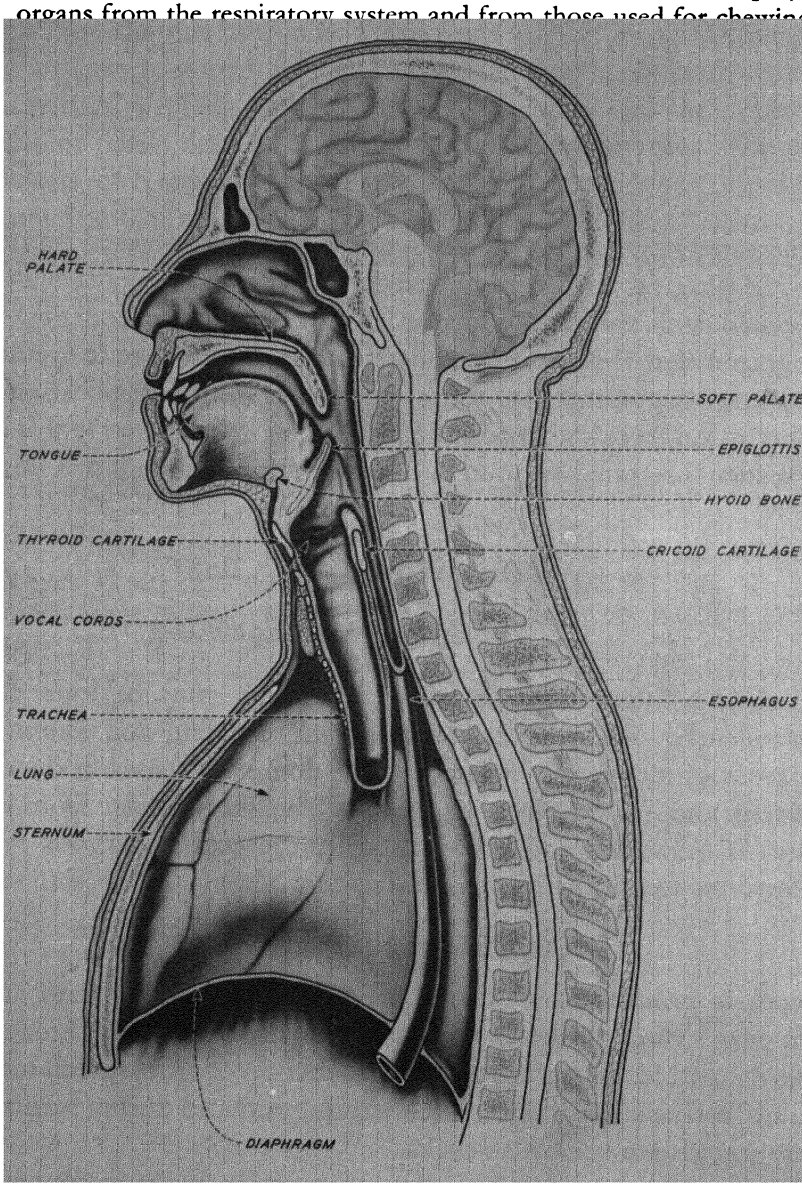


Fig. 85. Schematic median section of the head and thorax.

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Air is taken into the lungs by enlarging the chest cavity. This may be achieved by increasing the diameter of the rib cage by means of the thoracic musculature, or by lowering the diaphragm, shown in Fig. 85. Air is forced from the lungs by decreasing the diameter of the chest cavity or by contracting the abdominal muscles, thus forcing the viscera inward and the diaphragm upward.

The normal breathing rate in healthy human beings is considered to be about 16 times per minute. In speech the rate is usually slower and the ratio of exhalation time to inhalation is markedly increased [8] (*cf.* Ref. 5, p. 489). During speech the expiratory pressure is increased over that for quiet breathing [9].

In exhalation, air passes from the lungs into the trachea or windpipe. The entrance from the trachea into the throat cavity or pharynx is controlled by the valve-like structure of the larynx.

The larynx is constructed of cartilages and muscles which house the vocal cords. The vocal cords are two wedge-shaped lips of ligament and muscle tissue which lie at either side of the larynx, as shown in the figure.

In voiced speech sounds, the vocal cords are brought together and then set into vibration by forcing air between them from the trachea beneath [10]. The varying opening between the cords is called the glottis. In normal speech, the vibration rate rarely falls below 80 or 90 cycles per second. This is a rate far in excess of that which could be achieved by sequential nerve innervation [11] and is based upon the BERNOULLI principle, with a relatively constant muscle tension maintained for a given frequency of vibration.

In whispered speech the cords are partially closed and produce a turbulence in the breath stream as it is emitted from the larynx.

In most speech sounds the air is directed from the throat into the mouth cavity and thence past the lips. The tongue and lips are highly flexible in their shaping of the vocal cavities, and are also employed in forming constrictions and stoppages at various positions along the vocal tract. Both the tongue and the lips are also used to direct the breath stream across the sharp cutting edgeof the teeth in the formation of fricative sounds.

The pathway through the nose is controlled by the soft palate or velum. This structure is made of muscle and is located between the oral and the nasal sections of the throat, as shown in Fig. 85.

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(ii) *Innervation of Speech Musculature*

Language is essentially symbolic in character, and the meaningful units of speech represent a great variety of stimuli, such as objects, actions and relationships. A most fundamental step in the development of speech is the establishment of associations between such stimuli and motor movements of the speech musculature. Associations of this sort are formed by means of the cerebral cortex through which the highest order neurological functions of the body are directed [5, 12, 13, 14].

There are essentially two neurones or nerve units which direct nerve currents from the cerebrum to muscle fibres (*cf.* Ref. 13, pp. 327–333) The first neurone is known as the upper motor neurone and usually has its cell body located in the frontal lobe of the cerebrum. The upper motor neurones of the central nervous system connect to the lower motor neurones through the neurone juncture which is called a synapse. The lower motor neurones extend to the muscle fibres.

The cerebellum, located at the rear of the brain, is the chief coordinator of neuro-muscular movement (*cf.* Ref. 14, pp. 294–303). It may be thought of as essentially responsible for the feedback circuits in the nervous system. Associated indirect connections from the cerebral cortex and also from the sense organs through the cerebellum afford properly organized innervation to the speech musculature under widely diverse conditions (*cf.* Ref. 13, pp. 241–243; 333–335).

§ 2. Speech Sounds

Linguists have found it possible to analyze any particular language into a relatively small number of basic units or speech sounds [1, 15, 16]. In continuous speech these basic elements are not discrete; rather, there is a relatively continuous movement of the speech musculature from sound to sound.

The speech sounds differ somewhat in organic formation, of course, when pronounced by different speakers, and to a lesser extent when produced by the same speaker at different times. The sounds are normally classified according to the manner of their organic formation.

(i) *Vowels*

Vowel sounds are normally voiced; that is, the vocal cords are in vibration during their production. The various sounds are obtained

TABLE 1

IPA SYMBOLS AND KEY WORDS FOR THE VOWELS, ARRANGED ACCORDING TO TONGUE AND LIP POSITIONS

		Front	Central		Back
		Unrounded			Progressively less rounding
Symmetrical Vowels	High	i meet (6.44)			u pool (6.26)
		ɪ hit (10.27)			ʊ book (2.96)
		e sera (It.)	ɜˈ bird (0.80)	ə about	o beau (French)
		ɛ let (7.69)	ɜ bird (South Eng.)	ʌ cup (4.14)	ɔ law (4.15)
	Low	æ hat (6.89)	a patte (French)		ɒ hot (South Eng.)
		Progressively less rounding			Unrounded
Asymmetrical Vowels	High	y über (Ger.)			ɯ Fugi (Jap.)
		ʏ fünf (Ger.)			ɣ [ghɣr] (marathi)
		ø peu (French)			ɵ dum (Swed.)
	Low	œ peur (French)			ɑ father (6.52)
Diphthongs		eɪ late (4.78)	aɪ hide (7.58)		aʊ house (1.69)
		ɔɪ boy (0.19)	oʊ hope (4.74)		ju few (0.26)

by different shapings of the vocal tract, chiefly by means of the tongue and lips. The tongue positions in forming the vowels have been studied by X-ray techniques. Only in general terms are the same tongue and lip formations maintained by the same speaker and by different speakers in producing the same vowel sound [17, 18].

Table 1 shows the symbols employed by the International Phonetic Association to represent the vowels [16]. The vowels are arranged in this table according to their characteristic tongue and lip formations. The position of the tongue hump along the vocal tract is shown by the headings across the top, and the height of the hump is shown down the side. For each group of vowels shown the lip and jaw openings become greater as the tongue hump moves from high to low, and in the case of the lip-rounded vowels, the degree of rounding becomes less from high to low.

The unrounded front vowels and the rounded back vowels appear more frequently in most languages, and are called symmetrical vowels.

Nasal quality in the vowels is primarily controlled by the position of the soft palate, which determines the degree of opening into the nasal passageways. Most vowels in most languages are non-nasal. In phonetics nasalization is indicated by [~] as in [$\tilde{\text{ɔ}}$] and [$\tilde{\text{a}}$], French (longtemps) [l$\tilde{\text{ɔ}}$t$\tilde{\text{a}}$].

(ii) *Consonants*

The consonants usually involve greater constrictions than the vowels and are normally employed in conjunction with one or more vowels to make a syllable.

There are four general types of articulatory formations which are employed in producing consonants. These are listed as the headings across the top in Table 2. (1) – Plosives (or stops) are formed by blocking the speech mechanism so that no air may flow, and then an opening is suddenly created at some place along the vocal tract to form a puff of air. (2) – Fricatives are formed by blowing air through a narrow orifice or across the cutting edge of the teeth. (3) – The nasal consonants are formed by lowering the soft palate and blocking the mouth so that air must pass through the nose. (4) – The glides or semi-vowels are much like the vowels in that their formation is relatively open. A semi-vowel or nasal may be employed as the principal vowel of a syllable.

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TABLE 2

THE CONSONANT CHART (MODIFIED FROM IPA)

	Plosives (stops)	Fricatives	Nasals	Semi-Vowels
Two lips				
Voiceless	p–pit (2.54–1.24)	Φ–Schwester (Ger.)		
Voiced	b–bit (4.64–0.42)	β–saber (Sp.)	m–me (5.89–5.48)	w–was (9.38—)
Lip to teeth				
Voiceless		f–fill (3.96–1.37)		
Voiced		v–vest (1.25–4.23)	ɱ–invento (Sp.)	ʋ–westland (Dutch)
Tongue to teeth or gum ridge				
Voiceless	t–to (7.86–14.30)	θ–thick (2.02–0.04) s–see (5.46–3.13)		
Voiced	d–do (6.21–4.44)	ð–this (6.72–1.25) z–zoo (0.34–6.01)	n–no (4.99–12.52)	r–red (2.78–13.05) l–lateral (4.31–8.40)
Tongue to gum ridge and hard palate				
Voiceless		ʃ–shed (1.74–0.32)		
Voiced		ʒ–pleasure (0.02–0.01)		
Tongue to hard palate				
Voiceless	c–kotya (Hung.)	ç–ich (Ger.)		
Voiced	ɟ–nagy (Hung.)	j–yes (6.48–)	ɲ–ogni (It.)	ʎ–egli (It.)
Tongue to soft palate				
Voiceless	k–cap (5.55–2.85)	x–ach (Ger.)		
Voiced	g–gap (4.33–0.38)	ɣ–vagen (Ger.)	ŋ–sing (–3.57)	
Glottal				
Voiceless	ʔ–verein (North Ger.)	h–hat (5.75–)		
Voiced		ɦ–ahead		

Initial Compounds		Final Compounds	
pr (1.06)	pl (0.36)	nt (4.40)	ld (0.75)
hw (0.91)	kw (0.28)	nd (2.56)	rz (0.57)
st (0.87)	bl (0.23)	st (1.18)	ks (0.47)
tr (0.69)	sp (0.19)	ts (1.11)	kt (0.42)
fr (0.62)	kl (0.18)	nk (0.76)	rd (0.37)

The consonant symbols employed by the International Phonetic Association [16] are shown in Table 2. The organs employed in the formation of the specific sounds are shown along the side. In the voiced consonants the vocal cords are in vibration.

The [r] shown in this figure is formed with the tongue tip and takes several forms. In English *red*, it is a frictionless continuant; in General American *bird* [ɜ˞] it has the value of a stressed vowel; in Southern English *very*, it may be a one tap trill. In Italian, Spanish, and several other languages [r] becomes a tongue trill with various numbers of tongue to alveolar ridge flaps. In the uvular [R] of Parisian French, the soft palate is vibrated.

When two consonants occur together in speech the formation of one is very frequently affected by the other. For example, when English *what* is followed by *name*, the [t] of *what* is normally exploded through the nose rather than the mouth. Such effects are known as assimilation [19].

(iii) *Occurrence of Speech Sounds*

The different sounds of a language do not occur with equal frequency. There are various linguistic and phonological laws, some of which are known, which determine the phonetic structure of a language at any particular time in its history [20, 21] (*cf.* Ref. 3, pp. 346–391).

Studies have been made of the frequency of occurrence of sounds in American speech [22, 23, 24, 25]. The numbers shown in Table 1 for the vowels and diphthongs and in Table 2 for the consonants and consonant combinations are from the study of telephone conversations by FRENCH, CARTER and KOENIG. Because of their frequent occurrence, articles were excluded in these tabulations. The first number following the consonant is for the relative frequency of occurrence of the consonant in the initial position, and the second is for the final position.

(iv) *Speech Disorders*

Disorders of speech may have a functional basis, or may arise from organic defects in the auditory apparatus, the nervous system, or the motor speech mechanism [26, 27, 28, 29, 30].

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§ 3. The Acoustics of Speech Production

Three steps are involved in the production of a speech sound: the generation of a complex sound at a source somewhere in the vocal tract, the selective transmission of the components of this sound by the tract lying between the glottis and the lips, and the radiation of the modified sound from the lips.

In the case of vowels and semi-vowels a complex tone is produced at the glottis, from the modulation of the breath stream by the vibrating vocal cords. The tone consists of a fundamental frequency (which determines the pitch of the voice), and harmonics of this frequency. The fundamental varies widely between speakers, and for a single speaker during the course of his speech. Average values for American speakers with superior speech are approximately 130 cycles per second for men and 205 c/s for women [31, 32, 33, 34, 35]. A small child's voice may be pitched as high as 400 or 500 c/s.

Pitch inflection during speech and vibrato in singing are forms of frequency modulation of the glottal tone.

The harmonics of the glottal tone depend upon the wave shape of the volume velocity of the air stream, as modulated by the vocal cords. The vibration of the cords themselves has been observed in detail through high-speed motion pictures [36, 37] and stroboscopic pictures [38]. The pictures give some notion of, but do not show definitely, the shape of the volume velocity wave. FLETCHER (*cf.* Ref. 39, p. 50) and others [40, 41, 42] have concluded that the spectrum is approximately one in which the harmonic amplitudes vary inversely as the 1.5 power of the order of harmonic. This conclusion was reached from the departure of the shape of a vowel spectrum from that of simple resonances. Although it is certain that the harmonic amplitudes decrease toward higher frequencies, the exact law still awaits experimental verification.

The selective transmission of the vocal tract arises from the formation of constrictions by tongue and lips, and the separation of the vocal tract, thereby, into cavities. Such a system has resonances, which is to say that the transmission curve has maxima at various frequencies. A group of harmonics near a maximum will be strongly transmitted, and constitutes a region of energy concentration in the sound heard. Such a region is called a *formant* of the vowel (*cf.* Chap. 18, § 2). Different

vowels are made by moving the tongue, lips, and jaw, hence altering the resonant frequencies. The nasal cavity also takes part in the production of some sounds, when the passage from the throat into the nose is open. For these sounds, the nasal cavity is a branch line paralleling a part of the main vocal tract, with a secondary radiation point at the nostrils.

Loss of energy, through viscous friction, absorption by the walls, and radiation from the tract, produces damping of the resonances. This determines the widths of the resonant peaks and their relative heights.

It has been pointed out by DUDLEY [43] that speech may be regarded as a series of modulations. From this point of view, the continuous changing in the configuration of the vocal cavities, as they pass through the various speech sounds, is an amplitude modulation of the components of the glottal tone. The basic rate of this modulation rarely exceeds 8 or 10 cycles per second [11], and consists of a rather continuous flow in the form of the spectral pattern. Emphasis in speech is another form of amplitude modulation of very low frequency.

Radiation from the lips, for frequencies up to about 4000 cycles per second, is approximately that of a small orifice in a baffle. The higher frequencies are more strongly radiated than the lower, the ratio of pressure at a point in front of the lips to volume velocity emerging from the lips being approximately proportional to frequency.

Of the three steps in vowel production, only the selective transmission of the vocal tract plays a significant role in the differentiation of one vowel from another. The theory of vocal resonance has been treated by CRANDALL [44] and by BENTON [45], on a 'lumped constant' basis. More recently DUNN [46] has used distributed constants, and has shown that the frequencies of the first three formants of vowels may be calculated from the dimensions of the tract, as revealed by X-ray photographs such as those of RUSSELL [17, 47]. He finds that the dependence of the first and second formants of a vowel upon the throat and mouth cavities is mixed, although for some vowels a partial separation can be made which may be in either direction. For example, in [i] the first formant is more dependent upon the throat cavity, while in [ʌ] it is more dependent upon the mouth cavity. The third formant is usually due mainly to the second resonance of the

longer cavity. In some cases, however, it must be described in terms of three cavities.

Fig. 86 shows the frequency of the first formant plotted against that of the second formant, for the various vowels pronounced by one speaker. The unrounded (lips relaxed) vowels are shown by the

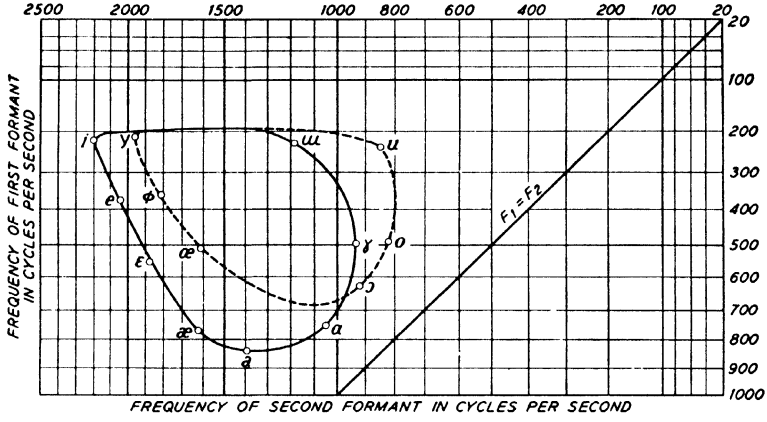


Fig. 86. The formant loops for lips-unrounded and lips-rounded vowels, by a single speaker (after POTTER and PETERSON).

solid loop, and the rounded vowels by the dashed loop. Several writers have pointed out that the relationships among the vowels in such a formant plot conform to the relationships among the vowels in the tongue and lip position diagrams which are employed by phoneticians [48, 49, 50, 51]. Both X-ray photographs [47] and theoretical treatment [46] have shown, however, that more than two parameters are required to specify a vowel in physiological terms.

In the production of voiced vowels and semi-vowels, as described above, a periodic source at the glottis is used. In whispered speech the vocal tract transmission is the same, and the source is again at the glottis, but the sound is random rather than periodic. The vocal cords do not vibrate, but are held in a partially closed position, and the random noise is produced by turbulence in the breath stream at this point. In the fricative consonants the same type of source is used, but the constriction at which it is generated may be at various points along the vocal tract. The same is true of the stop consonants, the

random sound being generated at the moment of release. When the source is at other points than the glottis, the configuration of the tract is less effective in modifying the spectral distribution of energy emerging from the lips.

§ 4. Artificial Speech

One of the earliest attempts to produce speech by artificial means was that of VON KEMPELEN, in 1791. His machine has been described and others summarized in a recent paper by DUDLEY and TARNOCZY [52]. The early efforts were directed toward mechanical imitations of the speech apparatus. In more recent times, PAGET [53] has made very successful mechanical models of vowel and consonant sounds. Others, notably HELMHOLTZ [54] and MILLER [55], have synthesized vowel sounds by supplying a tone at the proper frequency for each strong harmonic. STEWART [56] made the first electrical circuit to produce vowels, using two simple tuned circuits and a complex source, to produce two formants in desired positions and amplitudes. Along the same line WAGNER [57] has used four independent tuned circuits, but his best effects were obtained when additional suppression of harmonics between and above the formants was used. DUNN [46] has suggested that a more natural result would be obtained with independently controlled resonators, if they were connected as low-pass sections in tandem (separated by thermionic valves), instead of in parallel. He also introduced the 'electrical vocal tract,' in which a transmission-line type of circuit is used to represent the vocal tract, the separation into cavities being accomplished by the insertion of lumped inductances into the line. Such model cavities have more than one resonance each, as in the real case.

A different type of artifical speech maker, called the 'voder,' was introduced by DUDLEY [58]. It uses complex sources, both periodic and random, and produces the sounds by adjusting the amount of energy allowed to pass through each of a number of band pass filters. An extension of this principle, called the 'vocoder' [59], analyzes real speech into bands, transmits the variations of energy in each band over a line in relatively narrow frequency channels, then uses the transmitted signals to reconstruct speech at the receiving end, in the manner of the voder. A saving in transmission bandwidth is accomplished in this way.

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The artificial larynx [60] is a device for introducing a periodic complex source into the vocal tract, through the mouth, for those who have lost the use of their vocal cords. The sound is generated by a reed, actuated by the user's own breath. A similar arrangement has been demonstrated by FIRESTONE [61], in which a tube attached to a telephone receiver conducts sound into the mouth. In this case, various kinds of original sources can be used, and entertaining effects are produced when the modulations of speech are superimposed upon the original sound. WRIGHT [62] gets the same results with a small receiver attached to the throat.

The artificial mouth [63, 64] is used in testing microphones, and consists of a receiver made to imitate the impedance, directivity and power output of the real mouth.

In 'visible speech' [65, 66, 67, 68] the artificiality is not in the speech production, but in the method of perception. The result of a band analysis of speech is portrayed in such a way that the eye can (with training) perform the discriminating functions usually done by the ear. A horizontal time scale several seconds long is used, with the analysis of speech into frequency bands shown vertically. The brightness (or darkness) of the pattern at any point shows the intensity of sound at the corresponding time and frequency, thus revealing the formants and their variation with time. Deaf persons can learn to read the patterns. It has also been suggested that a deaf person can improve his speech by making the patterns from his own voice match those from someone with normal hearing. The widest use to date, however, has been in phonetic research, where the rapid analysis obtained at each instant of time is, for many purposes, a great advantage over the old methods of FOURIER analysis of each fundamental period of the sound. An example of a visible speech pattern may be seen in Fig. 87.

(B) PHYSICAL CHARACTERISTICS OF SPEECH

§ 5. Power of Speech Sounds

FLETCHER [39] gives the relative phonetic powers of the fundamental sounds as shown in Table 3. These data are composite values obtained from direct measurements with a calibrated microphone and amplifier system [69], combined with measurements of the relative amounts by

TABLE 3

PHONETIC POWERS OF SPEECH SOUNDS, RELATIVE TO THAT OF THE WEAKEST (FLETCHER)

ɔ	680	u	310	t ʃ	42	k	13
ɑ	600	ɪ	260	n	36	v	12
ʌ	510	i	220	dʒ	23	ð	11
æ	490	ɜˆ	210	ʒ	20	b	7
o	470	l	100	z	16	d	7
ʊ	460	ʃ	80	s	16	p	6
e	370	ŋ	73	t	15	f	5
ɛ	350	m	52	g	15	θ	1

which the spoken sounds need to be attenuated in order that they be mistaken 20 per cent of the time, or that they be just audible.

The vowels are the most powerful sounds and have a range of intensity of about 3 to 1. Diphthongs are not given in Table 3, but have about the same phonetic powers as the vowels which compose them.

According to FLETCHER [39], the average phonetic power radiated from the mouth of a speaker for the [θ] sound is 0.05 microwatts, and for the [ɔ] sound it is about 34 microwatts, representing a ratio of powers of 680, or an intensity range of 28 db for ordinary speaking. However, depending on their positions in a sentence and the emphasis desired, the vowel and consonant sounds may be produced with relative powers different from the averages given in the table. When this factor and the variations from talker to talker are taken into account, the phonetic powers encountered in conversational speech are found to vary from about 0.01 to 1000 microwatts of radiated power, a range of 50 db.

The peak powers of the speech sounds vary considerably with the type of voice. SACIA and BECK [69] show that representative values of peak power are in the order of 10 to 13 db greater than the phonetic power for vowel sounds.

Studies of the characteristic variations of intensity with time, and of the duration times of the vowel and consonant sounds have been reported by CRANDALL [70] and by STEINBERG [40].

§ 6. Spectra of Speech Sounds

Measuring devices which have been used in determining sound spectra include the HENRICI analyzer [55], photomechanical analyzers [71],

arrangements of parallel band-pass filters [72, 73, 91], heterodyne analyzers [74], acoustic diffraction spectrometers [75], tuned reed analyzers [76] and optical analyzers [77]. (Some of these are described in Chap. 17.) KOENIG, DUNN and LACY [78] and KERSTA [79] have described a sound spectrograph which portrays the formant positions and other significant characteristics of spoken sounds as shown in Fig. 87.

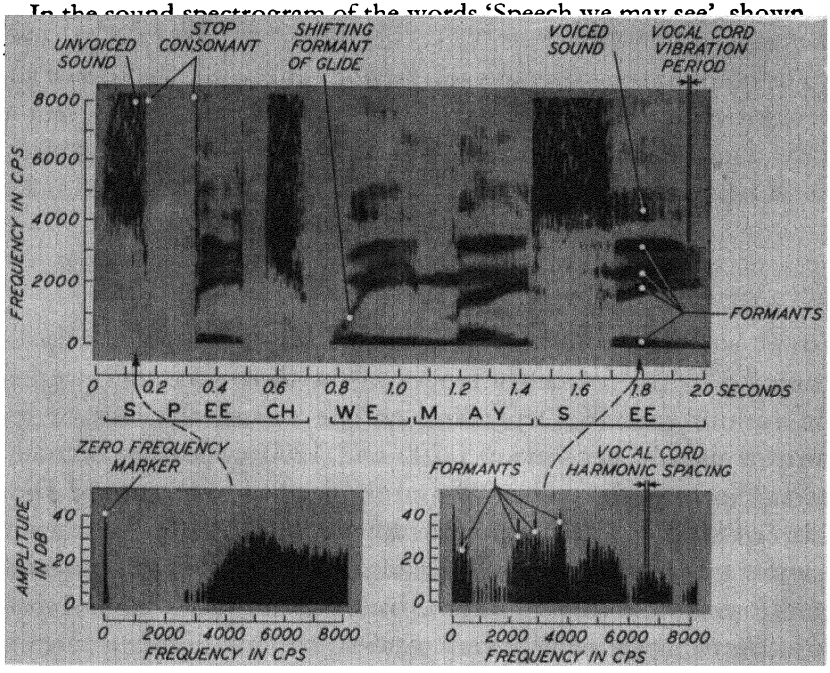


Fig. 87. Sound spectrogram and amplitude sections of the words, 'Speech we may see'.

to 8000 cycles per second on the vertical axis, and intensity by the darkness of the pattern marking. The spectrum at any given time may be portrayed by means of 'amplitude sections', two of which are shown in the figure, in which the relative amplitudes of the several components are plotted on a logarithmic scale against frequency. The two spectra were made at the points indicated by the arrows, for the sounds **s** and **ee**. In these sections, frequency is portrayed along the horizontal axis and amplitude is shown by the height of the pattern. In the section of the **ee** the fundamental and harmonics are resolved

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as individual spikes in the pattern. The spike at the extreme left edge of the pattern is introduced by the spectrograph as a marker of the zero frequency point of the scale. The fundamental frequency corresponds to the spacing in cycles per second between adjacent harmonics, and may be measured quite accurately on sustained sounds. The four principal formants are indicated on the section of the **ee**. The **s** sound, being a fricative consonant, does not show a regular harmonic structure in the section, nor pronounced formants in either the section or spectrogram, but instead exhibits a broad concentration of energy at frequencies above 4000 c/s.

For both the spectrograms and sections of Fig. 87, the speech signals were pre-equalized before analysis so that the amplitudes of frequency components at the upper end of the frequency range were increased 20 db relative to those below 500 c/s.

(i) *Spectra of Vowels*

Vowel sounds are generally characterized by three principal formants [67, 80]. The frequency ranges of the three formants overlap for the various sounds. Depending on the vowel, the first or lowest formant usually lies between 200 and 1200 cycles per second, the second formant between 600 and 3500 c/s, and the third formant between 1500 and 4500 c/s [81]. The lowest frequency formant generally has the greatest amplitude, with the second formant next, and the third formant the weakest. In the back vowels [**u, v, o, ɔ** and **ɑ**] the third formants are usually very weak so that those sounds are chiefly characterized by the two lowest formants. The formant frequencies and relative amplitudes vary considerably for any given sound from talker to talker, and to a less extent with successive utterances of the same sound by the same talker.

POTTER and STEINBERG [82] and PETERSON and BARNEY [81] have presented data on vowel spectra measured with the sound spectrograph. Frequencies of the first and second formants of vowels, for a single speaker, have been shown in Fig. 86. Table 4 summarizes averaged data from 76 voices reported by PETERSON and BARNEY on fundamental frequencies and frequency positions of the three principal formants for men, women and children, and also averaged relative formant amplitudes for several of the sustained vowel sounds.

TABLE 4

AVERAGE FUNDAMENTAL AND FORMANT FREQUENCIES AND AVERAGE RELATIVE AMPLITUDES OF FORMANTS OF SUSTAINED VOWELS.

		i	ɪ	ɛ	æ	ɑ	ɔ	ʊ	u	ʌ	ɜ˄
Fundamental Frequencies (c/s)											
	M	136	135	130	127	124	129	137	141	130	133
	W	235	232	223	210	212	216	232	231	221	218
	Ch.	272	269	260	251	256	263	276	274	261	261
Formant Frequencies (c/s)											
F_1	M	270	390	530	660	730	570	440	300	640	490
	W	310	430	610	860	850	590	470	370	760	500
	Ch.	370	530	690	1010	1030	680	560	430	850	560
F_2	M	2290	1990	1840	1720	1090	840	1020	870	1190	1350
	W	2790	2480	2330	2050	1220	920	1160	950	1400	1640
	Ch.	3220	2730	2610	2320	1370	1060	1410	1170	1590	1820
F_3	M	3010	2550	2480	2410	2440	2410	2240	2240	2390	1690
	W	3310	3070	2990	2850	2810	2710	2680	2670	2780	1960
	Ch.	3730	3600	3570	3320	3170	3180	3310	3260	3370	2160
Formant Amplitudes (db)											
	L_1	−4	−3	−2	−1	−1	0	−1	−3	−1	−5
	L_2	−24	−23	−17	−12	−5	−7	−12	−19	−10	−15
	L_3	−28	−27	−24	−22	−28	−34	−34	−43	−27	−20

The data given in Table 4 are averages of measurements for 33 men, 28 women and 15 children, all of whom were English speaking, with predominantly General American dialectal backgrounds. The average formant frequencies for men, women and children are given separately, showing large differences between averages for these three classes of talkers. The data on relative amplitudes of formants reported by PETERSON and BARNEY do not show large differences between classes, and have been averaged for all 76 talkers together in the table. The relative amplitudes are based on the amplitude of the first formant of [ɔ] as a reference. It was assumed that the total phonetic powers of the several sounds were as given in Table 3. Since the first formant is the most prominent, the relative amplitudes of the first formants of the several sounds would be nearly the same as the relative phonetic powers, except for slight corrections due to contributions of the higher formants to the total power. The average L_2 and L_3 values for the various vowels are also referred, on this basis, to the average amplitude of the first formant of [ɔ].

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The variations of observed values from the averages of Table 4 are frequently large. The standard deviations of measured values range from about 25 c/s for the first formant of [i] spoken by the 33 men, to about 250 c/s for the third formant of [ɔ] spoken by the 15 children. The variation of observed values of relative formant amplitudes are likewise large, with standard deviations of measured values of L_2 and L_3 ranging from 4 to 8 db for the various vowels.

The damping constants of the vocal cavities, as evidenced by the breadths of the formant peaks in the spectra of various vowel sounds, have been reported by several investigators. The damping constant μ_i in db per second is approximately equal to 27.3 times the bandwidth in c/s of the resonance, at points on the resonance curve 3 db below the peak response. LEWIS[41] and LEWIS and TUTHILL[42] determined damping constants ranging from 800 to 4000 or 5000 db per second in the spectra of sung vowels. STEINBERG reports damping constants of 1000 to 4000 db per second from measurements made on several spoken vowels[40]. TRENDELENBURG reports values for the logarithmic decrement Δ_i of resonant vibration of the mouth cavity when excited by a sharp impulse, varying from 0.10 for an [o] sound to 0.37 for an [u][83]. The logarithmic decrement Δ_i is approximately equal to the damping constant μ_i divided by 8.69 times the frequency f_0 of the peak of the resonance. When taken with the observed frequencies of resonance for the mouth cavity for the corresponding sounds, the decrements given by TRENDELENBURG give damping constants in the range from about 900 to 2000 db per second. Thus, if the vocal cavities were to be excited periodically at .005 second intervals, corresponding to a frequency of 200 cycles per second, the amplitude would be expected to decay to between 0.6 and 0.1 of its initial value during the interval. LEWIS reports that, in general, the lower values of damping constants are found on formants whose frequencies are less than 2000 c/s[41].

(ii) *Spectra of Semi-Vowels and Consonants*

The semi-vowel sounds [**l, r, m, n** and **ŋ**] show formant structures similar to those of the vowels in that their spectra contain relatively sharply defined regions of energy concentration. They are produced with vocal cord modulation of the breath stream and so have wave shapes which are periodic. The voiced fricative consonants [**v, ð, z**

and ʒ] show less clearly defined resonances in their spectra, with prominent concentrations of energy in the vicinity of the fundamental frequency. The corresponding unvoiced fricatives [f, θ, s, and ʃ] do not show the strong low-frequency component in their spectra, but otherwise have regions of energy concentration which are generally similar to their voiced cognates. Table 5 shows approximate characteristics of several semi-vowels and consonants.

TABLE 5

CHARACTERISTICS OF SPECTRA OF SEMI-VOWELS AND CONSONANTS

Sound	F_1	L_1	F_2	L_2	F_3	L_3	F_4	L_4
Semi-Vowels								
l	450	−8	1000	−21	2550	−20	2950	−24
r	500	−5	1350	−12	1850	−16	3500	−29
m	Fundamental	−11	1250	−21	2250	−23	2750	−30
n	„	−13	1450	−26	2300	−28	2750	−33
ŋ	„	−10	2350	—	2750	—	—	—
Fricative Consonants								
v	Fundamental	−18	1150	−37	2500	—	3650	—
ð	„	−18	1450	−27	2550	−45	—	—
z	„	−16	2000	−33	2700	−42	—	—
ʒ	„	−15	2150	−24	2650	−39	—	—
Stop Consonants								
b	Fundamental	−20	800	—	1350	—	—	—
d	„	−20	1700	—	2450	—	—	—
g	„	−17	Variable					

The columns headed F_1 to F_4 in Table 5 indicate the peak frequencies of the vocal resonances. They were obtained from data reported by KOPP and GREEN [67] and represent values for a single male speaker. The frequencies and relative amplitudes differ from voice to voice; the values presented in the table permit a comparison of the different sounds as spoken by one voice. In most of the sounds the frequency of the first formant was so nearly that of the fundamental that it was not clearly differentiated from the fundamental frequency by the analyzing means; in those cases it has been specified as the fundamental. The amplitudes in db given under L_1 of this table are derived from the amplitudes of the first formant as related to the first formant of the [ɔ] sound, as were the relative amplitude data of Table 4.

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The stop consonants [p, t and k], and their voiced cognates [b, d and g], are characterized by abrupt initial rises of energy, with their spectra showing components distributed throughout the voice frequency band for short periods of time. The regions of energy concentration are broad and poorly defined, and in cases of [k] and [g] they are quite variable in frequency, depending on the vowel associated with them [68].

(iii) *Transitional Effects in Speech Sounds*

The spectra of the various sounds vary to some extent depending on their positions in spoken words. The regions of energy concentration of vowels and consonants are influenced by the sounds preceding and succeeding them, being shifted up or down in frequency depending on the movements of the articulatory mechanisms required to produce the combinations of sounds [68, 84].

The frequency positions of the formants of vowel sounds which are observed in the steady state are sometimes not produced in connected speech, but instead some other set of formant positions is passed through in the transition between articulatory positions for preceding and succeeding sounds [84]. The nature of this movement, or 'gesture' has been described by POTTER and PETERSON in terms of the first and second formant frequencies [49]. The frequency positions of the first two formants alone appear to provide a major part of the essential information in the spectra of sustained vowels. SMITH [50] using Danish vowels, DELATTRE [51] with French vowels, and POTTER and PETERSON [49] using English vowels, showed that a measure of discrimination between vowels was obtained by plotting the frequencies of the second formants versus the frequencies of the first formants (see Fig. 86).

§ 7. Statistical Distributions of Power in Connected Speech

Because of the very irregular nature of connected speech, it is helpful in the engineering of communication circuits to have statistical averages of speech power and amplitude, measured over a wide sample of connected speech, and over many voices. The quantity usually measured is the *pressure* (actually excess pressure) at a point near the speaker. In point of time it may be an instantaneous value, or a peak,

average, or root-mean-square value in a fixed time interval. If the measuring point is located where the wave either has an approximately plane front or is spherically diverging, the pressure may be converted to intensity, or power per square centimeter of wave-front. The equation is $I = p^2/415.$, where I is intensity in microwatts per sq. cm, p is pressure in dynes per sq. cm, and the number 415 is the product of the characteristic impedance of air, the velocity of sound and a factor for reduction to the above units. If an interval of time is involved, p mist be the r.m.s. value in the interval, and I becomes the average intensity. By integrating intensity over the whole radiation area of the speaker (at a given distance), *total speech power* is obtained.

(i) *Distribution of Speech Pressures in Free Space*

The distribution of speech pressures in free space has been measured by DUNN and FARNSWORTH [85]. The measurements were r.m.s. pressures in 15-second intervals of connected speech, and were made in various frequency bands covering the range from 62.5 to 12000 cycles per second, as well as on the speech as a whole. The speech sample was chosen to have a representative variety of sounds and sound combinations, but only one speaker was used. For frequencies up to 5600 c/s, the pressure in front of the speaker is very nearly inversely proportional to distance (intensity inversely proportional to square of distance), and this holds down to very small distances provided the zero point be taken about 0.6 cm inside the lips *. For the frequencies above 5600 c/s, the mouth is more directive, so that for distances out to about 15 cm the pressure does not decrease so fast as is predicted by the inverse law.

The variation of speech pressure with direction in the horizontal plane, at a distance of 60 cm, is shown in Fig. 88 for several frequency bands. Since the scale of radii is inscribed in decibels, it holds for relative intensities as well as for pressures. The variation in the vertical plane, for the same distance and frequency bands, is shown in Fig. 89. For other distances, directions and frequencies, the reference [85] should be consulted. Other investigators [86, 87] have made similar, though less comprehensive measurements, with results in general agreement with those quoted.

* See Ref. 85, p. 189, for the effects of air puffs striking a close microphone, and how they may be avoided.

It may be noted from Fig. 88 and 89 that for angles up to about 75° in the horizontal plane, and between 90° upward and 45° downward in the vertical plane, there is no great divergence of the curves for the different frequency regions. Hence speech heard anywhere within these limits will sound natural and be almost as intelligible as

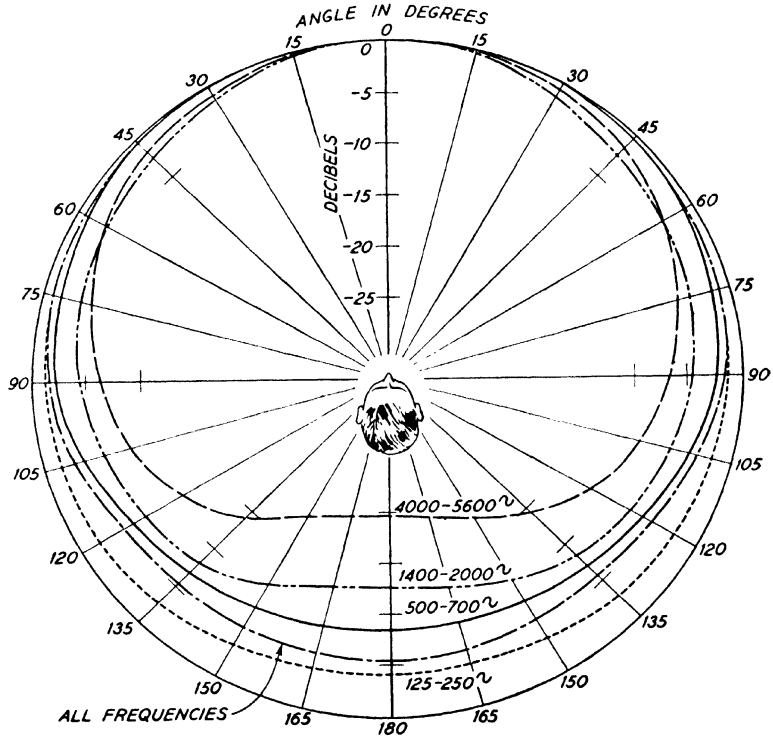


Fig. 88. Distributions of pressures in the horizontal plane, for several frequency bands in speech (after DUNN and FARNSWORTH).

when heard in front of the speaker. To the rear, however, the higher frequencies begin to fall off badly, resulting in poor intelligibility.

The spatial distributions which have been shown were measured in a room having walls treated to minimize reflections, and at points close enough to the speaker that the reflected energy was small compared with the direct.

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The average speech power radiated by a speaker is most accurately determined by integrating the intensities measured in the various directions. Fig. 90 shows the relation between the power so measured, and pressure at 30 cm in front of the speaker [85]. The curve covers

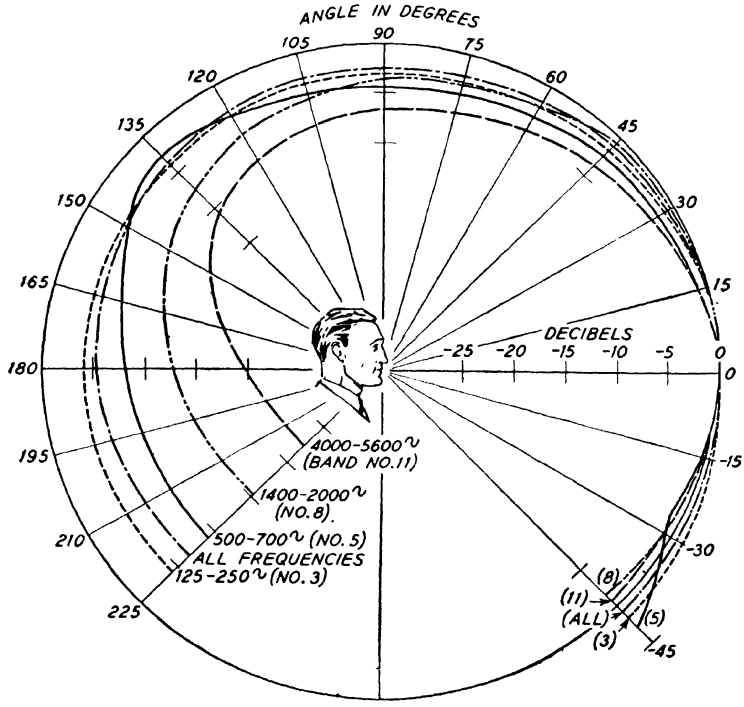


Fig. 89. Distributions of pressures in the vertical plane, for several frequency bands in speech (after DUNN and FARNSWORTH).

the various parts of the frequency range, while the relation for speech as a whole is 13.3 db. That is, if the total pressure at the 30 cm point is one dyne per sq. cm, then the total speech power radiated is 13.3 db above one microwatt. This is 2 db more than is obtained on an assumption of uniform radiation over a hemisphere. The 10 microwatts given by SIVIAN [88], from measurements by CRANDALL and MACKENZIE [89] and SACIA [90], must therefore be raised to 16 microwatts. DUNN and WHITE [91] found 34 microwatts for six men and

18 microwatts for five women. Perhaps the average total power of the average conversational speaker is about 20 microwatts. The corresponding pressure at 30 cm from the lips is slightly less than one dyne per sq. cm.

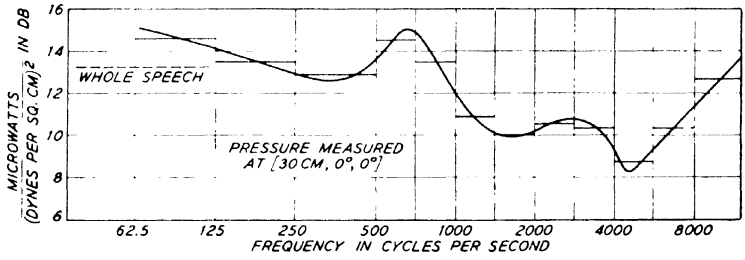


Fig. 90. Ratio of average speech power in the total field to mean-square of pressure at a single point, expressed in decibels (DUNN and FARNSWORTH).

Differences between individual speakers, however, are quite large. One group of eleven laboratory speakers [91] ranged from 8.1 to 91 microwatts. Another group of 146 speakers [88] covered a range of 30 db, as measured by a volume indicator. Since carbon microphones were involved in the latter tests, the measured range may have been greater than that of the speech used.

For departures from conversational levels, SIVIAN [88] reports 'confidential' speech as 10 db lower, and 'declamatory' speech 15 db higher. The total range of voiced connected speech is estimated by FLETCHER (*cf.* Ref. 39, p. 68) to be 0.1 to 1000 microwatts, with a soft whisper dropping to .001 microwatts. WOLF, STANLEY and SETTE [92] have found that a trained singer on a high note may reach an acoustic power of 1 watt. The peak power would be several db higher.

The powers for connected speech given above cover long intervals, and include the silent pauses. These silent parts have been estimated to occupy about 35% of the total time [90].

(ii) *Distribution of Speech Pressures at a Point*

The distributions of speech pressures at a point, in both time and frequency, were investigated by DUNN and WHITE [91]. All of their pressures were measured at a point 30 cm in front of the speaker's

lips. The continuous speech samples used were of a conversational type and level, and were in the English language as spoken by Americans. Fig. 91 compares three types of measurement on the whole band of frequencies, for a single speaker. One curve shows the percentages of instantaneous pressures exceeding any given level. The others are peak and r.m.s. pressures measured in one-eighth-second intervals, and show the percentages of such intervals during which the corresponding levels were found to be exceeded. The reference level for all is the long-interval r.m.s. pressure. The long average of instantaneous pressures is also shown, about 6 db lower than the r.m.s for this speaker.

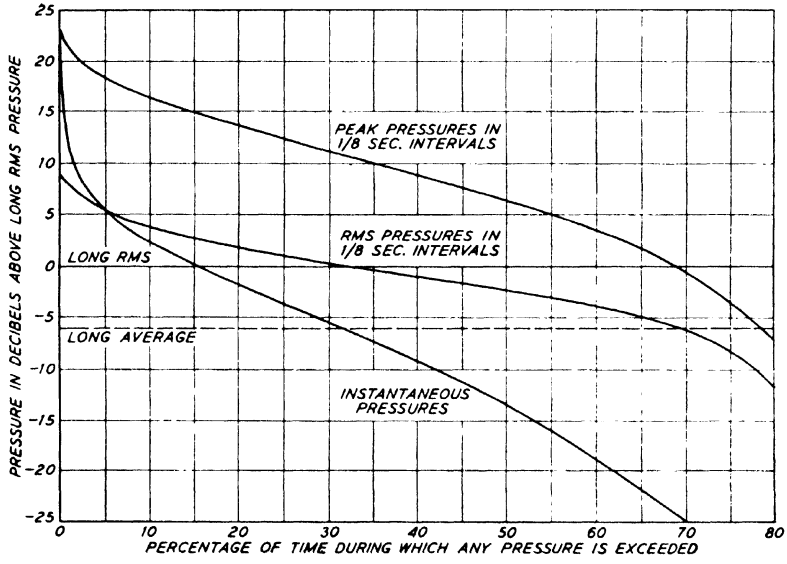


Fig. 91. Distributions of instantaneous pressures, and of peak and r.m.s. pressures in one-eighth-second intervals, from the speech of a single individual (DUNN and WHITE).

Distributions of r.m.s. pressures in one-eighth-second intervals, in both amplitude and frequency, are shown in Fig. 92 and 93, the former a composite from the voices of six men, the latter from five women. The measurements were in the bands of frequencies shown by the divisions of the frequency scale, and show the pressures exceeded, in the band, in certain percentages of the total number of intervals

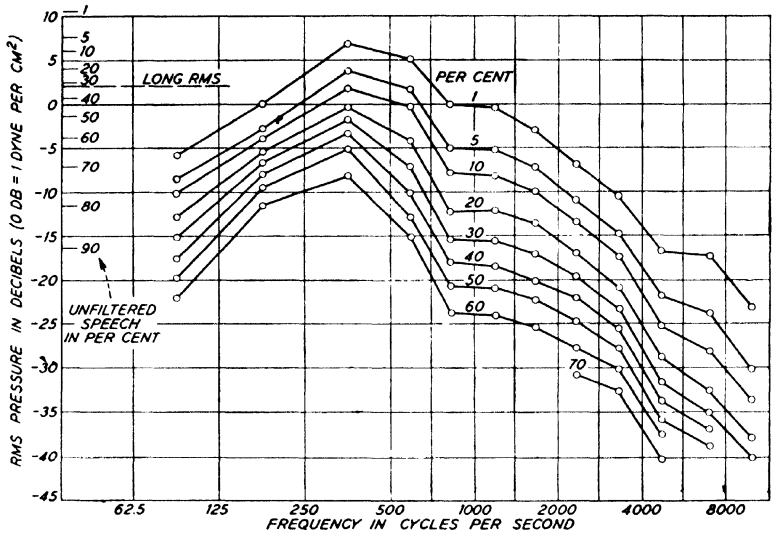


Fig. 92. R.m.s. pressures in one-eighth-second intervals, at 30 cm from the lips: composite from the speech of six men. Measurements were in bands indicated by the divisions of the frequency scale, and the percentages are those of intervals having r.m.s. pressures greater than the indicated ordinates (DUNN and WHITE).

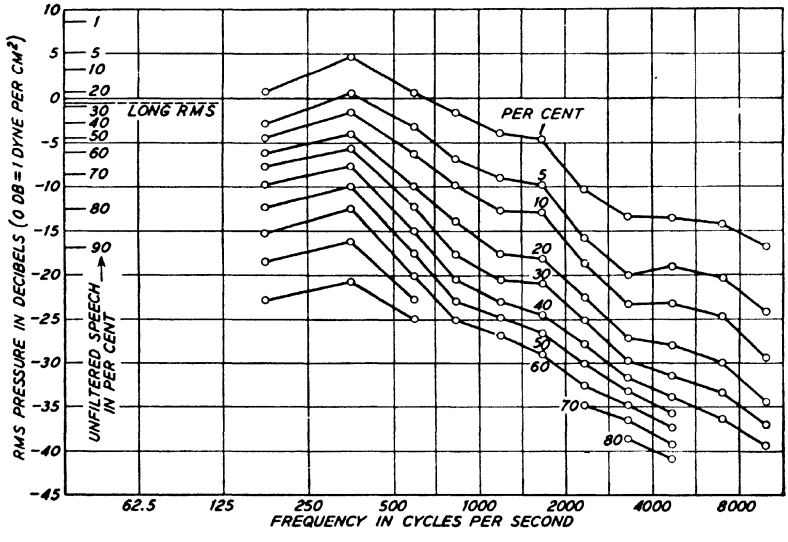


Fig. 93. R.m.s. pressures in one-eighth-second intervals: composite from the speech of five women (DUNN and WHITE).

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measured. Points for equal percentages in the different bands are connected by lines. The reference level in these figures is one dyne per sq. cm. The long-interval r.m.s. value is shown at the left, as well as a set of levels for the different percentages of short intervals in the speech as a whole (unfiltered).

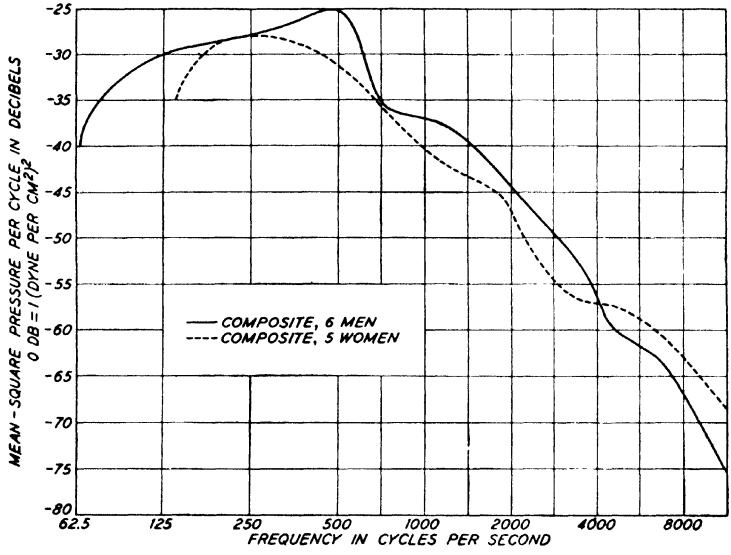


Fig. 94. Composite spectra of long-interval speech pressures, at 30 cm from the speakers' lips (DUNN and WHITE).

Fig. 94 shows the frequency distribution for r.m.s. pressures of connected speech as measured in 15-second intervals. In this case a reduction for bandwidth has been made, the square of measured pressure being divided by bandwidth in cycles per second. The smooth curves were drawn in such a way as to have the same integral of pressure-squared as the measured value, in each band. While these curves pertain to a conversational level of speech, the work of SIVIAN [88] shows that energy shifts toward the lower frequencies at low speech levels, and toward higher frequencies at high levels. Similar measurements to those of Fig. 94 were made by RUDMOSE *et al* [73]. Considering the fact that a more limited speech sample was used, their results for seven men are in good agreement with the curve for men in

Fig. 94. A paper by STEVENS, EGAN and MILLER [93] should also be consulted for methods used in this type of measurement.

The frequency distribution of average speech power may be obtained from Fig. 94 by applying the curve of Fig. 90.

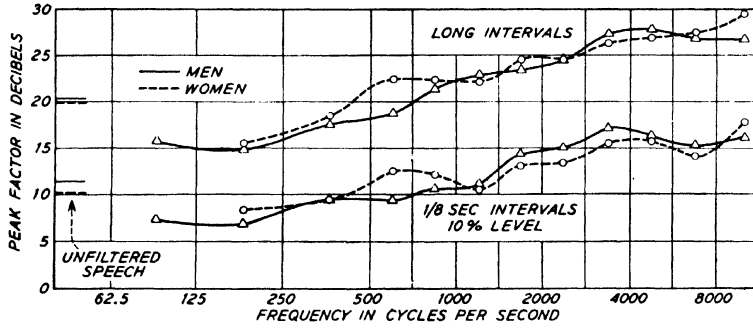


Fig. 95. Peak factors of speech at different frequencies, in one-eighth-second intervals and in long intervals (DUNN and WHITE).

Fig. 95 shows peak factors in connected speech, that is, the ratio of peak to r.m.s. pressure in specified intervals. The lower curves are for the upper 10% of one-eighth-second intervals. Applied to the r.m.s. levels of Fig. 92 and 93, at the corresponidng frequencies and percentage, they show the peak pressures found by DUNN and WHITE [91] to be exceeded in 10% of the intervals measured. The same factors apply within $\pm$ 2 db for percentages from 1% to 30%, for all frequencies below 5600 c/s, and from 1% to 20% for higher frequencies. At lower levels—those exceeded 40% or more of the time—the factors tend to become smaller. For 'unfiltered speech', the short interval factors shown (about 11 db) apply within $\pm$ 1 db to all percentages from 1% to 80%.

The upper curves of Fig. 95 are peak factors for long intervals (15 seconds) of speech. If the pressure-squared in Fig. 94 were multiplied by band width, then the application of these factors to the result would give the highest peaks of speech in the bands. Application of the unfiltered speech factor (about 20 db), to the 'long r.m.s.' levels of Figs. 92 and 93, gives the highest peaks of the unfiltered speech.

Another fact worthy of mention is the asymmetry of the peaks of

speech on the two sides of the wave. This asymmetry will be retained in an electrical system if the pick-up and transmission is reasonably free of non-linear, frequency, and phase distortion. The difference between opposite peaks varies with the sound, but HATHAWAY [94] has found it to reach 6 db or more. He also proposes to take advantage of it by proper poling of the microphone, in circuits which overload more quickly with one polarity than the other. DUNN and WHITE (*cf.* Ref. 91, p. 288) associated the higher peaks with the condensation phase of the acoustic wave.

(iii) *Speech Pressures at High Altitudes*

Speech pressures at high altitudes have been measured by CLARK *et al* [95]. They found that for a constant effort on the part of the speaker, the db change in r.m.s. speech pressure, in a given part of the frequency range, is approximately proportional to the logarithm of the ratio of barometric pressures. At a barometric pressure corresponding to an altitude of 35,000 feet above sea level, the acoustic level of vowel sounds fell off about 10 db, although varying somewhat with the vowel spoken and the frequency range investigated. On the other hand, fricative and stop consonants rose in level some 0 to 5 db, compared with sea level. The net effect is that the intelligibility of speech as a whole would change very little at high altitudes, if other factors (such as noise) were kept under control.

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CHAPTER 11

HEARING

For authorship, see p. 193

The translation of the pressure variations of sounds impinging on the ear into distinctive auditory sensations involves a succession of steps of varied character. These are effected by a number of specialized elements which, together, constitute the auditory system. The intent here is to present a broad picture of the overall system and some of the main features of the component elements. Further details are available in numerous books, including those listed in references 1 to 6.

The auditory system, shown in part in Fig. 96, includes the external and internal ears, separated by the middle ear, and a network of nerve fibres leading to and comprising the auditory sensory centres in the brain. The external and middle ears together amplify the pressure variations of air-borne sounds and transfer them to fluids within the internal ear. Here the pressure variations are sorted by the frequency-analyzer action of the cochlea, and encoded into pulses of electro-

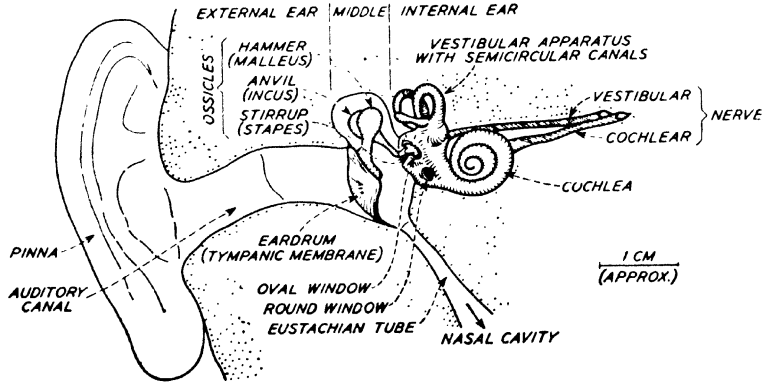


Fig. 96. Schematic section through the human ear (adapted [11] from BRÖDEL [6]).

chemical activity in the cochlear nerve fibres. The final effect of these pulses, acting through the neural system which extends to the brain, is the creation of patterns of excitation in the auditory receptive centres. The electrical manifestations of these patterns have been detected [7, 8] and mapped to a limited extent.

Like any communication system, the auditory system has an internal background of noise which limits the minimum level of external signals that can be detected. Contributing to it are the vibrations arising from circulation of the blood, breathing, muscle tremors and movements of the head. The ear is constructed, however, to respond far less efficiently to such vibrations than to air-borne sounds [9, 10, 11]. As a result, the ear can detect external sound pressures which are unbelievably small [12] (see Fig. 106).

(A) ANATOMY

§ 1. External and Middle Ears [1]

The external ear (Fig. 96) comprises the projecting pinna and the external auditory canal, which is terminated at the inner end by the eardrum. The walls of the canal are formed of cartilage and bone.

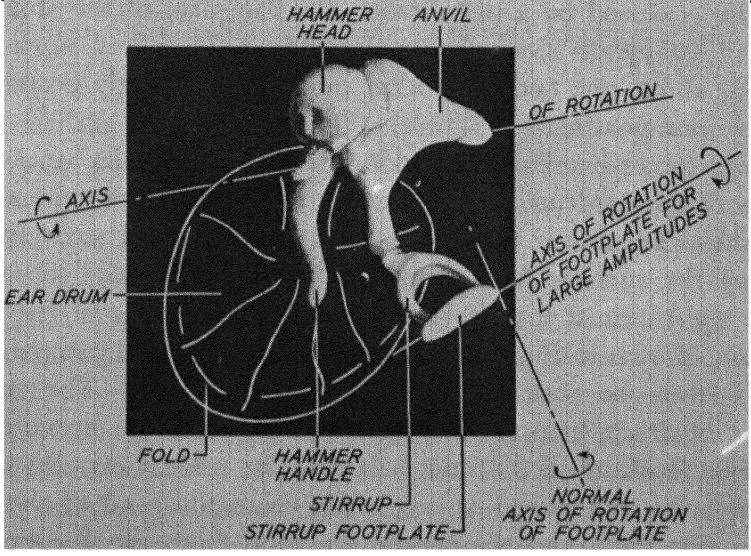


Fig. 97. View showing axes of rotation of the ear drum and ossicles (adapted [11] from BÉKÉSY).

The bony portion extends outward about 16 mm from the eardrum and the cartilaginous portion another 8 mm or so. The length, diameter and volume of the canal are, respectively, about 25 mm, 7 mm and 1 cm^3. The eardrum slants downward and inward and is shaped somewhat like a cone with an oval base. Its longest diameter is 9 to 10 mm and its shortest 8 to 9 mm. It is about 0.05 mm thick[10] and 85 mm^2 in area [13]. Its outside surface is deeply concave and the deepest point, the umbo, corresponds with the position of the lower end of the handle of the hammer (Fig. 97). The eardrum resembles a shallow rigid cone rather than a stretched diaphragm. A loose fold around the lower margin permits most of the drum to oscillate as a unit about an axis nearly tangential to its upper rim [10]. An exception is the small lax portion near the upper margin called the pars flaccida, which apparently is an important factor in the low frequency impedance of the drum (*cf.* Ref. 14, p. 42).

Behind the eardrum in the air-filled tympanic cavity of about 1.5 cm^3 in volume are the hammer, anvil and stirrup, linking eardrum to cochlea [15]. Approximately 23.0, 27.5 and 2.5 mg are the average masses of these ossicles [16]. In the inner wall of the tympanic cavity opposite the drum are the round and oval windows of the cochlea openings of about 2 and 3.2 mm^2, respectively [13, 16, 17]. The round window is closed by a membrane. The footplate of the stirrup and binding ligaments close the oval window. It moves normally by rocking about one end, but other motions, seemingly protective in purpose, are possible [9, 10, 18].

The hammer handle is embedded, lengthwise, in the inner surface tissues of the eardrum and its head is bound tightly to the anvil by ligaments. The lower tip of the anvil is also joined securely to the head of the stirrup. As a result, the entire system can vibrate as a single rigid unit at low frequencies [10]. The positions of the axes of rotation, shown in Fig. 97, are governed in part by ligaments which extend from the ossicles to the walls of the cavity. The centre of gravity of drum, hammer and anvil is close to their common axis of rotation, resulting in dynamic balance. The middle ear also includes two muscles, the stapedius, extending from the head of the stirrup to the side wall, and the tensor tympani, from the handle of the hammer to the wall of the cavity. Their contractions tend to move the stirrup in opposite directions [2].

§ 2. The Cochlea [1]

In appearance the cochlea somewhat resembles a snail shell (Fig. 96). It measures about 5 mm from apex to base and about 9 mm across the widest part of the base. Fig. 98 illustrates its internal construction.

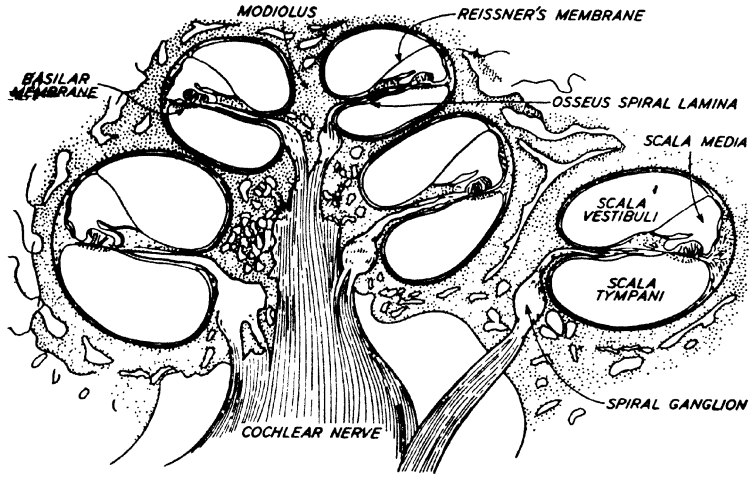


Fig. 98. Diagrammatic longitudinal section of the cochlea (adapted from GRAY's *Anatomy* [1]).

In essence, the cochlea is a double canal coiled up spirally for $2^3/_4$ turns about a bony axis, the modiolus, which is channelled to form a pathway for the cochlear nerve. Projecting outward from the modiolus is a bony shelf, the osseous spiral lamina, which follows the spiral windings of the canal and nearly subdivides it into two parts throughout its length. The subdivision is completed by the basilar membrane, which extends from the outer edge of the spiral lamina to the outer wall of the cochlea. Connecting these two subdivisions is the helicotrema, a small passageway at the apex of the modiolus. Its cross-sectional area [17] is about 0.25 mm^2. The position of the helicotrema is indicated at the extreme right of the upper part of Fig. 99, which portrays the cochlea as it might appear if uncoiled. The average length [20] of the basilar membrane is about 31.5 mm but varies over a range of some 10 mm. The width [21] of the membrane is about 0.10 mm

close to the basal end and reaches a maximum of about 0.50 mm not far from the apical end. The two main divisions of the canal are called the scala vestibuli and scala tympani. The round window, in the wall of the latter, and the oval window, in the wall of the former, lie close together at the basal ends of these ducts. The dimensions of the windows have been given previously. Dimensions of the canals are complex, but are available in numerous publications [2, 17, 22].

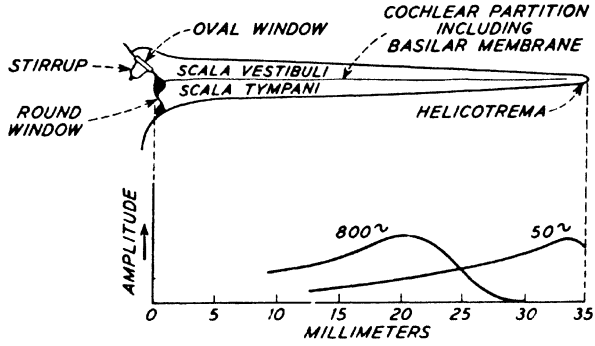


Fig. 99. Deflection of the basilar membrane (adapted [11] from BÉKÉSY [34]).

In addition to the basilar membrane, a second membrane (REISSNER'S) extends from the spiral lamina to the outer wall of the cochlea, which it meets just above the outer edge of the basilar membrane. The space between forms a third canal known as the scala media or cochlear duct. Although visible in Fig. 98, the details of this duct are more easily seen in Fig. 100. Filled with a viscous fluid, the endolymph, and having two flexible sides, the scala media is capable of movement between the other two large ducts. The latter are filled with perilymph, a thin, watery fluid which transmits to the scala media the pressures produced by movements of the stirrup. The basilar membrane, forming one pliable side of the scala media, supports the spiral organ of CORTI, which consists of a large assemblage of terminal nerve elements. The first elements of the neural system are the inner and outer hair cells. The inner cells, arranged in a single row along the inner edge of the basilar membrane, number about 3500 and have diameter of 12 microns [2]. (With a basilar membrane length of 35 mm or less, one of these figures appears slightly excessive). The outer

hair cells, numbering some 12,000 [1], to 20,000 [2], are arranged in three or four rows along the basilar membrane. Their diameter is about 8 microns [2].

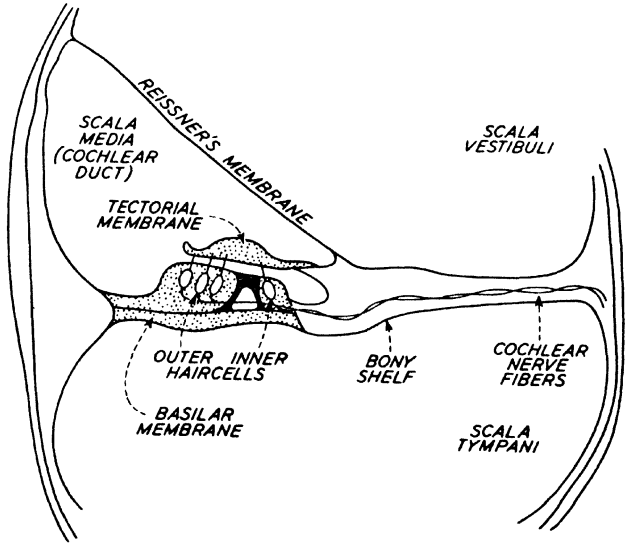


Fig. 100. Schematic section through the human cochlea (adapted [11] from STEVENS and DAVIS [2]).

Like all living cells, the hair cells are believed to be electrically polarized. Each cell is surmounted by hair-like processes whose upper ends are thought to be embedded in the tectorial membrane, another membrane just above the organ of CORTI (Fig. 100). As a result of this coupling, it seems probable that forced vibration of these parts can develop forces tending to stretch, compress or otherwise distort the hair cells and thus affect their polarization. These cells play an important part in starting the pulses of electrical activity which have been observed in the auditory nerve fibres [23] between the cochlea and the brain. There are several different types of fibres, some having many branches connecting a number of hair cells to the same main fibre [24]. From the hair cells, the nerve fibres pass through small canals in the osseus spiral lamina and enter the spiral ganglion of the cochlea, where their cell bodies are located (Fig. 98). The fibres then extend

downward along the axis of the spiral, where they become grouped to form the cochlear branch of the auditory nerve.

§ 3. Nerve System [25]

Shortly after emerging from the base of the cochlea, each nerve fibre of the cochlear nerve divides into two branches which terminate in two groups of cells called the ventral and dorsal cochlear nuclei. The dorsal cochlear nucleus lies in the pathways taken by the pulsing signals on their way toward the auditory receptive centre of the brain. The other nucleus is involved in the reflex aspects of hearing, but may have other functions as well. The nerve fibres entering the cochlear nuclei are called first order neurons. Those leaving it, to form the second link in the transmission chain, are second order neurons. The cochlear nuclei are junction points in which the first order neurons terminate in small attachments, called synapses, to the cell bodies of the second order neurons. The synaptic junctions transmit pulses in only one direction, *i.e.* from first to second order neurons. Even in this direction nerve pulses do not pass through the junction. They exert, instead, influences on the cell body of the second neuron which may or may not start new impulses. Several sets of such junctions intervene between the cochlea and the auditory receptive centres. Also, there are cross pathways from each cochlea to both sides of the brain [8, 26]. The auditory tract is a bewilderingly complex system [27], but many systematic features have been discovered. For example, the nerve fibres, from the different parts of the basilar membrane, terminate in an orderly fashion in correspondingly different parts of the cochlear nuclei [28]. It is believed that similar orderly arrangements extend throughout the entire auditory tract and enable stimuli in different parts of the cochlea to project their effects to corresponding different parts of the auditory receptive centres.

§ 4. Auditory Receptive Centres [25]

The surfaces of the cerebral hemispheres contain many irregular eminences, called gyri, separated from each other by deep depressions or fissures. One of these gyri, the anterior transverse temporal gyrus, contains the primary auditory receptive centre. Artificial stimulation

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of this region, if exposed, causes buzzing or roaring sounds to be heard [25]. Covering each hemisphere and forming the outer shell of each gyrus is a thin gray lamina called the cerebral cortex. In the auditory region this lamina is composed of alternate light and dark bands or sheets running parallel to the surface. The dark bands contain tremendous numbers of cells. The light bands and the internal light core contain many nerve fibres, which include the incoming fibres from the auditory tract and fibres interconnecting different parts of the cortex.

(B) PHYSICAL PROPERTIES

§ 5. Pressure Transformations

Sound pressures in the cochlea may exceed those to which the ear is exposed by as much as 100 to 1 at some frequencies. The increase results from several factors such as diffraction by the head, reflection at the closed end of the auditory canal, leverage of the ossicles and the transformer-like action of the relatively large eardrum and tiny oval window. The relation between open air sound pressure and pressure at the drum has been measured by WIENER and ROSS [29], and that between pressures at the drum and in the cochlea by BÉKÉSY [13]. The theoretical aspects of the latter transformation have been studied by ESSER [30].

§ 6. Acoustic Impedance of Ear

Table 1 shows mean values of the acoustic impedance of the ear obtained by METZ [14]. The measurements were made at a distance of 14–18 mm from the umbo of the eardrum, *i.e.* about one-third of the way in from the entrance of the auditory canal. The data for individual ears varied widely from these mean values.

The results of some earlier measurements were also reviewed in the same report. Other early measurements are those of INGLIS, GRAY and JENKINS who, employing a wide range of frequencies, determined the acoustic impedance of the ear as seen through the aperture of a receiver cap held to the ear and also the impedance looking into the entrance of the auditory canal [31]. Their data on the latter condition

TABLE 1

ACOUSTIC IMPEDANCE OF AUDITORY CANAL AS SEEN FROM POSITION ABOUT 16 MM FROM TYMPANIC MEMBRANE
(after OTTO METZ [14])

Frequency Cycles per sec.	Number of Ears	Mean Value of Impedance - Acoustic Ohms * Resistance Component	Reactance Component
384	41	40	−143
512	57	41	−117
768	55	25	− 61
1145	50	28	− 47

* The values obtained by METZ have been converted to acoustic ohms per sq. cm.

and those of METZ show that the magnitude of the impedance is dominated by negative reactance at the lower frequencies.

§ 7. Natural Period and Damping

The natural period of vibration of the intact ear lies within the range of 500 to 1500 cycles per sec. (*cf.* Ref. 2, p. 262). With an occasional exception [32, 33], it is generally agreed that the ear as a whole is highly, but not quite critically, damped. The isolated cochlea was studied by BÉKÉSY [34]. He noted, after removal of a stimulus, that the amplitude of each oscillation of the basilar membrane fell to about 1/5th that of the immediately preceding oscillation, indicating a logarithmic decrement of about 1.6 per cycle, about one-fourth of the value (2π) necessary for critical damping.

§ 8. Frequency Localization

Sounds impressed upon the ear cause the basilar membrane to oscillate between the scala vestibuli and the scala tympani. The oscillations of this membrane are not, in general, of equal amplitude along its length. For each single frequency tone there is a unique position along the membrane where the displacement is greatest (Fig. 99) [34]. These positions lie near the stapes at high frequencies and shift systematically toward the helicotrema as the frequency is lowered. Thus the stimulating action of different frequency tones, especially at low levels,

tends to be localized on different groups of hair cells and nerve fibres along the length of the basilar membrane [23]. From the hair cells to the cochlear nuclei these fibres maintain a systematic spatial arrangement which is believed to continue all the way through to the auditory receptive centres [28]. WOOLSEY and WALZL [7] have shown that electrical stimulation, by fine electrodes, of only a few auditory nerve fibres at a time in an exposed cochlea, excites a vertical band of tissue in the auditory centres of a cat. Stimulation of fibres near the stapes excited the most forward parts of the centres, while stimulation of fibres near the helicotrema projected to the posterior parts. Since the nervous system is preceded by a mechanical frequency analyzer in the cochlea, a similar localization in the cortex might be expected for tones applied to the ear as acoustic stimuli. In this case, however, the spatial patterns are broad patterns [7, 8]. This is consistent with BÉKÉSY's findings that the vibrational patterns of the basilar membrane are not sharply localized (Fig. 99). Some tests have also shown that each cerebral hemisphere may have at least two adjacent regions in which auditory patterns occur [7, 8, 35].

§ 9. Travelling Waves in Cochlea

With a low frequency tone applied to the oval window, BÉKÉSY observed a phase difference between the oscillations of the window and basilar membrane [36]. This difference, tiny for points on the membrane near the window, increased with distance, slowly at first and then more rapidly as the point of observation approached the position of maximum vibration. These observations signify the presence of a wave travelling from the stapes toward the helicotrema, with a decreasing phase velocity as the wave advances. Transit time measurements, using an electric spark as an impulse, [34] gave the results shown in Table 2. They indicate a speed of propagation ranging from several hundred to a few meters per second.

§ 10. The Cochlear Response [3]

Electric potentials are generated within the cochlea when sound waves are impressed upon the ear. A favourable location for the detecting electrode is the moist membrane of the round window. The

TABLE 2

TRANSIT TIME OF IMPULSE TRAVELLING ALONG COCHLEAR DUCT

(G. VON BÉKÉSY [34])

Distance from Stapes – mm	Time from Stapes to Point of Measurement – Milliseconds
20	0.13
23	0.56
25	1.00
28	1.49
30	2.00
33.5	3.89

other electrode is often placed on the muscles at the back of the neck. The potential difference between these points contains two components, one of which reproduces the wave-forms of the applied sounds. This one is called the cochlear response and its magnitude is proportional to that of the applied stimulus. WEVER (*cf.* Ref. 3, p. 142) ascribes the cochlear response to changes in the polarization of hair cells when the cells are distorted by movement of the cochlear fluids. The other component of the potentials detected is called the action potential and distinctive features which enable it to be distinguished from the cochlear potential have been found [2, 3, 37]. The wave form of the action potential bears no resemblance to that of the applied signal, and the intensity relationship is non-linear.

§ 11. Action Potentials [2]

A nerve fibre may be thought of as a semi-permeable membrane enclosing a solution of electrolytes and having selective properties with regard to the passage of ions. One characteristic property of nerve fibres is a difference of potential between external surface and interior. However, when a nerve is suitably stimulated, depolarization occurs and gives rise to a self-propagating pulse of negative activity, known as an action potential. Among the many reviews of the facts and theories of nerve excitation is that of KATZ [38].

During the life of a pulse, roughly 1 millisecond, re-excitation of the nerve is impossible. This 'absolute refractory' period is followed by a 'relative refractory' period, during which the nerve can be re-excited, but only by a stimulus larger than the normal minimum. A

total interval of about 20 milliseconds elapses before the nerve is restored to its original condition. These properties limit the pulsing rate of the individual fibres. Their pulsing rate is also sensitive to the magnitude of the stimulus. This is indicated by Fig. 101, which shows the rates observed by GALAMBOS and DAVIS in the auditory tract of cats, in response to acoustic stimulation with single frequency tones [23]. The pulses were detected with micro-electrodes believed to be in contact with cell bodies of second order neurons. It is seen that the pulsing rate can vary from several to several hundred per second as the magnitude of the stimulus is varied over a range of 30–40 db.

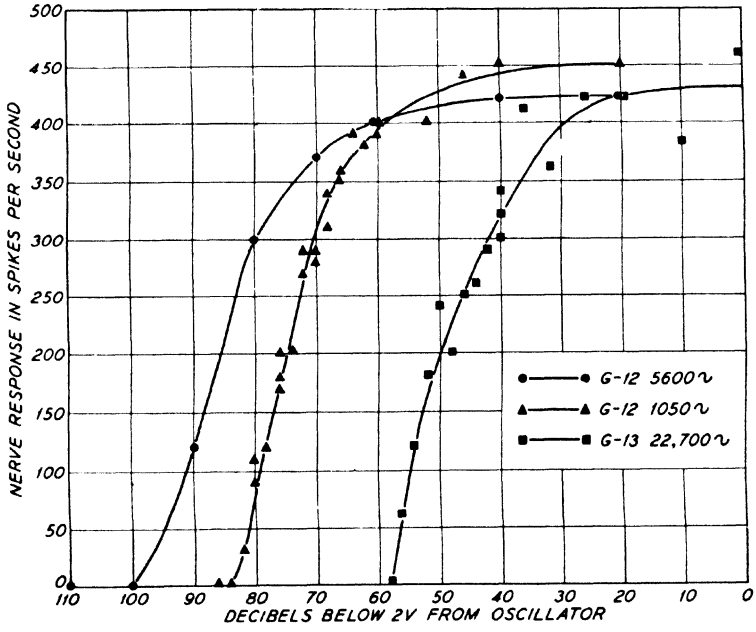


Fig. 101. Graph showing increase of discharge rate with increase of sound intensity for three different fibres of auditory nerve of cat (from GALAMBOS and DAVIS [23]).

In the above tests, each different unit on which the probe was placed was responsive to a wide range of frequencies at high levels of stimulation. At sufficiently low levels, the units responded only to frequencies within a narrow band of several cycles per second whose frequency position varied from one unit to another. The behaviour of the

individual units is illustrated by Fig. 102, which applies to a unit that responded best to a frequency of 7000 c/s. These observations, important to auditory theory, seem to be qualitatively consistent with what is known about the analyzing properties of the cochlea and the arrangements of the nerve fibres along the basilar membrane. Also

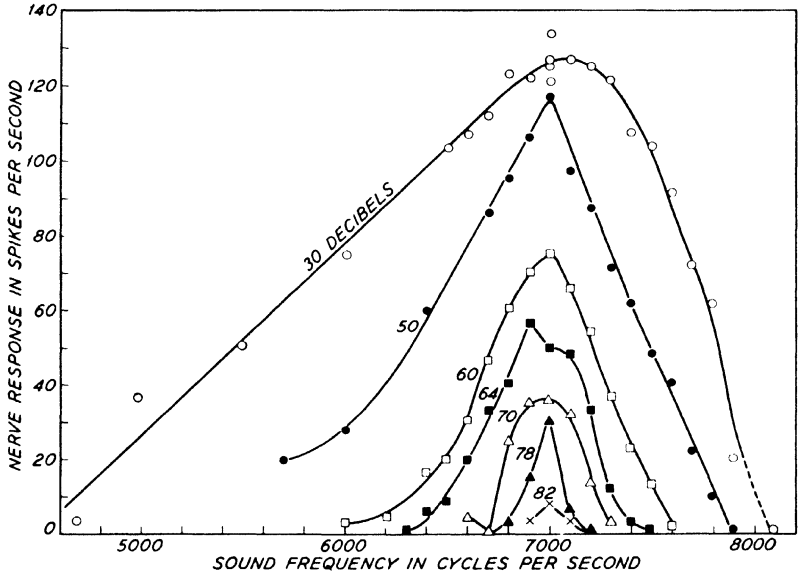


Fig. 102. Iso-intensity contours for a single nerve fibre, one which responds best to a frequency of 7000 cycles. Each contour shows how nerve responds to a fixed level of stimulus as frequency is varied. The number on each contour represents the level of the stimulus in db below a level arbitrarily designated as 0 db (from GALAMBOS and DAVIS [23]).

important is a further observation that the successive pulses are always separated by an integral number of cycles per second of the stimulating sound wave [23]. The variability of this coincidence amounted to about 0.25 millisecond.

§ 12. Equivalent Circuit of the Ear

Structures containing mechanical and acoustical elements are often represented by equivalent electrical circuits to facilitate the study of

their transmission properties. Although sufficient information is not yet available for specifying the nature and arrangements of all the analogous electrical elements of such a circuit to represent the ear, a number of approximate representations have been published. An early representation of the cochlea was given by WEGEL and LANE [39], a recent one by PETERSON and BOGERT [40]. OLSON [41] gave a circuit which includes the outer and middle ears in addition to the cochlea.

(C) CHARACTERISTICS OF HEARING

Some of the important characteristics of the hearing system are revealed by data which involve both physical and subjective tests. That is, the measurements include not only the usual physical types such as pressure and wave-form measurements, but also subjective tests in which an observer responds to loudness, pitch, or other aspects of the sensations experienced. Both physical and subjective measurements are important, and inadequate treatment of either may result in a loss of precision or misinterpretation of results.

Methods of measuring sound pressure accurately have been im-

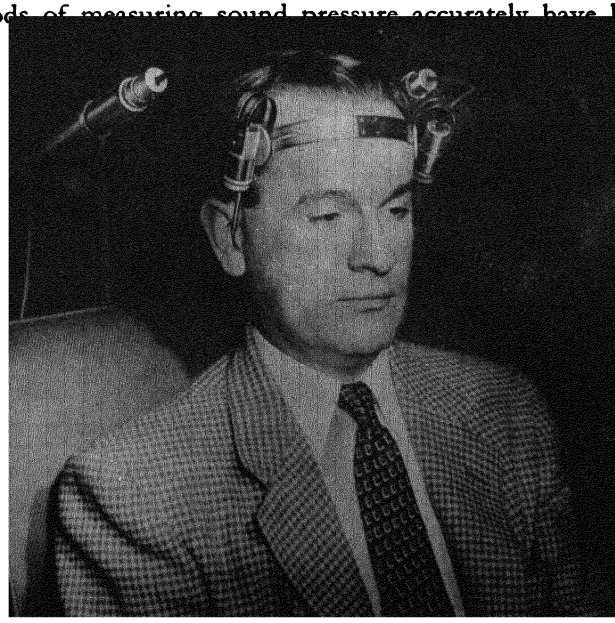


Fig. 103. Probe tube microphones for measuring sound pressures in the ear canals.

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proved in recent years, but, owning to diffraction effects, specifying the levels that finally reach an observer's ears is still a difficult problem [42]. Small probe tube microphones that have been designed to measure the sound pressure directly in the ear canal of an observer appear to offer one of the best solutions [43] (Fig. 103).

When the experiments involve the listener's responses to loudness, pitch, etc. the psychometric method becomes an important factor in the measurement. It is well recognized that precision in subjective measurements is dependent upon obtaining numerous observations in a carefully controlled psychological environment. A properly designed subjective test will enable the experimenter to collect large numbers of relatively independent responses in a short period, without unduly fatiguing an observer.

A number of different testing techniques [44], each having certain advantages, are in common use. For example, the method of 'average error', the method of 'constant stimuli' and the method of 'limits' are often used in auditory tests as well as various combinations and modifications of these techniques. In the method of average error, the observer can control the sound he hears and adjust the controls until he is satisfied that his task, such as a loudness balance or a threshold test, has been accomplished satisfactorily. In the 'constant stimuli' method the observer has no control and is given no part in the selection of the stimuli. He simply listens and responds 'louder', 'softer', or whatever is necessary to comply with instructions. In the method of 'limits', the stimuli are not under control of the listener, but are usually presen ted in an ascending or descending series and he responds at one point in the sequence. For instance, in an acuity test, the level of the signal would be diminished until the listener indicates he no longer hears it. For the ascending series the signal is started below threshold and increased until it is perceived.

The reader should be aware of the fact that the data for most of the subjective characteristics tabulated in the following pages depend to some extent upon the testing method. For example, results of clinical audiometric measurements of threshold acuity differ by 5 or 10 decibels from results obtained by the more carefully controlled procedures used in laboratory testing. Where a direct comparison of hearing data is required, a standardized test should be used.

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§ 13. Absolute Sensitivity

The normal ear of a person in his early twenties responds to tones covering a frequency range of approximately ten octaves, that is, from about 20 to 20,000 c/s. Near the extremes of the range where the ear is least sensitive, the sound pressure necessary to elicit a response is many times greater than is required between 2000 and 4000 c/s, the region of highest sensitivity. In this latter region the minimum audible field pressure for the acute ears of an experienced observer is less than 0.0002 dyne per sq. cm (10^{-16} watts per sq. cm).

(i) *Pressure Measurements*

Ear sensitivity is usually given in terms of the minimum audible ear canal pressure (M.A.C.) or the minimum audible field pressure (M.A.F.). The M.A.F. measurement is made on a plane progressive wave in a free field at the point where the observer's head will be positioned. The intensity of the signal reaching the auditory system depends upon the orientation of the head with respect to the source, and upon the frequency of the tone. These relationships have been reported by WIENER and ROSS [29]. An observer was placed in a known sound field and measurements made, by means of a search-tube microphone, of the sound pressure at the entrance to the ear canal and near the ear

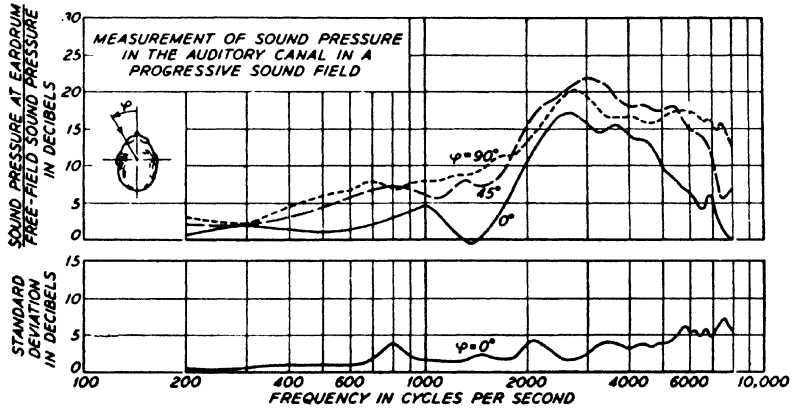


Fig. 104. Sound pressure at ear drum compared with sound pressure in a free field (WIENER and ROSS [29]).

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drum. The difference between the pressure level in decibels at the drum and the level in a free field is plotted as the ordinate of Fig. 104. The pressure at the drum can be as much as 20 db greater at some frequencies than the pressure in a free field. The difference between the pressure level at the drum and the level at the entrance to the ear

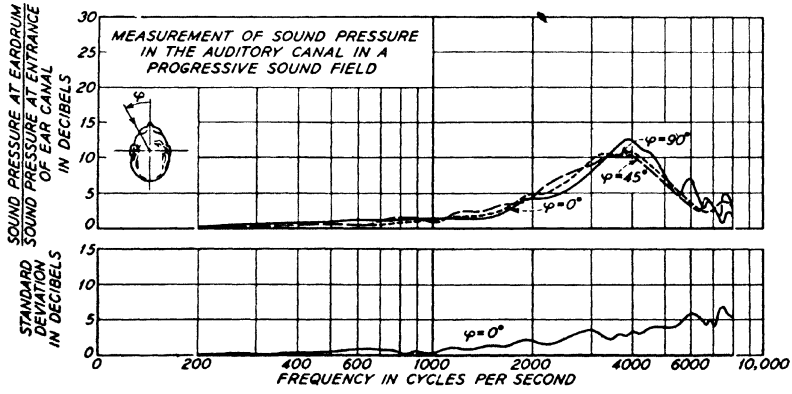


Fig. 105. Sound pressure at ear drum compared with sound pressure at entrance of ear canal (WIENER and ROSS [29]).

canal is plotted as the ordinate of Fig. 105. The pressure at the drum may be as much as 10 db higher than at the entrance to the ear canal. Obviously any data on minimum audible sound pressures should

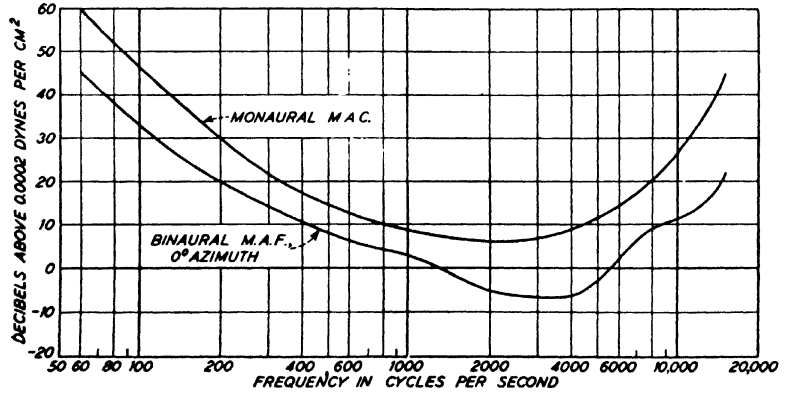


Fig. 106. Minimum audible ear canal (M.A.C.) sound pressure and minimum audible field (M.A.F.) sound pressure (SIVIAN and WHITE [13]).

specify where the pressure was measured and how the observer was oriented with respect to the source.

(ii) *Minimum Audible Ear Canal Pressures*

The data on minimum audible ear canal pressures are numerous, but represent such a variety of testing conditions that it is difficult to compare them. In some cases pressures were measured near the ear drum, in others at the entrance to the ear canal, and in still others the pressure developed by the transducer was measured when it was energized with the threshold voltage level but placed on a hard-walled cavity simulating the volume of the ear canal. After studying all available data, SIVIAN and WHITE [12] derived the average values shown by the monaural curve of Fig. 106. The ordinate is the monaural minimum audible pressure level in db from 0.0002 dyne per cm^2.

(iii) *Minimum Audible Field Pressure*

Tests on minimum audible field pressures made by a number of experimenters also have been collected and compared by SIVIAN and WHITE [12]. The binaural curve of Fig. 106 shows the average values which they derived. The ordinate is the binaural minimum audible free field pressure level when the observers face directly towards the source of sound.

(iv) *Minimum Audible Field Pressure for Speech*

When the long time r.m.s. pressure level of undistorted speech is about 8 db above .0002 dyne per sq. cm the loud speech sounds will be at threshold for trained observers with good hearing (*cf.* Ref. 45, p. 109).

(v) *Results of Hearing Surveys*

A number of hearing surveys have been made to determine the extent and degree of hearing impairment in the general population. Those made in the United States, for example, include the U.S. Public Health Service [46] survey in 1936, the World's Fair-Bell Telephone Laboratories [47] survey in 1939 and the San Diego Fair [48] Survey of 1948.

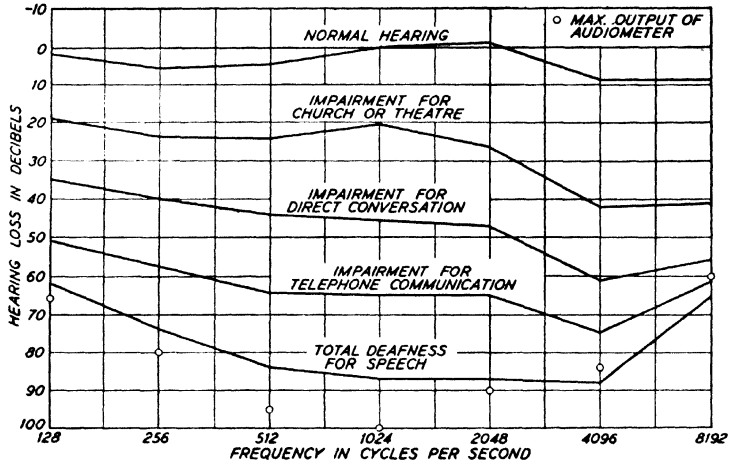


Fig. 107. Average audiograms for groups reporting different degrees of hearing (adapted [47] from BEASLEY [46]).

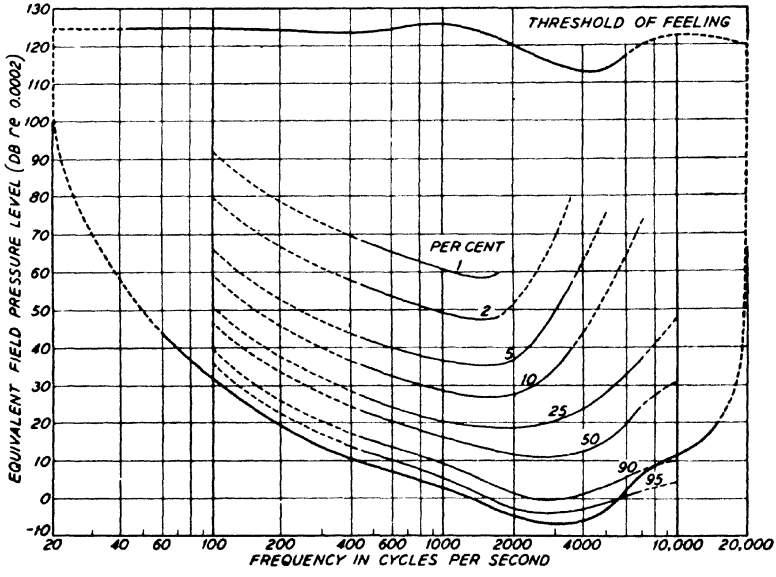


Fig. 108. Hearing contours showing the percentage of a population that require sound pressures above the contour to hear pure tones (STEINBERG, MONTGOMERY and GARDNER [47]).

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Fig. 107 shows the relation between hearing loss and the remaining capacity for interpreting everyday speech, as determined by the U.S.P.H.S. survey. This figure shows that a 25 db loss is usually attended by inability or difficulty in understanding speech in a church or theatre; that with a 45 db loss it is difficult to hear direct conversation; that with a 65 db loss it is difficult to hear over the telephone;

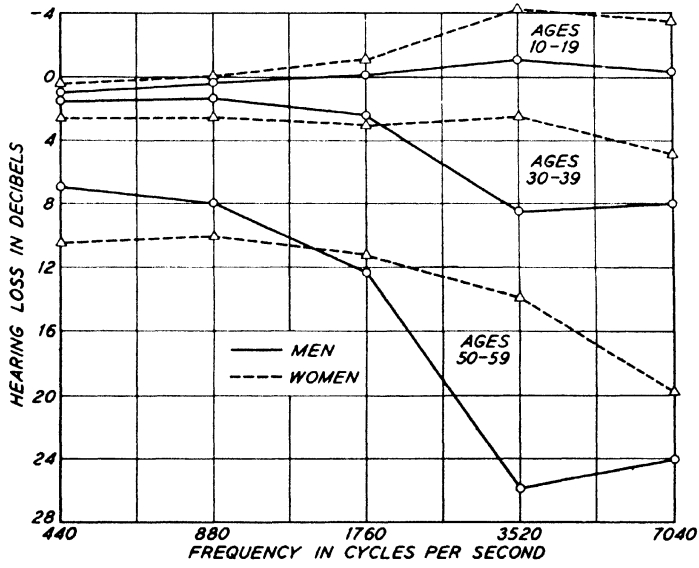


Fig. 109. Effect of age on the average hearing in a population (STEINBERG, MONTGOMERY and GARDNER [47]).

and that 85 db loss means complete inability to understand speech unless lip reading or other aid is used.

Fig. 108 shows the distribution of hearing loss for the 10–59 age group, as found by the World's Fair hearing tests. The bottom curve of this figure is the M.A.C. curve of Fig. 106 and the upper curve corresponds to a level in the region of feeling [49] threshold or total loss of hearing.

Fig. 109, also taken from the World's Fair Survey results, shows how age and sex affect hearing impairment. Women tend to show a greater loss at low frequencies while men show greater loss at high frequencies.

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§ 14. Differential Thresholds

An important fundamental problem in the study of the auditory sensory system is the determination of a just perceptible change in a sound. Most investigators have used relatively simple stimuli such as pure tones, and simple changes such as intensity of frequency, in measuring difference limens (D.L.), sometimes referred to as 'just noticeable differences' (J.N.D.).

(i) *Pure Tones – Difference Limens with Respect to Intensity*

The most extensive measurements have been made by RIESZ [50], whose results are shown in Fig. 110. The ordinate is the just noticeable difference in level in db.

The abscissa is frequency and the parameter is the level in db above threshold (sensation level). The stimulus used by RIESZ consisted of two tones, about three cycles apart in frequency, which were sounded simultaneously, and therefore gave rise to continuous loudness fluctuations in the observer's ear in the manner of beats. The level of one of the tones was increased from inaudibility to the point where

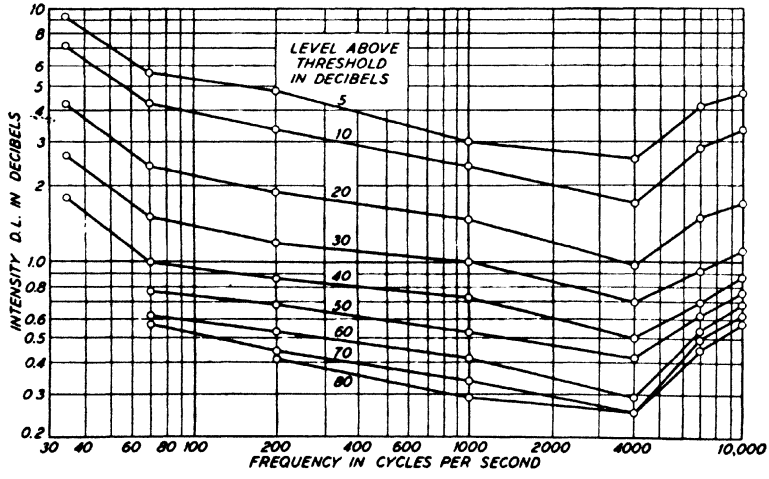


Fig. 110. Intensity difference limens for pure tones at different levels above threshold (from RIESZ [50]).

the observer could just detect the loudness fluctuations. Then the upper and lower limits, p_1 and p_2, of the sound pressure fluctuations for this condition were calculated and the difference limen in db was found from the relation: D.L. $= 20 \log_{10} p_1/p_2$.

More recent measurements by DIMMICK and OLSON [51], using tones of 1500 milliseconds duration, show a relatively constant difference limen in db for sensation levels above about 30 or 40 db. This is in disagreement with RIESZ's data which show substantial decreases in the difference limen at high levels (see Fig. 110). Moreover, the difference limens measured by DIMMICK and OLSON are substantially larger than those obtained by RIESZ. The differences in the results of these two studies and others reported by KNUDSEN [52], CHURCHER, KING and DAVIES [53], and HUGHES [54] are not easily explainable in terms of observer variability alone. Differences in the testing method and in presentation of the signals, especially their temporal sequence, exist and are matters of importance. MONTGOMERY [55] has made an attempt to assess some of the factors, but many of them are as yet imperfectly understood. The effect of an adapting tone just prior to an intensity D.L. test was studied by RAWDON-SMITH and STURDY [56], who showed that the D.L. could be changed by adaptation.

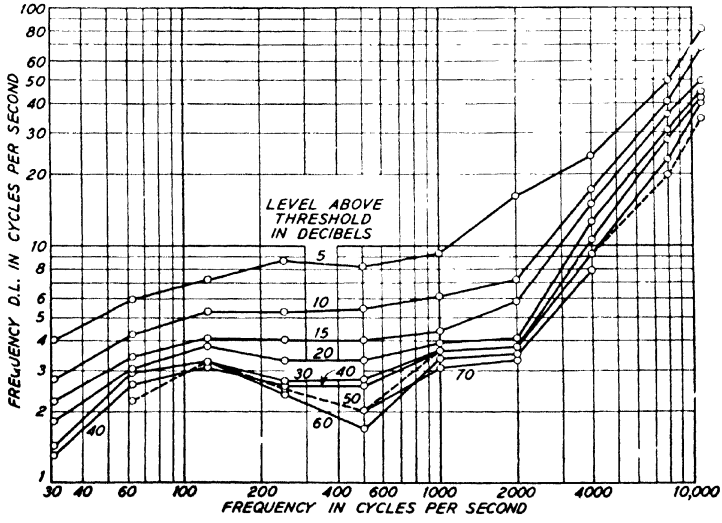


Fig. 111. Frequency difference limens for pure tones at different levels above threshold (from SHOWER and BIDDULPH [57]).

(ii) *Pure Tones – Difference Limens with Respect to Frequency*

Extensive data are available from a study by SHOWER and BIDDULPH [57], the results of which are summarized in Fig. 111. The stimulus used was a single-frequency tone, whose frequency was slowly and smoothly changed between two values. The just noticeable difference in frequency is plotted against frequency with level above threshold as a parameter. It can be seen from the figure that, roughly speaking, there is little change in the difference limen between 100 and 1000 c/s, but large changes occur at lower and higher frequencies. A similar pattern is found in other subjective tests and in physiological measurements in the cochlea.

(iii) *Complex Sounds*

Intensity difference limens for steady noises were measured by MILLER [58] who found a constant D.L. of about 0.4 db over the range from 30 to 80 db above threshold. KARLIN [59] found that very small changes in the frequency characteristic of a noise could be detected.

(iv) *Quantal Theory of Discrimination*

In view of the all-or-none character of neural phenomena, predictions have been made as to the possible discrete step character of the functions relating magnitude of sensation to magnitude of stimulus. BÉKÉSY [60] and later STEVENS [61] and others were able to show that in a measurement of an auditory differential threshold under certain special conditions, the results obtained are predictable on the assumption that changes in the magnitude of a sensation occur in discrete steps. In most tests, however, where these special conditions do not apply, the step functions, if real, are not in evidence.

§ 15. Masking

(i) *Masking*

The presence of one sound reduces, in general, the sensitivity of the ear to other sounds and a shift in threshold results. This phenomenon

is called masking, and a plot of the threshold shift for tones of different frequency is called a masking spectrum or masking pattern. Fig. 112 shows the masking patterns for a 1200-cycle tone obtained by WEGEL and LANE [39]. The level of the 1200-cycle tone in db above threshold is the parameter. These experimentally derived curves indicate the number of db that a single frequency tone must be raised above a

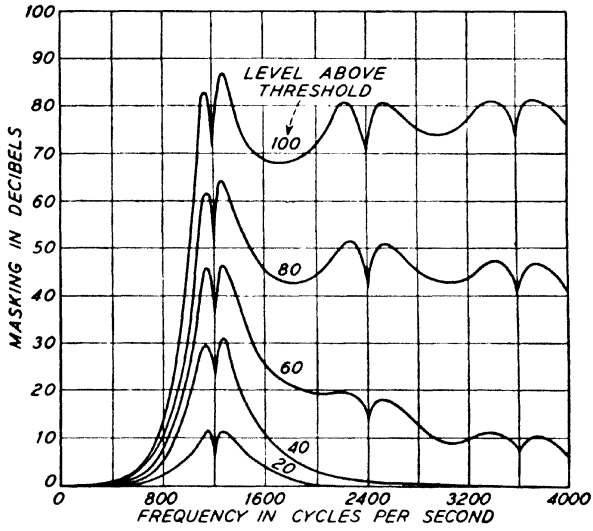


Fig. 112. Masking effect of a 1200-cycle tone on tones of other frequencies (WEGEL and LANE [39]).

'quiet' threshold value in order to be just detectable in the presence of the masking tone. Similar patterns for very narrow bands of thermal noise, measured by EGAN and HAKE [62], reveal interesting differences between noise and tone spectra. The masking spectrum of a noise does not exhibit sharp dips at harmonic frequencies, and the peak masking is higher. The masking patterns of wide bands of noise, whose distribution of energy does not exhibit abrupt changes with frequency, were studied by FLETCHER and MUNSON [63], who found that masking spectra can be obtained by measuring the intensity level per cycle of the noise as a function of frequency, and referring to the empirically determined relationships of Fig. 113.

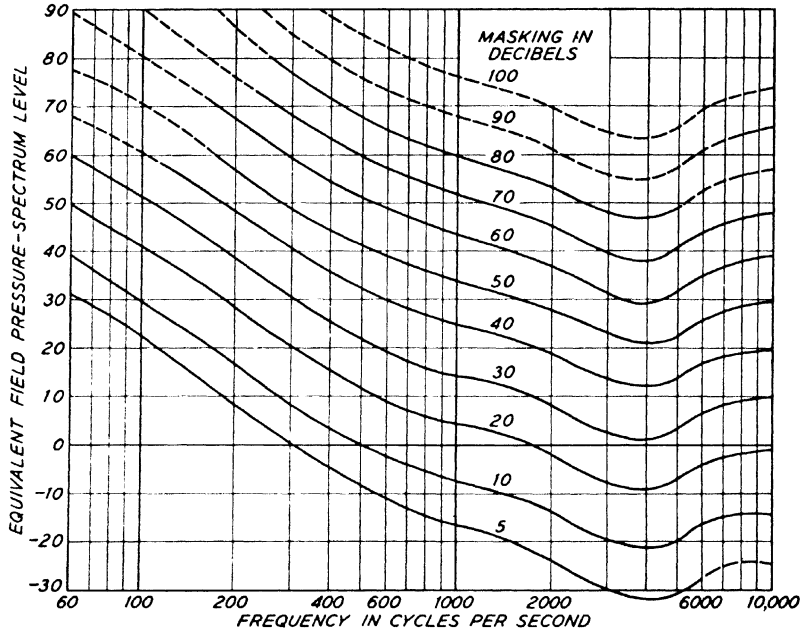


Fig. 113. Contours showing the masking effect of a noise (level given by ordinate) on pure tones (frequency given by abscissa) (FLETCHER and MUNSON [63]).

(ii) *Residual Masking*

The ear exhibits a loss in its sensitivity not only during the time a sound is heard but also in the period immediately following the cessation of the stimulus [64, 65, 66, 67, 68] This prolonged or residual masking effect can be measured by using an exploring or probe tone of short duration, for a threshold test before and after exposure to the primary sound. The magnitude of the residual masking effect is known to depend upon the level and the duration of the primary tone and upon the frequencies of the primary and probe tones. For moderate levels of a primary sound of 0.5-second duration, the return to normal sensitivity occurs within a few tenths of a second after the sound is cut off. For levels at or near the threshold of pain, a longer time is required and the ear is said to be temporarily deafened[69]. For excessive durations or levels of the primary sound, the ear may be damaged, in which case a greatly prolonged or even a permanent

loss in sensitivity may occur. Since some individuals are more susceptible than others to the damaging effects of excessive acoustic stimulation, the residual masking test at safe levels has been suggested as a means of detecting such susceptibility [70]. Placing these individuals in high noise level environments can then be avoided.

§ 16. Loudness of Sound

The loudness of a sound is defined as the magnitude of the auditory sensation which it produces. In general, this magnitude cannot be found directly by measuring the pressure level or spectrum of the unknown sound. Rather it is obtained by measuring the pressure level of an equally loud 1000-cycle tone, and deriving the corresponding loudness from an established empirical relationship [71]. The loudness level is determined by comparing the unknown sound with a 1000-cycle reference tone and adjusting the pressure level of the latter until the two sounds are judged equally loud. The level of the equally loud reference tone is called, by definition, the loudness level of the unknown sound. Once the loudness level of the sound is known, its loudness can be found by using Table 3. This relationship [72] is one in which the loudness corresponding to a loudness level of zero was arbitrarily called one loudness unit. The term 'millisone' is now in general use [73]. When the loudness level was increased 2.1 db above the 1000-cycle zero reference level of .0002 dyne per sq. cm, as measured in a free field, Table 3 shows that the average listener having a free field threshold of .0002 dyne per sq. cm had the impression that the loudness had doubled, that is, had increased from one millisone to two millisones. At + 3.9 db above .0002 dyne per sq. cm, that is, at a loudness level of 3.9 phons, the loudness doubled again to give four millisones. Proceding in this manner it was found that a loudness level of 40 phons gave a loudness of 1000 millisones, etc. The term 'phon' [74] is used as the unit of loudness level. At 1000 c/s the numerical value of the intensity level is identical with the number of phons. At other frequencies and for all other sounds, in general, this one to one correspondence does not hold.

(i) *Loudness of Pure Tones*

Fig. 114 shows the loudness levels of pure tones of different frequencies and pressure levels as determined by direct experiment [71, 75]. Each

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TABLE III

LOUDNESS IN MILLISONES VERSUS LOUDNESS LEVEL IN PHONS

Loudness Level	0	1	2	3	4	5	6	7	8	9
0	1.0	1.42	1.95	2.58	3.36	4.32	5.57	7.10	9.00	11.4
10	14.4	18.7	23.3	28.9	35.1	42.2	50.6	60.3	71.6	85.0
20	100	120	142	165	188	214	242	272	307	340
30	380	421	470	522	577	635	700	763	835	915
40	1,000	1,080	1,170	1,260	1,360	1,470	1,590	1,710	1,850	2,000
50	2,150	2,330	2,510	2,710	2,930	3,160	3,410	3,690	3,980	4,300
60	4,640	5,010	5,410	5,840	6,310	6,810	7,360	7,940	8,580	9,260
70	10,000	10,800	11,700	12,600	13,600	14,700	15,900	17,100	18,500	20,000
80	21,500	23,300	25.100	27,100	29,300	31,600	34,100	36,900	39,800	43,000
90	46,400	50,100	54,100	58,400	63,100	68,100	73,600	79,400	85,800	92,600
100	100,000	108,000	117,000	126,000	136,000	147,000	159,000	171,000	185,000	200,000

line is a contour of equal loudness level as indicated by the number above the respective contours. The zero loudness level curve corresponds to an M.A.F. curve (see Absolute Sensitivity) for a group of observers having a 1000-cycle threshold of .0002 dyne per sq. cm. The loudness level of any single-frequency tone for which a measurement of pressure level has been made can be determined from this chart. Once the loudness level is known, the loudness can be found by referring to Table 3.

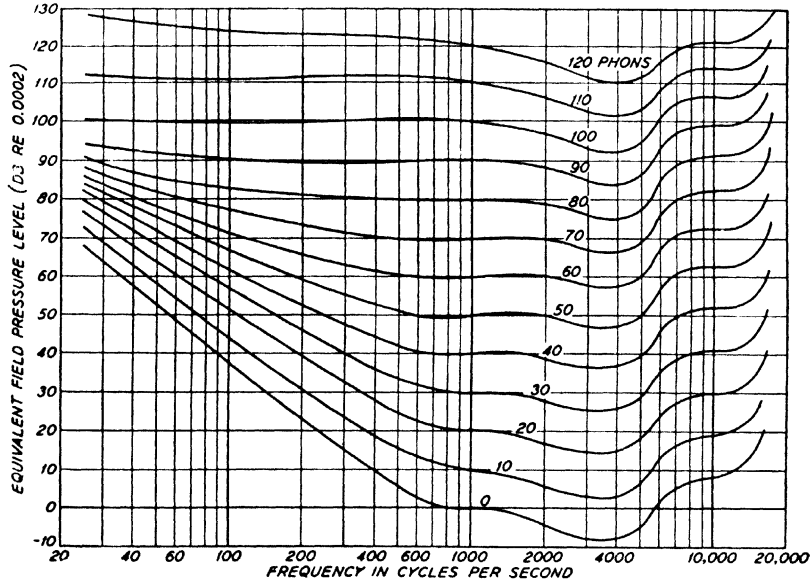


Fig. 114. Contours showing the loudness levels of pure tones (FLETCHER and MUNSON [71]).

(ii) *Loudness of Complex Tones*

When a pure tone stimulus is at low level, it excites a small number of nerve endings within a limited region of the basilar membrane. When the frequency is changed, this point of stimulation moves along the basilar membrane. If the intensity of the tone is increased, the excitation increases at the original point stimulated but also spreads over a wider region of the membrane. If a sound has more than one

frequency component, it excites more than one region of the basilar membrane. If these regions of excitation do not produce overlapping patterns, the loudness of the combination of tones can be obtained by summing up the loudness values that would result from each of the tones acting alone. If the regions of excitation overlap, the problem becomes more complex and cannot be obtained by a simple summation process [71, 76]. Fig. 115 shows the loudness level of complex tones having ten equally loud components that were separated by the frequency intervals indicated. The fundamental frequency for each

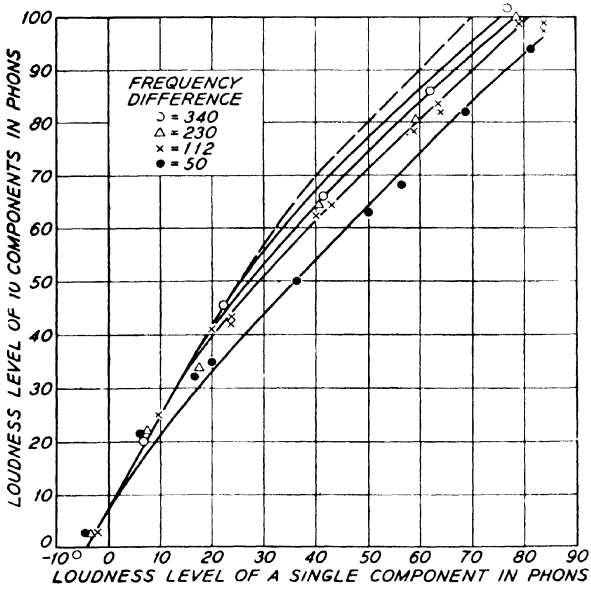


Fig. 115. Loudness tests showing the effect of decreasing the frequency separation between components of a complex sound (FLETCHER and MUNSON [71]).

group of tones was 1000 cycles per second. The upper dashed line shows the loudness levels corresponding to 10 times the loudness of a single component, and indicates the result which would be obtained were sufficient separation of components maintained to give insignificant overlapping of their activity patterns. Note that as the separation increases the loudness level curves approach this maximum value.

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(iii) *Loudness of Speech* [45, 47].

Fig. 116 shows the loudness of undistorted speech as a function of level above threshold. As would be expected from the wide energy distribution of speech, and from the fact that a large proportion of the basilar membrane is activated even at low levels, the initial rate

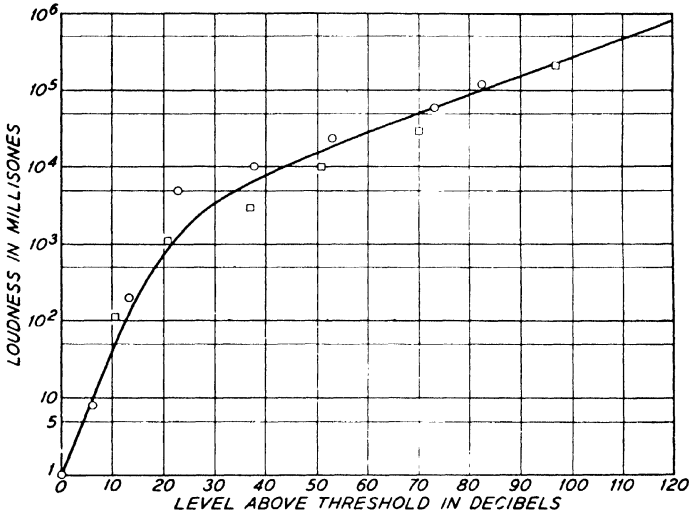


Fig. 116. Loudness of speech (FLETCHER and GALT [45]).

of increase of loudness is higher than it is for a 1000-cycle tone, but tapers off faster at higher levels. However, for the same level above threshold, the absolute magnitude of the loudness of speech always exceeds that of a 1000-cycle tone.

(iv) *Growth and Decay of Loudness*

The loudness of a sound does not reach an instantaneous maximum at the onset of a sound. Its magnitude depends in part on the duration of the stimulus. BÉKÉSY [78] has shown that loudness increases rapidly for a brief interval after onset, then gradually approaches a steady value and finally decreases slowly for long duration sounds. Since the mechanical elements of the ear are believed to be too highly

damped to account for the rate of growth of loudness, it is probably determined by the response characteristics of the neural components of the auditory system. Fig. 117 shows the relation between loudness

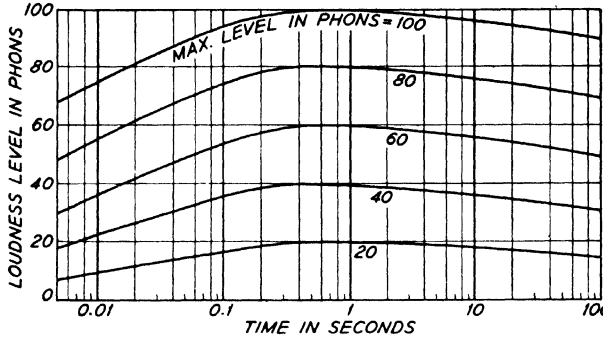


Fig. 117. Loudness level of tones as a function of duration time (MUNSON[79]).

level and the duration time, with the phon level of a one-second stimulating tone as a parameter.

§ 17. Pitch of Sounds [74]

Pitch is a subjective aspect of a sound which, in the case of pure, low-level tones, is primarily related to frequency. A measurement of the pitch of a sound is made by comparison with the pitch of pure tones that are 40 db above threshold.

(i) *Pure Tones*

Although the pitch of pure tones is primarily a function of frequency, it changes somewhat at high levels [80]. Present indications are that with most observers there is a tendency of pitch to rise with increasing level of high-frequency tones. A downward change is likely to occur if the level of a low-frequency tone is increased. A region of greatest change of pitch for low-frequency tones of high levels has been found to centre around 200 c/s [81]. The pitch of tones of medium frequencies varies very little with level.

Also of importance is the relation of pitch to frequency. STEVENS and VOLKMANN [82] have established a pitch scale by comparing the

pitch sensations of pure tones of various frequencies at levels of 40 db above threshold. The psychophysical procedures used to obtain this scale were those of fractionation and equal sensory intervals. For example, the observers were asked to adjust the frequency of a tone so that its pitch sounded 'half as high' as the pitch of a second, fixed, tone; or to adjust the frequency of a tone so that its pitch lay 'mid-way' between the pitches of two other, fixed, tones. These procedures gave consistent results and were combined to yield the scale of pitch shown in Fig. 118. The unit of pitch is called the mel.

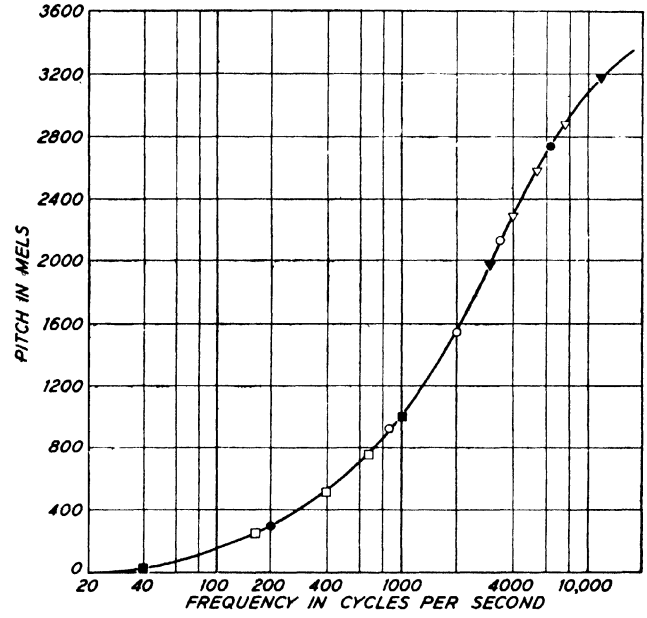


Fig. 118. Relation between pitch and the frequency of pure tones 40 db above threshold (STEVENS and VOLKMANN [82]).

Arbitrarily, the pitch of 1000 mels is assigned to a 1000-cycle tone of 40 db sensation level. The pitch scale is a ratio scale, hence a tone of say 2000 mels pitch sounds twice as high as one of 1000 mels.

(ii) *Complex Sounds*

Although the concept of pitch is defined in terms of the auditory sensation due to pure tones, it is matter of common experience that

it is possible to assign a pitch to some steady-state complex sounds and even transient sounds. If the complex sound consists of a band of frequencies lying close together and irregularly spaced, but about equal in level, the observer, when asked to assign a pitch to the sound, usually assigns one near the centre of the band (*cf.* Ref. 2, Chapter 12). If the sound has relatively few components, the ear is sometimes capable of assigning a pitch to the individual components.

The above rule breaks down when the complex sound consists of a sequence of components, differing in frequency by a constant but not too small amount. The pitch, in that case, is related to the difference frequency rather than to a pitch corresponding to some mean frequency of the components. Indeed, in a musical note, the fundamental can often be removed without any change in pitch; a musical selection rendered with many or all of the fundamentals removed is remarkably little affected. SCHOUTEN [83] has performed a series of interesting experiments bearing on the problem.

(iii) *Sounds of Short Duration*

The changes in pitch of short tones as the number of cycles presented to an observer is varied, from the limiting case of one cycle to a long train of cycles lasting several hundred milliseconds, have been studied by several experimenters [84, 85]. In general, it is found that the pitch is lowered as the tone is shortened but the change is not great until durations less than 10 or 20 milliseconds are reached.

§ 18. Other Attributes of Sounds

(i) *Additional Attributes*

Many, if not all, sounds elicit sensations that are inadequately described by the attributes of loudness and pitch alone. Most auditory sensations can be given more elaborate descriptions in terms of additional attributes.

In addition to being called 'loud' or 'soft' (loudness) or 'high' or 'low' (pitch), sounds can also be classified as to whether they appear 'big' or 'small' to the observer. This aspect, measured by the attribute of *volume*, has been quantitatively investigated by THOMAS [86] for pure

tones. THOMAS determined equal-volume contours, *i.e.*, functions which show how the level and frequency of a pure tone must be modified, on the average, so as to sound 'as big' as a standard tone to a group of observers. Words like 'tightness', 'hardness' or 'compactness' have been used to describe another attribute of sound, namely 'density'. STEVENS [87], has obtained equal-density contours for pure tones, and POLLACK [88] studied the density, volume and annoyance effects of bands of random noise.

While volume and density, in conjunction with loudness and pitch, describe the sensations due to pure tones reasonably well, additional attributes seem to be necessary to describe more adequately the sensations due to complex sounds.

(ii) *Consonance and Dissonance*

Part of the musician's vocabulary are the words 'consonance' and 'dissonance'. The terms 'smoothness' and 'roughness', which are used to describe not only musical sounds but also an attribute of all complex tones, have received attention from certain investigators. BÉKÉSY [89] has used two beating tones of equal level, sounded together at frequencies of 3000 and 3050 cycles per second, as a standard of roughness for comparison with other sounds. The roughness of the standard is varied by changing the level of the two-tone combination, higher levels producing the effect of greater roughness. MATHES and MILLER [90] have studied the variations in roughness of certain complex tones due to variations in the shape of the envelope of the wave.

§ 19. Localization of Sounds

The ability of man to localize sources of sound in space is a matter of everyday experience and great importance. A complete and quantitative picture is not yet available that explains the mechanism responsible for the fact that sources of sound can be localized binaurally, often with astounding accuracy, in direction (angular localization) and in distance (depth localization). Neither do we know why binaural listening enables one to perceive a difficult signal, such as speech in a noisy reverberant room, better than with monaural listening if the noise and speech do not originate at the same point. Although

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experiments have shown that some localization is possible using one ear only, better results are usually obtained when listening with two ears.

(i) *Angular Localization*

In general [91], observers listening to sources of sound in free space rarely confuse right with left, but their sense of direction is poor when the source is nearly opposite either ear. The accuracy of angular localization is much better for sound sources in the horizontal plane directly in front of the observer (and, in many cases, directly behind). The direct binaural clues which may serve to explain the ability to localize are the differences at the two ears in level and time of arrival or phase.

Steady-state differences in level and phase at the two ears occur due to diffraction of the incident sound by the head. This aspect of the problem has received comparatively thorough attention. FIRESTONE [92] has measured the differences in level and phase at the two 'ears' of a dummy, for pure tones and various separations between the source and the dummy head. WIENER [93] investigated experimentally the diffraction of an approximately plane wave by the human head, by determining the magnitude of the pressures at the entrance to the auditory canals of several observers relative to the free-field pressure. The source of sound was located in the horizontal plane through the observer's ears. The average pressure ratios at the two eardrums, determined as a function of frequency and angle of incidence, agree reasonably well with those obtained by SIVIAN and WHITE [12] from threshold measurements. The pressure ratio changes most rapidly with angle for the source directly in front (or back) of the observer. At high frequencies, there is evidence that the presence of the auricle (pinna) may aid somewhat in front-to-back discrimination. STEVENS and NEWMAN [94] investigated the ability of observers to localize actual sources of pure tones in a horizontal plane in the absence of reflections. The results showed that the errors made by the observers by confusing front and back quadrants exhibited an appreciable decline beyond about 3000 c/s. Apparently the observers were aided by the fact that the source, when directly behind, sounded appreciably less loud above 3000 c/s than when directly in front. This is most likely due to the diffraction effect of the pinna mentioned above. At medium

and low frequencies, no such shadow effect is to be expected [93]

Most experimenters draw the conclusion that phase differences at the two ears are primarily available—and used—as clues for the localization of tones of low and medium frequencies, whereas the level differences at the two ears are utilized for tones of the higher frequencies. In the transition region of intermediate frequencies near 3000 c/s neither clue is assumed to be fully operative and localization is rather poor. The localization of complex tones, noises, etc. is aided by the additional clues of changing loudness and quality with azimuth due to the frequency dependence of the head diffraction effect.

More than one primary clue is generally present in practical situations. Furthermore, it is well known that important additional (secondary) binaural clues exist. Experience in listening and expectations about the various familiar sources of sound aid materially in localizing. This appears to be especially the case in monaural listening. Last, but not least, we use movements of the head [95] when going through the process of trying to localize a source of sound in space. If the change in binaural clues so obtained corresponds with those to be expected from the movements of the head, localization is aided and a sensation of an actual source of sound 'out there in space' results. If this is not the case, as happens in some synthetic experiments, confusion often results. DE BOER and VAN URK [96, 97] have conducted a series of ingenious experiments involving binaural telephone systems and stereophonic sound systems, which confirm WALLACH's results [95] and lend great weight to the supposition that the observer's head movements are important in binaural angular localization. (*cf.* Chap. 16 § 2).

In an attempt to explain the successful localization of sound sources even in reverberant surroundings, KOENIG [98] and WALLACH [99] *et al.* have conducted a series of experiments to explore what these authors call the 'Precedence Effect' in binaural localization of short sounds coming from two different sources.

(ii) *Depth Localization*

Not much is known at present about the mechanism of depth localization. STEINBERG and SNOW [100] have found that the success with which observers can estimate the distance of a source of sound varies with level and the ratio of reflected to direct sound. The sound sources

with which the observers were presented in these tests were the virtual ones obtained by the use of a multichannel stereophonic sound system. Again, experience aids in such judgments. BÉKÉSY [101] has shown experimentally that noises or transients devoid of low-frequency components are generally judged to be more distant than those containing them. This is, to a certain extent, also true of speech.

§ 20. Hearing Impairment

The usual effect of deterioration within the auditory system is to raise the threshold of hearing. As a result, the sound pressure at some or all frequencies must be greater than normal if tones are to be heard. Hearing loss may thus be conveniently described in terms of the number of decibels that the threshold has been raised above the accepted normal threshold. A chart of hearing loss versus frequency is called an audiogram.

(i) *Measurement of Hearing Impairment*

The threshold tests reported in the section on Absolute Sensitivity represent the hearing acuity of a selected group of trained observers listening under carefully controlled and nearly ideal conditions. For the evaluation of the hearing acuity of the ordinary layman a less exacting test which lends itself to more general use has been developed. This is the so-called audiometer test, which may be of the clinical (single observer) type, or in a form designed for group testing. The clinical test usually employs single frequencies and thus makes possible a frequency analysis of the acuity of hearing. The group test may employ either recorded speech or tones as a source of test material. It provides a means of screening rapidly large numbers of individuals. For information on group test material and test procedures the reader is referred to articles by FLETCHER and STEINBERG [102], by HUDGINS *et al* [103], by REGER and NEWBY [104], and by STEINBERG, MONTGOMERY and GARDNER [47].

The pure-tone audiometric test is widely used by the otologist and audiologist for the measurement of hearing impairment. In this test the signal is interrupted by manually operating a 'tone interrupter key.' Hearing acuity is indicated by the ability of the listener to give

a properly synchronized return signal of some kind, such as operating and releasing a hand switch connected to a signal lamp, or by raising and lowering the hand or finger. Threshold is defined as the lowest level at which consistency is obtained between the tone interruptions and the listener's signal. For this test, threshold is usually approached from levels both above and below threshold for each frequency teted, and the average of the two taken as the test result.

A variation of the above threshold test is the pulse-tone technique initially used for mass testing at the New York World's Fair [47], and later adapted for portable group testing by REGER and NEWBY [104] and for clinical use by GARDNER [105]. For this test, the observer is presented with a group of one or more short pulses of tone. Threshold is defined as the lowest level at which the listener can report correctly the number of tones in each pulse group presented.

For the clinical and group hearing tests described above, the average level required to elicit threshold is appreciably higher than the levels shown by the M.A.C. curve of Fig. 106.

(ii) *Types of Hearing Impairment*

Hearing impairment may be divided into two main classes, namely, conductive losses and nerve or perceptive type losses. There is also a third class of impairment in which the peripheral organ is normal, but because of some abnormal functioning of the brain or for psychological reasons the individual does not hear properly. Since most impairments are either conductive, perceptive or a combination of the two, further discussion will be limited to these two classes.

(iii) *Conductive Impairment*

This type of hearing loss may be caused by a blocking of the external canal or by a mechanical malfunction of the ear drum or ossicular chain. Conductive type losses do not exceed 50 or 55 db. This is the maximum mechanical loss [106] that can be introduced before sound waves begin to reach the inner ear by way of the bones of the skull.

Fixation of the stapes at the oval window (otosclerosis) is a conductive type loss. Its alleviation by an operation known as 'fenestration' has received a great deal of attention. Beginning in 1876 with KESSEL,

through PASSOW, BARANY, JENKINS, HOLMGREN and SOURDILLE, the introduction of a new window or 'fenestra' into the lateral semi-circular canal was reduced in the 1930's to a one-stage procedure by LEMPERT. An excellent bibliography covering the development of this operation has been prepared by KELEMEN [107].

(iv) *Perceptive Type Impairment*

The magnitude of this type of loss may vary from a few decibels to total loss of hearing. It is usually caused by degeneration of the hair cells of the cochlea, of the nerve fibres which terminate on them, or of the auditory nerve. Because it is a degenerative type loss, it is generally believed there is no type of treatment, medication, or stimulation that will improve the hearing of anyone having true nerve deafness [106].

(v) *Response of the Defective Ear*

The type of the impairment present affects the response of the deafened ear in a number of ways. Thus the conductively deafened ear hears lower levels by bone conduction than the nerve-deafened ear, while the latter type of loss shows a more rapid rate of increase in perceived loudness as a function of level above threshold (loudness recruitment). The nerve-deafened ear can detect a smaller incremental change [108] in intensity than the conductively deafened ear at levels near threshold. Nerve impairment also differs from conductive deafness in that the ear does not return as quickly to its usual threshold sensitivity after exposure to high level sounds (delayed masking or fatigue). [109] Despite these and other differences in response, however, a subjective analysis of the nature of the impairment present in a particular case is subject to some uncertainty.

(vi) *Bone Conduction* [110]

This test goes back at least as far as CAPPIVACCI in the 16th century. In today's bone conduction test, the vibrations of an inertia type sound source (normally applied to the mastoid process) reach the inner ear by way of the bones of the skull, and in so doing detour the mechanism of the middle ear. If the middle ear is the site of the hearing loss, the bone conduction threshold sensitivity tends to be the same as that

obtained on a normal hearing observer. If the inner ear is defective, on the other hand, the loss in bone conduction threshold sensitivity tends to equal the air conduction loss in sensitivity and the result is interpreted as an indication of nerve involvement.

(vii) *Loudness Recruitment*

In 1924 POHLMAN and KRANTZ [111] reported that tonal gaps present at levels near threshold 'seemingly disappear quite completely' at higher levels. FOWLER [112] and HAM and GUTTMAN [113] reported that loudness increased more in the deafened than in the normal ear, as the level was increased above threshold. This greater than normal rate of loudness increase was verified by REGER [114] in tests on individuals

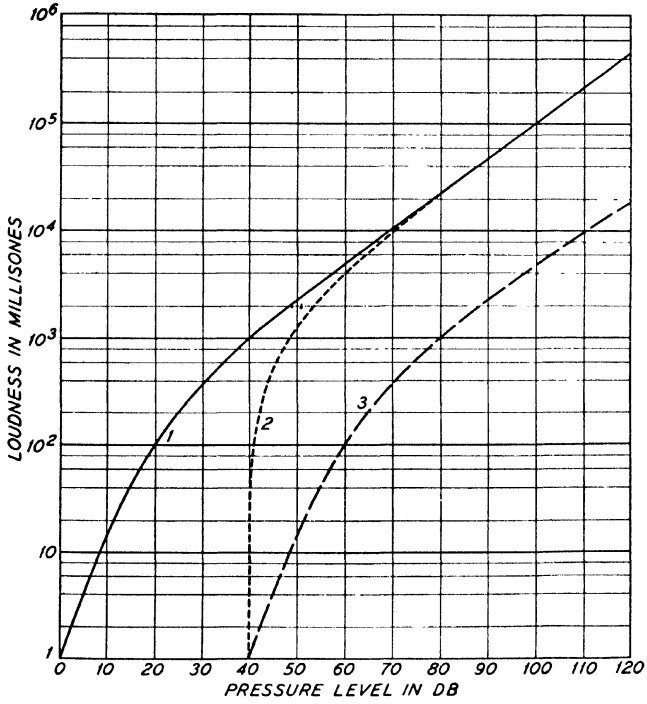


Fig. 119. Relation between the pressure level of a 1000-cycle tone and its loudness. (1) normal hearing; (2) 40 db perceptive impairment; (3) 40 db conductive impairment (STEINBERG and GARDNER).

with nerve deafness. The phenomenon was associated with nerve deafness by FOWLER [115] and STEINBERG and GARDNER [116]. The latter also gave an explanation of recruitment in terms of the loudness function, and showed that a similar effect occurs when the normal ear listens in the presence of noise.

Loudness recruitment can be demonstrated rather easily on individuals having one normal and one nerve-impaired ear. In such cases the pressure level of the tone on the normal ear is adjusted until it sounds as loud as a fixed level of tone on the impaired ear. For the conductively deafened ear the difference in the pressure levels for equal loudness in the two ears remains constant at all levels above threshold, as shown by curve 3 of Fig. 119. For the nerve-impaired ear the difference in the pressure levels for equal loudness diminishes as the levels are raised above threshold, as shown by curve 2. Curve 3 is a simple displacement to the right of the normal curve 1, so that for an equal number of decibels above their respective thresholds the loudness sensed by the conductively impaired observer is equal to that sensed by the normal observer. For the nerve-deafened observer, on the other hand, the loudness returns to normal, at least for moderate amounts of impairment.

(viii) *Residual Masking*

Immediately after listening to a primary acoustic signal, the sensitivity of an ear is temporarily reduced. This reduction in sensitivity variously has been called 'short duration auditory fatigue' [109], 'adaptation' [65] and 'residual masking' [117].

GARDNER [109] found that the residual masking effect was different for conductive deafness than for nerve deafness, and used it as a means of differentiating between the two types of impairment.

(ix) *Hearing Aids*

Modern hearing aids are pocket-sized acoustic amplifiers, consisting of a microphone, miniature vacuum tube amplifier and either a bone- or an air-conduction transducer. The bone-conduction unit is held against the mastoid process behind the ear by means of a headband. The air-conduction unit fits over the ear, or for the more widely

used miniature type, is coupled to the ear canal by means of a moulded earpiece.

Two rather comprehensive studies have been made to determine how nearly the frequency gain characteristic of a hearing aid should compensate for the frequency loss characteristic of the individually impaired ear. These two studies were made by the British Medical Research Council [118] and by Harvard University [119]. In general their findings are in agreement and indicate a hearing aid should have the following characteristics: (1) a uniform frequency response, *i.e.*, without marked resonant peaks or valleys, covering the range from 300 to 4000 cycles per second; (2) a tone control adjustment to give a flat, or a moderately rising response; (3) a smoothly graded gain control approximately logarithmic over a 40-db range; (4) a choice of models, the lowest powered to have at least 40–db acoustic gain available, the highest powered a maximum of 80 db, with provision for limiting the maximum acoustic pressures by compression or peak clipping, as required by and acceptable to the wearer.

Most individuals in need of a hearing aid would be adequately fitted by an instrument which met the above design objectives. A few, however, would remain who would find certain departures from a 'flat' frequency response of advantage in their particular cases.

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CHAPTER 12

SPEECH TRANSMISSION RATING

For authorship, see p. 193

The performance of a system for speech transmission may be expressed in terms of physical measurements, such as single frequency response characteristics, or in terms of the results of various types of intelligibility tests involving speakers and listeners. Both procedures have been used, and considerable progress has been made in relating the results of intelligibility tests to the measured characteristics of systems.

(A) INTELLIGIBILITY TESTS

In addition to their extensive application in the rating of transmission systems, intelligibility tests have also been employed in the evaluation of speakers and listeners during communication. Such tests have been employed in the study of individuals with speech disorders [1], and in the selection of personnel for communication under special conditions [2]. Listeners with various types of hearing loss, or those with foreign language backgrounds, have been compared with normal subjects [3].

§ 1. Test Material

A large number of test lists have been devised for intelligibility measurements [4, 5, 6]. At one extreme are the individual speech sounds and monosyllables, which form the most elementary discrete units of speech; at the other extreme are sentences and other forms of connected speech material.

Lists of meaningless monosyllables have received wide application in transmission rating tests. Trained speakers and listening crews are required, since the individual speech sounds in the syllables must be

recognized, spoken and recorded. The nonsense monosyllables may be of several types: consonant-vowel, vowel-consonant, or consonant-vowel-consonant, in the simpler cases. More complex lists may involve initial and final consonant combinations.

The relationship of the intelligibility of various types of speech material has been studied by FLETCHER and STEINBERG [4]. Several of the relationships are shown in Fig. 120. Curve (*a*) of this figure shows the percentage of individual speech sounds correctly identified (sound articulation) in the CVC type of nonsense syllables, as a function of the percentage of complete syllables correctly recognized. Unless indicated otherwise, the term syllable articulation refers to this latter percentage. In (*b*) the intelligibility of words containing various numbers of syllables is plotted against syllable articulation under the same conditions. Curve (*c*) shows the number of sentences correctly heard as a function of syllable articulation.

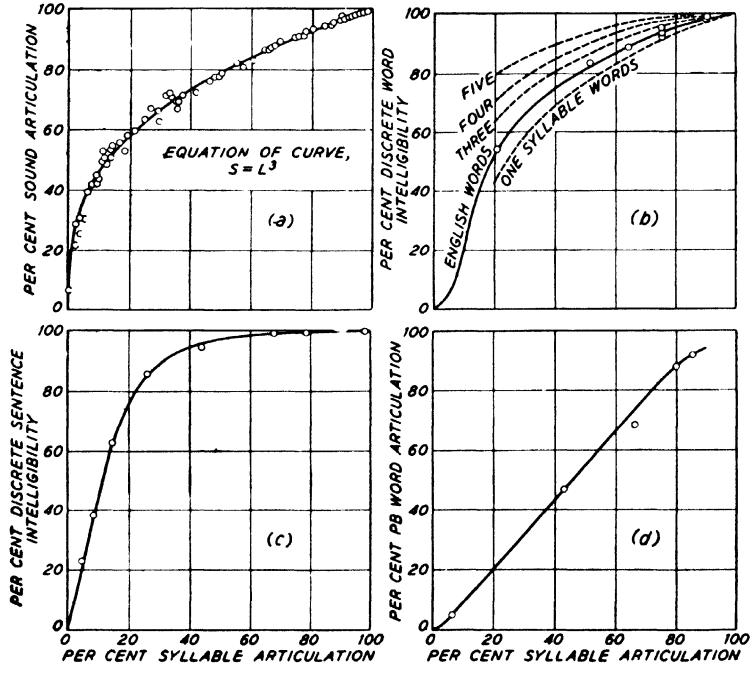


Fig. 120. The relationships of the intelligibility of various types of speech material (after FLETCHER and STEINBERG, and EGAN and WIENER).

A large number of monosyllabic word lists for tests in English have been prepared by the Psycho-Acoustic Laboratory of Harvard University [5, 6]. In these lists, which are referred to as PB lists, the frequency of occurrence of the various speech sounds is based upon DEWEY's frequency count for printed material [7], and an attempt was made to maintain a phonetic balance or equivalence, from list to list. The intelligibility of these PB words versus syllable articulation with equivalent transmission system conditions is shown in (*d*) of Fig. 120 [8].

§ 2. Field Tests

Under field conditions, speech transmission circuits present a wide and complex range of variables. With the introduction of a new kind of instrument or circuit, service conditions may be modified in an unpredictable way, depending upon the user's reactions. A basic criterion for rating a circuit is its transmission performance when in actual use, as a link in the extremely complicated and variable communication channel between the brain of one telephone user and the brain of another.

A transmission rating system is described by MARTIN [9] and MCKOWN and EMLING [10] in which the repetition rate, observed in normal flow of conversation over a circuit, is correlated with an equivalent volume loss. The number of repetitions requested by normal telephone users per unit time, when using the circuit in actual service, is taken as a measure of circuit performance, on the basis that this number is a direct quantitative measure of the success with which telephone users carry on a conversation. The effects of volume loss, noise, distortion, and sidetone have been evaluated on various types of circuits, in terms of equivalent volume loss on a reference circuit, by means of repetition rate tests.

(B) FACTORS AFFECTING INTELLIGIBILITY

The data presented in this section have been selected to show the effects on intelligibility of a number of more or less basic types of distortion that occur in transmission systems. Distortionless transmission is assumed to be that provided by an air path one meter long, in the absence of noise and reverberation. A reference system having

transmission characteristics equivalent to those of a one meter air path is referred to as an 'orthotelephonic' system [11]. In practice it is simulated by means of suitably calibrated electrical systems composed of microphones, amplifiers, frequency equalizing networks and earphones or loudspeakers. The elements of the system are individually calibrated so that their combined response is comparable with the response of the reference air path [11, 12]. Provision is also made for introducing distortion in measurable amounts.

§ 3. Received Level

The articulation observed with an experimental testing crew using a high quality speech transmission system varies between zero for speech reproduced at the threshold of audibility and about 98 or 99 per cent at received levels 60 or more db above threshold. The solid line curve of Fig. 121 shows the relationship between syllable articulation and received level as reported by FLETCHER and GALT [13],

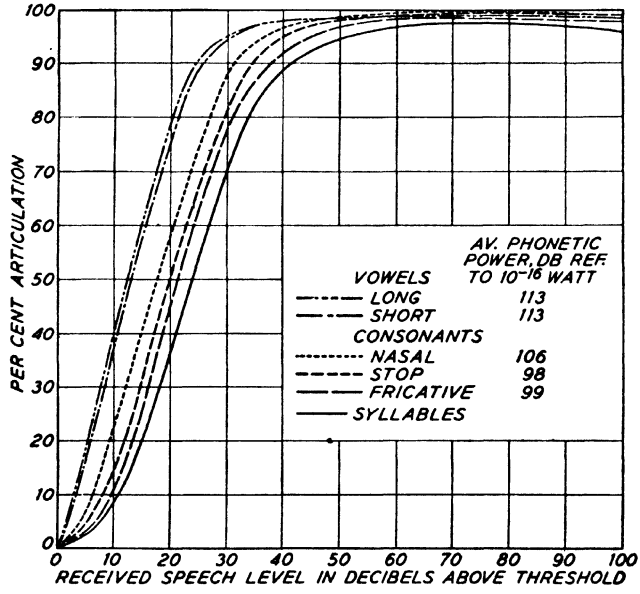


Fig. 121. Syllable and sound articulation *versus* received level (FLETCHER and GALT, and STEINBERG).

for data obtained with an articulation testing procedure described by FLETCHER and STEINBERG [4]. The articulation increases rapidly with increasing level above threshold, up to about 40 db. For received levels greater than about 50 db above threshold, the articulation remains relatively constant at about 95 to 98 per cent. The articulation observed in normal face to face conversation, in a quiet well-damped room, is about the same as the maximum values of Fig. 121. If the speaker and listener are separated by a one meter air path and the speaker radiates an average speech power of 10 microwatts, the level perceived by a listener having normal hearing will be about 62 db above threshold.

The broken line curves of Fig. 121 show averaged characteristics of sound articulation versus received level, as reported by STEINBERG [14] for five classes of speech sounds, namely: long vowels **[ɑ, e, i, o, ɔ, u]**, short vowels **[æ, ɛ, ɪ, ʌ, ʊ, w, j]**, semi-vowels **[l, m, n, ŋ, r]**, stop consonants **[b, tʃ, d, g, h, dʒ, k, p, t]**, and fricative consonants **[f, s, ʃ, st, θ, ð, v, z, ʒ]**. The differences in level above threshold for these five classes, at a value of sound articulation of 90 per cent, correspond approximately to the relative phonetic powers of the sounds in the several classes. Thus, when the speech level is less than about 50 db above threshold, the articulation of the various sounds depends to a considerable extent upon the relative powers with which they are normally spoken.

§ 4. Noise

(i) *General*

One of the effects of noise at low or moderate levels is to shift the threshold of audibility of other sounds heard in the presence of the noise [14]. If the masking produced by a noise is substantially uniform over the range of frequencies which are important to the recognition of speech sounds, its principal effect is simply a shift in the curve of articulation versus level above threshold. Fig. 122 given by FLETCHER and GALT [13] shows sound articulation measurements on a system with and without noise. The response of the system versus frequency is given in the upper part of the figure, along with the spectrum of the noise heard by the listener. The sound articulation is plotted against received speech level, referred to the threshold of audibility when

there was no added noise. It may be seen that the addition of the noise had an effect on the observed articulation which was equivalent to a threshold shift of 14 or 15 db, and that the shapes of the characteristics

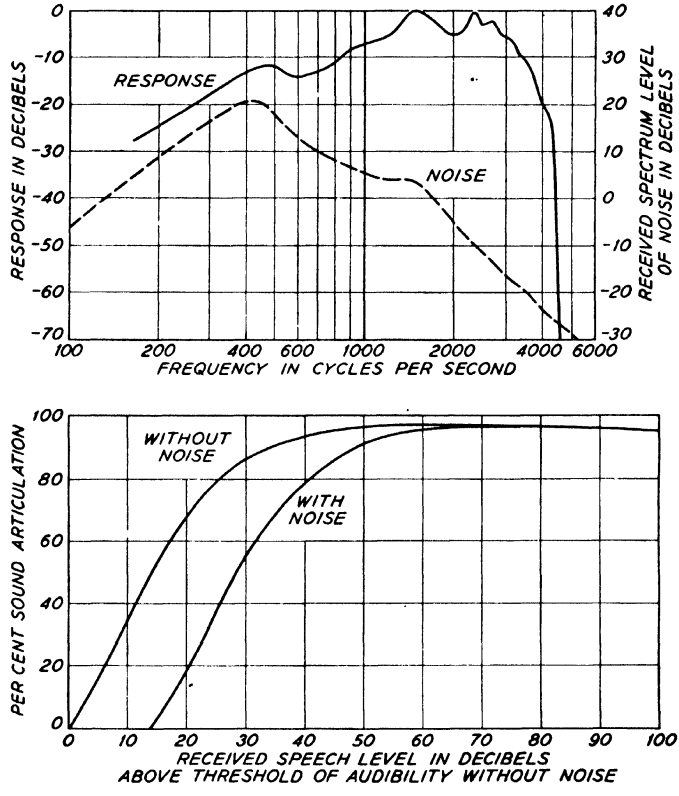


Fig. 122. Sound articulation *versus* received level, for a transmission system with and without noise (after FLETCHER and GALT).

with and without noise are nearly the same, even though the spectrum of the added noise shows considerable deviation with frequency.

With stronger interfering noises, the maximum observed articulation values are less than would be indicated by a simple shift of system gain. Modulation takes place in the ear between the noise and speech when both are at high levels, resulting in decreased articulation scores. EGAN and WIENER have reported on a series of articulation measure-

ments in which relatively loud noises like thermal noise were used [8, 15]. Their studies also included tests with various frequency band widths, and with noises strong enough to cause threshold shifts of the order of 70 db.

When the interfering signal is a tone, the masking produced in the ear is restricted to a limited portion of the frequency range, and the effects on articulation involve both a shift in the curve of articulation versus system gain and a change in its shape [14].

(ii) *Deafness*

Slight or moderate deafness has an effect on articulation similar to that of noise. If the hearing loss is uniform with frequency, the articulation to be expected is similar to that observed when the normal curve of articulation versus level is shifted by an amount corresponding to the hearing loss.

(iii) *Room Reverberation*

Room reverberation may be regarded as a type of noise in its effect upon articulation. If the reverberation time is short enough, that is if the reflected energy from one speech sound is reduced to threshold before the next sound is well advanced, then the effect upon articulation is negligible. If one sound persists into the next, however, it interferes with the recognition of the sound, in the same way that noise does. Articulation data for different amounts of reverberation have been reported by STEINBERG [14].

§ 5. Frequency Distortion

Speech intelligibility may be affected seriously by the unequal transmission of different parts of the frequency range. There is an infinite variety in this type of distortion, but some forms that can be specified easily have been investigated more or less thoroughly.

(i) *Band Width*

The importance of the different regions of intensity and frequency to the recognition of speech has been determined by articulation

tests, using essentially distortionless test circuits into which electrical filter networks and different amounts of attenuation were introduced. Figs. 123 and 124, given by FRENCH and STEINBERG [16], show the effects on syllable articulation of low-pass and high-pass filters having

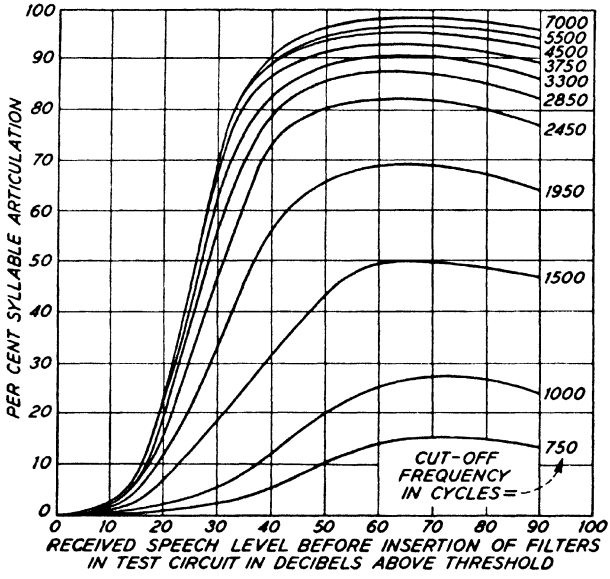


Fig. 123. Syllable articulation *versus* received level, for systems having indicated low-pass filter cut-off frequencies (after FRENCH and STEINBERG).

various cut-off frequencies. The curves of Figs. 123 and 124 are smoothed results of a series of articulation tests, and show composite data taken with both men's and women's voices. The filters used introduced negligible attenuation in their pass bands, and caused practically complete suppression of speech components above the cut-off frequencies in the cases of the low-pass filters, and below the cut-off frequencies in the cases of the high-pass filters. A value of 0 db orthotelephonic response on these characteristics corresponds approximately to a received speech level of 60 db above threshold before the insertion of the test filters in the circuit.

When the several speech sounds are considered separately, considerable variation is observed in the characteristics of articulation versus

filter cut-off frequency. FLETCHER gives such characteristics for 36 different sounds [17]. STEINBERG has grouped the sounds into five classes, as shown in Fig. 125, and also shows separate characteristics for men and women [14]. In these articulation tests, the level of received

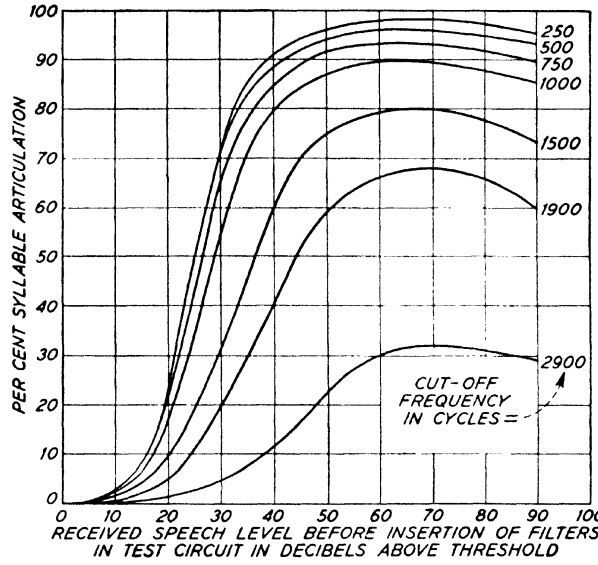


Fig. 124. Syllable articulation *versus* received level, for systems having indicated high-pass filter cut-off frequencies (after FRENCH and STEINBERG).

speech before the filter was introduced was maintained at 70 db above threshold. The characteristics for low-pass and high-pass cut-offs for women's voices always intersect at higher frequencies than for the men's voices, indicating that the higher frequency ranges are more important for recognition of women's speech.

The range of greatest importance to the recognition of the sounds may be taken as the frequency range through which the curves of Fig. 125 bend down. For the case of syllable articulation, which involves all the speech sounds, this range is from about 200 to 8000 cycles per second. For the different classes of sounds, the ranges are smaller, being as follows: long vowels, 200–2500 c/s; short vowels, 300–3000 c/s; semi-vowels, 500–4000 c/s; stop consonants, 750–5000 c/s; fricative consonants, 1000–8000 c/s.

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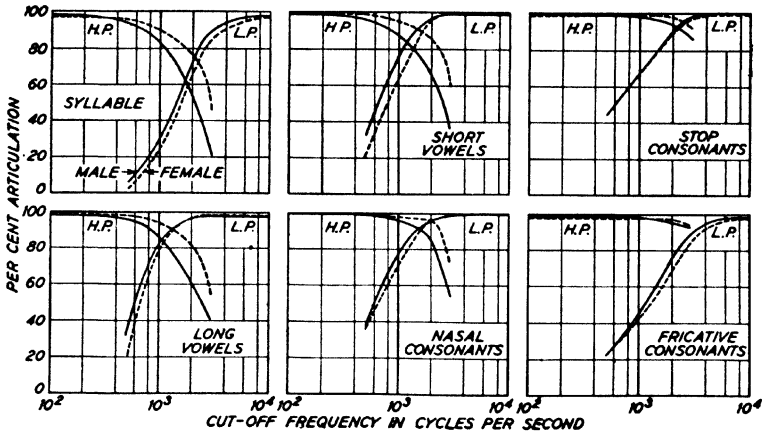


Fig. 125. Articulation *versus* high- and low-pass filter cut-off frequencies for different classes of speech sounds (STEINBERG).

(ii) *Resonances*

Another type of frequency distortion is the presence of resonances in the transmitting system. In Fig. 126 two such cases are shown, as reported by STEINBERG [14]. The curves on the left are the transmissions: 1 the flat system, 2 and 3 the resonant systems, both of

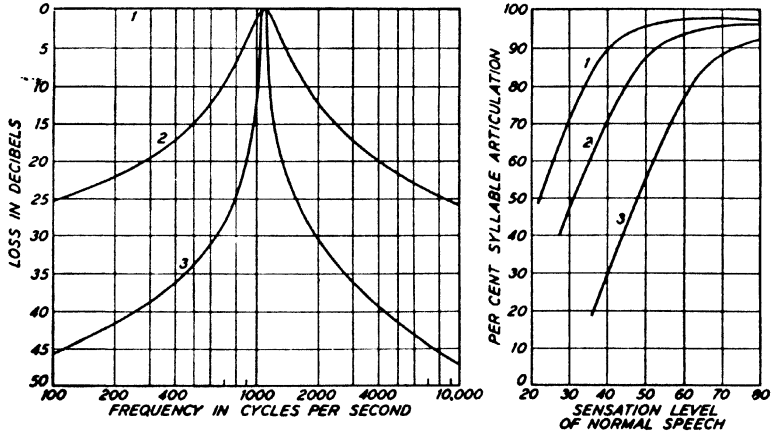


Fig. 126. Transmission curves and corresponding articulation curves, for a flat system and for two resonant systems (STEINBERG).

which peak at 1100 cycles per second. The damping constants are respectively 10000 and 1260 db per second (logarithmic decrements 1.04 and .132). Syllable articulation curves are given on the right, plotted against the sensation level of normal speech (level above the threshold of the flat system). The chief reason for the low articulation scores of the resonant systems, at the lower speech levels, is the attenuation of large portions of the frequency spectrum on either side of the peak. This brings the sounds which depend on these frequencies near to or below the threshold of audibility. Even when the optimum level is used for each case, however, the resonant systems are appreciably lower in articulation. Several other examples of resonant systems are given by FLETCHER and GALT [13].

(iii) *Frequency Shift*

Various ways of producing frequency shift have been used in studies of its effects on speech intelligibility. The effect on syllable articulation of multiplying component frequencies by a common factor, as reported by FLETCHER [17], is shown by the upper curve in Fig. 127. Lists of articulation syllables were recorded on phonograph records and the records were then played back at speeds of rotation from 0.5 to 1.5 times the speed used in recording. Duration times, fundamental

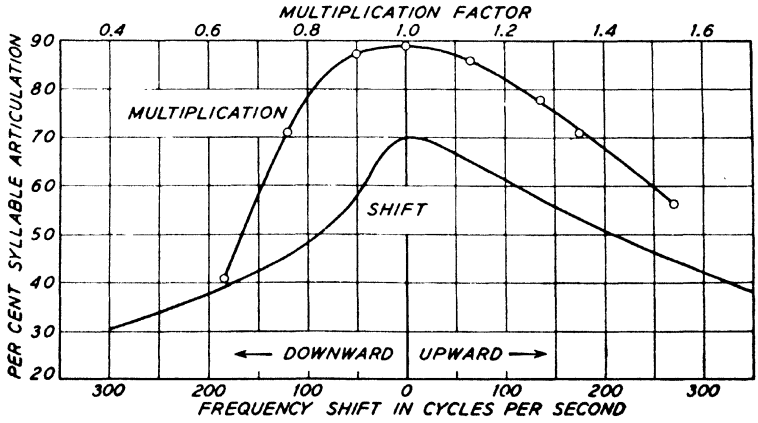


Fig. 127. Effects on articulation of frequency multiplication and frequency shift (after FLETCHER).

frequencies and frequencies of the formants are, of course, changed. In addition, certain of the component frequencies are shifted into frequency ranges in which the response of the equipment or the ear is reduced. The results of Fig. 127 show that a speed change of 10% or less has relatively little effect on articulation. For greater changes, the articulation decreases markedly. The effects on vowel quality have been discussed by CHIBA and KAJIYAMA [18].

The lower curve of Fig. 127 shows the effects of shifting frequency by a carrier frequency modulation and demodulation process. In this case, duration times are maintained but harmonic relationships are changed by the shift in frequency. The results show a marked decrease in articulation for frequency shifts of 100 to 200 c/s. This corresponds to a multiplication factor of 1.1 for frequencies in the range from 1000 to 2000 c/s. The much smaller losses shown for this factor by the upper curve suggest that the preservation of harmonic relationships and ratios of formant frequencies is of importance to speech intelligibility. There is some ambiguity in this comparison since the optimum articulation for the lower curve is about 19% less than that for the upper one. This difference arises because a carbon microphone having a resonant response was used in the carrier circuit tests.

Another method of producing frequency shift is by introducing hydrogen or helium into the vocal tract. This preserves duration time and fundamental frequency but shifts the frequencies of the formants upward. Ratios of these frequencies, however, are preserved. BÁRÁNY [19] reports vowel articulation values of 73% with hydrogen in comparison with vowel articulation of 93% with air, or a loss of 20%. He estimates that the formant frequencies were elevated by somewhat more than a factor of 2. From the upper curve of Fig. 127, a much larger loss would be expected from elevating the formant frequencies by a factor of 2. Probably the preservation of duration time and fundamental frequency in the hydrogen case accounts for the smaller observed loss.

Other methods of interest in this connection have been described but quantitative results have not been given. One is a so-called speech stretcher described by Joos [20]. This device holds both fundamental and formant frequencies fixed but doubles or quadruples duration times. Another device is described by SCHOTT as a playback for visible speech [21]. It can hold duration times fixed and shift either the

fundamental or the formant frequencies or both. Shifts of a fixed number of cycles per second or fixed ratio may be made. It is possible also to hold the fundamental and formant frequencies fixed and change duration time.

§ 6. Phase Distortion

Effects on intelligibility and quality, of the type of phase distortion that occurs in loaded lines and low-pass filters, have been reported by STEINBERG [22]. Such devices generally have curved phase characteristics. Distortion arises from differences in the slope of the phase characteristic in different parts of the frequency range. One part of the frequency range tends to be delayed with respect to the other. The expression $[(\partial\beta/\partial\omega)_f - (\partial\beta/\partial\omega)_{min}]$ is taken as a measure of the delay distortion [22], where β = phase shift in radians, and $\omega = 2\pi f$. The first term represents the slope at frequency f and the second term represents the minimum slope.

Fig. 128 shows the effects on sound articulation of various amounts of delay distortion of the type occurring in loaded lines. The delay

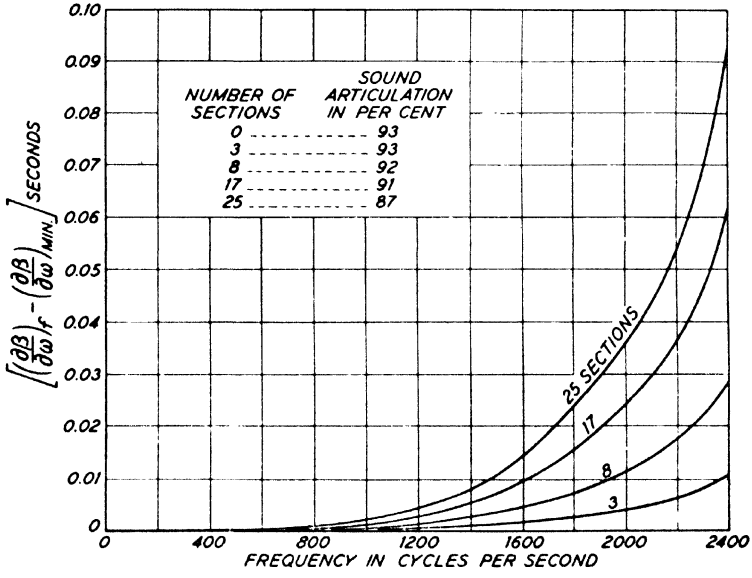


Fig. 128. Effects of delay distortion on articulation (after STEINBERG).

distortion was obtained by means of all-pass network sections arranged so that different numbers of sections could be used. The attenuation was equalized to 2500 c/s and a 2400-cycle low-pass filter having negligible phase distortion was associated with the network.

Articulation tests were also made on a system containing one or more 5000-cycle low-pass filters in series. In all cases the attenuation was equalized to 5000 c/s. Fig.. 129 shows the delay distortion and observed articulation results.

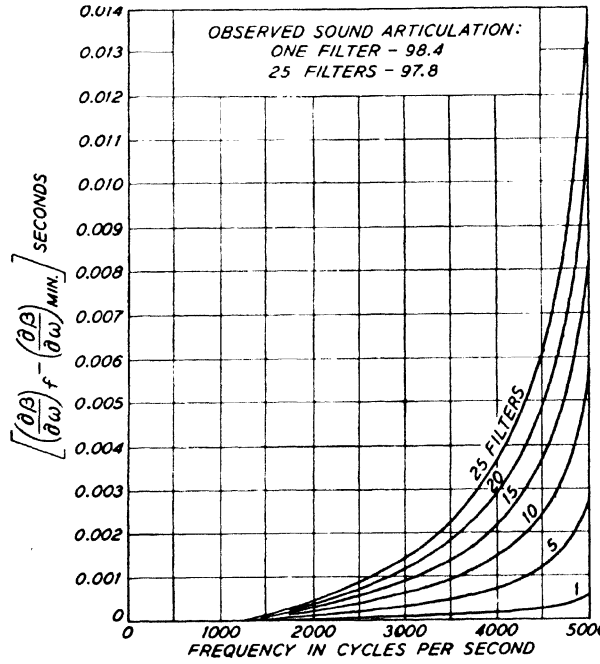


Fig. 129. Articulation and delay distortion for 5000-cycle low-pass filters (STEINBERG).

In addition to the effects on intelligibility, delay distortion of the above type is usually accompanied by audible transients which may be a source of considerable annoyance to the listener. The effects are caused by components of different frequencies arriving at different times. The obtrusion depends on the type of signal, the amount of

delay distortion, and the frequency range in which it occurs. For speech, the delay distortion of one section of the all-pass network (Fig. 128) was noticeable. The delay distortion at the cut-off frequency of 2400 c/s was .006 second. When three sections were used, the distortion was easily noticed. For the 5000-cycle low-pass filters, some number between five and ten filters in tandem caused just noticeable delay distortion. The distortion was easily noticed with 20 filters in tandem. The above figures depend somewhat upon the attenuation characteristic as the cut-off frequency is approached, since small amounts of attenuation reduce the noticeability of the effects. In general, the audible effects of delay distortion with music are very much less noticeable than with speech. For example, the delay distortion caused by 25 of the 5000-cycle low-pass filters in tandem was not noticeable when single notes were struck on the piano, or when a melody was played.

LICKLIDER [23] has found that interaural phase has an effect on the intelligibility of speech heard in the presence of noise. If speech and noise signals were mixed and led to separate receivers on the two ears of an observer, an appreciable increase in word articulation was obtained if the phase of one (but not both) of the signals was reversed in one of the receivers. The effect was small if the articulation was 70% or more; but when the noise was so severe that the in-phase articulation fell to 30%, it could be increased to 50% by reversing the speech phase in one ear. Practical use of the effect can be made when speech is being heard over earphones, in a noisy place, by reversing the connections of one of the receivers.

§ 7. Amplitude Distortion

In non-linear distortion, the instantaneous output amplitudes of a system are not proportional to the input amplitudes. The effect may be either compression or expansion, but in either case a decrease in articulation results, under optimum receiving conditions (*cf.* Ref. 14, p. 136). Under less than optimum conditions a compression may increase articulation, owing to an increase in the relative levels of the fainter sounds.

MATHES and WRIGHT [24] have described a system in which the envelope of the speech wave is compressed, but instantaneous

amplitudes remain undistorted over short intervals of time. This avoids the appearance of intermodulation frequencies, and the articulation remains high, although speech sounds slightly unnatural. Naturalness may be restored at the receiving end by expanding, the combination being called a 'compandor'. The advantage over a flat system lies in the higher level of the weak sounds while the signal is passing through a noisy environment (*e.g.* a radio wave).

A further modification of the compressor is described by BLACK and NORMAN[25]. It reduces circuit gain very quickly when amplitude increases, but requires a longer time (up to a second) for recovery when amplitude decreases. This maintains a high general level of speech compared with subsequent noise, but retains naturalness by avoiding changes from one sound to the next.

LICKLIDER[26] has investigated the effects of peak clipping (overloading). Cutting off all amplitudes higher than one-tenth of the input peaks (20 db clipping) left the high articulation of 96% for PB word lists, under quiet conditions and with sufficient listening level. Under noisy and low level conditions clipping reduced articulation, but it could be raised by reamplifying. In Fig. 130, for example,

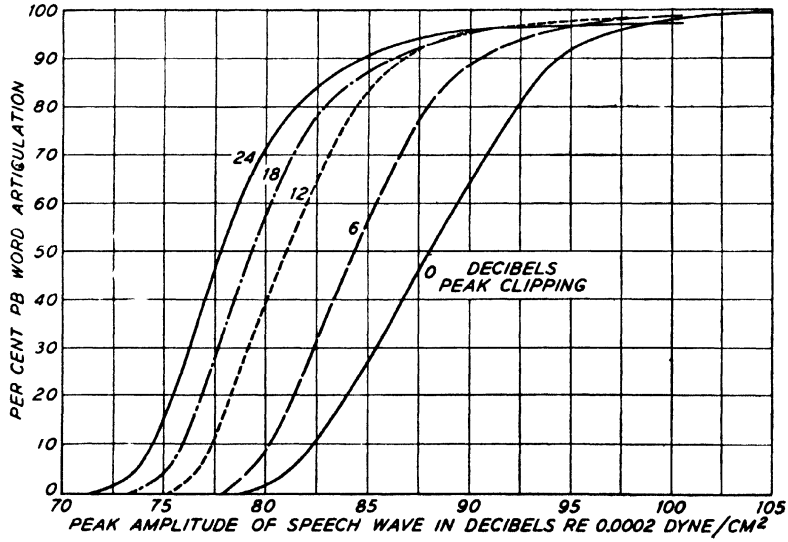


Fig. 130. Articulation for different amounts of peak clipping, plotted against peak level of the clipped speech wave (LICKLIDER).

the noise level at the receiving end was such that unclipped speech had zero articulation when output peaks were at 80 db above .0002 dynes/sq. cm. (loudness level of 80 phons). By clipping the peaks 24 db and amplifying by this amount, thus keeping output peaks constant, the articulation was raised to 75%. The effect is due to the amplification without distortion of weak consonant sounds important for understanding the speech. At high peak levels, the clipped speech is slightly less intelligible than the unclipped.

The effects of centre clipping [26] are much more serious, as shown in Fig. 131. In this case amplitudes up to a certain level are reduced to

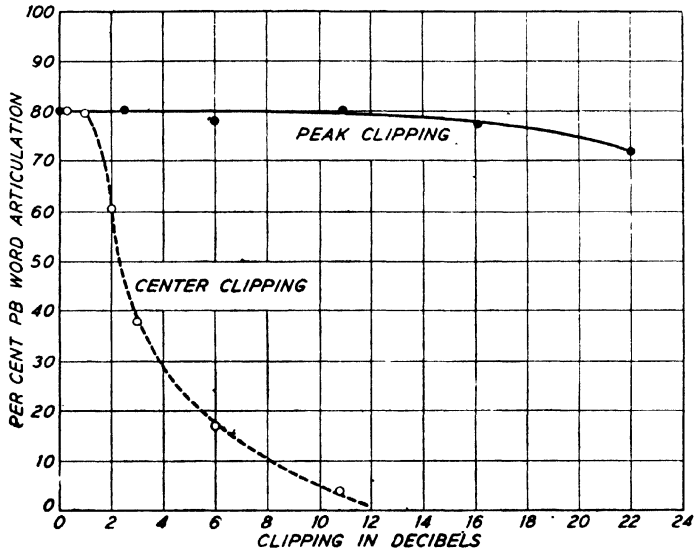


Fig. 131. Effect on articulation of different amounts of peak clipping and center clipping (LICKLIDER).

zero, while higher amplitudes are treated linearly. The clipping level is expressed in db below the input peaks. The curves of Fig. 131 start at only 80% articulation, due to noise at the speaker's location. Receiving conditions were good. The reason for the rapid fall with centre clipping is that weak consonant sounds are removed, and nothing can be done to restore them.

The effects of linear rectification [26] are shown in Fig. 132. Positive and negative amplitudes are treated differently, but for each side by

itself the treatment is linear. The abscissa of the curve is the positive input-output slope, the slope for the negative side remaining at 45°. Thus 45° represents no rectification, 0° half-wave, and —45° full-wave

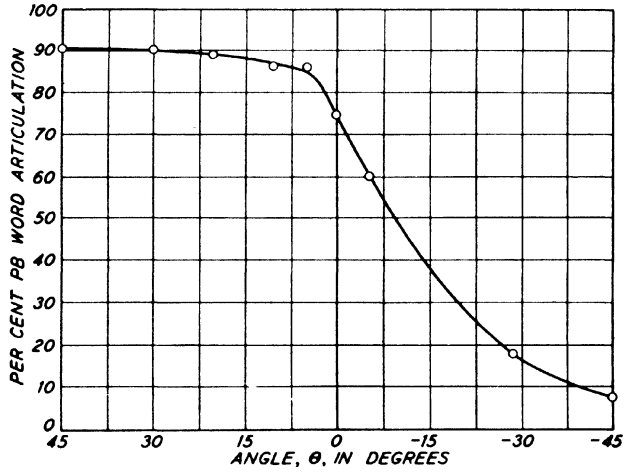


Fig. 132. Effect on articulation of linear rectification plotted against the slope of the input-output curve of one side of the speech wave, compared with +45° for the other side (LICKLIDER).

rectification. There is little drop in articulation until the half-wave condition is approached, then a rapid drop until little intelligibility is left at full-wave rectification.

The above types of amplitude distortion have a greater effect on the naturalness of speech than on the intelligibility [26]. They also are more detrimental to intelligibility if noise of a non-impulsive type is mixed with the speech before the distortion takes place, since the noise and speech then intermodulate. With impulsive noise, peak clipping may help by reducing the noise peaks.

Power may be saved by clipping speech and reamplifying, before using it to modulate a radio wave or other carrier [27]. For 24-db clipping, some 14 db of carrier power is saved without loss of intelligibility. This type of saving can also be had at high altitudes [28], although for maximum efficiency it is necessary to remove frequencies below 500 cycles per second.

LICKLIDER and POLLACK [29] have investigated complete, or infinite,

peak clipping, combined with differentiation or integration of the speech wave. The differentiation and integration are equivalent to frequency distortion, rising or falling, respectively, at 6 db per octave of frequency increase. Infinite clipping reduces the wave to a series of flat-topped pulses between axis crossings. Differentiation produces an axis crossing at each maximum and minimum of the original wave, and thus increases the number of such crossings produced by higher frequencies mixed with lower. Integration reduces the number of crossings.

The effects of different combinations of the three types of distortion are shown in Fig. 133. The rise of all curves shows the effect of in-

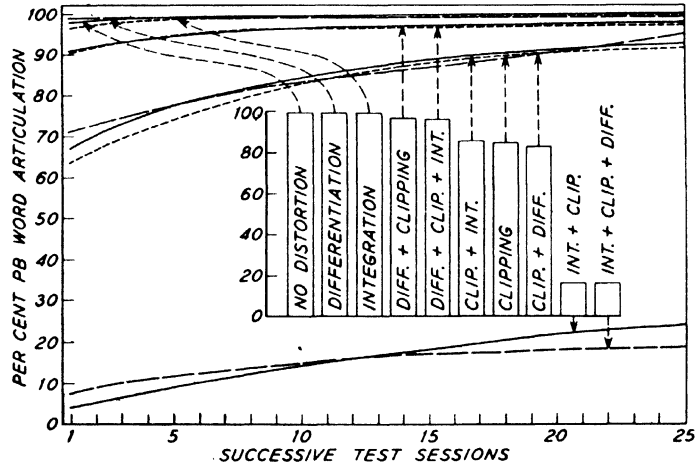


Fig. 133. Effect on articulation of differentiation, integration, and infinite peak clipping, in various combinations (LICKLIDER and POLLACK).

creasing observer experience, but at any given session the different curves are comparable. An average articulation for each curve is shown by the height of the corresponding bar in the inset figure. Differentiation or integration, alone, had little effect on articulation (although both affected naturalness). Infinite clipping, alone, reduced articulation at the beginning to 63%. Even this is surprisingly high for so extreme a distortion, for it would permit conversation with little difficulty. The most important result, however, is shown when

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the wave is differentiated before clipping, leaving an articulation of 90% for inexperienced observers. Subsequent integration does not increase articulation significantly, although it does restore considerable naturalness. If integration takes place before clipping, articulation is very low, and subsequent differentiation helps little, if at all. The advantage of infinite clipping is that it reduces the amplitude range to a simple on-off, thus permitting transmission by a relay type of action.

All amplitudes were reduced to zero for short periods of time at the beginning of test syllables, in tests reported by STEINBERG [22]. This is a type of distortion which may occur when voice-operated relays are not fast enough. The result, for the average initial consonant, was a linear reduction of 5.3% in the consonant articulation for each .01 second of time removed, up to a total of .11 seconds.

Repeated interruptions, in which speech is completely cut off from the listener for short periods regularly repeated, have been investigated by MILLER and LICKLIDER [30]. Results are shown in Fig. 134, plotted against frequency of interruption. The different curves are for different percentages of the total time during which speech was heard. For very low interruption rates, several words are heard in

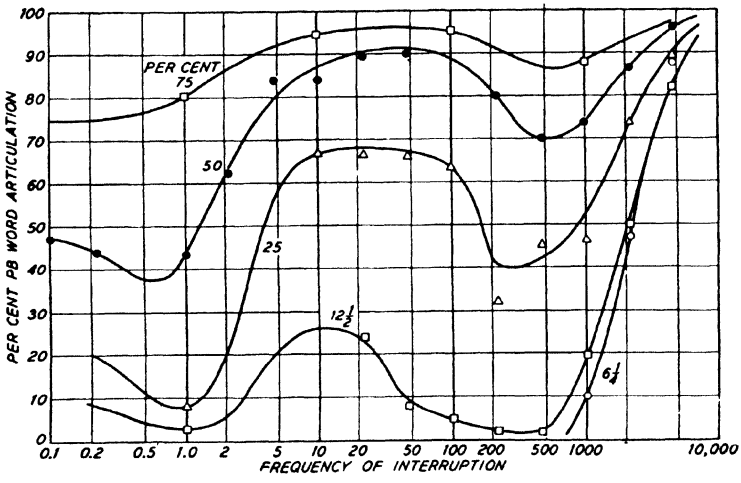


Fig. 134. Effect on articulation of interrupting speech at regular intervals. Percentages are those of the time of each interval which is still occupied by speech (MILLER and LICKLIDER).

each interval, hence the articulation is about the same as the percentage of 'on' time. As the 'on' period begins to approach the length of a word, articulation falls because few words are heard completely. It rises again as each word begins to be sampled several times. There is a dip at interruption rates between 100 and 2000 per second, which is attributed to the intermodulation products of speech and interruption waves, constituting a noise which is most damaging in this range. At higher rates, all curves rise to high values. Even when only .0625 of the time is left for speech to be heard, an articulation of 90% is obtained for rates of 6000 or more periods per second. This is what makes possible time-division systems of multiplex telephony, where one conversation occupies only a fraction of the time, the remaining time being used for sending other conversations over the same circuit.

§ 8. The Computation of Intelligibility

The effect upon intelligibility of many of the variables in transmission systems have been studied in various laboratories. Sufficient data are available to afford a systematic treatment of the main factors involved, so that it has been possible to develop methods for the prediction of the performance of a system when its measured characteristics are known and the conditions under which it is to be used are specified.

The basic relationships involved and procedures for computing the performance of a particular system under given conditions have been treated in some detail [13, 16, 31]. The approaches described are somewhat different, but all involve certain basic concepts in speech and hearing.

The methods which have been developed have afforded considerable success in the computation of the performance of a large variety of transmission systems [13].

References p. 300/301

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CHAPTER 13

MICROPHONES

For authorship, see p. 193

(A) GENERAL THEORY

§ 1. Introduction

Microphones belong to the class of substantially linear electroacoustic transducers designed to convert acoustical energy into energy of an electrical nature. The sound waves first actuate a mechanical structure of some kind * (*e.g.* a diaphragm) which is part of a mechanical vibratory system [1]. By means of an electromechanical transduction mechanism [1], coupling is effected to a pair of electrical terminals. By virtue of the fact that important classes of electromechanical couplings are bilateral, *i.e.*, capable of transmitting energy in both directions, the analysis and performance of many microphones is closely linked to that of sound generators. (See Div. V.) Indeed, the reciprocity theorem, wherever applicable, affords a valuable means of calibrating microphones in absolute terms by making use of their properties as sources of sound.

An electroacoustic transducer is said to be *passive* if all the power delivered by the transducer to the electrical (acoustical) system, to which it is coupled, is derived from power absorbed by the transducer from the acoustical (electrical) system on the opposite side. This does not prohibit the presence of sources of power within the transducer, such as polarizing voltages or currents, provided they do not release power to the electrical and acoustical systems to which the transducer is coupled. Such transducers are characterized, in general, by recipro-

* Devices based on a different principle, such as thermal microphones, are excluded from this discussion of mechanical microphones.

city. In *active* transducers, on the other hand, such power is supplied. Among them are the carbon microphone and the electronic microphone (see below). Active transducers are generally characterized by uzequal transmission in the two directions [2].

§ 2. Theory

The model of a linear passive electroacoustic transducer considered in a general theoretical treatment given by FOLDY and PRIMAKOFF [3] consists of a closed surface S (only part of which need be movable), in contact with a medium capable of propagating sound. Connection to the electrical circuit is made by means of a pair of terminals and the rules of conventional electrical circuit analysis apply. Interest is centred therefore on the acoustical side of the transducer and on the transduction mechanism.

In general, the surface S can vibrate in many modes. The transducer may be said to have as many mechanical degrees of freedom. The theory given by FOLDY and PRIMAKOFF [3] covers this general case. However, in many instances it is proper to assume a distribution of the normal component of the velocity on S which is the same for all driving and coupling conditions. The theory given by FOLDY [4] for this case is much simplified, but still retains sufficient generality for most purposes. A single mechanical degree of freedom is the consequence of this assumption—and hence a pair of mechanical 'terminals' can be specified.

(i) *Electromechanical Transduction*

On the surface S, points of which are specified by the radius vector $\vec{r}$ (from an arbitrary origin), there exists a sound pressure $p(\vec{r})$ and a velocity component $v_y(\vec{r})$ in the direction of the normal y to the surface element dS. There results a force F and velocity V at the mechanical terminals. The electrical quantities are E, the voltage across the electrical terminals, and I, the current flowing into them. The transduction of energy between these two pairs of terminals is effected by a linear electromechanical transducer. A pair of well-known linear steady-state relations exists between the two pairs of variables F, V, and E, I, which may be written as follows:

$$\left.\begin{aligned} E &= Z_b\, I + k V \\ F &= k' I + z_o\, V \end{aligned}\right\} \qquad \text{(IV, 1)}\ *$$

The parameters Z_b, z_o, k, k', which will, in general, be functions of frequency, are characteristics of the electromechanical transducer. Z_b is the blocked ($V=0$) electrical impedance [5], z_o the mechanical impedance for electrical open circuit (blocked mechanical impedance) [5] ($I=0$); k and k' are the electromechanical force factors [5] (or transduction coefficients). Explicitly, k' is the force developed per unit current when the mechanical structure is blocked; k is the open-circuit voltage developed per unit velocity.

(ii) *Coupling to the Acoustic Medium*

The mechanical quantities F and V are connected with the acoustical field quantities $p(\vec{r})$ and $V_y(\vec{r})$ as follows: [4]

$$\left.\begin{aligned} F &= \int_S p(\vec{r})\, \bar{\psi}(\vec{r})\, \mathrm{d}S \\ V_y(\vec{r}) &= V\, \psi(\vec{r}) \end{aligned}\right\} \qquad \text{(IV, 2)}$$

where $\psi(\vec{r})$ is a function of position independent of the excitation of S and $\bar{\psi}(\vec{r})$ is its complex conjugate. For any value V the velocity distribution $V_y(\vec{r})$ over S can be obtained, provided $\psi(\vec{r})$ is known. The driving force F is obtained from the pressure distribution $p(\vec{r})$ and a knowledge of $\psi(\vec{r})$.

(iii) *Radiation*

Radiation of sound takes place since the surface S is moving. The sound pressure $p(\vec{R})$ at an arbitrary point P with radius vector $\vec{R}$ can be computed by using an appropriate GREEN's function [3, 4]. This function can be shown to exist in all cases of practical importance and it is possible to derive important relationships without its explicit knowledge. This GREEN's function gives essentially the pressure contribution at P due to an element dS vibrating with unit normal velocity.

* The sinusoidal time dependence exp ($i\omega t$) is suppressed throughout and the customary complex notation is used; $\omega = 2\pi f$ is the pulsatance. The positive directions of E, I, F, V are chosen so that, when there quantities are essentially positive, power tends to flow into the transducer at both terminal pairs.

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The total pressure $p(\vec{R})$ at P is obtained by multiplying the GREEN's function by $V_y(\vec{r})$ and integrating over S. With the process of radiation, there is associated a mechanical impedance as 'seen' from the mechanical terminals, the so-called radiation impedance z_r (see below and Chap. 1 § 15).

(iv) *Diffraction*

When a microphone is introduced into a sound field, the sound pressure $p(\vec{r})$ on its surface S will, in general, be different from the free-field pressure $p_{ff}(\vec{r})$ which existed at that point prior to the introduction of the instrument. This effect is due to diffraction. Sound diffraction results in distortion of the wave front of the incident wave due to the fact that part of the incident energy is re-radiated or scattered by the obstacle, thus interfering with the incident wave field. In accordance with the general laws of radiation this will take place to an appreciable extent for all but very long wave-lengths, where $\lambda \gg L$, where L is a significant linear dimension of the transducer surface S and λ the wave-length. On the other hand, for very short wave-lengths the obstacle S will cast a sharp 'shadow' in the direction of the propagation of the incident wave in the sense of geometrical optics. The sound scattered by S will then be in such magnitude and phase as to interfere destructively with the incident field in the 'shadow' zone, and, generally, to reinforce the incident field on the 'illuminated' side of S. Moreover, diffraction causes directivity, *i.e.*, the output of the microphone will depend, in general, on the orientation of the device with respect to the direction of the incident wave.

The total pressure $p(\vec{r})$ on the surface S can be shown [3, 4] to be the sum of three contributions: first, $p_{ff}(\vec{r})$, the pressure in the undisturbed incident wave; second, $p_s(\vec{r})$, the pressure scattered by S when it is held rigid, and third, $p_r(\vec{r})$, the pressure contribution due to the fact that, in general, S is not rigid but moves with velocity $V_y(\vec{r})$. As can be seen from Eq. (IV, 2), this third and last contribution is proportional to V.

To obtain the force F acting on the mechanical terminals, it is only necessary to integrate the three pressure contributions suitably over S and one obtains an equation of the form

$$F = F_0 - V z_r \qquad \text{(IV, 3)}$$

Only the first two pressure contributions above give rise to the force F_0, whereas $p_r(\vec{r})$ gives rise to the term $-V\mathfrak{z}_r$. F_0 is the force acting at the mechanical terminals when S is held rigid. In THÉVENIN's sense, then, the coupling of the transducer to the acoustic medium is described, as seen from the mechanical terminals, in terms of the blocked-motion force F_0 of a mechanical generator * and its internal mechanical impedance $\mathfrak{z}_r$, which is the radiation impedance, by definition. Substitution of Eq. (IV, 3) into the second of Eqs. (IV, 1) yields

$$F_0 = k'I + (\mathfrak{z}_0 + \mathfrak{z}_r)V \qquad \text{(IV, 4)}$$

For most practical microphone structures used in air $\mathfrak{z}_r \ll \mathfrak{z}_0$, and F may be replaced by F_0 in Eq. (IV, 1). This means that the mechanical driving force is substantially equal to the force across the mechanical terminals when the surface S is held rigid. It is therefore important to examine the diffraction effect caused by S under these conditions by measuring the pressure distribution on S, held rigid, and comparing it with the free-field pressure.

Exact solutions of the diffraction of plane waves by rigid finite obstacles and expressions for the resulting pressures on the surface are very few. RAYLEIGH [6] has solved the spherical problem and SCHWARZ [7] has tabulated the diffraction effect on the surface of a sphere in phase and magnitude (*cf.* Chap. 1, § 17). Numerical solutions have also been obtained for the problem of sound diffraction by a rigid circular disk of zero thickness, for normal incidence, using spheroidal wave functions [8]. WIENER [9] has compared these computations with experiment and obtained, in addition, experimental data for oblique incidence. Rigid circular and square plates as well as cylinders and cubes have been investigated by SIVIAN and O'NEIL [10] and MULLER [11] *et al.* These authors also developed an approximate theory (related to KIRCHHOFF's optical diffraction theory) which was used to compute the diffraction effect at the centre of the faces of the obstacles. The experimental approach is in many cases the only feasible one, especially in practical design problems. The principle of dynamical similitude applicable to diffraction theory is frequently used to solve a problem by model experiments [12]. In cases where the diaphragm is mounted in a recess (*e.g.* condenser microphones), it may be practically

* F_0 is independent of V.

useful to separate the effect of this cavity (cavity resonance) from the total diffraction effect [12].

(v) *Response* [5, 13]

The voltage response (sometimes called sensitivity) of a microphone for a given frequency is defined as the open-circuit voltage developed per unit sound pressure, and is often expressed in logarithmic units relative to some convenient reference level. There are two important ways to specify this pressure. If the pressure is specified as that uniformly activating the active part of the microphone surface S (diaphragm) then one speaks of the 'pressure response.' * If, on the other hand, the pressure is specified as that existing at the microphone location prior to its insertion into the sound field, one speaks of the 'free-field response'. To specify this type of response unambiguously it is necessary to choose an acoustical centre and axis on the transducer, and to specify the incident sound wave. The difference of the free-field response and the pressure response of a pressure microphone, expressed in decibels, is often referred to as the 'free-field correction' [13].

Other types of response are frequently defined by considering electrical quantities other than the open-circuit voltage. In the majority of cases, the response phase angle [13] is of minor importance and only the magnitude of the response is considered.

(vi) *Directivity*

The directional response pattern [5] (directivity pattern) of a microphone describes, at a given frequency, the variation of the free-field response with the angle of incidence. The 'directivity factor '[5], of a microphone, and the 'directivity index '[5], are other useful measures of microphone directivity. They can be computed from the directional pattern [14]. Transducers obeying the reciprocity theorem have the same directivity factors on receiving and transmitting [4].

(vii) *Noise*

The sound pressure which can be linearly transduced by any microphone is, of course, limited (a) by overloading on the upper side,

* It should be noted that this concept is applied only to pressure microphones (see below).

and (b) on the lower side, by inherent noise. After noise due to associated preamplifiers, stray pick-up, etc., has been allowed for, there still remains thermal noise [15] generated in the resistive component of the effective microphone impedance as seen from the electrical terminals. For electrical open circuit and a temperature of about 20°C the root-mean-square noise voltage in a frequency band one cycle per second wide is given by

$$E_{\text{noise}} = 1.27 \cdot 10^{-10} \sqrt{\text{Re}(Z)} \text{ volts/c/s.} \qquad \text{(IV, 5)}$$

where Re(Z) is the real part of the microphone impedance in ohms. For a non-dissipative transducer, *i.e.*, where Z_b and ζ_0 are pure imaginaries [4], there still would be in evidence the noise associated with the radiation resistance which can be interpreted as being due to the random motion of the molecules of the medium in thermodynamic equilibrium with the transducer. The sound pressure which may be thought of as being the cause of the open-circuit noise voltage of a microphone is called the 'threshold pressure' [5]. WEBER [16] has computed the threshold pressure in a band approximately 10 kc wide for microphones of various types.

(viii) *Reciprocity*

The electroacoustic reciprocity theorem [5], as applied to the free-field case, asserts essentially that, for a given frequency, the ratio of the steady-state free-field response as a microphone (due to a spherical wave generated by a simple source at a point $\vec{R}$ a large distance away) to the response of the transducer as a sound generator * measured at $\vec{R}$ is independent of the nature and construction of the transducer. This ratio equals $2d\lambda/\rho c$ in magnitude, ** where d is the distance between $\vec{R}$ and the (arbitrary) acoustic centre of the transducer, ρ the density of the medium and c the velocity of propagation of sound. In order to satisfy the theorem, a transducer must meet the necessary condition [3, 4] $|k| = |k'|$†. This imposes a requirement on the electromechanical transduction mechanism. Another condition to be satisfied

* Defined here as the ratio of the sound pressure generated at $\vec{R}$ to the current energizing the transducer as a sound source.

** In cgs units; if the responses are to be expressed in terms of volts and amperes, the above ratio of the responses must be multiplied by 10^{-7}.

† In appropriate units.

has to do with the velocity relations on S. Most, if not all, electroacoustic transducers in use today utilizing reversible electromechanical couplings obey the reciprocity theorem to a high degree of approximation. McMillan [17] has shown, however, that transducers using combinations of certain types of individually reciprocal couplings or certain combinations of individually reciprocal transducers may not satisfy the reciprocity relations.

Schottky [18], Ballantine [19], MacLean [20] and Cook [21] have discussed various aspects of the theorem. Its importance lies to a considerable extent in the fact that it may be used to obtain the response of electroacoustic transducers in absolute terms without an acoustic standard.

The electroacoustic reciprocity theorem can be recast to cover specifically the case where the pressure response is involved. To insure uniform pressure over the diaphragm, the transducer is coupled to a cavity whose acoustic impedance ζ is so chosen as to constitute an effective acoustic 'short circuit' [20], [21]. The ratio of the pressure response of a transducer satisfying the reciprocity conditions to the response as a sound generator, so terminated, is given then by $|\zeta|^{-1}$, in cgs units.

(ix) *Efficiency*

Although microphones are not primarily designed to transfer a maximum of power, there can be formulated certain theoretical relationships [4] involving suitably defined efficiencies [5] which are of practical importance.

(x) *Equivalent Circuits*

It is a matter of importance that, in practice, acoustic networks are frequently inserted between the transducer surface S and the medium, and that ζ_0 is the mechanical impedance of a network containing acoustic elements. This is done to attain a specific design objective, in most cases a uniform response over a wide frequency range, by adding effectively new degrees of freedom. The lumped element approach is usually satisfactory in analyzing these networks.

In applied acoustics it is customary to represent a microphone with

its associated acoustical and electrical networks by an equivalent electrical circuit. This circuit can be arrived at either by using generalized LAGRANGE's Equations [12, 22] or by inspection, using suitable systems of electromechanical and electroacoustic analogies [23, 24, 25].

(xi) *Measurements*

This important subject, too extensive for inclusion here, is surveyed in detail by BERANEK [26] (see also Div. I).

(B) CLASSIFICATION OF MICROPHONES

There are many ways of classifying microphones. Besides active and passive, there are also *mechanical*, *thermal*, and *optical* types whose operation depends upon some aspect of the sound wave such as pressure, particle velocity, condensation, temperature or index of refraction. The following discussion will be concerned mainly with mechanical types which are in most general use.

§ 3. Types of Electromechanical Transduction Mechanisms

Microphones of the passive linear type can be classified electrically according to the electromechanical coupling utilized. Of the types in use today most all of them satisfy the condition $k = -k'$ where k and k' are the electromechanical force factors appearing in Eq. (1). This means that generally for such transducers the reciprocity law holds. In particular, for electrostatic, piezoelectric and electrostrictive coupling, $k = k'$. Here the force factors are pure imaginaries and inversely proportional to frequency. As a result, the open-circuit voltage generated is proportional to and in phase with the mechanical displacement. For electromagnetic and magnetostrictive coupling, $k = -k'$. Here the force factors are real and independent of frequency. As a result, the open-circuit voltage is proportional to and in phase with the mechanical velocity. A short description of these and various other voltage generating mechanisms is given below.

(i) *Electrostatic Coupling*

In the condenser microphone [27], a pair of closely spaced conducting

plates, one fixed and the other a movable membrane, provide a condenser of variable capacity. The condenser is charged by the application of a d.c. voltage through a high resistance network. Displacement of the movable membrane by an applied force varies the capacity and consequently the condenser voltage (for constant charge). The open-circuit alternating condenser voltage is proportional to the membrane displacement.

(ii) *Piezoelectric Coupling*

The piezoelectric effect is used in crystal transducers [28, 29]. Deformation of a crystalline material, such as quartz or Rochelle salt, by an applied force results in electrical charges appearing on opposite surfaces. No bias or polarization is necessary since the signs of the charges are dependent upon the direction of the deformation. Electrodes placed on a pair of these surfaces collect the charge and provide an open-circuit voltage which is proportional to the displacement.

(iii) *Electrostrictive Coupling*

The application of an electric field to certain dielectric polycrystalline materials [30, 31] results in an appreciable deformation which is proportional to the square of the electric field and independent of the field direction (electrostriction). For microphone use, the material is polarized by the application of a high initial voltage; after removal of the external field the polarization is retained. An incremental deformation by an external force will then be accompanied by a corresponding incremental change in polarization. The resulting alternating electromotive force is proportional to the displacement for small amplitudes.

(iv) *Electromagnetic Coupling*

Transducers utilizing this type of coupling are of two forms [23]. In the first, often called moving armature or magnetic type, the armature is part of a magnetic circuit containing one or more air gaps. A suitable steady magnetic flux is provided. Variations in the length of the air gaps caused by the armature displacement produce corresponding

variations in the magnetic flux linking a suitably placed pick-up coil. The resulting electromotive force is proportional to the armature velocity.

Most microphones with electromagnetic coupling, however, use the second form, often called moving conductor or electrodynamic type. A conductor is placed in an essentially uniform magnetic field. Motion of this conductor gives rise to an electromotive force which is proportional to the rate at which the conductor links more or less flux lines to its electrical circuit and hence is proportional to the velocity with which it moves. Moving coil [32] and ribbon microphones [33] are examples of this form.

(v) *Magnetostrictive Coupling*

The application of a magnetic field to certain magnetic materials, notably the nickel-iron alloys, is accompanied by a deformation which is proportional to the square of the flux density and independent of the field direction (magnetostriction) [23]. For microphone use, the material is magnetically polarized. The voltage generated in a coil coupled to the magnetic circuit is dependent upon the rate of change of flux and hence proportional to the rate of change of deformation by an external force for small displacements.

(vi) *Variable Resistance Coupling*

Although microphone theory has emphasized the passive linear transducer and its reciprocity aspect, there is an active irreversible type which is important because of its universal use in telephone practice. In the carbon microphone, the variable resistance principle is employed [1]. A cell containing carbon granules is used in which the displacement of a moving electrode in contact with the granules causes a change in the cell resistance. A battery in the circuit supplies a biassing current through the cell. As the cell resistance is varied by an applied force, the voltage drop across it is made to vary. The alternating voltage is proportional to and in phase with the displacement of the moving electrode. Similar variable resistance effects can be

obtained with single and multiple points of carbon or metallic conductors, liquid cells, flexible carbon wafers, etc [34].

Another active transducer whose output voltage is proportional to the displacement of the moving element is found in the electronic microphone [35]. Here the position of an electrode in a vacuum tube is controlled by the applied force through an interconnecting mechanical structure. A vacuum seal must, of course, be provided.

§ 4. Acoustic Classification

Mechanical microphones can also be classified according to the manner in which the surface S′ of the responsive element is coupled to the acoustic medium.

(i) *Pressure Microphone*

A pressure microphone is characterized as one where essentially only one side S' of the closed surface S is movable. The typical example is a diaphragm structure mounted in a housing. If, in addition, the open-circuit mechanical impedance of the diaphragm z_0 is high compared with the radiation impedance z_r, the force F at the mechanical terminals is given by $F = \int_{S'} p_{ff}(\vec{r})\,\vec{\psi}(\vec{r})\,\mathrm{d}S$. The integration is carried out over the diaphragm of area S' and it is assumed that diffraction effects are negligible. Such a microphone is, of necessity, non-directional and measures the free-field pressure p_{ff} in the sound wave. The electromechanical transduction mechanism and the controlling element in the mechanical impedance are chosen so that the ratio of the electromotive force to the sound pressure is essentially independent of frequency over the operating frequency range.

(ii) *Pressure Gradient Microphone*

A pressure gradient microphone of order one is a microphone where the active parts of the surface S are essentially symmetric about a plane. The typical example is the microphone with an active element lying entirely in one plane, with both sides accessible to the sound

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wave (ribbon microphone *). Under the assumptions made above for the pressure microphone, the driving force is obtained by substituting the free-field pressure difference at the two sides of S' into the integral above. This difference is proportional to the pressure gradient taken in the direction of the normal to S'. This makes the microphone a bidirectional device with a cosine characteristic for plane progressive waves. For plane and spherical waves, the pressure gradient is proportional to frequency and particle velocity, hence the name velocity microphone is sometimes used. However, the controlling mechanical element and the electromechanical coupling is chosen so that the ratio of the electromotive force to the sound pressure is independent of frequency. A pressure gradient microphone of order one is bidirectional [5]. Pressure gradient microphones of higher order [36] can be compounded by using two pressure gradient microphones of the next lower order and connecting the electrical outputs in opposition.

§ 5. The Controlling Mechanical Elements

Microphones are sometimes classed according to the element in control of the mechanical system [1]. The selection of this element is dependent upon the manner in which the mechanical driving force F depends on the sound pressure p and whether the open-circuit voltage E_0 is proportional to the displacement x or velocity $\dot{x}$ of the moving member.

In a pressure microphone where $F \sim p$ and, say, $E_0 \sim x$, it is necessary, in order to keep the ratio E_0/p constant, to hold F/x constant. Hence, $F/\dot{x} \sim (i\omega)^{-1}$, *i.e.* the mechanical impedance must be a pure stiffness and the device operated below resonance (*stiffness control*). If, however, a transduction mechanism is to be used in which the open-circuit voltage is proportional to $\dot{x}$, $F/\dot{x}$ must be held constant over the operating range (*resistance control*). A first order pressure gradient microphone, where $F \sim i\omega p$ and, say, $E_0 \sim \dot{x}$, is generally operated above mechanical resonance, so that $F/\dot{x} \sim i\omega$ (*mass control*).

§ 6. Thermal Transducers

In this type of transducer, advantage is taken of thermal effects produced by the sound wave. The most common form (hot-wire

* However, if one side of the ribbon is acoustically shielded, the device has the configuration of a pressure microphone.

microphone) utilizes the cooling effect of the air particles moving back and forth across a wire or grid heated by an electric current [37]. A bridge circuit is used to detect the resultant change in resistance which is proportional to the square of the particle velocity. For microphone use, a polarizing stream of air may be passed across the heated element to increase the sensitivity and reduce the double frequency distortion. The alternating component of the voltage across this variable resistance is then nearly proportional to the product of the stream velocity, the particle velocity, and the cosine of the included angle.

Other thermal microphones [38] take advantage of the temperature changes in a sound wave. Corresponding electrical changes are reproduced in a probe of small heat capacity.

(C) MICROPHONE CHARACTERISTICS

§ 7. Carbon Microphones

Because of its high sensitivity and simple and rugged construction, the carbon cell is used in telephone transmitters [39] (see Fig. 135). The cell is actuated by a diaphragm exposed on one side to the sound field. Resonances in the operating frequency range are utilized to achieve high sensitivity. For a battery current of about 0.06 amperes the sensitivity is of the order of -40 db (re 1 volt dyne/cm^2) and the

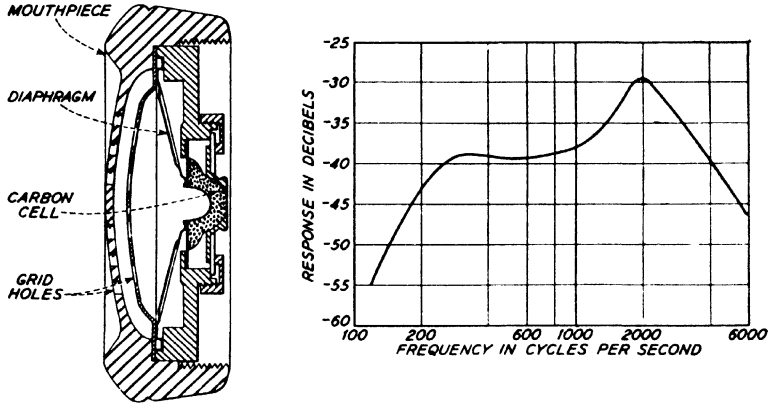


Fig. 135. A carbon microphone for telephone use. In the voice range the sensitivity is better than −40 db (re 1 volt per dyne/cm^2) (JONES [39]).

impedance is about 40 ohms. The sensitivity is not constant, but varies somewhat with position of the instrument and the 'packing' of granules [40]. The sensitivity increases with the sound level, making it possible to achieve some discrimination against ambient noise. When ambient noise conditions are severe enough to cause mechanical overload, attenuating acoustic screens are used to reduce the sensitivity so that the talker can be understood if his speech level is above the noise. For simple public address systems, high sensitivity microphones employing 'non-burning' carbon granules are used directly in series with a battery and a moving coil loud speaker.

Practically all carbon microphones are subject to distortion due to the inherent non-linearity of the carbon cell. Also, contact and burning noises occur, so that where linearity and quiet operation is essential, other microphones are preferred. However, the double button microphone [41] has seen considerable use. In this microphone, the distortion is reduced by using a matched pair of carbon cells in a push-pull coupling circuit which cancels the distortion products of even order.

§ 8. Condenser Microphones

A typical condenser microphone [40] consists of a stretched stainless steel diaphragm spaced about 0.001″ away from a grooved back electrode (see Fig. 136). Its mechanical system is stiffness controlled. Damping of the diaphragm resonance is accomplished by motion of

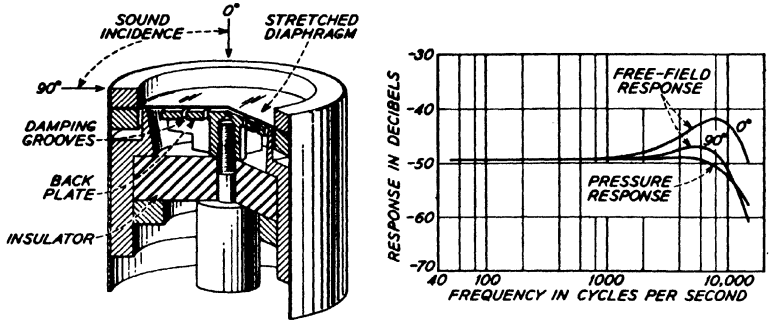


Fig. 136. A high quality condenser microphone for laboratory or studio use. A sensitivity of −50 db (re 1 volt per dyne/cm² open circuit) is typical. It is shown how the free-field response deviates from the pressure response at high frequencies due to diffraction and cavity resonance.

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the displaced air passing through the air gap between the electrodes. With a polarizing voltage of 200 volts, the sensitivity is about −50 db (re 1 volt per dyne/cm^2). Microphones of this type are quite stable and are generally used for accurate laboratory measurements [40]. They are often fitted with calibrated tubular probes for exploring sound fields in confined spaces. The high electrical impedance (about 40 $\mu\mu$f capacity) of this microphone makes it necessary to provide a preamplifier (to which the microphone is generally attached) so that a low impedance output can be obtained for use with the long cable runs often required.

For special applications, such as improved low-frequency response and improved signal-to-noise ratio, condenser microphones have been used in bridge circuits [42] and modulation schemes [43]. Some of these circuits are coming into commercial use.

§ 9. Crystal Microphones

The most commonly used crystal units are the bender and twister bimorph * types [1]. Fig. 137a shows how charges appear on a single crystal in tension. The charges reverse sign in compression. Fig. 137b

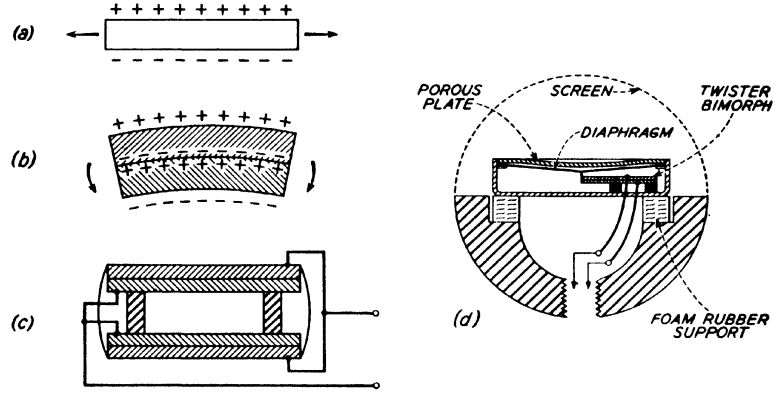


Fig. 137. Crystal microphones: (a) a single crystal in tension showing the charges developed (the signs reserve in compression); (b) two crystals of opposite polarity cemented together form a bender bimorph unit; (c) a directly actuated crystal cell formed by a pair of bender units exposed only on one side to the sound field; (d) diaphragm-actuated crystal using a twister bimorph (De Boer [1], Beranek [40]).

* The term bimorph originates with the Brush Development Company, Cleveland, Ohio, U.S.A. [40].

shows two crystals of opposite polarity cemented together and stressed in flexure. The upper crystal is in tension and the lower is in compression so that opposite charges appear on the outer surfaces. A microphone can be made in the form of a direct actuated cell as shown in Fig. 137c by using a pair of such bimorph elements and allowing only one side of each to be exposed to the sound pressure. Cells are sometimes stacked in multiple to reduce the electrical impedance. A much used construction employing diaphragm and a twister bimorph (held at 3 corners and actuated at the fourth) is shown in Fig. 137d. These microphones are pressure operated and stiffness controlled. Since the acoustical impedance is very high, damping at the resonant frequency can be accomplished by a porous disk of sintered bronze which also acts as an effective wind screen. The sensitivity of Rochelle salt crystal units is about −50 db relative to 1 volt per dyne/cm^2 on open circuit. The capacity is about .002 μf, a value which allows the use of moderate lengths of cable. Operation is limited to temperatures between −40° to 130°F. Above 130°F Rochelle salt loses its piezoelectric properties permanently. Units are sealed for protection against humidity changes.

A relatively new synthetic piezoelectric material, ammonium dihydrogen phosphate—A.D.P. for short—does not have many of the limitations of Rochelle salt and is therefore replacing it in many applications. MASSA [44] has described a small pressure microphone using A.D.P. expander bars. Since the sensitivity is quite low, of the order of −93 db (re 1 volt per dyne/cm^2 open-circuit), and the impedance high, these units are used in conjunction with an adjacent preamplifier.

§ 10. Ceramic Microphones

These microphones are operated essentially like their crystal counterparts [45]. For barium titanate bimorph units, the sensitivity is about 10 db down from Rochelle salt crystal units of similar construction. However, the ceramic units are quite unaffected by temperature and humidity.

§ 11. Moving Coil Microphones

This microphone is constructed with a thin aluminium diaphragm to which is fastened a light coil of edge-wound aluminium ribbon [46] (see Fig. 138). The coil projects into an annular gap in the magnet structure.

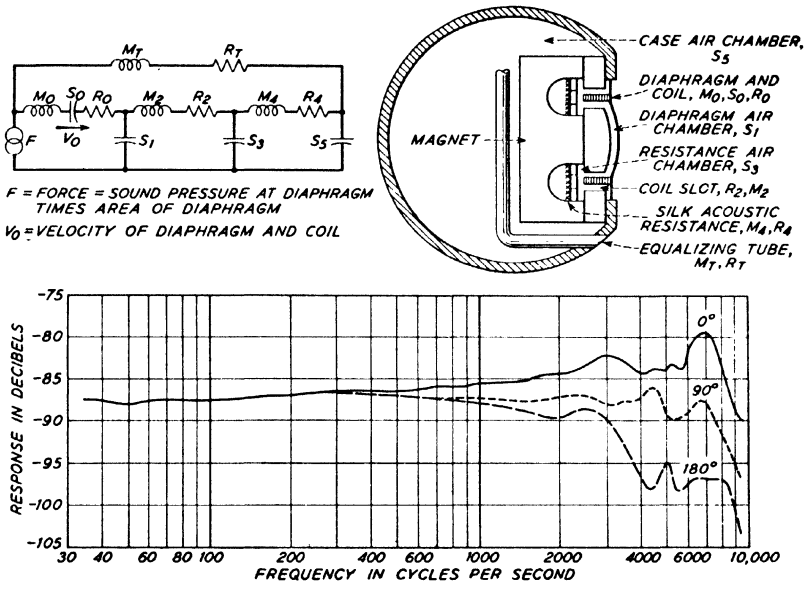


Fig. 138. A moving coil microphone with its acoustical and mechanical system represented in electrical circuit notation. The response is relative to 1 volt per dyne/cm² (MARSHALL and HARRY [46]).

The mechanical system is resistance controlled. Resistance is provided by the motion of trapped air through confining passages or acoustic resistance screens. To extend the low-frequency range, a pressure-equalizing tube leading to the back of the instrument introduces a low-frequency resonance. The open-circuit sensitivity of this type of microphone is about −85 db (re 1 volt per dyne/cm²) at a coil impedance of about 25 ohms.

The inductor microphone, an example of which is discussed by OLSON [24], is similar to the moving coil pressure type except that the diaphragm is attached to a single conductor moving in a straight gap in the magnet structure. Because of the low impedance of the conductor, a transformer is mounted in the microphone housing to provide higher output impedance levels. For 250 ohms the sensitivity at this impedance level is about −80 db (re 1 volt per dyne/cm²).

A moving coil velocity microphone described by DE BOER [1] is similar in construction to its pressure prototype except that the magnet

structure is altered to allow free access to the back of a mass controlled diaphragm.

§ 12. Ribbon Microphones

The ribbon velocity microphone [24, 33] is a pressure gradient transducer. It employs a single conductor consisting of a corrugated ribbon of aluminum foil suspended vertically in the gap of a magnet structure. The ribbon impedance is essentially a mass reactance. The construction, as shown in Fig. 139, is such that the back of the ribbon is also exposed to the sound wave, the magnet structure acting as a short baffle. The high-frequency limit is reached as the distance from the edge of the baffle to the rear of the ribbon approaches a wavelength. At this frequency (and at integral multiples) the equalization of pressure causes response nulls. In addition, since the transducer element is mass-controlled, the response, normally flat, begins to fall at the high frequencies as the ribbon starts to act as a pressure device because of diffraction.

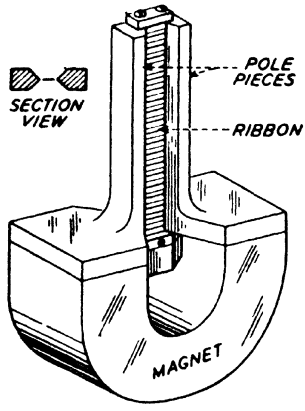


Fig. 139. Ribbon velocity microphone (BERANEK [40]).

At low frequencies, the pressure sensitivity of a velocity microphone increases at short distances from the sound source, that is, for the same average pressure as at greater distances, the gradient is much steeper and the particle velocity correspondingly greater. In a standing wave pattern, the velocity microphone indicates maxima at velocity loops (pressure nodes) and minima at velocity nodes (pressure loops) [33, 47]. Because the ribbon element is extremely light, the microphone is provided with a windscreen to protect it from damaging air currents encountered outdoors. It is quite useful in broadcast studio work because of its bidirectional characteristics. The sensitivity is about -80 db (re 1 volt per dyne/cm^2) with a transformer providing an output impedance of about 250 ohms.

The ribbon microphone can also be operated as a pressure device, resistance controlled [24]. In this case the back of the ribbon is closed off by a terminating acoustic resistance network.

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§ 13. Hot Wire Microphone

This instrument devised and developed by TUCKER and PARIS[37] consists of a HELMHOLTZ resonator with a grid formed of a 0.0006 cm diameter platinum wire bent into the shape of a grid and placed at the entry of the neck into the main body of the resonator, and a current passed by the terminals through the platinum wire until it is just below red-heat. Under these conditions a very slight motion of the air through the neck is sufficient to change the resistance by a measurable amount. The change in resistance in the alternating draught which takes place in the neck when the resonator is responding to a tone, partakes both of a steady drop, and of a complex periodic change containing the fundamental and harmonics of the frequency of the air motion. It is possible to measure the amplitude of the response, either by means of a vibration galvanometer tuned to the fundamental of the resonator, or else by measuring the average resistance of the hot wire during the motion, by a simple WHEATSTONE Bridge method (Fig. 140a).

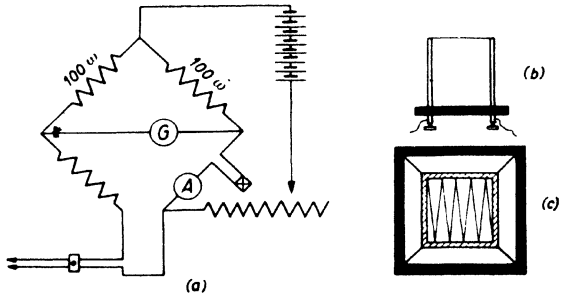


Fig. 140. Hot-wire microphone (a) circuit (b) simple wire (c) grid type (RICHARDSON [47a])

For small velocities it is found that the steady drop in resistance is proportional to the square of the particle velocity; for larger signals this relation tends to become linear [47a], or to the intensity of the sound therein. The grid is more sensitive, *i.e.*, shows the greater resistance change to a given amplitude of air motion, as the heating current is increased and the wire diameter reduced. Normally, the grid is kept just below red heat. Apart from its use in analysis, the instrument, as

its name indicates, serves as a sensitive detector for sounds of the frequency of the resonator.

The cooling of a hot wire by the alternating velocity in a stationary wave system forms a convenient method of plotting velocity amplitudes in a fluid column (*cf.* Chap. 3, § 3c).

Two suitable types of mounting for a thin platinum or nickel wire are illustrated (Fig. 140b, c). By the use of such a device in conjunction with a manometer for pressure amplitude the impedance at points can be measured.

§ 14. Differential Microphones

The property of pressure gradient microphones to discriminate in favour of sound sources located in the vicinity of the transducer, relative to waves originating far away, has already been discussed. If such an instrument is used close to the lips of the talker, it will discriminate against ambient noise. In addition, its bidirectional characteristic provides a further advantage [48].

Carbon differential microphones have been designed for use in noisy locations. Some are strapped across the user's upper lip (see Fig. 141). The instrument is stiffness-controlled, providing a uniform frequency response for close-up speech sounds. For sounds from distant sources (noise) it has a lower response which rises with frequency.

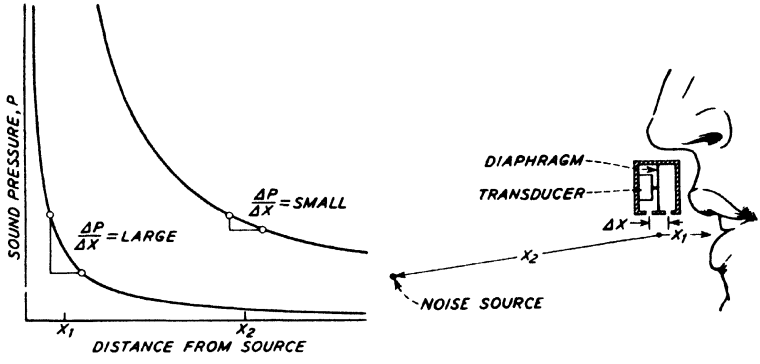


Fig. 141. A differential microphone of the military type for use under conditions of high ambient noise. Close-up speech sounds are favoured because of their steep pressure gradient even though the actual pressure of distant sounds may be higher.

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Some differential transducers are referred to by their designers as dipole microphones. In one form, the microphone is made an integral part of a telephone operator's headset [49]. A pair of tubes leading out from the two sides of a diaphragm structure measure the pressure gradient close to the lips of a talker. BEAVERSON and WIGGINS [50] have described a second order gradient microphone employing a single diaphragm structure with multiple ports. With respect to a first order gradient microphone, the directionality is improved and the differential discrimination increases, but the sensitivity is relatively lower.

§ 15. Directional Microphones [24, 40]

Useful measures of the directivity of microphones are the directivity index and the directional pattern. In many applications, directivity is a desirable feature; moreover, it is often desired that the directional pattern be of a particular shape. The obvious way to achieve high directivity is to use instruments which are large compared with the wavelength. The higher the ratio of dimensions to wave-length the higher is the directivity. For many applications, however, it is desirable that the directional patterns be independent of frequency. This can be realized by using suitable microphone types or combinations of them whose dimensions may be small compared with the wavelength. But in practice, only moderate degrees of directivity have been achieved. This class of devices will be discussed first.

Ordinary pressure or zero order gradient microphones are non-directional in the low and medium audio frequency range. Above that range they tend to become somewhat directional as diffraction effects make themselves felt. Pressure gradient microphones of order one or higher have bidirectional characteristics (Fig. 142) of the figure-8 type. The ratio of the response at a given angle θ with respect to the response in the direction of maximum output is proportional to $\cos k\theta$, where k is the order of the gradient microphone [5]. This holds where the path-length difference between the two sides of the movable element is small compared with the wave-length. At higher frequencies the directional characteristics are altered due to diffraction.

To obtain cardioid or other desirable unidirectional [5] characteristics (see Fig. 142) a pressure microphone and a gradient microphone are combined. This can be done by using two separate units and combining

References p. 326–328

their electrical outputs [46]; by using a ribbon instrument one-half of which is operated as a pressure gradient device and the other as a pressure operated unit [24]; or by exposing one side of the movable element of a pressure gradient microphone to the incident sound

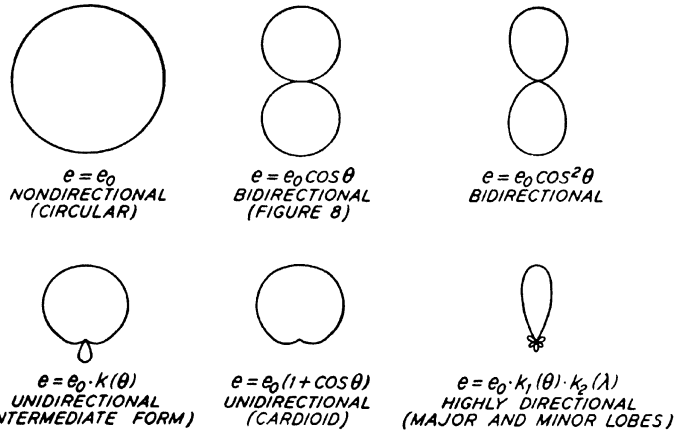


Fig. 142. Directional characteristics of various microphones. In the top row the patterns from left to right are those of zero, first and second order gradient microphones, respectively. The first two patterns in the bottom row are representative combinations of zero and first order gradient microphones while the final pattern is typical of highly directional microphones of dimensions large compared with the wave-length λ.

field not directly but through suitable phase shifting networks [24]. Still other combinations are possible [36] and are used [51]. Most of these devices have suitable controls which permit transition at will among a variety of directional patterns. Suitable precautions must be taken to control the amplitudes and phases of the outputs of the individual microphones in a combination.

For the class of devices whose dimensions are not small compared with the wave-length, high degrees of directivity can be realized. The directional patterns, however, will depend on frequency. The use of large transducers, or arrays of small ones, employment of horns, reflectors and lenses, result in appreciable directivity when the ratio of a significant dimension to wavelength is large [24]. Parabolic reflectors focussing sound into a microphone are used to advantage to secure high directivity [52] when the aperture is large. Acoustic delay lenses [53] can be employed to accomplish similar results. A difficulty

often encountered in such reflector or lens arrangements is that the directivity of the microphone proper affects the directivity pattern of the combination. This can be avoided in the case of a lens by the use of a shielding horn (see Fig. 143c). A single microphone which terminates an array of tubes of progressively increasing length has been successfully employed by MASON and MARSHALL [54] (see Fig. 143).

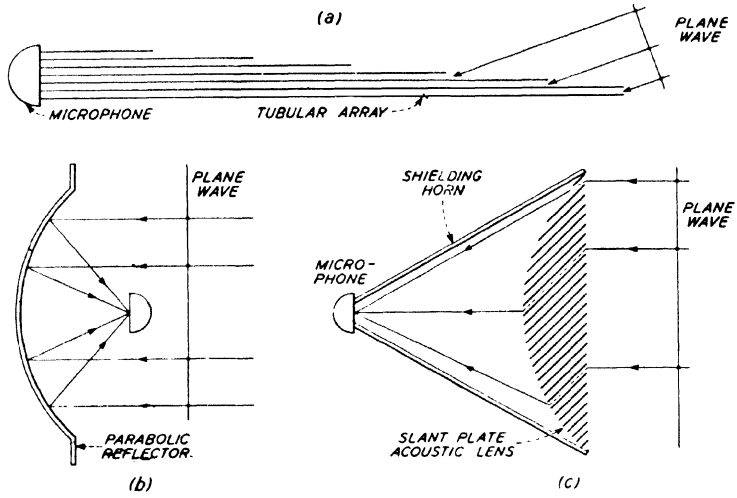


Fig. 143. Microphones in which the directivity is dependent on large structural dimensions: (a) the tubular microphone; (b) the parabolic reflector microphone; (c) the lens microphone with a shielding horn (MASON and MARSHALL [54], KOCK and HARVEY [53]).

Maximum response of the device is obtained along the axis since in that direction sound waves reaching the microphone through the various tubes arrive in phase; in other directions the various contributions do not add in phase and so result in decreased response.

§ 16. Miscellaneous Microphones

Sound-powered telephones [55] use high efficiency transducers in pairs for both talking and listening over communication lines in which no batteries or amplifiers are employed. Generally, these units are

balanced-armature magnetic units whose resonances are utilized to attain high sensitivity.

Throat 'microphones' [56] are unaffected by airborne sound. A pair of units is strapped to the talker's throat and actuated by the motion of the flesh. Although the articulation efficiency is poor, they are of some use in military and aircraft applications in locations with high ambient noise.

§ 17. Sensitivity Ratings

To illustrate some of the various sensitivity ratings applied to microphones by their makers, the following examples are given. A certain 25-ohm moving-coil microphone developes $5.6 \cdot 10^{-5}$ volts on open circuit, when placed in a sound field of 1 dyne/cm^2. The sensitivity is said to be -85 db (re 1 volt per dyne/cm^2). A step-up transformer is then included in the housing so that the output impedance becomes 25,000 ohms. The sensitivity is now -55 db (re 1 volt per dyne/cm^2). Taking the impedance into account, in either case the available power [5] into a matched load is $3.14 \cdot 10^{-11}$ watts or -75 dbm (re 1 mw per 1 dyne/cm^2). Since close talking levels in the broadcast studio are of the order of 10 dynes/cm^2, the so-called 'effective output level' of this microphone is said to be -55 dbm (re 1 mw per 10 dynes/cm^2). ROMANOW and HAWLEY [57] have discussed the system rating [13] of microphones which are part of a transmission system.

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DIVISION FIVE
SOUND REPRODUCTION

CHAPTER 14

LOUDSPEAKERS AND TELEPHONES

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§ 1. Electroacoustic Sources in General

(i) *Introduction. General Survey of Applications and Problems*

At the end of every electroacoustic line there is a loudspeaker or telephone as acoustic source, whose function is to transform the electrical into sound energy. Today there are hundreds of millions of loudspeakers in use, *e.g.* in sound-film cinemas, in broadcasting and television as well as in telephone systems. This is in fact the weakest element in the system and so determines the overall quality of reproduction. The problem of getting electroacoustic radiation free from distortion is in fact a rather difficult one. On top of the general troubles inherent in a mechanised system coveringmany – up to eight – octaves must be added the fact that greater efficiency is generally expected of a transmitter than of a detector of sound (microphone). Also the radiation of low frequencies is more difficult than that of high frequencies (SCHOTTKY's law). However, present-day progress in the design of acoustic transducers is such that one can attain quite high fidelity. To keep within the scope of this book it is not possible to give details of all these manifold developments of which, indeed, many must be regarded as outmoded for loudspeakers and telephones have passed the peak of their highest development. We shall restrict our description to the most common and endeavour to describe some characteristic types.
Books are available which give further details[1–5].

(ii) *Historical Survey of Electroacoustic Transducers*

In 1861 PHILIP REISE, a teacher in Frankfurt-am-Main, first transmitted the human voice electrically. As transducer he used the belly of his violin, which was excited magnetostrictively through a knitting needle pressed on it and working in a coil. (A model of this apparatus is to be found in the Deutsches Museum at Munich.)

In 1876 GRAHAM BELL discovered the first practical form of telephone receiver which, until the discovery of the carbon microphone by HUGHES, served also as microphone.

In 1878 WERNER V. SIEMENS built an electrodynamic transducer with pot-magnet and moving coil[6]. Whereas the telephone with its close attachment to the ear requires little energy and was soon adopted, the loudspeaker required more energy to fill a complete room with sound which was then attained in some fashion by means of a number of high-current microphones in parallel or a *charcoal amplifier*. For this purpose large telephone systems with horns were used having their resonance near the middle of the gamut and exhibiting both linear and non-linear distortion. Loudspeakers of good quality first developed alongside the thermionic-valve amplifier.

In 1924 RIEGGER discovered the low-frequency loudspeaker without horn resonance, formed of a large membrane in a baffle using the dynamic principle. This made possible a wide response band with much less linear and non-linear distortion than had been previously possible. By having the resonant frequency low it became possible to make the fall in the diaphragm velocity balance the rise in the radiation resistance. RIEGGER realised this principle in the Blatthaller. Independently and shortly after the publication of RIEGGER's manuscript, KELLOGG and RICE patented the low-frequency cone loudspeaker which, on account of its simple construction, has now replaced the Blatthaller[7].

From the commercial point of view other solutions are of less importance, *e.g.* the magnetic loudspeaker that, with pleated membrane, was conceived by GERLACH in 1930 and is now employed as a free-swinging source or as magnetic 'quadripole' system. The electrostatic or dielectric loudspeaker was used in 1922 by ENGEL, VOGT and MASOLLE as so-called Statophone or Oscilloplan for sound films. It is today employed, on SELL's principle, with counter-electrodes

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roughened by sand-blasting, as a high-frequency loudspeaker. The piezoelectric loudspeaker is also used as high-frequency accessory and as so-called soft-speaker under the pillows of patients in hospitals. The corrugated and ribbon speakers of GERLACH, which also work on the dynamic principle, have proved too complicated in construction to last.

In 1888 FERDINAND BRAUN discovered the thermophone, wherein sound is produced by heating a thin wire with an alternating electric current. This is now only used for calibration purposes. The principle of the singing arc, discovered in 1897 by T. H. SIMON, has no technical application. Finally the ionic loudspeaker[8], together with one working on the principle of the point-discharge, has too poor a frequency response, or bad transduction, to be used. Of irreversible loudspeakers should be mentioned that of JOHNSON RHABECK and the compressed air loudspeaker. The first works by the electrostatic attraction of a conducting band through a rotating semiconducting cylinder when an electric potential is established between them. The latter[9] has two comb-like insertions in a compressed air channel and a funnel which approach each other rhythmically and so modulate the air-stream. With this arrangement one can get very great intensities but, on account of the inertia of the moving parts and the consequent hissing noise, the quality is not very good.

Among telephones, the magnetic type is the one in most general use to-day; it has for a long time been standardised. For special purposes telephones working on dynamic, piezoelectric, electrostatic and induction principles have been built. Quite recently the magnetic type has been improved in sensitivity and frequency response, *e.g.* by increasing the areas of the pole pieces and by separating the acoustic and magnetic diaphragms.

(iii) *Principles of the Conversion of Electric into Acoustic Energy*

Electroacoustic sources can be divided into reversible and irreversible (or relay) systems. The latter take the energy from a source (*e.g.* of compressed air) and use the electrical energy as control only. Reversible types turn the electric energy directly into acoustic and can be used indifferently as senders or receivers. The performance of the latter is not very good, but fortunately the ear is very sensitive so

that great outputs are not usually required, those common in low-current technology sufficing.

The conversion occurs in two stages; from electrical to mechanical and from mechanical to acoustical, so that we shall first deal with the electromechanical system and then the radiation process, which must, in most cases, involve a membrane to convert the mechanical forces into sound pressures (*cf.* §§ 3 & 4).

(iv) *Quality of an Electroacoustic Source*

The quality of a transducer depends on its efficiency (sensitivity), its linear amplification (frequency band viz. phase-fidelity and response) the non-linear amplification (harmonic content) which also concerns its dynamics, its directivity, its capability of being loaded, as well as its constructional characteristics (*e.g.* weather protection, weight, unit construction, etc). Finally, the expenditure which is necessary to secure a given solution often plays in practice an important role. According to the application envisaged (telephone hydro-sender or music reproduction) the requirements under these headings can differ much.

(v) *Sub-divisions of Types*

The electrodynamic loudspeaker, which is the most common, is made commercially in types with maximum output from $\frac{1}{2}$ to 25 watts, larger ones of 150 watts for special usage. Great sub-division in this respect is not worth while since an output increment of 3 phons (that is double), is about the minimum discernible step.

Such a division is unnecessary for telephones, which have to fill always the same ear space with sound.

(vi) *Linear Law of an Electroacoustic Transducer*

We shall consider here the reversible transducer.

Contrary to the quadratic relation between electric and mechanical quantities which is appropriate to the high-current working of motors and dynamos, in electroacoustics we must have linear characteristics if we want to avoid non-linear audible distortion[10]. This can be obtained if we apply to the system a polarisation field or excitation H_0, large

compared to the alternating field H. The force is then, *e.g.* proportional to $(H_0 + H)^2 = H_0^2 + 2\,H_0H$ if we neglect the square of H. This means replacing at the driving point the parabolic force (F): current (i) relationship by its tangent, which is permissible with the small values of H which we meet in acoustics. In this way we prevent a frequency doubling which would otherwise, without the excitation H_0, appear.

(vii) *Relation between Mechanical and Electrical Quantities*[11]

The following equations hold between mechanical and electrical quantities for a reversible sender-receiver.

Sender; $F = Mi$. Receiver; $e = Mv$.

where M is the electromechanical coupling factor, dependent on construction data, but having the same value in both equations, for a reversible transducer. For other transduction principles, however M is different in the two types (*vide infra*, ix).

(viii) *Relation between Acoustical and Mechanical Quantities*

The transforming of mechanical into acoustical energy is quite different for incoming and outgoing radiation. The SCHOTTKY law of receiving energy is pertinent between acoustic transmission and reception in the free field, but in the pressure capsule another law holds. For the radiation one can write the general law;-

$$P_{ac} = \varkappa v^2$$

where P_{ac} is the acoustic output, $\varkappa$ the radiation resistance and v the membrane velocity. The radiation resistance is in general very dependent on frequency, has different values according to the order of the radiation (1st, 2nd, etc.) and can only be calculated for ideal cases.

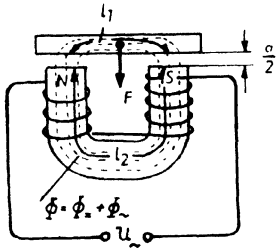
Inside the capsule, in which one cannot properly speak of 'output', the so-called pistonphone law holds;-

$$p = \frac{\triangle V}{V} \chi\, p_0 = \frac{S\xi}{V} \chi\, p_0$$

where p is the sound pressure in the chamber of volume V when a piston (or equivalent membrane) of area S has a displacement ξ; χ is the ratio of specific heats and p_0 the atmospheric pressure.

(ix) *Transducer Principles and Electromechanical Coupling Factors*

a. The *electromagnetic principle* requires that when electric currents of sonic frequency act through the intermediary of an electromagnet with n windings coupled to a membrane, a force F should act on an airgap of length a (Fig. 144).



$l_e = l_1 + l_2$

Fig. 144. Electromagnetic principle.

If φ is the magnetic flux, the electromagnetic coupling coefficient is

$$M_m = \frac{\partial \varphi}{\partial a} = -\frac{\varphi}{a + l_e/\mu}$$

with μ the permeability of the iron and l_e the path-length of the magnetic lines of force. When the resistance of the iron can be neglected in comparison with that of the airgap, we have

$$M_m = n\frac{\varphi}{a}$$

Advantages of this system are that one can work with the forces from very small permanent magnets and that the windings since they are stationary can be made bigger and of higher impedance than, *e.g.*, those in the moving coils of the dynamic systems. Disadvantages are the small available space (about 10% of the airgap) which on account of the necessity to avoid saturation means heavier iron pieces and the hysteresis losses which occur at higher frequencies. In loudspeakers which must employ larger amplitudes, non-linear distortion results.

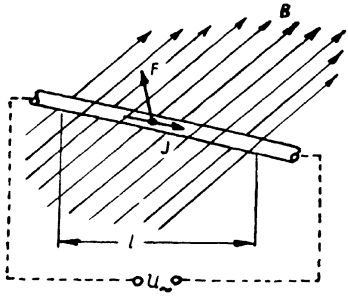


Fig. 145. Electrodynamic principle.

The magnetic principle is best for middle frequencies and in different types is most used for telephones.

b. The *dynamic principle* is that in which a current-carrying conductor experiences a force in a magnetic field (Fig. 145). If B is the magnetic induction in the airgap and l the conductor length the electromechanical coupling coefficient is

$$M_d = B.l$$

Advantages of this system are that the available space can be made quite large since no chattering can occur.

Also the system, if made rigid, is free from non-linear effects so long as the conductor moves in the homogeneous part of the magnetic field[12]. The resistance can be made mostly ohmic so that adaptation to an internal ohmic resistance, *e.g.* of the amplifier valve, is readily possible.

Disadvantages of the principle are that a very powerful permanent or separately excited magnet must be used.

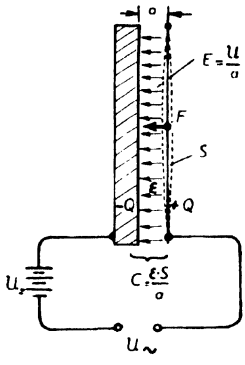


Fig. 146. Electrostatic principle.

This system lends itself best to the reproduction of low frequencies of large amplitude and is accordingly much used for cone loudspeakers. It is also used on ribbon microphones and loudspeakers, and blatthallers (p. 250).

The systems can be tuned low without fear of chattering, *i.e.* loosely suspended.

c. The *electrostatic principle* depends on COULOMB's law. If an electric field of intensity f is set up between two electrodes they repel each other. Variations in this force are produced by an applied alternating field of pulsatance ω (Fig. 146). The electromechanical coupling coefficient is

$$M_\rho = \frac{\bar{E}}{\omega}$$

It is possible so to write the fundamental equations that no dependence on frequency is shown by this factor.

$$F = \bar{E}\, C.u = \bar{E}\, q \text{ and } i = \bar{E}.\, C.v \text{ or } u = \bar{E}\, \xi$$

Here C is the capacity, q the charge, u the tension and ξ the amplitude between electrodes.

Advantages are that the moving parts can be made very light and equal force over the whole surface is possible. Disadvantages are that a much smaller pressure for a given electrical input is obtained than by using the former principles[13]. Chattering and non-linear distortion are possible with too narrow an airgap. On the other hand a large airgap demands dangerously high voltages, if an intense field has to be set up. In application, an account of the narrow airgap low

frequencies are poorly expressed, but high frequencies even into the ultrasonic are well treated on this principle.

d. The *piezoelectric principle*[14] depends on the effect discovered by the brothers CURIE in 1880 (see Chap. 20). A crystal being characterised by elastic modulus E, piezo-modulus σ, dielectric constant ε has length l, thickness d and breadth b. In the longitudinal effect, charges

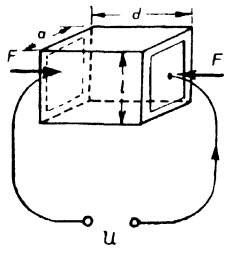


Fig. 147a. Piezoelectric longitudinal effect.

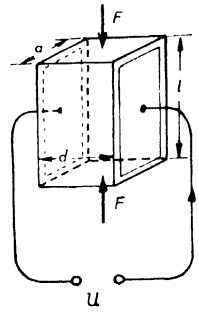


Fig. 147b. Piezoelectric transverse effect.

and forces arise on surfaces to which the potential is applied (Fig. 147a), but in the transverse effect on the faces perpendicular to these (Fig. 147b). The electromechanical coupling coefficient is for the longitudinal effect.

$$M_{\mathrm{pl}} = \frac{\sigma E}{\varepsilon \omega}$$

independent of crystal dimensions. In the transverse effect

$$M_{\mathrm{p}\pi} = \frac{Eb}{C\omega} = \frac{\sigma E d}{\varepsilon \omega l}$$

Advantages are the absence of additional electric potentials, simplicity of construction and the forces are greater than from corresponding electrostatic systems, but so is the crystal stiffness. As the dielectric constant of the material is very high (up to 1000) useful capacities can be formed on quite small dimensions. Disadvantages are the inflexibility of the crystal so that at low frequencies its specific impedance is ill-matched to the ρc of air. Sensitive crystals (like Rochelle salt) are temperature-dependent and easily broken when used as senders.

In application, piezoelectric systems are used for special telephones, high frequency loudspeakers ('tweeters') and pillowspeakers, but mostly for underwater loudspeakers (See Chap. 20).

Magnetostrictive and electrostrictive transmitters are not normally used in air and so will not be discussed further.

§ 2. Loudspeakers

(i) *Characteristic Parameters for the Transduction of Electrical into Acoustic Energy*

a) *Efficiency*[4].

α. *Calculations*

Whereas the efficiency of a reversible microphone – generally only a fraction of one thousandth – is not of much moment, as one can increase its sensitivity notably by small increments of applied potential, that of a loudspeaker (several per cent) is more important since increments of output on electrical amplifiers working on the public a.c. mains are more costly.

In what follows we shall calculate the performance of a loudspeaker in terms of its acoustic, mechanical and electrical data.

The efficiency η of an electroacoustic source is equal to the ratio of acoustic output (P_a) to electric input (P_e). It is convenient to divide this into two parts η_{am} and η_{me} of which the first expresses the ratio of P_a to P_m (mechanical power) and the other of P_m to P_e

$$\eta = \eta_{\mathrm{am}} \cdot \eta_{\mathrm{me}} = \frac{P_a}{P_m} \cdot \frac{P_m}{P_e} \qquad \text{(XIV, 1)}$$

But the radiated *acoustic energy* $\mathrm{P_a} = \mathrm{z}\,\dot{\xi}^2$ *(vide* p. 335*)* and the *mechanical energy* $\mathrm{P_m} = Z_{\mathrm{m}}\,\dot{\xi}^2$,

Z_{m} being the mechanical impedance of the membrane and the *electrical energy* $(R_e + R_w)\,i^2$,

R_e being the external (generator) resistance, R_w the apparent loudspeaker resistance and i the current through it.

Substituting in (XIV, 1) we derive

$$\eta = \frac{\mathrm{z}\,\dot{\xi}^2}{Z_m \dot{\xi}^2} \cdot \frac{Z_m \dot{\xi}^2}{(R_e + \mathrm{R}_w) i^2} \qquad \text{(XIV, 2)}$$

Eliminating F with the formula $i = \frac{F}{M}$ and a mechanical OHM's law; $F = \dot{\xi} Z_m$; one gets i as $\dot{\xi} Z_m / M$. Putting this, in which M is as before, the electromechanical coupling coefficient in (XIV, 2)

$$\eta_i = \frac{\varkappa}{Z_m} \cdot \frac{M^2}{(R_e + R_w^*)} \qquad \text{(XIV, 4)}$$

Now consider the efficiency formula:

$$\eta_i = \frac{\varkappa}{Z_m} \cdot \frac{Z'_{el}}{Z_m} \qquad \text{(XIV, 5)}$$

If all the mechanical impedances, R_m, Z_m, Z'_{el} are equal, the efficiency is 1 and electrical energy is transformed entirely into acoustic, but this is difficult to ensure in practice.

The efficiency goes up with the ratio of radiation resistance $\varkappa$ to mechanical resistance Z_m, but this is made up of the system resistance Z_s, the impedance of the vibrating air mass M_s and the radiation resistance. It is, therefore, desirable to make the system impedance as small as possible. If it could be made zero then at high frequencies when the radiation resistance exceeds the vibrating air mass $\frac{\varkappa}{Z_m} = \eta_{am} = 1$. In general Z_m is much greater than $\varkappa$ which one tries to enhance, *e.g.* with a horn; in any case Z_m in resonance can be quite small and so, *e.g.* in underwater acoustics, high efficiencies can be reached. But in loudspeakers the moving parts are generally dense compared to air so that over a large range of frequency the mass resistances are greater than those concerned in radiation.

Further, the efficiency is greater as Z'_{el}/Z_m increases. With large electromechanical coupling-coefficient M, *viz.* with very small mechanical resistance Z_m, (obtained *e.g.* by lightly damped resonance), the electrical resistance R_w^* can become so great, due to the mechanical feedback $R'_w = M^2/Z_m$ that $R_e + R_w$ can be neglected in comparison therewith. This would be possible if the electrical impedances R_e and R_w were very small. The imaginary part would vanish at electrical resonance. If the output resistance were small then the usually not inconsiderable current heat loss would disappear. Then

$$\frac{Z'_{el}}{Z_m} = \eta_{me} = 1 \text{ and } \eta = \frac{\varkappa}{Z_m}$$

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The coupling factor M, which should be made as large as possible, is in fact limited in both magnetic and dynamic cases by magnetic saturation, in the dielectric case by the sparking potential. Also in the electrostatic and magnetic cases chattering of the membrane at high M can occur when its flexure exceeds one third of the airgap. Also too great a force of attraction sets up a negative stiffness which changes the tuning of the system.

With electrical mismatch $R_w = \infty$ (idling) then according to eqn (XIV, 4) $Z_{el} = 0$ and $\eta = 0$, also with mechanical mismatch (braking), $Z_m = \infty$, Z_{el} and $\eta = 0$. The optimum is reached when the system itself has no mechanical resistance $Z_s = 0$, *i.e.* if $Z_m = \chi$ and only an impedance Z_{el} due to electrical feedback arises, which by selection of M and $R_e + R_w$ can be made nearly equal to χ.

β. *Example*

Fig. 148 shows the mechanical resistance and efficiency of a cone

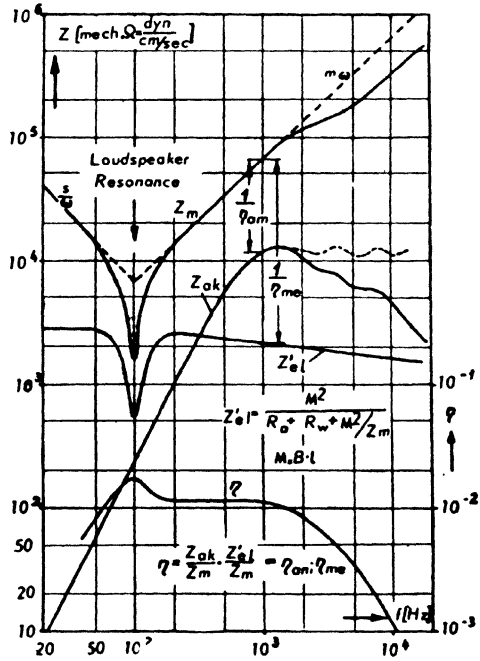


Fig. 148. Mechanical impedance Z and efficiency of a dynamic cone-loudspeaker of radius R = 10 cm.

loudspeaker of 10 cm radius. The mechanical impedance Z_m is made up of m, stiffness s/ω below resonance and inertance $m\omega$ above resonance. The resistance of the vibrating air mass and the radiation resistance are neglected in this figure as they are small compared to the system resistance. The mass-resistance of the loudspeaker comprises the reactance offered by the membrane mass and the coil. At higher frequencies only the inner portion of the membrane oscillates so that practically only the reactance of the coil is operative. The radiation resistance is, in this cane, replaced by that of a piston of equal size in an infinite baffle, but at high frequencies it should be smaller than this for then the membrane acts as a radiator of higher order and some parts of it are shortcircuited by others. The electrical resistance Z_{el} arising from the mechanical resistance is a minimum at resonance for then Z_m is small, but in practice it also falls at high frequencies since the transducer impedance of a loudspeaker grows somewhat under the action of its inductive component $i\omega L$. From the three logarithmically inscribed curves of this figure η can be calculated graphically by adding on coordinate paper the radiation resistance and the electrical feedback resistance Z'_{el} deriving from the mechanical resistance Z_m and reading off its reciprocal. It appears that the efficiency in the middle of the frequency scale, where the radiation resistance matches the square of the mechanical resistance, is nearly constant but falls off at high frequencies. In the frequency curves this fall is not so strongly expressed, because the concentration of flux in the axis of the loudspeaker, in which region the sound pressure is usually measured, results in more energy being radiated from this axial than from annular regions.

γ. *Performance at Larger Dimensions*

In the case of a cone loudspeaker with large surface one can approximate the sound-radiation to that of a radiator of zero-order. At low frequencies its radiation resistance grows with the fourth power of the radius ($\chi \sim x^4$ when x represents the scale of linear dimension ratio). Outside resonance the mechanical resistance Z_m equals the system impedance Z_s, which however in the transmission region is a mass-impedance m. As this is proportional to x so is the mechanical resistance at constant frequency. The mechanical feedback resistance of the electric circuit $Z'_{el} = M^2 (R_e + R_w + M^2/Z_m)$ can be reckoned away

References p. 377–382

from resonance with electrical matching ($R_e = R_w = R$) as $Z'_{el} = M^2/2R$. But as $M = Bl$, therefore $M^2 \sim x^2$ and $R = l\varrho/S$, where l and S are the conductor length and cross-section, respectively, so $R \sim 1/Fx$ and $Z'_{el} \sim x^3$. Putting these in equation XIV, 5 we have

$$\eta = \frac{\tau}{Z_m} \cdot \frac{Z'_{el}}{Z_m} \sim \frac{x^4}{x^3} \cdot \frac{x^3}{x^3} \sim x. \qquad \text{(XIV, 6)}$$

so that the efficiency is *ceteris paribus* proportional to the linear dimension. There is also the fact that in large types one can attain greater values of the induction B with which the efficiency increases quadratically. So large loudspeakers have greater efficiency than small[15]. But this applies only in the middle of the scale in which the membrane moves as a whole. At high frequencies it is better to use a smaller system; only a portion of the membrane moves and it is then better to use a smaller coil of consequent smaller mass-reactance[16].

The higher modes of such a diaphragm are shown by the CHLADNI method of strewn sand in Fig. 149, but the calculation of the radiation resistance in such cases is not practicable. At high frequencies therefore the loudspeaker efficiency is no longer calculable by equation XIV, 5, but one can measure it experimentally in a free field or in an

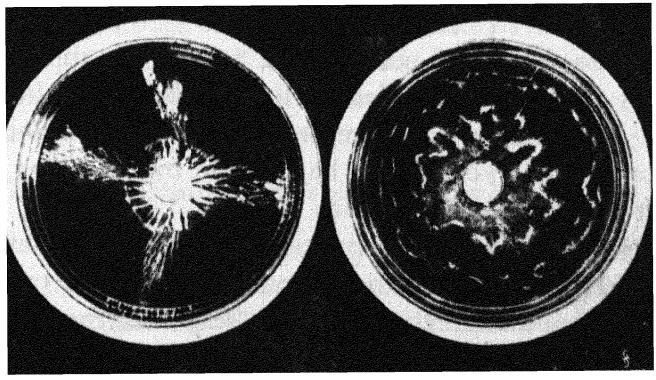


Fig. 149. Vibration forms of a membrane at different frequencies, 450 and 4310 c/s.

anechoic chamber by integrating the sound pressure over an enveloping surface. HOPKINS and STRYCKER[17] use a parameter called 'loudness rating' or 'loudness efficiency factor'. With the aid of this

parameter they describe the polar characteristics of a loudspeaker at 10 frequencies between 300 and 3000 c/s. BÜCHMANN and KÜPFMÜLLER[18] take a mean value over the same range which they call 'norm-efficiency'.

The efficiency in the common dynamic loudspeaker is about 2–5%; in special loudspeakers 10%, in horn loudspeakers 30–50% and in resonant systems, *e.g.* in underwater sound, nearly 100%.

b) *Sensitivity (Transmission Factors)*[19]

To determine the Response-Frequency Characteristic, the efficiency is not usually chosen, but instead a linear quantity called 'Absolute Response'

$$E = \frac{\text{sound pressure}}{\sqrt{\text{electric loading}}} = \frac{p\sqrt{R_m}}{e}$$

In this definition the direction (axial) and distance (1 or 5 ft or a multiple thereof) must be specified. In Germany the sensitivity is expressed in microbars per (watt)$^{\frac{1}{2}}$, in English-speaking countries in positive or negative decibels in relation to a threshold; O db = = 1 microbar per (watt)$^{\frac{1}{2}}$ [19a] or 2.10^{-4} μ bar per (m.W)$^{\frac{1}{2}}$ [19b] or 1 μ bar per Volt.[19c]

c) *Transmission Factor*

Sometimes also the sound pressure due to a given applied voltage is quoted as $A = p/e$. If it is desired to equilibrate two loudspeaker types using this parameter, one must also give the resistances of the transducers since the voltage can be changed by shunts or windings of different dimensions for the same loading.

As R_w^* and the resistance of the current source R_e enter the expression, (eqns XIV, 4 and XIV, 5) – for the efficiency and so for the transmission factor – and can change the frequency response, the transmission should only be measured with the appropriate amplifier *in situ* and the proper voltage on the grid of the final valve. There is in fact an electromechanical feedback and one can *e.g.* damp the loudspeaker resonance by a corresponding low resistance termination of the generator resistance R_e.

Just as with microphones a distinction is made between the field and the capsule transmission value, so we have in electroacoustic

radiation beside the field transmission of loudspeakers the capsule transmission of telephones. As terminal of the telephone a hard-surfaced pressure chamber or artificial ear of 2 c.c. volume with impedance like that of the ear can be used[20, a, b] or else with a microphone and narrow probe the sound pressure can be measured when the telephone is held against the human ear (*vide* p. 335, 373).

(ii) *Matching of the Loudspeaker to an Electrical Circuit*

a) *Final Valve and Loudspeaker*

The impedance of thermionic valves to which the loudspeaker is connected is mostly ohmic and in triodes is about 1K, but higher in pentodes unless they are brought down by back-coupling. The dynamic loudspeaker has a slowly increasing apparent resistance with frequency, so that a matching transformer must be placed between the final valve and the loudspeaker. In triodes the harmonic content becomes smaller as the terminal resistance is made greater, so that the loudspeaker impedance is made twice or four times the valve impedance. In pentodes the non-linear distortion is least when the external resistance is 5 to 8 times smaller than the internal resistance of the valve. Because the loudspeaker impedance goes up on account of its inductive component at high frequencies the region of large distortion in pentodes is reached. BOUCHEROT's circuits enable (Fig. 187) this to be reduced[21]. To reduce the inductive component in the loudspeaker impedance a second compensation coil (short-circuit ring) can be built round the magnet core[22].

b) *Coupling Several Loudspeakers to one Amplifier*[22, a, b]

In broadcasting and sound film equipment, with always one loudspeaker to each amplifier, the dimensions can be so chosen that the best output is obtained. With ELA (= electroacoustic) systems when many loudspeakers may be coupled to one amplifier it is usual in Germany to make the exit amplifier by pentode coupling of so small resistance (R_e about $\frac{1}{8}$ R_w) that even at maximum output there is 100 volts across the terminals[23]. Then one can put many loudspeakers – like a nest of lamps – in parallel if each has a resistance of (100 volt)^2 /P watt, where P is the nominal power of the loudspeaker. The leads to the loudspeaker having wave resistances of several hundred ohms,

even at high frequencies, a capacitive short-circuit supervenes and the ohmic resistance of the leads absorbs considerable power. There must then be a further transformer at the loudspeaker itself to adapt the lead impedance to the small coil impedance.

c) *Damping by Electrical Short-circuit*

In resonance the loudspeaker impedance $R_w^* = R_w + M^2/Z_m$ becomes so big that its mechanical resistance is small. If the generator has low impedance, the connecting impedance approximates in resonance to the idling state; current and driving force become low and the resonance peak in the frequency curve disappears[24]. With the stoppage of the driving voltage the loudspeaker experiences a microphone effect and executes damped vibrations because of the strong magnetic counterforce on short-circuiting the low-resistance generator[25]. For aperiodic damping the generator resistance must be $R_e = M^2/2Z_m - R_w$. Also one can so select the mutual inductance of the loudspeaker transformer that at resonance the loudspeaker short-circuits to this value. The damping is more easily reached as the magnetic induction B or electromechanical coupling coefficient M is greater since with large M the action of the small mechanical resistance Z_m on the apparent total loudspeaker impedance $R^*_w = R_w + M^2/Z_m$ is not so great.

d) *Electromechanical Countercoupling*[26]

To compensate the fall in the response curve feedback circuits are used, whereby a potential is induced by the motion in a second coil which regulates the input potential. Such circuits make possible a compensation of the movement of the coil but not of the sound radiation, for the oscillatory properties of the membrane come into this. To prevent the inductive influence of the auxiliary coil on the loudspeaker coil, compensation through additional transformers can be attained[27].

e) *Equivalent Electrical Circuits*

The mechanical parameters of a loudspeaker can be calculated in terms of equivalent electrical circuit elements, using the dynamic and magnetic analogy $S = M/L^2$; $m = C.M^2$ and $Z = M^2/R$. In this way a mechanical parallel circuit becomes an electrical parallel circuit, *viz.* a mechanical

mode ΣF is replaced by an electrical $\Sigma i = 0$, and a mechanical antinode $\Sigma v = 0$ by an electrical $\Sigma m = 0$[28,29]. The action of the air can also be expressed by an electrical analogue[30], in which the radiation resistance $z = M^2/R$ and the vibrating air mass $m_s = C.M^2$, which for a radiator of zero order is connected in series. For one of first order there is also an inductance $L = M^2/s$ in parallel with a resistance. Fig. 150 shows the relevant circuit and Fig. 151 the corresponding circle diagram for resonance in air and *in vacuo*. In such a way the efficiency of tunable underwater detectors can be estimated[31].

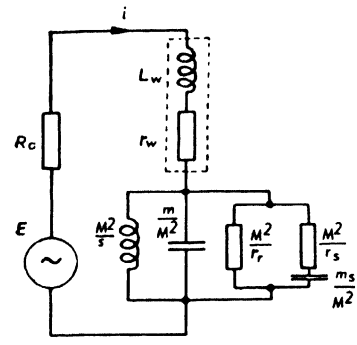


Fig. 150. Equivalent electric circuit of a loudspeaker.

The usefulness of such simple equivalent circuits for common loudspeakers is not very great, seeing that with the sub-division of the membrane at high frequencies and other influences the equivalent elements are not constant but frequency-dependent in a very complex and unpredictable way. Generally such analogies do not reduce the

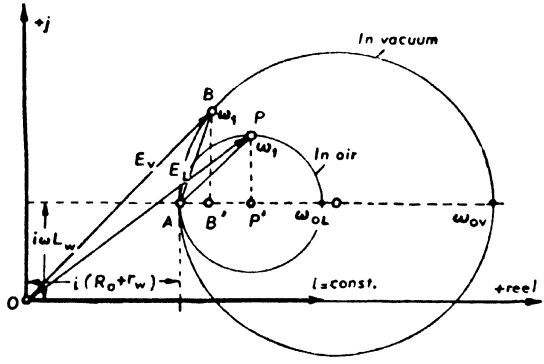


Fig. 151. Vector diagram for the equivalent circuit in the neighbourhood of resonance (constant current).

labour of calculation but one gets a deeper understanding of the mechanism of the acoustic system. Only to those who are more accustomed to work with electrical rather than with mechanical symbols will this (in any case, incomplete) demonstration appear more deceptive.

f) *Electric Switches*[31a]

In combinations of loudspeakers there is an electrical switch provided whose purpose is to divert electrical low and high frequencies respectively to the low – and high-frequency loudspeaker. The resistances

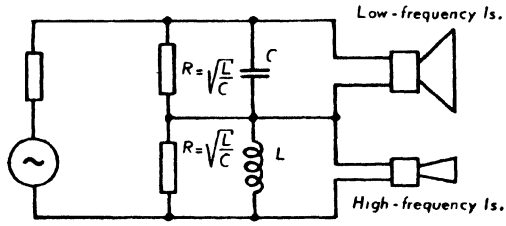


Fig. 152. Electric Switch.

(R) of the high and low frequency systems are equal. The inductance (L) and capacitance (C) are to be so chosen that $R = \sqrt{(L/C)}$.

The impedances make the input resistance independent of frequency. The resonant frequency determined by L and C forms the cut-off.

§ 3. Transformation of Electrical into Mechanical Energy

(i) *General*

The driving system has the object to turn electrical into mechanical energy and then with the aid of the membrane to transduce this into acoustic energy. The driver can be considered as an electromechanical 'quadrupole' whose entry has an electric and exit a mechanical impedance. The electrical impedance can be mainly inductive (magnetic), capacitive (dielectric or piezoelectric) or as well resistive (dynamic) and changes with the mechanical load from the electrical feedback resistance $R^1_w = M^2/Z_m$. The mechanical impedance (Z_m) depends on the movement of the mechanical mass-elements and the sound radiation and can have a complicated variation with frequency as the membrane passes into higher overtones. Within limits one can make the simple assumption that the mechanical impedance behaves as a vibrating particle-mass, *i.e.* as $Z_s = r + i(s/\omega - m\omega)$ and make another approximation that in resonance the particle experiences only viscous resis-

tance, below resonance only stiffness s, and above resonance only mass m. The influence of the air can also be expressed in viscous and mass terms (*cf.* p. 347).

The relation between the mechanical and electrical impedance is expressed by the electro-mechanical coupling coefficient M(*cf.* p. 335) whose value also characterises the sensitivity of the system. The way in which the energy in its different manifestations passes through the different impedance systems has already been described. (p. 339). In what follows only the technical details of the systems used to attain the various ends will be described.

(ii) *The Electromagnetic Driving-System*

The simple asymmetric system of Bell (See Fig. 144) has the disadvantages of likelihood of chattering and non-linear distortion. The former can be obviated in the so-called free vibrator (Fig. 153a)[31b], the latter within limits by a symmetrical construction or so-called quadrupole system (Fig. 153b), as the quadratic terms which excite the second overtone are thereby removed. The symmetrical system also

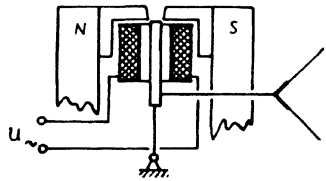


Fig. 153a. Electromagnetic loudspeaker (free vibrator).

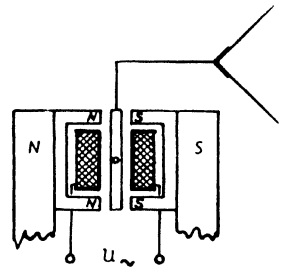


Fig. 153b. Quadrupole system.

reduces the danger of magnetic saturation since the magnetic flux in the stationary state is absent. The armature lies in the null branch of the magnetic bridge and so need not be made massive. Generally the whole of the membrane is not made of ferromagnetic material as this would make it too heavy; accordingly a little rod is placed between armature and membrane but this may exhibit resonance at high frequencies.

The magnetic free vibrator (Fig. 153a) can have greater amplitudes

as it cannot chatter, but even so it can show non-linear distortion under large deflection when the armature leaves the homogeneous magnetic field.

It is best to provide separate paths for the steady and alternating fluxes as the permanent magnet has large losses. SEIBT shows how one can employ an airgap across which the magnetic alternating flux can bridge over the permanent magnet.

The coil, since it is fixed, can be heavy and of high impedance, so that a magnetic loudspeaker can be connected to the final valve without a matching transformer; also the magnet is cheaper than one for a dynamic system so that it can be used as free vibrator for utility broadcasting equipment.

(iii) *The Dynamic Driving-System*

The magnetic airgap in the Blatthaller (Fig. 154) ribbon loudspeaker and pleated loudspeaker was a long slit. To-day cone loudspeakers with pot magnets are mostly used, having a circular airgap. In this

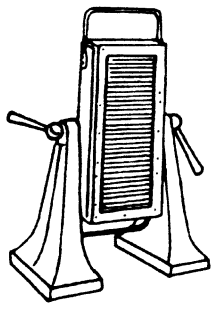


Fig. 154. Blatthaller.

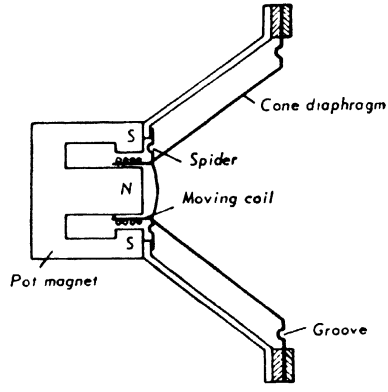


Fig. 155. Dynamic loudspeaker.

is located the moving coil which is retained in place by a spider and the stiffness of the diaphragm attachment which abuts on the so-called loudspeaker basket[32] (Fig. 155).

a) *Pot Magnet*[33]

In small types the airgap is about 1 mm. For permanent magnet

systems pot magnets are standardised[34] with which one can build loudspeakers of 0.5 to 25 watt. The larger types have also large airgap induction (up to 16,000 gauss) as the core diameter grows with the square of the radius but the airgap only with circumference and so is less easily saturated. Recently so-called inner-core magnets of alnico or ticonal magnetised in a preferential direction have been introduced

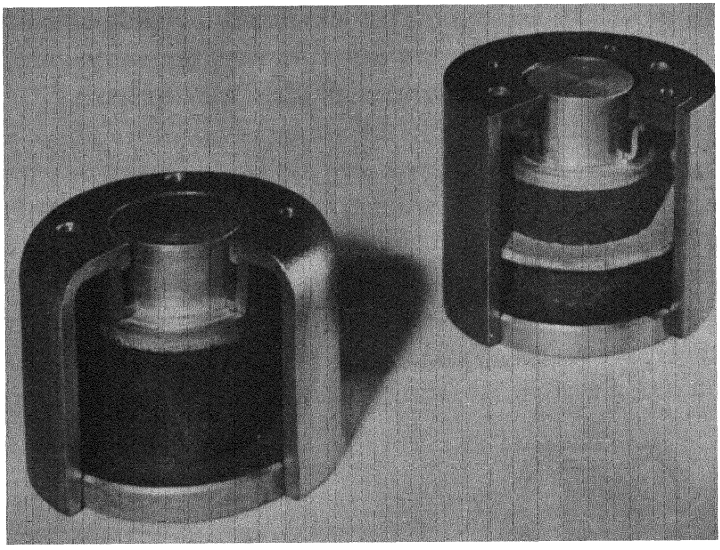


Fig. 156. Inner core magnet.

(Fig. 156). On both sides of the airgap is a disturbing field that can carry up to 50% of the total magnetic energy. To keep out dust and iron filings the driving systems of better loudspeakers are enclosed in a textile pouch. Externally excited loudspeakers are often employed, *e.g.* in broadcasting apparatus. In these the inductance of the exciting coil is often used for anode screening.

b) *Moving Coil*[35]

The moving coil should have low resistance (*ca.* 5–20 ohms) as then the filling-factor is greater and the coil can be made readily. Older types which, to obviate a transformer, were of high resistance have

not survived as the anode current causes a one-sided pull and heating.

Also the high resistance loudspeaker, like the capacity of a long cable, short-circuits the high frequencies. Of the airgap only 20 to 30 % is used for the actual copper coils. The rest is used for the support of the coil and the necessary adjustment. Recently attempts have been made to fill the coil space more completely and to let the coil—in silicone oil—slide directly over the iron of the magnet[36]. The coil former is made of paper in simple loudspeakers; at high frequencies the coil collar cracks. In larger types it is therefore made of hard paper, calite or aluminium. To prevent non-linearity, the coil is often made big enough that it invades the stray field on both sides.

c) *Spider*

The spider and the grooves, both internal and external, stamped therein serve to retain and centre the membrane (Figs 157a and b). The spiders, usually made of pertinax having a stiffness of several kg/cm, are clamped to the coils and membranes. Good loudspeakers must have a low natural frequency f (*ca.* 40 c/s, p. 332) and so lightly tensioned. As the dimension d of a system vibrating under its own weight is known to

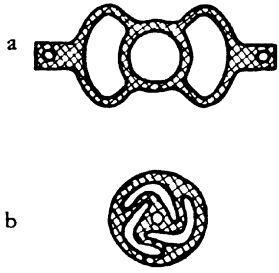


Fig. 157. Spiders for holding the cone-membrane (a) external, (b) internal.

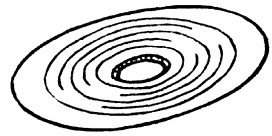


Fig. 158. Centering diaphragm.

be that of a pendulum of the same frequency, *i.e.* $d = 25.6/f_0^2$ cm., this would make $d = 0.15$ mm for 40 c/s. *i.e.* as big as the airgap of a telephone. This is not possible with a dynamic system, so – instead of a spider – a diaphragm provided with grooves is employed (so-called centering diaphragm; Fig. 158). At the outer edge of the cone diaphragm grooves are dug to make it flexible; specially low-tuned membranes have instead of this an edge of soft goatskin.

(iv) *Dielectric Principle*[37]

The pressure amplitudes that one can get from the dielectric principle are small compared with those attainable with magnetic or dynamic systems[38] but they can have large surfaces. VOGT[39] uses a differential loudspeaker (Oszilloplan Fig. 159). The diaphragm in an aluminium alloy and the opposing electrodes are perforated and carry a potential of 1 kilovolt and an airgap ranging from 0.3 mm on the outer edge to 1.5 mm on the inner. According to SELL[40] one can use sand-blasted perforated plates covered with a damping foil.

That the electrostatic loudspeaker is not common is due to the fact that large amplitudes cannot be obtained with it; which would mean large surfaces to radiate low frequencies. Because of its small electromechanical coupling the feedback resistance M^2/Z_m is small

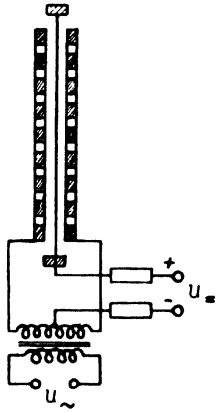


Fig. 159. Electrostatic loudspeaker.

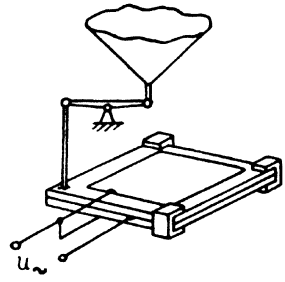


Fig. 160. Piezoelectric loudspeaker.

compared to its capacitive reactance and the electrical matching to the ohmic resistance of the output valve is bad at low frequencies, when too the added hydrodynamic mass may be greater than the membrane mass.

(v) *Crystal Loudspeaker*[41]

Generally a 'bender' is used. Two crystals are fixed together in bimetal form and provided with a diaphragm (Fig. 160). Under applied potential the upper plate expands and the lower contracts

or *vice versa*. This gives a form of leverage which leads to a large displacement of the ends but owing to heat production and danger of fracture, such a system cannot give large outputs (Fig. 161).

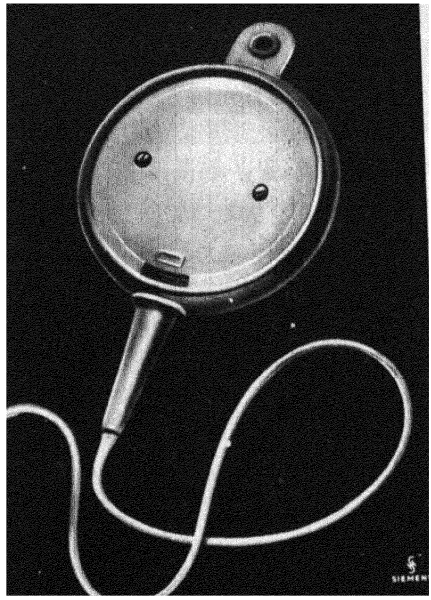


Fig. 161. Cushion loudspeaker (crystal system).

§ 4. Transformation of Mechanical into Electrical Energy

(i) *Sound Radiation*

a) *Radiation Resistance*

If a surface moves to and fro at sonic frequencies, with velocity ξ, it radiates energy with intensity dependent on its surface and construction and on the frequency. In general the radiated power $P = \zeta\dot{\xi}^2$ with ζ = radiation resistance. Besides this there is the reactance of the diaphragm and the added mass of air. Whereas the resistance is low at low frequencies and thereafter rises, the reactance reaches a maximum and falls at high frequencies. In general the radiation resistance of the diaphragm vibrating in an overtone mode is too difficult to calculate,

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but RAYLEIGH and RIEGGER have worked it out for the cases shown in Fig. 162, *viz.* (a, b) radiators of zero and first order, (c) membrane in infinite baffle. In the shaded regions of $kr < 1$ the wattless output

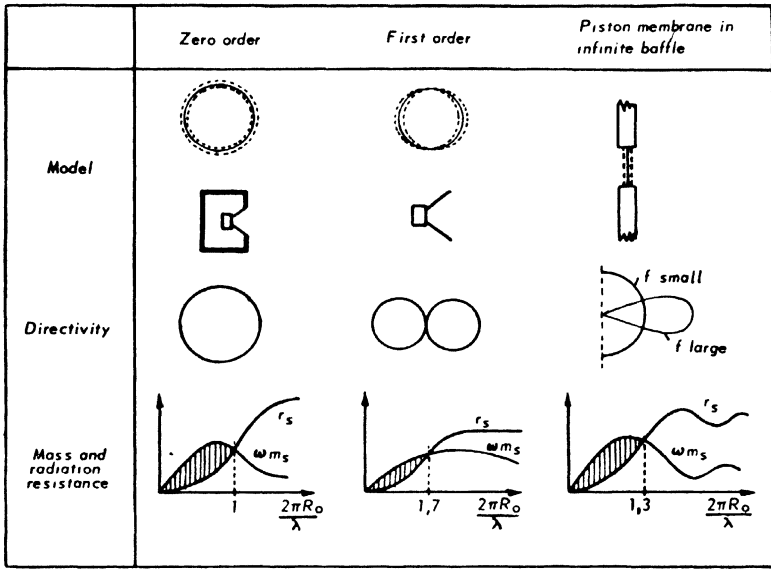


Fig. 162. Radiation resistance and mass reactance of a simple acoustic source.

$m\omega\dot{\xi}^2$ in a radiator of zero order exceeds the useful power $r\dot{\xi}^2$ which means that on purely radiation grounds one must with smaller membranes at low frequencies expect more wattless than useful power. As the radiation resistance at low frequencies is small, the diaphragm then needs greater velocity amplitude to radiate a given power. Fig. 163 shows how the pulsation volume varies with frequency for a sender of zero order producing constant pressure amplitude. At low frequencies the wave-lengths are so large that the pressures in the air each side can be equalised; thus at low frequencies the air moves to and fro with the membrane velocity but it is not much compressed –

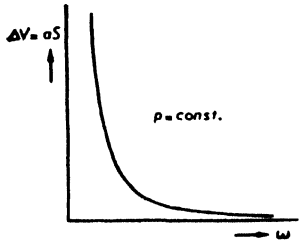


Fig. 163. Useful pulsation volume $\triangle V$ in relation to pulsatance ω for constant sound pressure.

sound pressure small, and acoustic power $p\dot{\xi}$ small. This compensation can occur with loudspeaker systems tuned low in the bass, for their diaphragms have large velocities, but if a horn is attached to the diaphragm to prevent 'spilling' of the air over the edge quite a large sound pressure may be produced at low frequencies, *i.e.* the source has a high radiation resistance. This quantity is to be integrated over the whole surface and not measured in one place or direction only if the radiation resistance is to be a true measure of output.

b) *Directivity*

Variation with direction occurs when the source is wide compared to the half wave-length or when, for a source of higher order, neighbouring parts give rise to sound pressures out of phase which therefore interfere in certain points. The first case arises with loudspeakers and horns only at high frequencies, the second in the bass region of loudspeakers without baffles and loudspeakers with acoustic delay lines radiating on one side only (*cf.* p. 362).

(ii) *Elements of the Radiation*

a) *Cone Diaphragm*

The commonest form of diaphragm is a cone, driven centrally and having great stiffness. At frequencies up to about 400 c/s the cone vibrates as a whole between the coil (held by the spider) and the fixing of the groove, but at higher frequencies this fundamental vibration is overlaid with partial modes, whereby the loudspeaker like a violin becomes a source of higher order. Radial (unsymmetrical) (Fig. 164) and concentric (symmetrical) nodal lines can then be observed.

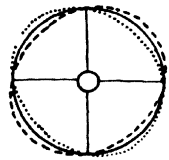


Fig. 164. Vibration form of cone diaphragm.

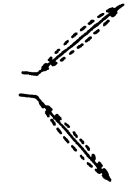


Fig. 165. Son rauque.

Exact calculation of the vibration over the shell must include at least the flexural dilational motions and lead to differential equations of higher order whose solutions are very complex[42] and approximations

to which are not of much value. In membranes which can unroll there appears at a certain intensity an undertone [43] (*son rauque*), due to cracking of the membrane, of half the fundamental frequency (Fig. 165), which can be mitigated by replacing the bevelled case type of cone with one which cannot unroll.

Effect of Material

In general one can state; A material which is light enough ($\rho \to 0$) and hard enough ($E \to \infty$) not to give partial vibrations does not exist. Degussit has the highest sound velocity ($\sqrt{E/\rho} = 10{,}000$ m/s). Thin membranes are liable to flutter, to produce undertones and to clatter; loudspeakers with thick heavy diaphragms have slight linear and non-linear distortion but low efficiency [44]. Membranes are generally made of paper pulp and moulded. If they possess little internal damping they are liable to exhibit strong resonance, but with too much, *e.g.* by admixing wool and grass to the pulp, the high frequencies are suppressed. It is best not to make the whole diaphragm of damping material but only the outer zone as this reduces the overtone structure and the tendency to undertones. Impregnating the membrane with a softening reduces the groove stiffness and raises the damping; also cuts off the reflected waves from the edge which in turn hinders the breaking up of the frequency curve and the formation of undertones [45]. At the highest frequencies only the inner part of the cone vibrates and the mass factor is somewhat smaller. Circular ridges are impressed on the diaphragm so that at high frequencies the outer part is separated from the inner. Membranes unaffected by acid and damp are made also of vinifol; or of triacetate leaf, unaffected up to 140° C (*c.f.* the Tropodyn loudspeaker of SIEMENS-HALSKE) [45a].

There are also types with double cones and coils [46]. At low frequencies the bigger coil drives the bigger cone and at high frequencies the smaller coil drives the smaller cone, the critical frequency being determined by the acoustical, electrical and mechanical networks. To radiate high frequencies a small calotte or second small cone is clamped over the coil [47].

b) *Sound Baffles and Housings* [48]

α. *Baffles*

As phase differences of 180° are set up on opposite sides of a vibrating

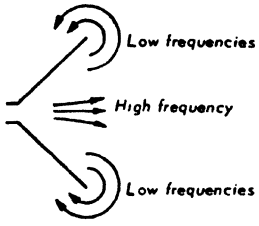


Fig. 166. Acoustic short circuit at low frequencies of a vibrating membrane without baffle.

membrane – excess pressure and defect – these can interfere destructively if the wavelength is large compared to the membrane diameter (Fig. 166). A baffle is therefore used, otherwise the music will sound thin, *i.e.* the low frequencies are poorly radiated. The size d of such a baffle can be calculated from the equation $f_0 d = 100$ where f_0 is the lowest frequency to be radiated, usually a normal baffle is 90 × 110 cm for a critical frequency f_0 of 100 c/s.

β. *Housing*[49, 50]

Internal and external diffraction are distinguished[51]. The former depends on the detour which the sound from the back has to make to reach the front, the latter on the surface which obstructs the radiation in the rear space (Fig. 167). The latter can be understood by considering

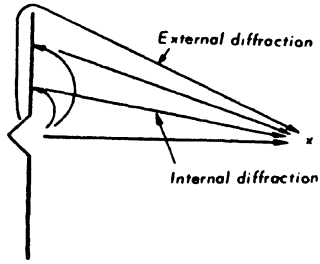


Fig. 167. Baffle diffraction.

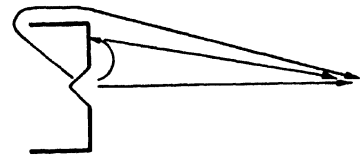


Fig. 168. Housing (For the same internal diffraction the external diffraction is less than with a baffle.)

FRESNEL zones. So it happens that of two housings having the same path-length from behind to before, the one which has a smaller front side (Fig. 168) is not so useful at low frequencies as one with a plane baffle. This happens in many broadcasting equipments as a defect in comparison with a loudspeaker mounted in a baffle.

γ. *Bass Reflexcabinet*[52]

Complete closure of the space behind the loudspeaker is not usually possible as the stiffness of the air cavity would then make the loudspeaker's natural frequency too high, but a slit can be made in the casing to prevent this rise, so turning the housing into an Helmholtz resonator. If this natural frequency is below the limit of reproduction,

the resonator like any resonance system has a phase lag of 180° and the radiation to the rear assists that to the front ('bass reflex cabinet'). If the volume of the box is small its air stiffness is large and the natural frequency of the Helmholtz resonator goes up. In such a case one can improvise a collar (as shown by broken line in Fig. 169) which raises the mass of the vibrating air and lowers the natural frequency. The calculation of the opening follows from a well-known formula (eqn. 33, p. 13).

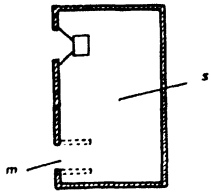


Fig. 169. Bass-reflex cabinet construction.

δ. *Corner Loudspeakers*

If a loudspeaker is placed in the corner of a room, its radiation is many times reflected and its radiation resistance at low frequencies is much improved. (RIEGGER's Patent). Latterly KLIPSCH[53] has used this construction in special cases.

ε. *Experimental Horn*

A horn[54] can notably increase the radiation resistance and so the efficiency. By adopting a wave equation due to WEBSTER[55]

$$\ddot{\varphi} = c^2 \left\{ \frac{\partial^2 \varphi}{\partial x^2} + \frac{1}{S} \frac{\partial S}{\partial x} \cdot \frac{\partial \varphi}{\partial x} \right\}$$

one can, from the cut-off frequency $f_0 = \frac{\mu c}{4}$ raise the radiation resistance considerably by using a horn whose cross-section is given by $S = S_0 e^{\mu x}$, The calculation is true only for an infinitely long horn. If it is cut short at under 45°, no reflection occurs at the far end. Nevertheless for a frequency of 100 c/s one needs a width at the neck of 2 cm, at the opening of 2.00 m and a length of 2.4 m.

ζ. *Conical, Hyperbolic and Spherical Wave Horns*

Conical horns have a very slow rise of radiation resistance with frequency and so are hardly used in loudspeakers. Hyperbolic horns show a steeper rise. In calculating spherical wave horns[57] this type of wave is taken as basis instead of a plane wave. For the same cut-off frequency, they are shorter than exponential horns and pass asymp-

totically into a plane surface like that of ordinary baffles and sound more natural than these because of the continuous rise of radiation resistance.

η. *Folded Horns*

To reduce length, boxed horns are often used, *e.g.* by OLSON[58] to radiate the low frequencies from the rear, the high frequencies from the front of a cone loudspeaker, and so attain an efficiency of 50% (Fig. 170). So also small boxed tin horns are used for 'inter-com' and 'walkie-talkie' equipment in which low frequencies do not matter so much as in music. The cost however of these is not less than for a large hornless system, but the efficiency is greater and one needs only midget amplifiers with the advantage of portability, *e.g.* in motor vehicles.

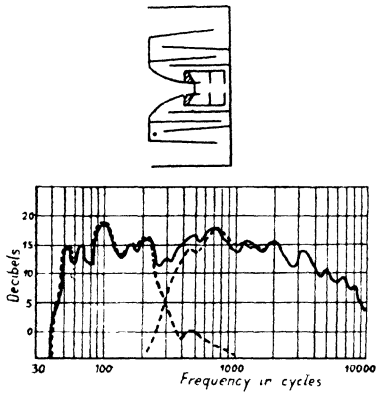


Fig. 170. Construction and response curve of a compound horn loudspeaker.

θ. *Adaption of Horn to Loudspeaker*

At the vertex of the horn a pressure capsule is accommodated whereby the pressure is transformed into velocity (Fig. 171) and better matched to the specific impedance of the air. It is better to introduce spacers in the shape of an onion, to modify the growth of the cross-section and to ensure a rigid diaphragm up to the highest frequencies, which involves building it as a calotte.

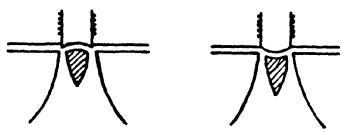


Fig. 171. Pressure chamber system.

ι. *Directivity of Horns*

RAYLEIGH[59] pointed out that a horn begins to form a beam when its aperture equals the half wave-length of the sound. This property was further examined by OLSON. At high frequencies this effect no longer increases so that to fill a space with radiation many loudspeakers in parallel are necessary; this takes the form of a multicellular loud-

speaker (Figs. 172 and 173). Sound lenses are also, in jalousie fashion,

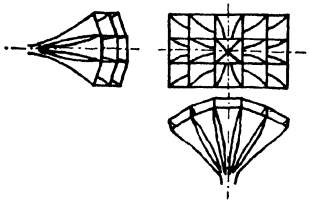


Fig. 172. Multicellular horn.

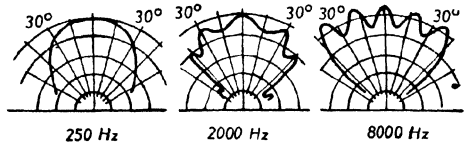


Fig. 173. Polar curves of multicellular horn.

arranged in front of the horn [61] (Fig. 174). On the broader parts of the lens elements the sound is retarded and propagates more in spherical

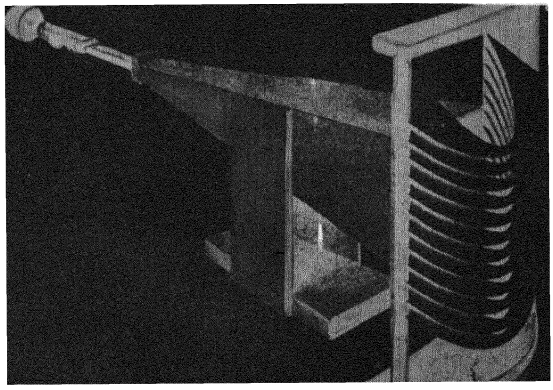


Fig. 174. Lens and horn combination.

than plane wave fashion. With such a lens one can, *e.g.*, increase the diffracted angle from 7° to 30°.

κ. *Groups of Loudspeakers* [61, 62]

It is often better only to radiate in a determined part of the total space, *e.g.* that in which the audience sits, and in this case groups of equipolar loudspeakers can increase the directivity (Fig. 175). This relative intensity, for such a group of n, at any point, is given by

$$I = \frac{\sin(n \times \frac{1}{2} kd \cos\alpha)}{n \sin(\frac{1}{2} kd \cos\alpha)}$$

where k is the wavenumber, d the inter-distance of the individual loud-

speaker, α the bearing of the location from the group-axis. At high frequencies most of the energy tends to be found in a plane perpendicular to the group centre. At the same time the energy is bunched into the median plane, theoretically more so at high frequencies. The amplification factor for the sound pressure along a l meter line is 2.4



Fig. 175a. Group of six loudspeakers.

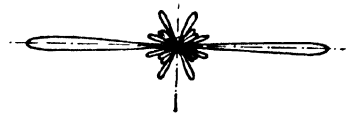


Fig. 175b. Polar curve of loudspeaker group.

$\sqrt{f}\,l$ where f is in kc/s[63], but at high frequency the various parts of the loudspeaker diaphragm do not vibrate in phase and at low frequencies the resistance of a radiator of first order grows with the third power of its surface so that the system has a straight line frequency response like that of a single loudspeaker. The bunching also results in less power.

A line of loudspeakers 1.5 m long has a bunching angle of about 20° in the middle of the musical scale, but this angle can be reduced by using several arrays.

λ. *Uni-directional Loudspeaker-groups*

There are occasions when no sound should be radiated to the rear, when, *e.g.* as in the Berlin Forest Theatre, this would give rise to a disturbing echo. This is easily done at high frequencies by encasing but at low frequencies diffraction supervenes. It is possible to introduce delay mechanisms into the sound going out towards the rear so that it interferes with that from the fore side[64]. Such acoustic delay lines consist of an air cavity to give stiffness and a damping material to introduce friction (Fig. 179), which like an electric circuit with capacitance and resistance delay the sound by a determinate time constant.

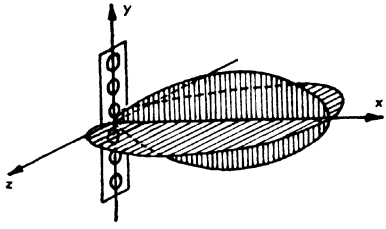


Fig. 177. Construction of three-dimensional polar curves for loudspeaker group.

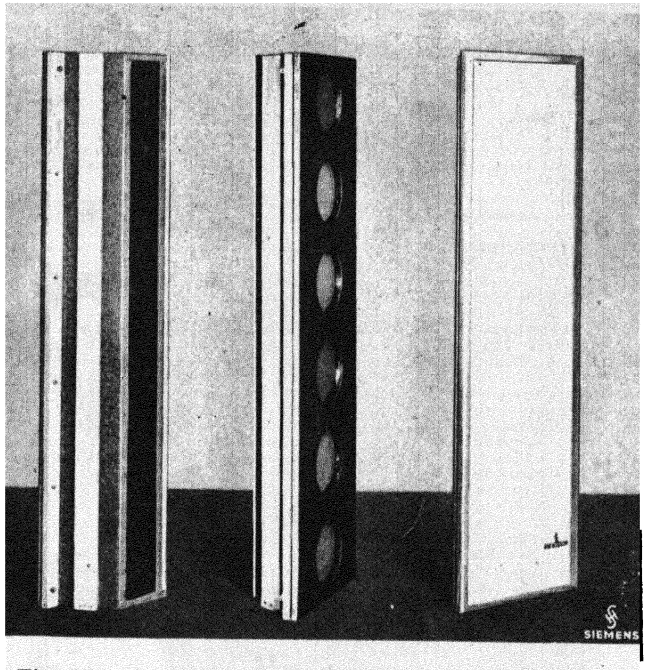


Fig. 176. Siemens loudspeaker group (front and rear).

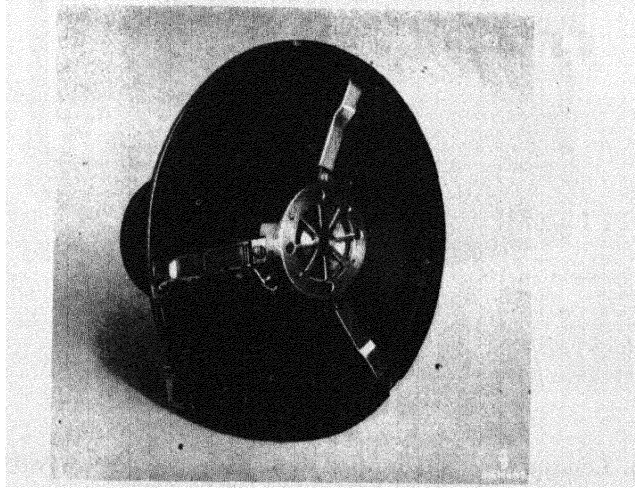


Fig. 178. Coaxial loudspeaker (12.5 watt).

Such loudspeaker or arrays of loudspeakers have a kidney-shaped polar curve (Fig. 177) and so do not propagate sound behind even at low frequencies.

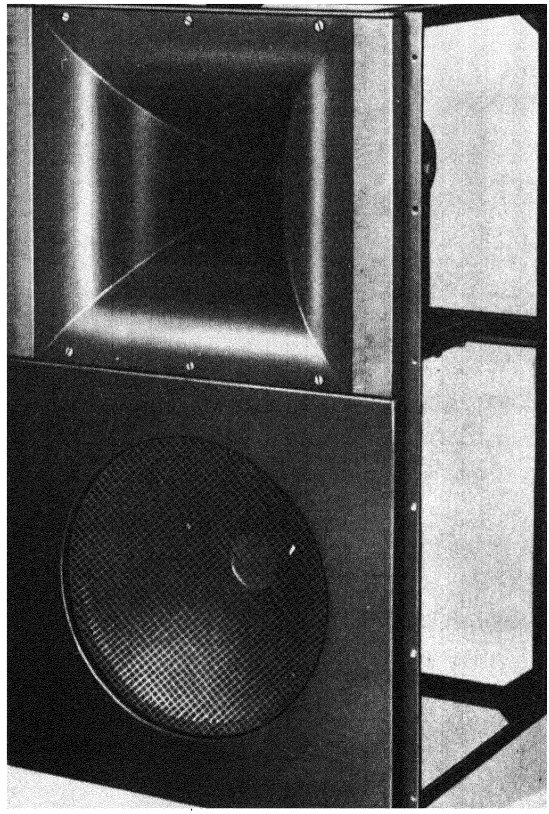


Fig. 179 (a). Loudspeaker combination with high-frequency spherical-wave horn (Siemens-Klangfilm), front view.

μ. *Loudspeaker Combination*

As the frequency characteristics of loudspeakers are very different it

:st to cover the whole gamut with three loudspeakers, one for the , one for middle and one for the upper frequency band. As at

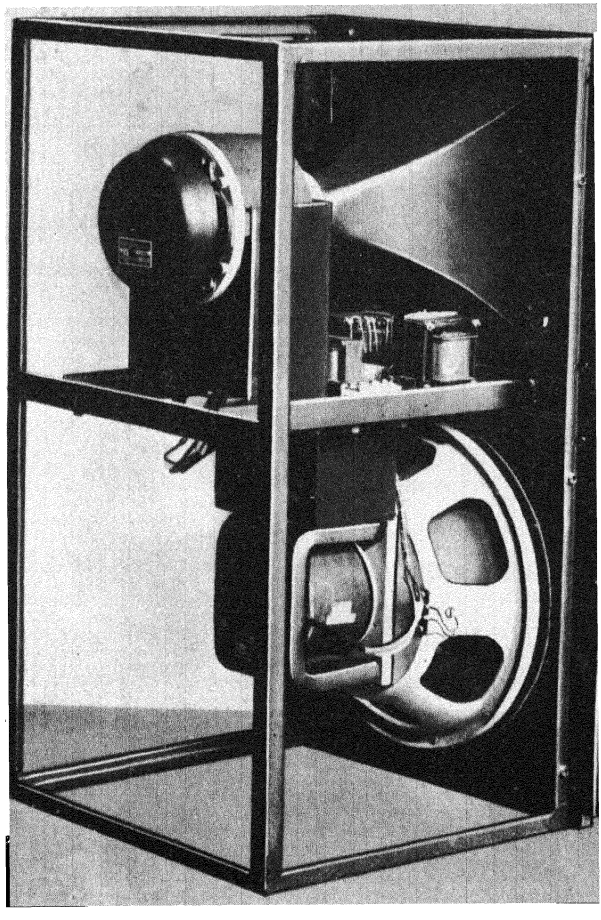
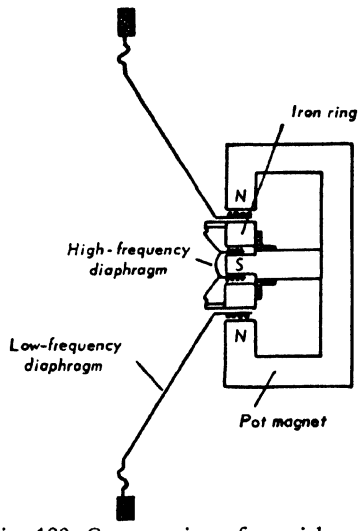


Fig. 179(b). Combination of Fig. 179(a), rear view.

frequency only the central portion of the cone of a large loudker moves, it is better to reduce the mass of copper to be moved

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in special high frequency speakers ('tweeters'). Fig. 178 shows such a combination with tweeter axially mounted; Fig. 179 shows the Eurodyne[65] with the spherical-wave horn on the high frequency loudspeaker; the two components have a common magnet[66]. For large cinemas a combination such as Fig. 181 is used[66a]. To propagate the bass a similar type of horn is used, giving resonant-free reproduction, high output and small depth of construction.

Fig. 180. Construction of coaxial combination with single magnet.

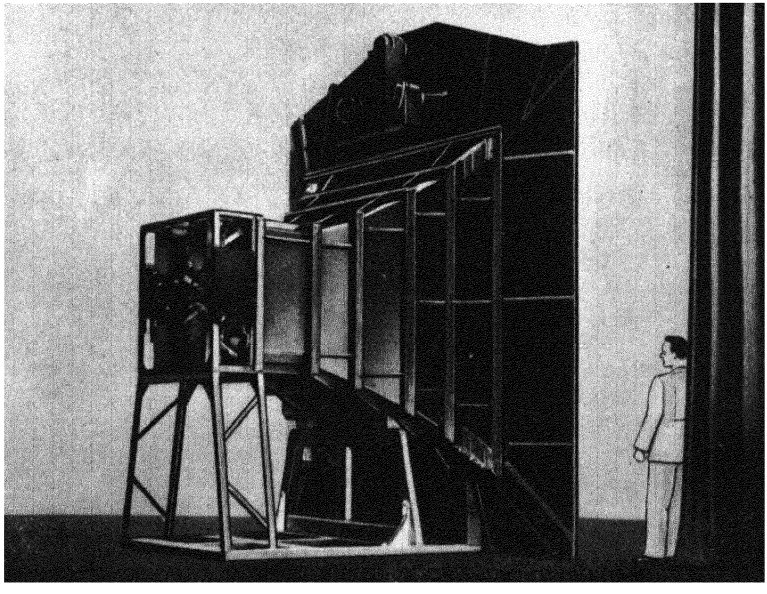


Fig. 181(a). Combination for large cinemas (Euronor). (Siemens-Klangfilm), rear view.

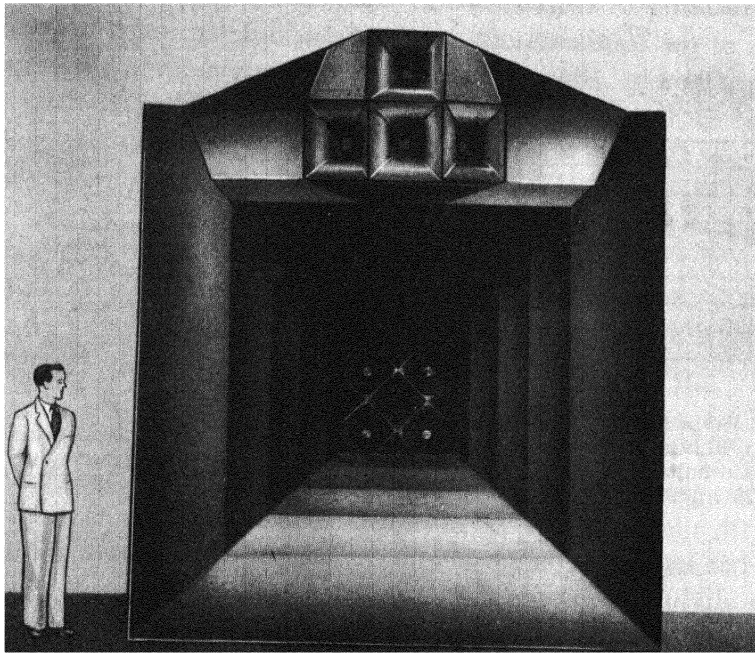


Fig. 181(b). Combination of Fig. 181(a), front view.

§ 5. Measurements of Loudspeaker Characteristics

In testing loudspeakers objective or subjective methods can be used. In the main these agree but it is difficult to translate fine differences in the objective measurements in terms of audible characteristics.

(i) *Frequency Curves*

A good criterion for an acceptable loudspeaker is the frequency; response curve, measured in the open or in an anechoic chamber. Fig. 182 shows such objective curves which were reproduced as audible differences in subjective tests (*vide* p. 344).

(ii) *Transients*[65] (*cf.* Chap 18, § 32).

When a tone is suddenly applied there occur at frequencies, for which the response is a maximum, a gradual rise and at a response minimum

an overpassing of the steady-tone response (Fig. 183). Oscillograph records of the transients of a brief tone at different parts of the scale appear as in Fig. 184. A narrow but deep crevasse in the response

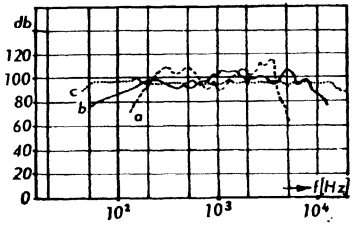


Fig. 182. Response curves of loudspeakers.
a) Old horn type.
b) Common cone type.
c) A high fidelity combination.

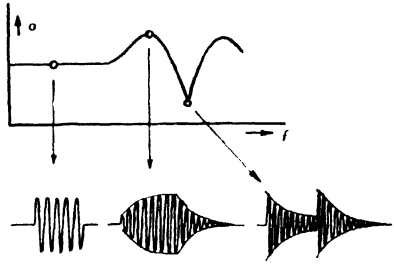


Fig. 183. Relation between response curve (above) and transient curve (below).

curve has less effect on speed and music, which also have transients, than a slight, or trough above or below normal over a wide band. Fig. 185 shows the intensity of the transient phenomena at different epochs after cutting off the tone. In order that the transient behaviour of a loudspeaker in speech or music shall not be heard, the amplitude

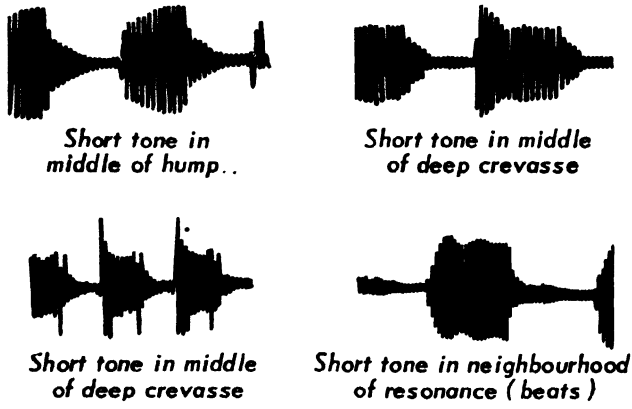


Fig. 184. Transient phenomena in different parts of frequency curve (DARRÉ).

after 10 ms must have sunk 30 db. Fig. 186 shows the influence of phase δ or group time $d\,\delta/d\omega$ [66]. In the bass where the diaphragm is not subdivided there is a close relation between this factor and the frequency curve.

(iii) *Electrical Impedance:* $R_v^* = R_v + \frac{M^2}{Z_m}$

The electrical impedance is measured on an impedance bridge and because of the inductance of the moving coil rises with frequency.

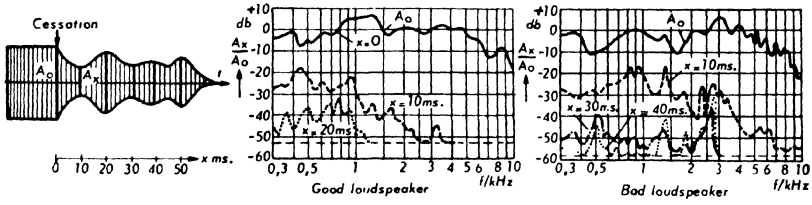


Fig. 185. Response curves of amplitude decrement at times after switching off exciting current (J. MOIR).

Depending on how highly the loudspeaker is damped there occur at resonance greater or smaller increases of impedance about M^2/Z_{res}. Fig. 187 shows the trend of resistance in the complex plane.

(iv) *Non-Linear Distortion*

Non-linearity of the mechanical restoring force, feedback of the magnetic circuit, inhomogeneity of the magnetic field are faults in the driving system together with some in the diaphragm which lead to non-linear distortion – and the latter may vary from frequency to

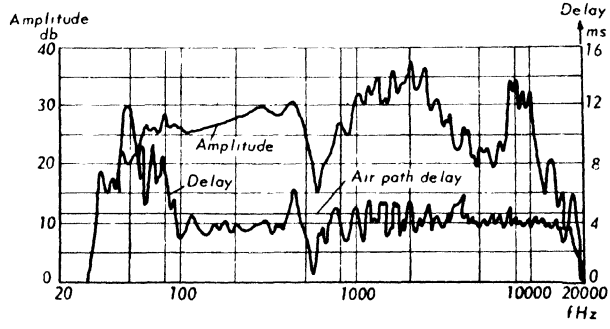


Fig. 186. Amplitude and delay time of loudspeaker (EWASKIO and MAWARDI).

frequency because of the natural resonance of the diaphragm. The harmonic distortion[67] can generally only be measured from point to point. The difference-tone factor and the modulation-tone factor can

be measured continuously. The first is better for the high frequencies, the second for the low. Fig. 188 shows in spatial form relative to modulated f_o and modulating frequency f_m the modulation factor m (modulation peaks) of a loudspeaker. Better loudspeakers at moderate power show only a few per cent harmonic distortion.

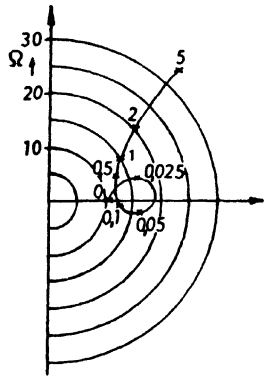


Fig. 187. Impedance of loudspeaker.

(v) *Polar Curves*

The sound pressure in the forthright direction up to 10 kc/s is propagated uniformly from a cone loudspeaker, but in directions to one side there is a falling off since owing to symmetry all parts of the membrane do not aid each other. At an angle of 35 λ/R the first minimum occurs, R being the membrane radius. Fig. 189 shows the polar curve

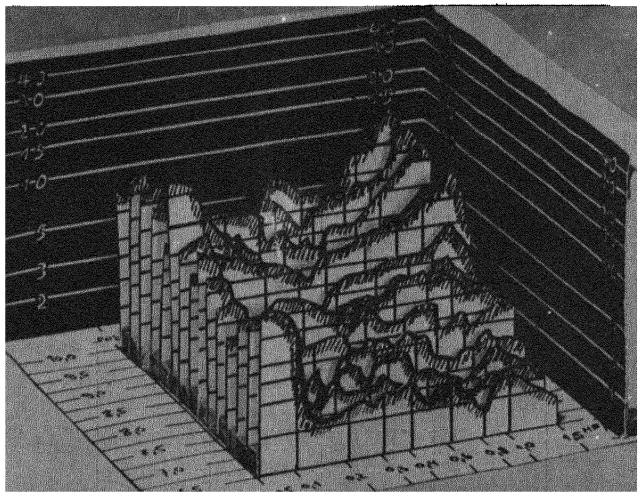


Fig. 188. Modulation distortion of a dynamic loudspeaker, (f_o to rear, f_m to right, m vertical).

of a common cone loudspeaker; Fig. 173 that of a multicellular system, which is used to obviate the sharp directional properties.

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(vi) *Subjective Estimation*

To get more detail one must submit the estimation of a loudspeaker to listening by a larger number of persons. Subjective errors in preliminary trials are to be cut out as far as possible. The errors can be artificially increased by iteration, recording the loudspeaker sound on a magnetophone, so that at each recording the error is magnified, on the understanding that the magnetophone has better quality than the loudspeaker.

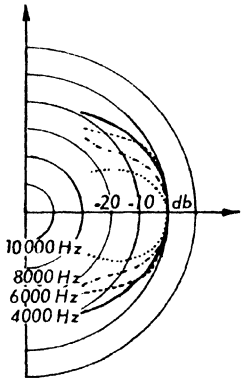


Fig. 189. Polar curves of cone loudspeaker.

The style of recorded speech or music has an influence on the subjective assessment; the same loudness and frequency band should be involved in the comparison. A narrow band being better than a wide one[68], a better loudspeaker may sound worse with a wide-frequency band than a common one with a narrow-frequency band.

The position of the source in a room has a big effect on the assessment and with good loudspeakers may make more difference than small differences in physical make-up.

§ 6. Location of Loudspeakers

(i) *Influence of the Room*

The position of a loudspeaker in a room can influence the sound reproduction in various ways. We have already noted that a loudspeaker in a corner propagates the low frequencies better than one in the body of the hall. The properties of the room, especially reverberation time and the manner in which loudness falls off, have a great effect. If the fall of loudness in the first 50 ms is small the direct sound is favoured ('good definition'), speech sounds precise and the orchestra stands out well. With exponential reverberation an impression of fullness is attained. The diffusion factor is also important (*cf* Chap. 5).

(ii) *Pseudostereophony*

The unnatural impression of the sound of a large orchestra emerging from a small loudspeaker is disturbing; it is better to have two loudspeakers in parallel arranged at some distance from each other. If the loudspeakers are at different distances from the listener, there occur in a lagged chamber physiological effects whereby the listener has the impression of an increase of loudness (AIGNER-STRUTT effect[69]).

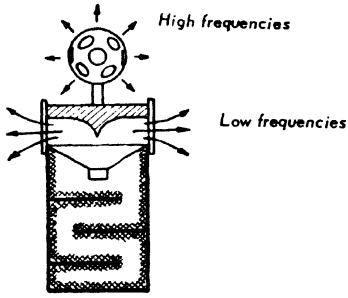


Fig. 190. Arrangement of loudspeakers; spherical-wave radiator and low-frequency cabinet (KÖSTERS).

There are a number of arrangements, so-called pseudostereophones[70], in which separation of the frequency bands on different loudspeakers or indirect radiation gives the binaural spatial effect on a single channel. The loudspeaker combination due to KÖSTER-HARZ[71] of the Nordwest Deutscher Rundfunk (Fig. 190) gives this impression. Herein a sphere with 12 tweeters radiating in all directions is mounted over a low frequency loudspeaker which radiates one sideways. The response curve of this system is so arranged that not only is the output independent of frequency on the axis but in all directions within the room on the average a smooth frequency curve is attained.

(iii) *Decentralised Equipments. Haas Effect*

If the acoustics of a hall are improved by a public-address system, that is, with loudspeaker and source in the same chamber, many single loudspeakers can be distributed among the audience; but then sight and sound are differently orientated[72]. This can be prevented by having a central public-address system of sharply directive loudspeakers or an array which operates at such power that to the audience there is great contrast between direct and reverberant sound. In this way in rooms of reverberation time up to 10 sec complete audibility is possible. If the loudspeaker sound comes to the listener 5 to 50 ms later, and the direct sound is not louder than it by more than 10 db, the operation of the public-address system is not noticed. (HAAS effect *cf.* Chap. 5.)

§ 7. Telephones

(i) *General*

A telephone has various uses, the main one being to transmit speech over a long distance; a frequency band 300 to 3400 c/s suffices for this[73]. The transmission of music which in the first instance was done by detector apparatus and heard for a considerable distance on telephones is now seldom concerned with headphones, which are mostly used for listening to noises or testing broadcasts, *e.g.*, on mixing consoles in studios, where the whole range is listened to on telephones. For deaf-aid systems[74](p. 272) and audiometers specially light telephones are used. For indicators, *e.g.*, on a.c. electrical bridges, very sensitive telephones are built[75]. Resonant telephones can also be used for underwater listening. Sometimes they are inserted for special purposes in pressure chambers as sound sources or thermophones.

The transducer principle herein is fundamentally the same as for loudspeakers, for which the electromagnetic coupling coefficients have already been given, only the order and magnitude are different. This is because one does not demand the same frequency characteristics with a telephone as with a loudspeaker. The telephone is shut off by the ear so that it is easier to drive it at low frequencies with small power.

(ii) *Sound Output and Transduction Factor*[76]

One can scarcely speak of the efficiency of telephones as they only give acoustic wattless output. With the sound field cut off by the ear it is quasi-stationary. The telephone works in a closed chamber of volume V and shut off by a mechanical impedance. The artificial ear has such a terminal which corresponds with the natural ear.

The transduction factor of a telephone in an acoustically hard chamber is calculated thus;-

Pressure causes a change of volume ΔV given by

$$\frac{p}{p_0} = \chi \frac{\Delta V}{V}$$

χ being the ratio of specific heats and p_0 atmospheric pressure. If

S is the surface and ξ the displacement of the pressure membrane under force F related to the current i by

$$F = Mi$$

and i to the applied potential u and apparent resistance R^* by

$$i = u/R^*$$

As also $\Delta V = \xi S$; $\omega\xi = F/Z$; Z mechanical impedance. Hence the required factor

$$\frac{p}{u} = S\frac{M}{2}\frac{\chi p_0}{\omega V R^*}$$

which is the greater as S and M increase and Z decreases, *i.e.*, as the mechanical restraint $S/\xi = Z\omega$ is made small.

This restraint, which at low frequencies, is equal to the membrane stiffness s, cannot in practice be made very small because of the danger of chattering. From this it follows that even with small stiffness the resonant frequency would be too low.

To ward off outside disturbing noise and to reduce the mechanical pressure on the ear, rubber pads[76] can be drawn over the headphones which reduce the background noise in noisy locations by as much as 15 db.

(iii) *Types of Telephone in Production*

Only the most significant driving systems will be discussed.

a) *Electromagnetic System*

The common electromagnetic system[77] (Fig. 191) has been improved

Fig. 191



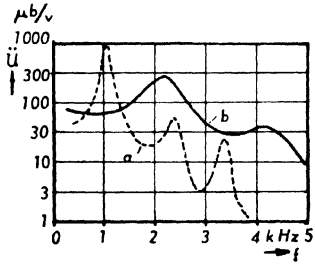


Fig. 192

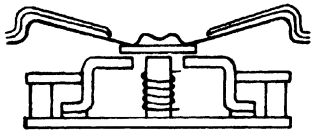


Fig. 191 and 192. Simple electromagnet telephone in relation to a ringpole system.

with passage of time in that the pole shoe for the magnetic flux is now made of good-conducting material. The alternating flux is given its own magnetic path and airgap parallel to the permanent magnet. In a further development[78] (Fig. 192) the core is made smaller and the diaphragm of aluminium, thus enabling in connection with a concentric drive the fundamental frequency to be raised from 1000 c/s to 2000 c/s. Above this the diaphragm subdivides and more resonances appear. By selecting the magnet material and the dimensions of core, driving system and permanent magnet, 91 % of the total alternating flux, against an original 19 % can be made to pass through the core.

A further advance in the development of a quadrupole system; another German invention produced a wider frequency band with greater sensitivity. The core was small and the special diaphragm was of plastic stiffened by grooves. With such systems batteryless systems can be built – *e.g.* in mines where on account of the possibility of explosions the requirement of supply batteries is undesirable – whereby the same system can act as sender and receiver coupled with trumpets for ear and mouth. Such systems are used instead of speaking tubes on board ship.

A more recent development is the 'ringmagnet phone'[79] (*cf.* Fig. 193). It has a symmetrical magnet with a light magnetic core and light coupled diaphragm of stiffened plastic, stiffened round the edge

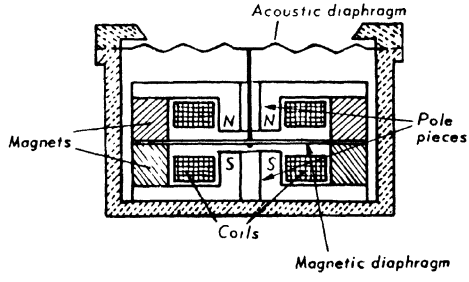


Fig. 193. Construction of ring-magnet 'phone.

by corrugations. With a modern Alnico magnet the magnetic induction is raised from 5000 to 8000 gauss. By increasing the pole cross-section tenfold the electromechanical coupling coefficient M can be raised fifteen fold. To reach the higher resonance the stiffness must be 15

times that of the normal telephone, yet because of its rigidity the acoustic membrane drives a volume twelve times as big as the magnetic diaphragm. So the sensitivity and band width are greater than in the

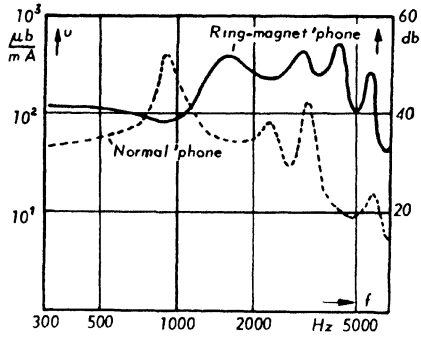


Fig. 194. Sensitivity of ring-magnet 'phone and of common type.

common system (*cf.* Fig. 194), which produces a case damping[80] of — 1.1 neper against — 0.1 to — 0.4 neper for the latter.

Such sensitive telephones permit thin cables to be used; they can serve for telephony with soft speech and enable the hard of hearing to telephone. Also with the 'electric porter' a speaking connection between the listener and the person waiting outside can be assured.

A similar loudspeaking telephone was built in U.S.A.[81]. By abandoning the electromagnetic drive from the middle towards the edge it is possible to get a greater poleshoe cross-section, *i.e.* longer

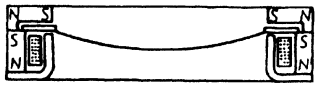


Fig. 195. Ring-anchor 'phone.

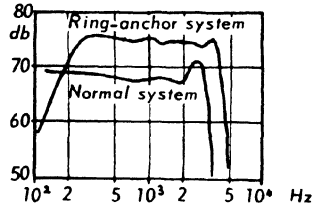


Fig. 196. Response of ring-anchor 'phone compared to that of normal type.

magnetic flux and so greater electromagnetic coupling coefficient. The diaphragm is of calotte-form and remains rigid at the higher frequencies (Fig. 195 shows the construction and 196 the response curve).

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Midget systems are used for deaf aids. With an olive-shaped case the telephone can be directly pushed into the meatus[82].

Magnetic systems are also built as deaf aids or as bone-conduction receivers (*cf.* Chap. 11). The armature swings freely in space and the case is laid directly on the mastoid[83].

The magnetic telephone[84] is mostly used in telephony as a capsule with 60 ohm d.c. resistance and 200 ohm impedance at 1000 c/s. As headphone it has per shell 2000 d.c. ohms and 7000 a.c. ohms at 1000 c/s.

b) *Dynamic System*[85]

Dynamic systems are used up to 5 kc/s and are built like high frequency loudspeakers, calotte diaphragms provided with moving coils. Owing to the large magnet they are heavier, *i.e.* less sensitive for a given magnet weight. By electric filters one can match the frequency response up 10 kc/s[86].

c) *Piezoelectric System*

The piezoelectric system is built in the same way as the corresponding microphone (*cf.* p. 317) They employ crystal benders, provided with diaphragms and have the advantage of extreme lightness[87].

The thermophone, electrostatic and induction principles[88] are scarcely ever employed as telephones.

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CHAPTER 15

SOUND RECORDING

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§ 1. Introduction

Sound-recording[1] has the object of reproducing a sound sequence once given after a given time interval and at a given location. As such it is used in news transcription (telegraphone, dictaphone), for relaying music (sound disc or film), and for technical purposes *e.g.* for the control of machine-tools. All sound-recording equipment employs a moving carrier, disc, film, magnetic tape or wire on which the sound-track is inscribed by a special operation, and of which copies can be made, in order to be played back in a second operation.

The quality of a sound record depends on the usual electroacoustic factors (frequency band response, non-linear distortion, dynamics, stereophonic and room-acoustic effects) and especially on the constancy of speed of the carrier. Different methods of assuring constant speed are in use depending on the application. In general a relative frequency-warble ($\Delta f/f$) of 3% can be heard but the permissible value varies very much depending on whether piano, song, string or dance music is concerned. The amount of sound diffusion in the auditorium also plays a part. Up to the present three fundamentally different types of recording have reached a considerable stage of perfection; mechanical (disc, foil, tefiphone), photoelectric (sound film, Philips-Miller system) and magnetic (magnetophone with tape, disc or wire). Each of these has certain advantages and suitability for certain applications.

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§ 2. Mechanical Recording

(i) *Historical*

EDISON in 1877 first produced a sound recorder. A stylus, attached to a diaphragm with a trumpet, inscribed a groove of varying depth (hill-and-dale, Fig. 197) in tune with the sound amplitude. The reproduction was done with the same apparatus. The change from drum (phonograph) to disc (gramophone) made possible the manufacture of copies from the matrix by pressure—impossible with the drum — and reduced the required storage space. Up to the present there is no other recording apparatus which affords such a simple way of making copies.

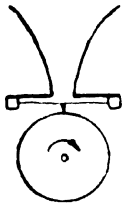


Fig. 197. Principle of hill-and-dale recording.

In 1887 BERLINER invented the transverse or sideways recording, in which the movement of the stylus took place radially to the disc (Fig. 198). Electroacoustic equipment for recording and reproduction appeared about 1925. In mechanical recording the musicians must be very close to the trumpet and the small amount of sound energy picked up must be amplified by mechanical resonance. Electroacoustic amplification, on the other hand, permits whole orchestras to be located at a distance to give the proper acoustical conditions pertinent to the room in which they play and with a balanced frequency band. Again in the reproduction with electrical amplifiers one can safely let the sound fill a large room as, for example, of a whole cinema with the early needle-recorded sound film. Before this for such cases one tried to use the auxetophone (*cf.* p. 333)—a sound-controlled jet of compressed air—and in this way to reach a purely mechanical reinforcement.

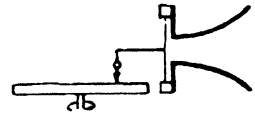


Fig. 198. Principle of transverse recording.

(ii) *Frequency Response of Disc*

As the radiated sound power $P = r_s \dot{\xi}^2$ and in an exponential horn the radiation resistance r_s is constant above the cut-off frequency, it would be desirable in purely mechanical recording on disc to keep the velocity $\dot{\xi}$ constant for equal pick-up sound pressure. If a is the deflection, $\dot{\xi} = \xi\omega$, requiring the deflection to be so great at low frequencies that

the stylus would overrun into neighbouring grooves. Formerly the cutting equipment was given a response curve that gave constant *amplitude* (ξ) below 250 c/sec, constant velocity ($\dot{\xi}$) above.

In magnetic recording the voltage ($E = M\dot{\xi} = n\dot{\xi}\ \partial\varphi/\partial a$) is proportional to velocity and so must be given a certain distortion correction. (M is here the magnetic coupling coefficient, *cf.* Ch. 14, p. 335.)

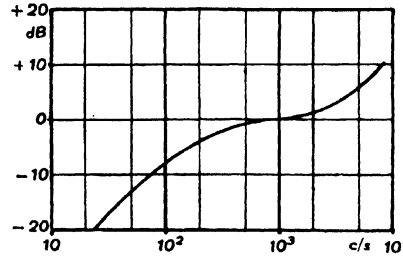


Fig. 199. Cutting characteristic of normal groove (at 78 revs/min).

Nowadays one uses a greater cutting amplitude at high frequencies so that they have a level frequency response at constant velocity[2]. In this way a greater signal-to-needle noise ratio is attained, resulting in a closer approach to constant-amplitude cutting which corresponds to the characteristics of a crystal pick-up.

(iii) *Preparation of the Disc*

The preparation of the disc takes place in three stages.
(1) a wax or lacquer plate is cut out,
(2) a matrix is prepared by electro-plating,
(3) the disc is pressed.

(iv) *Magnetic Recording*

In artistic high-fidelity recording a monitoring operator must control the auditorium acoustics and the positioning of the microphone in relation to the sound sources. Nowadays the programmes are first recorded on magnetophone (*vide infra*, section 4). This has the advantage that the entertainment can be heard again and in case of failure or mistakes can be repeated. 'Cutting' of programmes is so made possible and further one can make, with the aid of a preliminary jockey, long-playing records with groove-separation dependent on relative intensity. Finally, cutting the individual discs can be better

controlled against vibration and temperature variation, and with the easily portable magnetophone one can record anywhere. The condition for this is that the magnetophone (high-frequency type) exceeds in fidelity the sound disc, if some reduction in quality is not to ensue.

(v) *Original Equipment*

For the magnetophone a disc is used that was formerly made of wax about 3 cm thick but is more often today a foil with a metal underlay. Then a cutting apparatus works on the foil rotating on a wheel at 78, 45, or 33⅓ revs per min, being pushed gradually from the circumference towards the centre so as to cut a spiral groove several hundred metres long.

(vi) *Cutting Head*

The cutting head which makes the groove on the wax plate or foil can have a magnetic, dynamic or crystal drive, Figs. 200, 201 (see also Chap. 14), which moves the cutting stylus of sapphire rhythmically

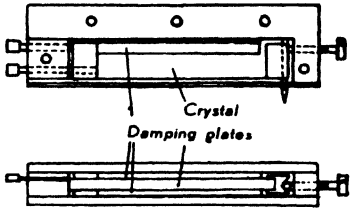


Fig. 200. Piezoelectric cutter.

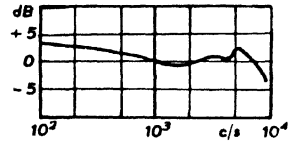


Fig. 201. Frequency response of piezoelectric cutter.

with the sound vibrations. Usually it is a magnetic symmetrical system in which an armature vibrates between two or four magnet poles when the electric current passes. Either the system is tuned above the recording gamut (Telefunken[3]) or is strongly damped in grease[4] (Fig. 202). Finally also electromechanical feedback is found, having smaller distortion and a smooth response curve up to 20 kc/sec.

Western Electric Co., have built a dynamic hill-and-dale cutter on the principle of the dynamic loudspeaker (p. 336); Fonofilm, Copenhagen[5], a transverse cutter on the principle of a moving coil instrument (Fig. 203). Its frequency response runs smoothly from 100 c/sec. to 25 kc/sec. (Fig. 204). It is possible to hear the recording on the feedback

coil while it is being made. A feedback amplifier free from phase distortion is requisite.

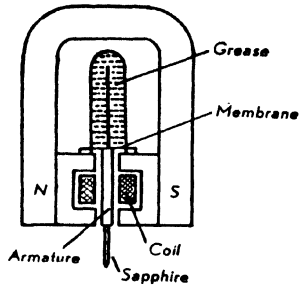


Fig. 202. Electromagnetic cutter.

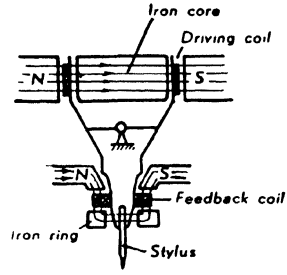


Fig. 203. Fonofilm cutter.

(vii) *Cutting Stylus*

Different styli are employed for normal and micro grooves. The normal grooves are about 120μ broad and 80μ deep, with 250μ

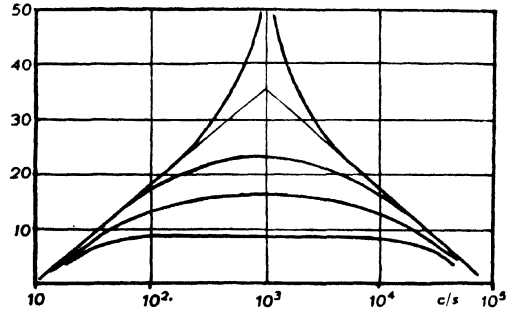


Fig. 204. Frequency response of fonofilm cutter for various damping factors.

spaces; the micro groove 70μ broad and 50μ deep with 140μ spacing. So the normal has 4 grooves to the millimetre, maximum deflection 60μ; the narrow has twice as many with 35μ maximum deflection. Seen from the front the cutting stylus has a triangular profile and from the back is ground to a tetrahedron Fig. 205), so that two sharp edges are formed on the sides. The rear ground angle of the stylus must be sharper than the angle between the sound groove and the interspace

so that, with the sideways motion of the cutter, the lacquer will not be crushed but cut away. The parings are sucked away in an airstream.

(viii) *Playing Time*

The playing time of a large commercial disc running at 78 revs/min is 4.5 minutes. If only the large amplitudes are cut at normal groove separation and the rest at closer spacing, this time can be increased to about 8 minutes. This is secured by a jockey forerunner on the magnetophone which controls the cutting head (*cf.* Fig. 206). On long-playing discs the speed is reduced from 78 to $33\frac{1}{3}$ revs/min, which enables most musical pieces to be played without break.

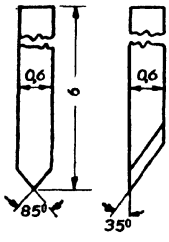


Fig. 205. Sapphire cutting stylus.

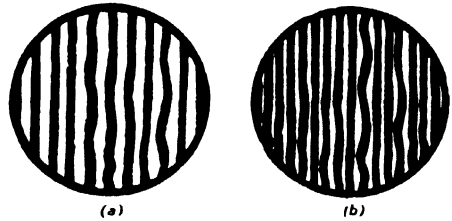


Fig. 206. (a) Normal disc with constant mean groove separation. (b) Long-playing disc with mean groove separation proportional to loudness.

(ix) *Vertical v. Transverse Grooves*

In 1932 and 1933 the original hill-and-dale recording was revised for high-fidelity discs[6]. Meanwhile this had only been used for dictation apparatus, because of the space-saving and long-running possibilities, with telephone quality of reproduction. The improvement in quality compared to the existing discs which had been reached by 1932 was due to the use of a better technique for galvanising and of the use of plastics without grain for the matrix. Plastics are so soft that a purely mechanical reproduction, imposing large forces on the needle, is not possible without spoiling the plate, so that only electrical reproduction can be used. It appeared that the hill-and-dale type has no longer any advantage over the transverse groove, for danger of spilling into neighbouring grooves and the longer running of the former was not so important that it could replace the transverse type, commercially established, and would in any case require a different form of pick-up.

(x) *Record Duplication*

No copies can be made from the original cutting. A hard metal plate must be prepared. By galvanic means a metal pressing of the original wax is made (Fig. 207). First the original is made conducting by steaming in vacuo or by cathode-sputtering, then it is laid in a galvanising

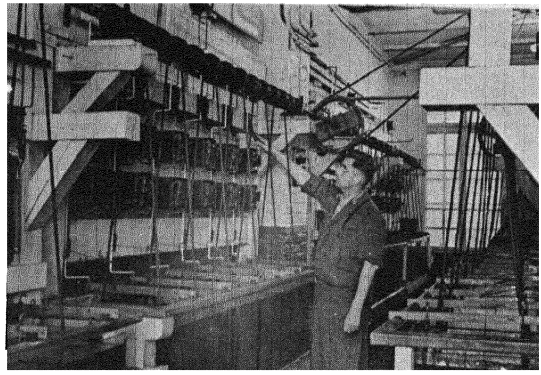


Fig. 207. Galvanising baths.

bath and by electrolysis another stronger metal layer is put on (metal master). From this negative a positive (metal mother) is made from which further negatives (working matrix) can be produced. The mother is kept in the archives of the gramophone company for further use when needed, while the working matrices go to the press room for final pressing of the actual discs to be sold.

The press consists of two steel plates in which the two matrices for the two sides of the discs are laid. A piece of the material is placed between the two matrices and is subjected to 80 atmospheres pressure (Fig. 208). The press forms are, in the meantime, heated by steam to about 150° C, and cooled afterwards by cold water so that the disc is not deformed on removal. The whole operation takes about one minute. In a large press room 100,000 discs can be made in a day, a rate of manufacture which is not reached in any other type of sound record making.

The compressed material consists of a mixture of one part shellac and three parts rock dust which gives a certain hardness and grinds the steel needle in the groove. If one wants a noiseless disc, but only

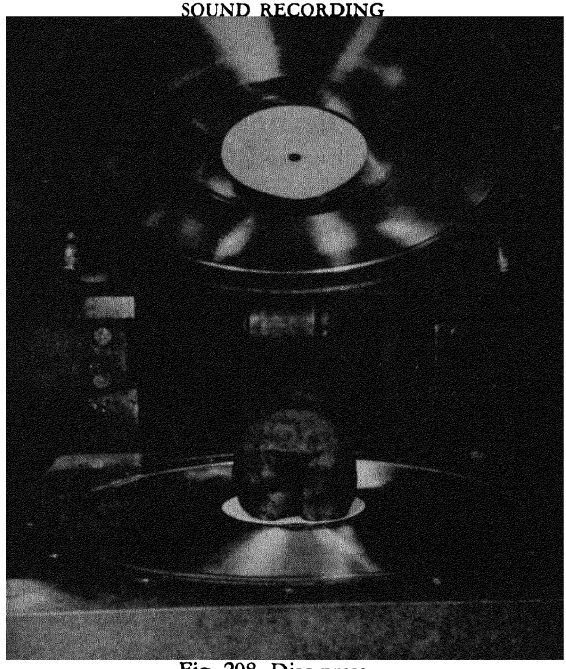


Fig. 208. Disc press.

to be used with electric pick-ups, a polymerised mixture of vinyl chloride and vinyl acetate, such as vinylite[8], is employed.

(xi) *Playing of Records*

Either mechanical or electrical pick-ups can be used.

a) *Mechanical Reproduction*

In the mechanical type a lever system works on a diaphragm that is operated through a pressure capsule from a horn. A number of resonances can intrude, but with a sufficiently long horn (several times folded on itself) one can reproduce over the range from 100 to 3000 c/sec. At higher frequencies, diaphragm and lever no longer move rigidly and do not completely follow the impulse applied to them. Because of their short horns, cabinet gramophones do not reproduce the lower frequencies (Fig. 210).

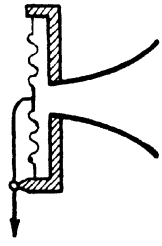


Fig. 209.
Mechanical pick-up.

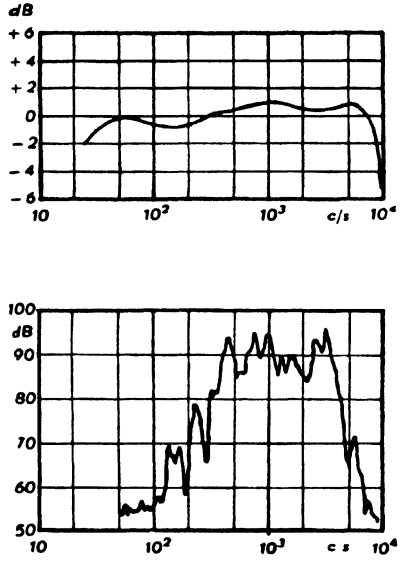


Fig. 210. Frequency response of mechanical reproduction.

b) *Electrical Reproduction with Pick-up*

Either a magnetic, electrodynamic or piezoelectric system is used in electrical reproduction. The magnetic system is built like the corresponding cutting head; a small core, surrounded by a coil, vibrates in the gap between two or four magnet poles. The system has a low cut-off frequency at which the mass of the pick-up swings against the armature suspension (vibration resonance). At the upper cut-off frequency the needle or its holder is in resonance with the core mass. Owing to the needle being tightly held in the groove, this resonance does not affect the response curve, but only reduces the mechanical impedance. In a very light magnetic system (Siemens ST8) the lower vibration is about 20 c/sec., the upper cut-off at about 10 kc/sec.[9]. On older pick-ups the lower resonance is higher in pitch and can exert considerable stresses on discs.

Electrodynamic pick-ups are less sensitive than magnetic but show less distortion and core resiliency, which spares the disc[10]. The Orthophone pick-up has a smooth frequency response up to 10 kc/sec, rising beyond to 14 kc/s, the effective needle-mass being 4 to 5 mgm.

Crystal pick-ups[11] give a potential proportional to amplitude and

so, in relation to magnetic and electrodynamic types, fall off at high frequencies. They have a capacity of 100 to 1000 pF and, for protection against humidity, are embedded in grease. The crystal pick-up has the advantage that it can be coupled directly without transformer or pre-amplifier to the usual radio apparatus. Its response is such that it improves to a certain extent the frequency characteristic in the cutting of a disc (*cf.* Fig. 199). So it is much in evidence although the electrodynamic system, at anyrate at high power, is more suited to high quality reproduction. The pick-ups are usually interchangeable for normal and micro grooves, but with a special sapphire which can be adjusted to the disc by tilting.

Ceramic pick-ups[12] have the property of being unaffected by temperature—whereas Rochelle salt melts at 56° C— but they are, on the other hand, insensitive and have not so good a frequency response.

Electronic pick-ups[13] are made of tiny thermionic valves in which one electrode vibrates. The internal resistance of the valve changes in sympathy with the sound amplitude. The characteristics of some pick-ups are shown in Fig. 211.

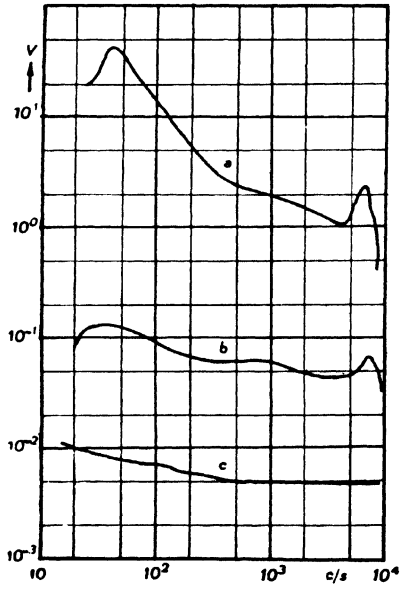


Fig. 211. Characteristics of different pick-ups at constant velocity: (a) crystal C = 3300 pF: R = 0.5 MΩ; (b) electromagnetic R_i = 200 Ω; (c) electrodynamic R_i = 1.5 Ω.

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c) *Needle*

Of special importance is the part of the pick-up which comes in contact with the disc[14]. The normal needle has a radius of curvature of 50μ, the needle for micro grooves 25μ. With these small surfaces, pressures of more than 1000 atmos. can act on the grooves, especially at the vibration resonance (pressures up to 22,000 atmos.). If the needle is very hard (diamond) it does not wear, but the disc does. Therefore softer needles (replaceable steel needles) are used and the disc material contains a grinding medium so that the needle after one revolution will have completely followed the profile. Permanent needles (sapphire) can only be used if the pressure of the pick-up on the disc is low[15]. An older type of magnetic pick-up, for example, exerted a load of 200 g, one with built-in sapphire needle only 20 g. 6 g is about the lower limit to prevent the needle jumping out of the groove. The restoring force on a good pick-up is less than 10 g wgt. for a deflection of 60μ.

Thin needles are resilient and therefore soft. Wooden and plastic needles preserve the discs but do not reproduce the higher frequencies so well.

There are also pick-ups which can be used indifferently for hill-and-dale or transverse grooves [15a].

d) *Drive*

In mechanical gramophones the drive is by clockwork motor, controlled by a centrifugal governor. In electrical reproducers electric motors are used. Synchronous motors are liable to hunt. D.C.–A.C. motors have centrifugal governors operating mechanical brakes, speed being reduced by a worm gear. Brush motors should be electrically suppressed. Commonly A.C.-driven asynchronous motors are used in which the phase is varied by a condenser or braking copper short-circuit coil. As they free wheel they change revs. with frequency, but are nearly independent of voltage. The drive takes place by switchable friction wheels which grip the turntable and turn it at 78, 45 or $33\frac{1}{3}$ revs/min. Motors should be sufficiently sprung that any vibration is not taken up by the pick-up[16] and should have sufficient torque (300–400 g cm) that the frictional loading by the needle cannot vary the speed by more than 0.7 revs/min. The vertical and lateral varia-

tions in the truth of the grooves should be less than 0.5 mm[17]. The pick-up must be so mounted with reference to the turntable that in all positions it lies nearly tangential to the groove, otherwise harmonic distortion up to 5% may ensue[18].

Automatic disc-changers can deal with up to 10 discs with possibility of stopping and re-starting. The radio-broadcasting has synchronising double-disc players to make it possible to follow one disc with another without intermission.

e) *Distortion and its Measurement*[19]

(1) Linear distortion of a cutter or pick-up is measured by the method of BUCHMANN and MEYER[20]. If parallel light falls on a disc and one looks at a given angle, one sees a band of light of breadth b, from which one can estimate the mean particle velocity from

$$\bar{\dot{\xi}} = \frac{b\pi n}{60}$$

where n is the rev/min, *e.g.* 78.

The groove is to a certain extent resilient to high frequencies, resulting in some loss in reproduction[21].

2. Non-linear distortion can occur with magnetic super-saturation of the cutting apparatus. It also occurs if the radius of curvature of the recording curve on the disc is less than that of the stylus. This happens specially at low playing-speeds (near the centre of the disc) and with large amplitudes. So on the radio, for example, the record is played from the centre outwards for then the needle still has a fresh point at the centre.

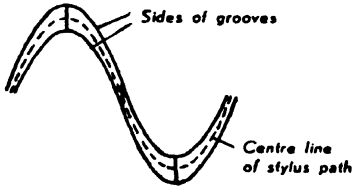


Fig. 212. Pinch effect.

Non-linear distortion is measured in the usual way for harmonic-distortion factor, difference-tone factor or modulation factor[22] (*cf.* Chap. 14).

Such distortion can also occur owing to the pinch effect (*cf.* Fig. 212). If for example at high frequencies near the centre of the disc with a worn needle the radius of curvature of the disc is greater than that of the grooves, the needle is pushed up at places. Also a false positioning of the tone

arm can result in the grooves being obliquely scanned, which causes distortion.

f) *Background Noise*

The signal-volume range of disc-recording is limited by the greatest permissible deflection and the background noise.

At low frequencies there are vibrations due to the motor[23]. Even if these lie below the threshold of the ear, they can modulate higher audible tones[27]. Going to higher frequencies, disc noise comes in which is to be ascribed to filings from the cut medium being dragged along (Fig. 213). The material has an average unevenness of $3 \cdot 10^6$ cm. The

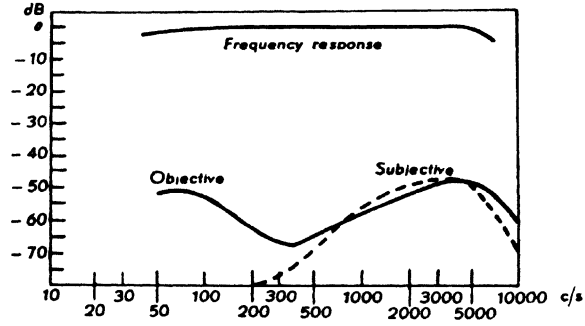


Fig. 213. Useful and idle potential in reproduction from disc.

scratch noise can be eliminated by low-pass filters, but this reduces the quality of the reproduction. As this needle noise is worse in the pianissimi one tries with a variable-mu valve to shunt the noise filter if an audible signal amplitude ensues. Then the noise is not disturbing, as it is acoustically masked by the high-frequency signal[24].

The sound volume ratio can be much greater if discs without cutting material[25] are used, but these can only be used with an electric pick-up.

The commercial exploitation of the disc[26] can be gauged if it is noted that 300 million were bought in U.S.A. in 1947. The technical preparation of the disc represents only part of the price. In large musical works the fees of the artists are costliest. A consequence of the mechanical method is that one cannot introduce 'cuts' because the join always gives a jerk when played. The easy reproducibility and serviceability are advantages which enable discs to be used for record libraries—and distribution among a large populace. Research on stereophonic reproduction has also been made[27] (See Chap. 16).

g) *Sound Foil Apparatus* [28, 29]

If a disc is not to be duplicated, as *e.g.* in isolated radio-reporting, one uses sound foils. These are plastic materials (Dekelith or Lacquer) which are laid on a carrier, *e.g.* a metal plate. Cutting is done with steel, as the foil is not so soft as heated wax and a sapphire would be broken by irregularities in the foil. With some plastics, subsequent thermal hardening is possible. The replaceable steel needle means a cutter of somewhat greater mass so that the response curve is cut off with a resonance peak at 5000 c/sec. The power is somewhat greater than with the black disc, as there are no filings. Before the introduction of the magnetophone 80,000 foils were prepared daily in Germany for radio reports.

h) *The Tefiphone*

In the tefiphone the recording medium is not a disc but originally a normal 35 mm gelatine film, in which more than 100 grooves have been cut side by side. The film can be joined into an endless band, so that it can be played continuously from either side. Playing times up to 24 hours can thus be attained. It is now possible to bridge over the intersection without an audible disturbance, so that a multiplication of the records is possible (Fig. 214). The films are boxed in bookform[30].

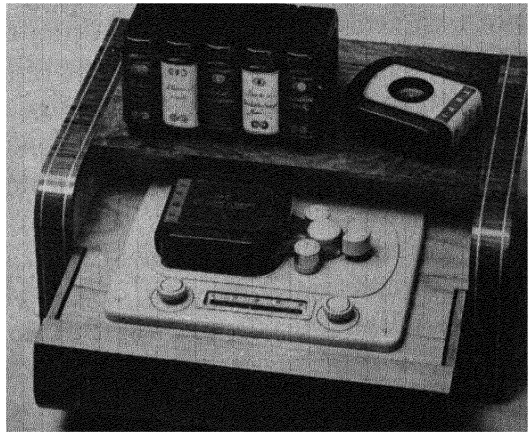


Fig. 214. Tefiphone.

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§ 3. Optical Sound Reproduction [31]

(i) *Historical*

In 1857 L. SCOTT recorded sound vibrations on a soot-covered drum. Reproduction was not possible. DUDDELL in 1900 used his oscillograph to record fluctuations in brightness of the singing arc and in 1901 RUHMER invented the photographone. An arc lamp modulated in brightness by the current from a microphone gave a stepped trace which was explored with a selenium cell. In 1902 DUDDELL produced a toothed trace.

The output in loudness and quality which these equipments possessed was brought to a certain stage by thermionic amplifiers after the first World War.

In 1922 the invention firm of Triergon (VOGT, ENGEL, MASOLLE) made a sound film in Berlin. The kathodophone was used as microphone, and a glow discharge tube as variable light source; reproduction was by photoelectric cell and statophone (*cf.* p. 332) [32]. After the first use of the sound film with and without amplifier in the disc apparatus (needle sound film), in 1928 the purely optical apparatus emerged. After that it was applied on a very large scale by many firms (Klangfilm, Western Electric, RCA, etc.).

(ii) *Principal Equipments*

The sound carrier is in all optical equipments a beam in which fluctuations in brightness take place rhythmically with the sound. The fluctuation can be photographed on a film, developed and fixed or produced by a chopping mechanism which produces an opaque layer on the film partly cut off (Philips-Miller system). The reproduction follows by light reflected or transmitted through the film which, falling through an optical system on a photoelectric cell, produces a potential which is amplified and applied to a loudspeaker. The talkie film is the principal application of the optical arrangement. Standard film is 35 mm, microfilm 16 mm wide, on which a track 2.54 or 2.23 mm wide, respectively, is inscribed.

(iii) *Types of Film*

There are two possibilities, (a) variable density or (b) variable area of strip (Fig. 215a).

The variable density is produced by a change of blackening over

the whole track breadth, so that the transmission varies in the direction of the

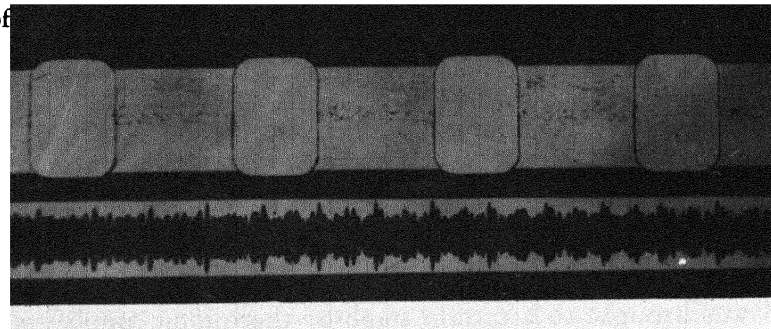


Fig. 215a.

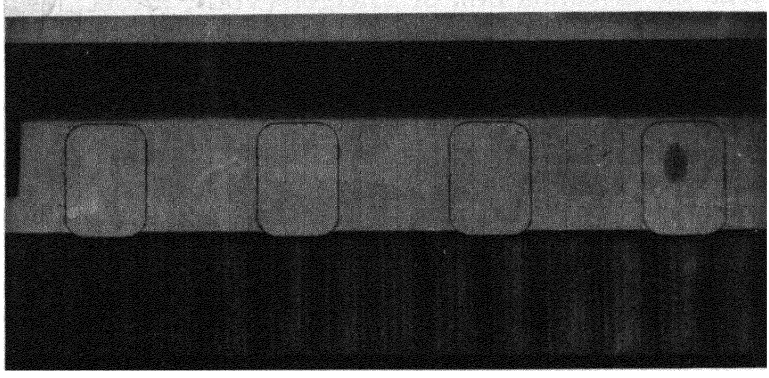


Fig. 215b. Specimens of film with (a) variable area (b) variable density sound track.

track-width is black but this all over equally (Fig. 216). In the former it is necessary in preparing copies to secure exact proportionality between light and blackening, but this is not so necessary in the latter wherein the requirement is only that the track-width is properly controlled.

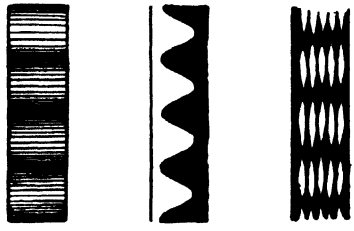


Fig. 216. Various light-tracks; (a) variable density, (b) variable area—single track, (c) variable area—multiple track.

Two types of variable density track are possible, push-pull A and B (Fig. 217) and corded track (Fig. 218). The variable area can be single, multiple and as push-

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pull (A and B) or corded (Figs. 216, 217). The object and properties of these tracks will be discussed below. Different light-control apparatus is needed for the two main types, but the same reproducing gear can be used for both.

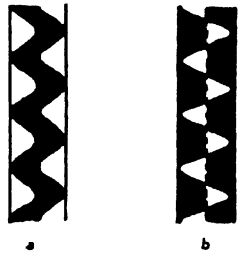


Fig. 217. Push-pull tracks (a) A type (b) B type.



Fig. 218. Corded track.

(iv) *Recording*

The electric energy coming from the pick-up microphone controls a light beam through a special instrument, which illuminates the film in synchronism with the sound. So, according to the method, one or other of these typical tracks is made.

a) *Light Control Apparatus*

Various light controlling devices have been employed in the past. Either the light source itself can have its brightness modulated with the sound or a constant source can be chopped to give a variable density or deflected to give a variable area track.

α. *Active Light Modulation*

The discharge tube can take various forms for sound recording *e.g.* ultrafrequency lamp or Aeolight or 'light-spray' with oxycathode, as well as the wolfram arc which had already been used by RUHMER. Finally the Braun tube has also been used; but all are weak, inconstant, require a supplementary potential and have a short life, so that in practice only the electromechanic light-control methods have prevailed.

β. *Passive Light Modulation*

Passive light-control uses a constant source. A lamp driven by A.C. must have such a heat capacity that it does not cause hum. Hum can

also be caused by gas modulation. In the path of the light the control is fixed; a Kerr cell, vibrating shutter or mirror. An electrically heated stylus polishes the groove [4a].

Kerr Cell

The effect discovered by KERR in 1875 showed that many substances, *e.g.* nitrobenzene, become doubly refracting under the influence of an electric field. The path difference for a beam which traverses the length l of an electric field is $\partial = BlE^2$, B being a material constant. For linearity, an A.C. E must be polarised by a large D.C. field. KAROLUS

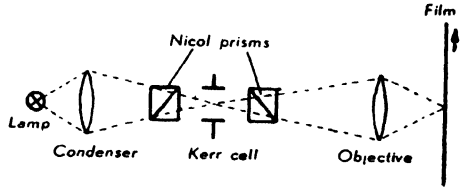


Fig. 219. Kerr cell scheme.

first used the Kerr effect in optical recording using a cell formed of a condenser with 100μ electrode separation, 4 cm long and 1 to 5 mm slit length, filled with nitrobenzene. About 500 volts D.C. was applied to the condenser plates. It has been found a disadvantage that in use the nitrobenzene becomes impure, so a flushing cell is built in which the liquid can be renewed. The light characteristic, being not quite straight, requires compensation. The Kerr cell is only seldom used; it produces variable density track.

γ. *Controls with Moving Blade*

Light Valve

The Western Electric Co. has a system in which a metal band carrying an electric current in a magnetic field vibrates on the electrodynamic principle and covers a light slit in so doing (Fig. 220). In practice two bands vibrate out of phase one on each side of the aperture, which when fully open is 25μ wide. The bands are tuned to 9500 c/sec. Frequency compensation is assured with electric filters. The light-chopper uses one hundredth of a watt

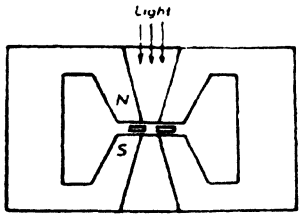


Fig. 220. Light valve.

with a resistance of 12 ohms and gives a high illumination. It is however liable to disturbance and requires a nice adjustment in practice.

String Oscillograph

This is made by the Selenophone Co. in Vienna. A vibrating string, actuated by electrodynamic drive, cuts off more or less the optical image of a slit. The covering can take place in the parallel or perpendicular position to the slit and so produces one or other of the typical tracks. The 2 cm long wire has an undamped natural frequency of 14,000 c/sec. The resistance is 0.5 ohm and the maximum current 1 amp.

8. *Light-Control with Rotating Mirror*

RCA Galvanometer

In a magnetic symmetrical so-called quadripole system a reed moves and turns a mirror, which is held on a stretched strip of phosphor bronze (Fig. 221). As the centre of rotation of the mirror lies only

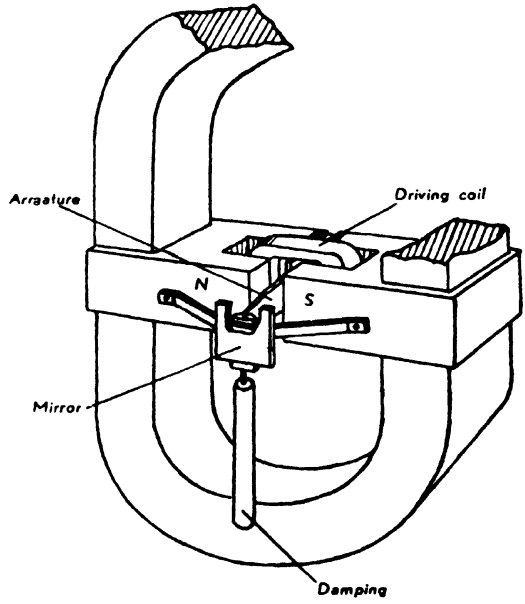


Fig. 221. RCA galvanometer.

0.5 mm from the sloping edge of the magnetic core, a mechanical transformer ratio of 10 to 1 is possible. The natural frequency of the vibrating system, damped by plastic, is about 10 kc/sec.

Light Tape (Lichthahn)

The Klangfilm Co. has devised two types, a high-fidelity one working on the electrodynamic principle and a more robust one on the magnetic principle.

In the electrodynamic Lichthahn (Fig. 222), as in a vibration galvanometer, a mirror is suspended between two current-carrying wires in a magnetic field. The resonant frequency is damped by oil and lies about 10 kc/sec. To compensate for the variation with temperature of the viscosity of the oil, a damping vane is attached to a bi-metal spring, which reduces the damping oil-aperture at higher temperatures.

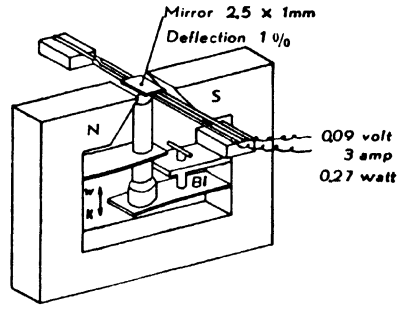


Fig. 222. Dynamic Lichthahn.

In the magnetic Lichthahn (Fig. 223) a mirror is fixed to an iron armature which vibrates on stretched steel wires between the poles of a magnet. The magnetic Lichthahn is tuned to about 9 kc/sec, is insensitive to overloading and is applied to transportable apparatus giving a multiple variable area track.

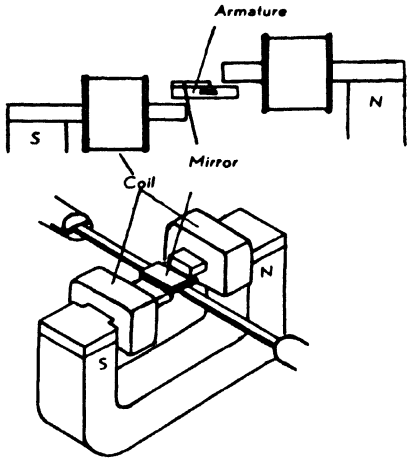


Fig. 223. Electromagnetic Lichthahn.

b) *Tone Optics*

In order to produce a 'light-tone' with any of the control apparatus described above or to pass it over a photoelectric cell, one needs an optical system giving a slit image. It consists of a light source, condenser, slit and objective. For source one uses a gas-filled low voltage lamp with thick wolfram filament. Its image can be formed in the

slit Fig. 224a or in the objective (Fig. 224b). Nicol prisms are placed before and behind the Kerr cell. The source forms an image in the entry pupil of the objective. In the Lichthahn and in the RCA galvanometer the image is a slit, which, according to the type of variable area track, has one or more teeth, moving over the slit under the action of the rotating mirror, and by the intervention of the objective, the resulting image is brought onto the film.

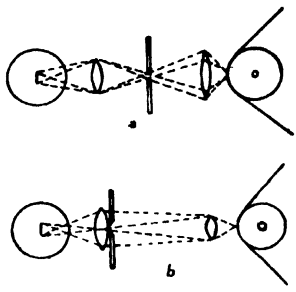


Fig. 224. Basic types of sound optical systems; (a) image in slit-plane (b) image in objective.

In the light valve the light source in the slit builds an image on the film. In the recording, by-paths for the light are arranged generally so that the record can be controlled by direct observation and monitored through the optical path.

c) *Technical Aspects of Film Recording*

Acoustic Recording

Sound recording cameras are made by various firms. They contain the stock of spools and the run-off spool for the film, an electric driving motor and a feed regulated by flywheel and damping, the light source, optical system and light chopper, with noiseless film shutter. There is also a monitoring spy hole as well as a headphone for listening to the optical transduction. The film and optical system must of course be securely boxed against light-entry.

The light control (section 1), which is different according to the type of record, is connected over a mixer or controller tothe recording microphone, which is usually remote from the other recording apparatus. The recording should be done in an acoustically adjusted studio. The sound engineer follows the action on a mixing console with a distortion meter and a control loudspeaker and regulates the power to the limiting value of about 40 db. Also on the console there are so-called quality controls, with which a certain desired frequency range can be set up. For the first recording at present the magnetophone principle is generally used (*cf.* § 3) while the copies are made optically.

Development and Copying

The exposed film in the sound camera must now be developed and

fixed. This usually is done in special copying rooms where automatic temperature and concentration regulation of the developer secures equal overall development of the film. The γ conditions (*cf.* p. 407) must be fulfilled.

To prepare a talkie film one must collect various tracks (music, noise and speech) in a mixer to play over on a single track. The monitor has a number of drums from which the sounds are picked up by photoelectric cells and, in a given proportion, are applied to a new sound camera. The necessary reverberation is then produced artificially in an auditorium and added, since the original is recorded as far as possible without reverberation. The 'cutting' (deletion of certain portions) is done beforehand, and the rejoining must be made at an angle in order that the change of light intensity over the join should be gradual and not sudden. This can also be done by opening an aperture which gives an indistinct sinusoidal transversal blackening of the film. The prepared film, many specimens of which are to be hired out to the talkie cinemas, is now copied many times. For this process special machines are built. The picture must pass in jerks through the gate and be projected while still, but the sound, which is staggered 20 frames from the vision, must pass continuously through the copying gate. For this a continuous motion combined with an interrupted motion is provided by a suitable drive. The shrinking, which the film undergoes in the photographic processing, is also taken out in the copying machine.

(v) *Reproduction of Optical Sound Film*

Reproduction takes place by the film being illuminated by transmitted or reflected light, the light falling on a photoelectric cell and, after amplification, the potential being applied to a loudspeaker. An image of the slit must be cast either on the film or on the light source. As pick-up takes place directly on the film the photoelectric cell must either lie inside the moving spool or the light beam must be turned round the axis. This can be done, for example, with a glass rod in which the light suffers total reflection and is led out (Fig. 225).

a) *Photoelectric Cells*

Of the different types of light-sensitive cell, resistance cells of selenium and copper oxide are, owing to their insensitivity to high frequencies, only used in recording (the latter type) and not for reproduction.

Alkali cells require a potential of about 100 volts and can be filled with a rare gas or be evacuated. The vacuum cells have a wide frequency response but are less sensitive, so that today gas-filled cells are generally used. The h.f. potential drop through the cable can be made up in the amplifier. The photoelectric cell is joined to the first amplifier valve, like a condenser microphone, but the grid leak resistance should not be so large since the photoelectric cell acting as generator has a

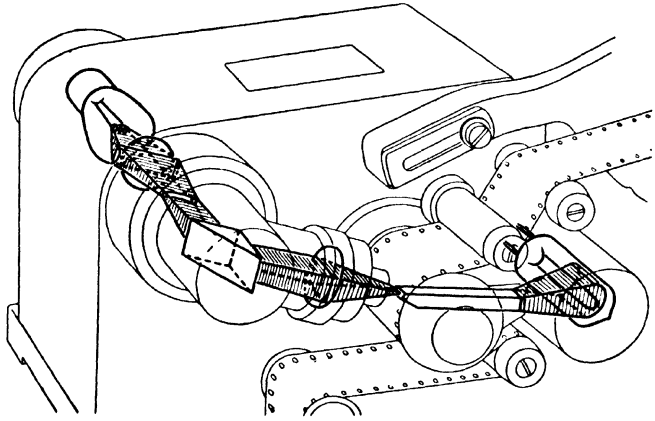


Fig. 225. Optical arrangement for sound film ('Europa'), showing path of light from source through slit onto photoelectric cell.

lower resistance than the condenser microphone. The connecting cable to the amplifier should not be too long and should be capacitance-free and not show any microphonic effect.

b) *Acoustic Reproduction*

A multi-stage amplifier passes the amplified photoelectric current to the loudspeaker. In cinemas the loudspeaker combinations usually stand behind the sound-transparent projection screen. The reverberation in the reproduction hall must be small because some such will already have been added in the optical recording. In talkie films, the quality of the apparatus is more important than small differences in price, consequently quality designs with great reliability have been developed.

c) *Stereophonic Reproduction*

A great improvement in sound recording can be attained with stereophonic reproduction (*cf.* Chap. XVI). According to FLETCHER a stereophonic reproduction up to 6 kc/sec is better than a plain one up to 12 kc/sec. The reproduction sounds more natural and, as with vision, gives a verisimilitude and localisation otherwise lacking, which raises the artistic effect.

Two or more channels are required and the effect depends on intensity[33], speed[34] and timbre differences[35]. Many microphones must be employed, many tracks inscribed on the film and the result relayed over many loudspeakers. Despite research in U.S.A.[36], Holland and Germany, the difficulties are so great that the stereophonic film has not been much exploited in practice.

(vi) *Distortion in Optical Recording*

a) *Constancy of Speed*

The usual sources of distortion in electroacoustic apparatus like amplifier hum, thermionic noise, non-linear amplification will be set aside. In the optical process constancy of running speed is as important as in all recording. The film can deviate in three dimensions, while it is led, especially in the reproduction, unevenly through the Maltese cross action. There must therefore be a mechanical filter containing springs, masses and dashpots to compensate for all deviations that arise in the sound film for example, from the perforations exciting an unwanted frequency of 96 c/sec.

b) *Hum*

Hum can be caused optically if the A.C. supply to the lamp is not sufficiently smoothed.

c) *Slit Effect*

Linear distortion is produced when the slit is so large in relation to

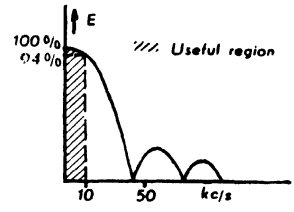


Fig. 226. Diffraction from slit.

the film speed that at the highest frequencies different light-intensity values appearing in the sine wave fall simultaneously on the slit. The sensitivity varies as the diffraction function $\sin \alpha/\alpha$ (Fig. 226). With a 10μ slit and a film running at 45 cm/sec this loss is only 6% at 10 kc/sec, but at the speed of miniature films (16 cm/sec) it is 44%.

d) *Adjustment Errors*

Distortion can also arise from faulty adjustment; when the slit is out of place the picture is askew.

If the recording and pick-up slits of the variable density system are not exactly parallel there is a loss at high frequencies. Errors of fractions of a degree are noticeable.

In the variable area system skew slits produce non-linearity. With multiple tracks (Fig. 216c) this distortion is inversely proportional to the number of teeth.

e) *Goldberg Rule*

Non-linear distortion can arise when in the photographic development the blackening (S) is not proportional to the illumination(E) (Fig. 227). As the film is developed first as negative and then as positive the incorrect development can be compensated. According to the GOLDBERG rule, the gamma product $\gamma_{pos.} \times \gamma_{neg.}$ is unity if γ is the slope $\partial S/\partial E$.

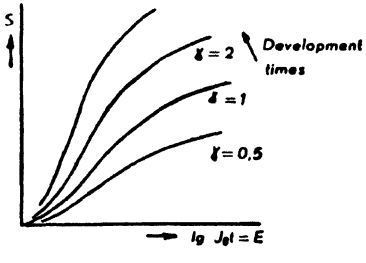


Fig. 227. Photographic blackening S in relation to illumination E.

f) *Thunder Effect*

On the variable area track fogging of the curves can occur at high frequencies where the individual teeth lie close together, due to diffused light, a phenomenon called 'thunder effect'. It is especially heard with hissing speech, but may be obviated somewhat by using blue light. 4% of this effect is permis-

sible. Partly in consequence of this effect the frequency response is cut off at 7 kc/sec.

g) *Noiseless Reproduction*

The granularity in the transparent part of the film produces a noise in the reproduction of the variable area film, which has been corrected by a blackening of this part not enough to blur the frequency variations on the developed portion. This can be attained if a slit with special teeth is allowed to move in front of the film so that in fortissimi a great part and in pianissimi a smaller part of the film is available for pick-up (Fig. 228). So the sound is properly reproduced while the false light is cut off.

By the movements of the walls of the slit a new frequency is recorded, *i.e.* that corresponding to the rate of change from *ff* to *pp* and back,

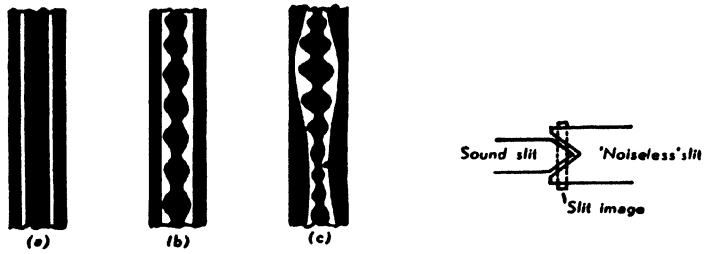


Fig. 228. Mode of operation of noiseless equipment.

but it is so low that the amplifier cuts it off. In the light valve system this action is obtained by a dynamically driven toothed slit (Fig. 229); in the RCA galvanometer by two iron reeds vibrating in opposite phase.

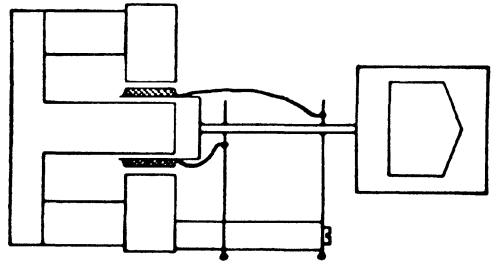


Fig. 229. Slit of noiseless equipment.

(vii) *Other Equipment*

a) *Miniature Film*

Besides the normal film of 35 mm with 2.54 mm track running at 45.6 cm/sec there is one of 16 mm with 2.23 mm track at 18.3 cm/sec. The latter is fireproof and so fulfils police regulations in most countries. There are no new optical problems involved in its use and it uses less material, which brings it in great demand by amateurs and expeditions. Because of its slower speed it shows a greater loss at low frequencies than normal films.

b) *Philips-Miller System*[37]

In this a film strip is cut with a cutting stylus out of a black layer over which it runs. In this way a double toothed track is obtained and no developing or fixing is required. Difficulties arise from the thunder effect and lack of uniformity in the film material, which for different thicknesses of the strip (known as 'Philimil strips') results in variations of the cutting breadth. Pick-up is by the same method as for other sound films.

The stylus has an angle of 87° which gives a magnification of 40, *i.e.* to cut out a sound track of 2 mm requires only an up and down motion of the stylus of 50μ.

c) *Optical Recording with Reflex Pick-up*

It is possible to reproduce from sound strips by reflex light, but this device has not been used in practice. One application is the optical name-announcer which on the German Railways is used on the public address system to announce the names of places. A transparent film runs over a rotating mirror wheel and is illuminated. The reflected light passes through an optical system on to a photoelectric cell.

§ 4. Magnetic Tape Recording[38]

(i) *Historical*

In 1898 POULSON magnetised a steel wire—actually a piano wire 0.5 to 1 mm diam.—with alternating magnet fields of audio frequencies and, running the wire alongside a coil, reproduced, with the help of the

remanent magnetism, the alternation as sound in a telephone. This arrangement called Telegraphone was used to record distant speech, but the quality was not so good that it was much used. This magnetophone was more widely in demand after the introduction of the electronic amplifier, as with the other types of reproduction.

Instead of steel, wire tape was used, *e.g.* for the time announcement over the telephone system. From about 1936 a German patent (D.R.P. 500,900) of PFLEUMER called Magnetophone has been used with iron oxide powder on film strip as sound carrier. This development was especially due to the A.E.G. firm in Germany, with the IG-Dye Co. of Ludwigshafen making the tapes. In 1941 VON BRAUNMÜHL and WEBER improved the magnetophone so that the wiping-out and picking-up were done by h.f. alternating currents[39]. A dynamic amplification up to 60 db was obtained. After the Second World War magnetophones were built in many countries in many different forms, on the development and application of which more will be said below.

At the present day magnetic recording is certainly of the highest quality with respect to frequency response and sound-volume ratio and has the following advantages; possibility of cutting, of immediate reproduction and of cancellation; best quality in relation to frequency band, power and non-linear distortion. Also tape is less liable to damage by vibration and heat than disc apparatus. The weakest link is still the magnetic carrier whose quality is variable. Disadvantages are the necessity for rewinding and the mechanical somewhat labile character of the tape rolls. The difficulty too of making a great number of high-fidelity copies can also be considered unfortunate.

(ii) *Physical Basis*

The principle of magnetic recording involves a sound carrier (Fig. 230) of magnetic material which is moved past an electromagnet (sound head) with a certain velocity and there magnetised by the amplified microphone current passing through the sound head. The carrier retains the signal as remanent magnetism until it is again passed with the same velocity in front of another sound head. In the coil of the latter a potential is induced, amplified and relayed from a loudspeaker

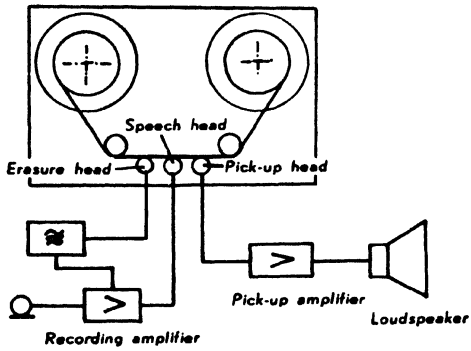


Fig. 230. Scheme of magnetophone equipment.

in its original sound form. The remanent magnetism can be wiped off after the reproduction by applying a larger magnetic field, so that the same carrier can be used often.

The first magnetophone had the carrier magnetised transversely so that the lines of force crossed the wire perpendicularly, for a coil with an iron core on each side of the wire served to magnetise and pick-up the induced potential (*cf.* Fig. 231a). At the present time so-called longitudinal magnetisation is mostly used (Fig. 231b). The magnetising

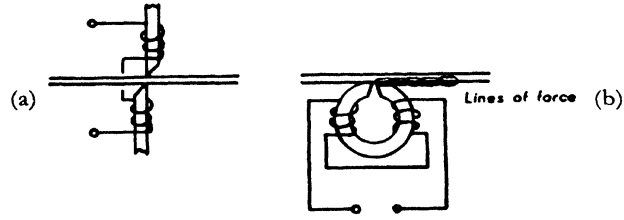


Fig. 231. Transverse and longitudinal magnetisation.

head (for recording, pick-up or erasure) is only on one side of the tape and the lines of force run from one pole to the other of the head, *i.e.* along the tape, only at the entry and exit points are they perpendicular to the surface, *i.e.* transverse to the tape. The

magnetic relationships in the constant and h.f. field magnetophones are set out below[40].

(iii) *Constant Field-Magnetophone*

The operation of recording is best understood by considering the B — H characteristic of the carriers

(a) Erasure of the text takes place by applying a constant field of value H_L to the tape, so that it reaches the saturation point D of the characteristic (Fig. 232). In this way the small magnetic fluctuations imposed on the tape are wiped out. If the tape then passes the head and the constant field is removed it is in the magnetic state represented by A.

(b) Recording follows, but in the state represented by A the B: H curve is not straight, so that large sound amplitudes would give rise to non-linear distortion. To obviate this a constant reverse field HG must be applied to bring the working point C on to the linear part of the magnetic characteristic (Fig. 233). If while the tape is run past the

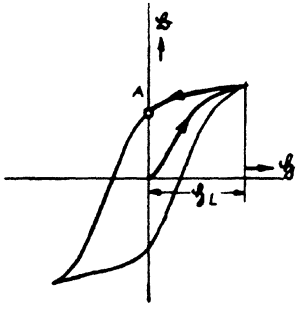


Fig. 232. Constant-field erasure.

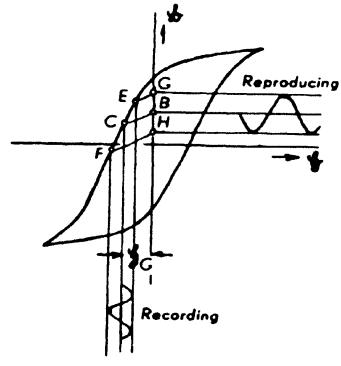


Fig. 233. Recording and reproduction with constant field.

head the magnetic field of the pick-up is removed, the points E C F move to G B H.

(c) Reproduction. In this state the tape runs past the pick-up head and induces an e.m.f. in the coil, which represents the sound.

(iv) *H.F. Magnetophone*

(a) In the erasure head the magnetic tape is subjected to a h.f. field which magnetises it in both directions of polarisation up to saturation. When the tape however has retreated from the erasure head the field becomes proportionately smaller as the leakage flux gets weaker (Fig. 234). The tape is thereby completely demagnetised and the wire-noise is reduced as the tape has no longer any magnetic potentials.

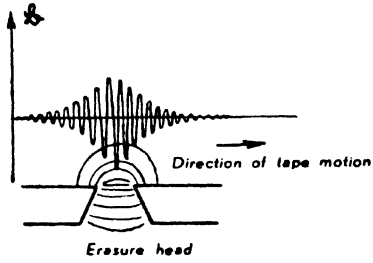


Fig. 234. High frequency erasure.

(b) Recording. The sound A.C. is added to a h.f. potential, so that there is no longer a branch of the magnetisation curve to act as working characteristic but a mean line between two branches with a relatively large region of linearity. Therefore the non-linear distortion of the h.f. magnetophone is less than that of the steady field magnetophone.

(c) Reproduction takes place without the intervention of the h.f. just as in the steady field type.

(v) *Heads*

For erasure, recording and listening the tape is carried in front of various heads (Fig. 230). They are made of ferro-magnetic material with an air-gap at the carrier and have coils, which are linked with the magnetic field of the tape. With longitudinal magnetisation the heads have a more or less iron-enclosed magnetic path, which at the tape is broken by a gap. The heads are ringshaped and carry two coils. They have accordingly, like a two-limbed transformer, a small stray-effect, as an external field separately excited envelops both coils, the induced potentials of which cancel. Besides there is an iron-shielding against stray flux.

a) *Erasure Heads*

The erasure head has the broadest gap (400μ) and is supplied with 5 watt h.f. (about 60 kc/sec) power.

b) *Recording Head*

The speech head has a gap of 40μ. Its gap can be wider than that of the pick-up head—as during the travel of the tape, the recording is mainly governed by the magnetic lines of force of the pole shoe edges. This head is made of mu-metal and to reduce the harmful remanence and non-linear distortion has a supplementary airgap.

c) *Reproducing Head*

In the listening head, however, the gap should be as small as possible in order that the carrier may be as finely scanned as possible at the highest frequencies, for, in this as in the sound film, there is a loss of sensitivity if the complete wave-length falls within the gap width. The width is commonly 20μ, which with the traversing speed of 77 cm/sec of the high-fidelity magnetophone is adequate for the highest audible frequencies. This head must be made of a material of maximum permeability in order to keep its magnetic resistance as low as possible and to pick up as much as possible of the flux from the tape.

(vi) *Sound Carriers*[40a]

a) *Tapes*

In order to store a maximum of magnetic energy in the tape, the coercivity must be large and the permeability small. The grain size of iron oxide is 0.1 to 0.5μ. The powder layer has a thickness of 20μ, the carrier itself 30μ. It has a breaking strength of 2 kg and is spooled in rolls of 1000 m for 22 mins playing at a speed of 77 cm/sec. It is non-inflammable. In the L-tape, the iron particles do not form a moulded layer but are distributed over the whole band. The width of the tape is 6.5 mm and the sound volume ratio diminishes with width.

At high frequencies there is a loss of sensitivity on account of the demagnetising field of the elementary magnets, formed by the single grains which are so close together at high frequencies that they partly cancel each other's fields. This loss differs for the different types of tape. The tapes can be clamped together by a solution or by adhesive tape. For sound film purposes perforated magnetic tapes of width 35 mm or split, perforated, magnetic tapes of 17.5 mm are used.

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b) *Wires*

For certain special purposes wires of 0.08 mm diameter are used. They can be rolled on a small spool of 7 cm diameter to give a playing time of about one hour. If the wires become torn, they can be knotted together.

c) *Plates*

The magnetic material can also be compressed into a plate on which—just as with sound discs—a magnetic head can scan the sound, impressed in spiral grooves (Dimaphone[40b]).

(vii) *Driving Gear*

In contradistinction to the sound disc but like the sound film, there are three driving axes in the magnetophone equipment, the supply spool, the wind-up spool and the capstan which determines the speed of the tape passing the scanning point.

There are magnetophones with 3 different motors, one for winding, which spools the tape on the drum, one to run off the tape and one to carry the tape at constant speed past the heads. On rewinding, which can occur at a higher speed, the first two motors reverse their functions. There are also arrangements which use a single drive geared by belting to the two winding drums which need different speeds according to the amount of tape in situ. Care must be taken with inertia masses and, if necessary, mechanical filters to make the running free from all disturbance. On the magnetic film recorders, *e.g.* Siemens Klangfilm, the running is so constant that frequency modulation is less than 0.1 %[40b].

On the Dimaphone, a dictation apparatus, a disc rotates with constant velocity. The wave-lengths get smaller towards the centre just as on gramophone discs. The quality is poor but good enough for dictation.

a) *Playing Speed*

High-fidelity magnetophones like those used by the broadcasting companies have a tape speed, V, of 77 cm/sec; for film purposes $V = 45.6$ cm/sec and for magnetophones of lower quality $V = 38$ or 19 cm/sec.

b) *Running Fluctuations*[27]

Fluctuations in the running speed depend greatly on the smoothness of the drive (centering of spools and wobbling of tape). According to DIN 45511 (German Standards), magnetophone apparatus is placed in four different qualities:

Class	Mean fluctuation over 1 min.	Short period fluctuation over 10 sec.
I	0.2 %	0.2 %
II	0.3 %	0.3 %
III	0.8 %	0.3 %
IV	2.0 %	0.5 %

(viii) *Frequency-Response and Amplifier*

In recording, in order to have a speech-potential at the microphone proportional to the flux Φ, one requires a constant current i in the pick-up head at all frequencies,

$$\psi = n\Phi = iL$$

As the speed head usually has an inductance L, the potential therein must rise with frequency; $U_s = i\omega L$. Therefore its amplifier must have a rising characteristic.

The listening head must have an amplifier of opposite characteristic, because its potential is

$$U_b = n\frac{\partial \Phi}{\partial t} = n\frac{\partial \Phi}{\partial x}\cdot\frac{\partial x}{\partial t} = n\frac{\partial \Phi}{\partial x}\cdot V.,$$

$V = \mathrm{d}x/\mathrm{d}t$ being the tape speed. But with sinusoidal potential the flux distribution over the tape is $\Phi = \Phi_0\, \mathrm{e}^{\,i\omega x}$ and so

$$\frac{\partial \Phi}{\partial x} = i\omega\Phi_0\, \mathrm{e}^{\,i\omega x} = i\omega\Phi$$

Therefore $U = ni\omega\Phi V$, *i.e.* the potential must be greater as the frequency goes up, which means that the amplifier requires a falling characteristic. In practice, one must compensate the action of the demagnetising field of the tape at the highest frequencies. Moving the band a short distance from the head can cause a decrease at high frequencies[40c].

The distortion over the whole gamut in good magnetophones is so reduced that the fluctuations from 10 to 15 kc/sec are less than $\pm$ 2 db.

In the h.f. magnetophone an h.f. oscillator must be built into the amplifier whose generated potentials overlay the speech potentials.

(ix) *Distortions and Disturbances*[41]

Besides the usual troubles which also afflict other copying apparatus, especially electroacoustic transducers, a special 'copy-effect' can arise whereby the places of high modulation on the tape are affected by small modulations from the magnetic field, giving the impression of reverberation. These phenomena are time—and temperature—dependent.

Noises and background noise can arise from mechanical and magnetic unevenesses, the latter also from unsymmetrical erasure and pre-magnetising of the tape. Also if the tape in running does not pass squarely over the head, modulation follows the resulting weakening of the magnetic flux.

(x) *Magnetophone Designs*

A very high-fidelity—but expensive—installation is marketed by the A.E.G., especially for broadcasting studios (Fig. 235). For stereo-



Fig. 235. Magnetophone Type T 8 N of the A.E.G.

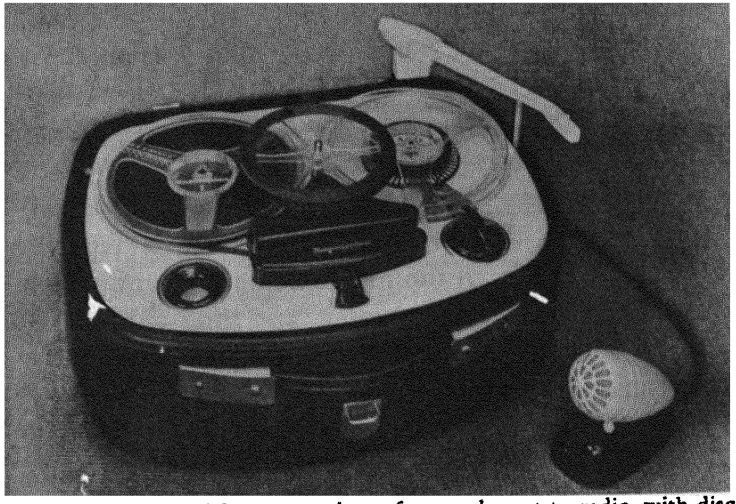


Fig. 236. Commercial magnetophone for attachment to radio, with disc pick-up (AEG).



Fig. 237. Commercial wire magnetophone (Siemens-Halske, Wien).

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phonic purposes one can split the ribbon and use two tracks passing over twin heads[42]. Various cheaper designs, some with lower tape speeds (38 or 19 cm/sec), by various firms are on the market. Wire magnetophones, some with disc-players are available, *e.g.* by Webster, Chicago, and Siemens-Halske (Fig. 237). On these the same head is often used for recording and reproduction. Erasing is mostly done at 50 kc/sec. For half-anhour's playing time the spool is about the same size as that of a type- writer ribbon.

The Dimaphone (*vide supra*) is used for dictation with a disc. The narrow track results in a small signal-volume ratio but this is compensated by electrical means.

(xi) *Applications of Magnetophones*

Besides broadcasting, magnetophones are used for a number of purposes, *e.g.* on the railway, for weather and time announcements and in the film industry[43] (Figs. 238, 239). They are not much used in the home because of price-competition with the much improved gramophone. Magnetophones with variable tape speed are used to decipher

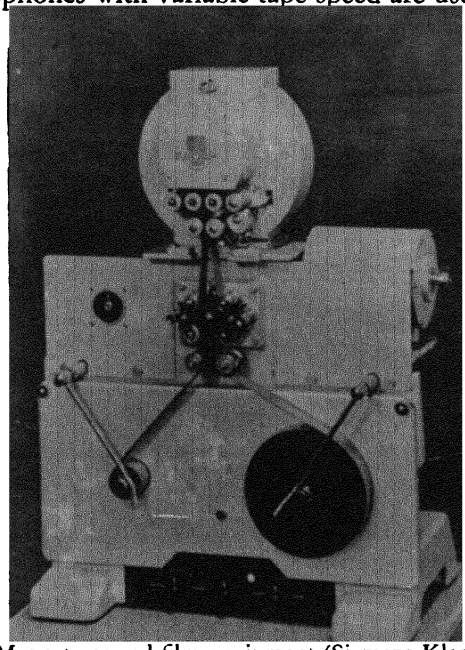


Fig. 238. Magneto-sound-film equipment (Siemens Klangfilm).

urgent telegrams and for coding and decoding signals; for recording results on calculating machines, for search-tone analysers, for repeating sound sequences on an endless tape and for microsecond measure-

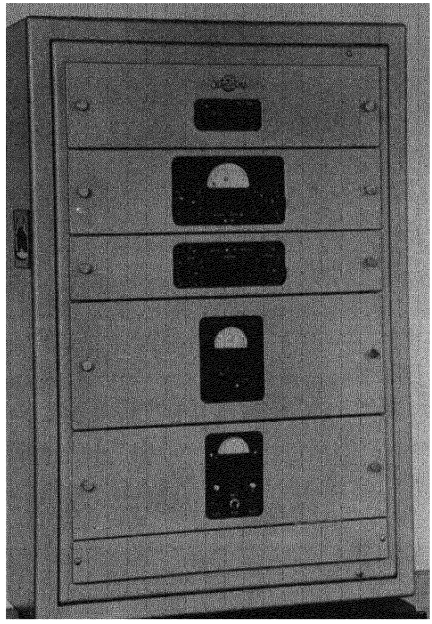


Fig. 239a. Amplifiers for magnetophone equipment, front view (Siemens Klangfilm).

ments in ballistics, this apparatus is very adaptable. It is also being used in conjunction with the adaptation of the Haas effect in the acoustics of buildings (*cf.* p. 135)[43a].

For film-cutting, a reproducing arrangement with time extension by rotating heads can be used[44].

(xii) *Copying*[45]

Various 'mother' and 'daughter' contact copies can be made, also direct copies. The daughter copy must be made of robust material. The virgin tape and the original are subjected to a magnetic field of 100 to 60 kc/sec and the copy is then a mirror image. That brings in a difficulty if the text lies to one side of the central line,

e.g. in stereophonic recording. In copying, the tape speed is 3 m/sec (Fig. 240).

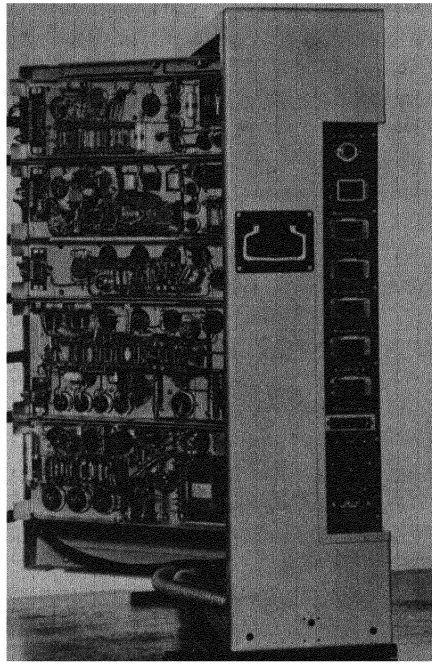


Fig. 239b. Magnetophone equipment (back partly opened).

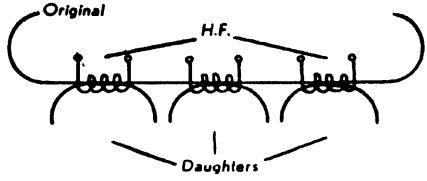


Fig. 240. Copying magnetic tape records.

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CHAPTER 16

STEREOPHONIC REPRODUCTION

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§ 1. Introduction

An essential part of the perception of sound is a space impression as to the origin of the sound. One of the original functions of the sense of hearing is that of a warning against danger, and it is important not only to know that there is danger, but also from what direction it comes and to some extent also how close it is. But not only in the case of danger the directional hearing is important. Without directional hearing a discussion between several persons would be practically impossible. When listening to a concert it is possible to identify the separate instruments or groups of instruments, not only by their timbre, but also by the direction from which the sound comes. It is also very remarkable that it is possible to concentrate our attention on the sound from a certain direction, and in this way to put into the background of our consciousness the sounds from other directions, which are considered as undesired at that moment.

§ 2. Binaural Hearing

For localizing a source of sound with respect to an observer 3 coordinates are needed: the distance, the direction in the horizontal plane and the angle to this plane (*cf.* Chap. 11, § 19).

(i) *The Distance*

A possible explanation for the observation of the distance of a sound source is given by VON BÉKÉSY [1]. The result of analysing the sound field of a pulsating sphere is that the sound pressure at different distances varies as $\frac{1}{r}$ (r = distance to the sound source) but it is

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independent of the frequency, whereas the velocity of the sound depends not only on distance, but also on frequency. The velocity of the different harmonics in a non-sinusoïdal sound varies therefore with the distance. According to some theories of hearing [2], the ear also responds to the velocity of sound and this means that the timbre of the observed sound changes with the distance.

It is experimentally confirmed that only distances smaller than 150 cm can be accurately determined by the ear.

In closed spaces the effect of distance *variation* can be observed because then there is a change in intensity and in the ratio of direct sound to reverberation. When the intensity, or (and) this ratio increases, the ear has the impression that the sound comes nearer.

It is clear from the above that when using a velocity microphone in electrical-acoustical sound transmission the impression of distance may be preserved; this will be imposssible with a pressure type of microphone.

The most important feature in localizing the sound source is the perception of direction in the horizontal plane. This depends on the collaboration of both ears. If the source of sound is situated in the perpendicular plane bisecting the axis of the ears, the sound waves reach both ears at the same time and with the same intensity.

When, however, the sound comes from a direction α, it first reaches one ear, and only after a certain length of time with another intensity (bending of the sound waves around the head) the other ear. (Fig. 241).

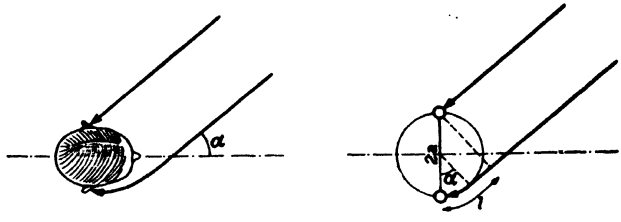


Fig. 241. Sound coming from a direction which makes an angle α with the vertical bisecting plane of the head reaches the two ears with a time difference of $t = \frac{-}{c}$ (c is velocity of sound). The wave which bends around the head is also weakened.

Our sense of hearing is extraordinarily sensitive to those differences in time and intensity, which it interprets as angular deviations from the bisecting plane.

(ii) *Time Difference*

The time difference can easily be calculated from the difference in distance and the velocity of sound (Fig. 241 and 242).

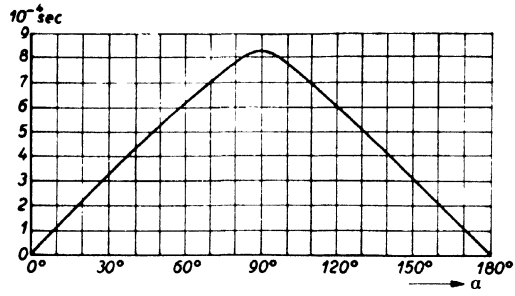


Fig. 242. Time difference t as a function of the angle α.

For a sphere with radius a this time difference Δt is

$$\Delta t = \frac{a(\alpha + \sin \alpha)}{c}$$

(c = velocity of sound). For a head of normal size the maximum value of Δt is $8 \cdot 10^{-4}$sec. It has been found experimentally that an angular deviation of 3° (corresponding to $\Delta t = 3 \cdot 10^{-5}$ sec) can be observed.

(iii) *Intensity Difference*

The intensity difference is not so easy to calculate. For a pure tone it depends on the frequency and radius of the sphere [3], increasing with both.

SIVIAN and WHITE [4] have measured the intensity difference on both ears for S.H.M. sound waves, bending around a human head. For a composite sound wave (speech) the calculated differences are given in Fig. 243.

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It may be seen that for angles (α) between 0° and 50° the intensity difference in decibels increases practically linearly with the angle, with a maximum value of about 7 db.

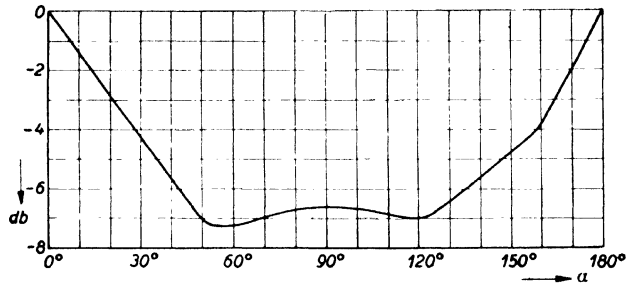


Fig. 243. Average intensity difference (in db) at the right and left ear as a function of the angle α upon hearing speech.

(iv) *Combination of Time and Intensity Difference*

Can any combination of time and intensity differences result in directional hearing? An answer to this question can be given by listening to two loudspeakers, placed in a horizontal line, reproducing

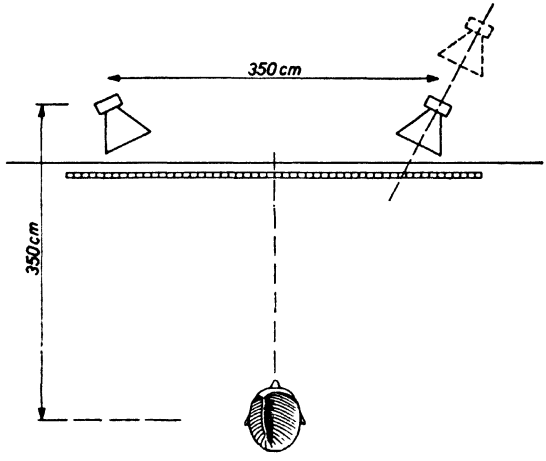


Fig. 244. Arrangement of the loudspeakers in the experiments concerning the cooperation of time and intensity differences.

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the same sound source, and by varying the intensity of both speakers, or the distance to the listener (Fig. 244).

If the sound of both loudspeakers reaches the ears without time difference and with the same intensity, it is clear that a listener observes the 'sound image' in the middle of the line connecting the two loudspeakers. An increment of the intensity or a time difference in favour of one of the loudspeakers shifts the sound image from the centre towards this loudspeaker.

In Fig. 245 the angular deviation of the sound is given as a function of time difference only. The dotted line in this graph shows the angular deviation by natural hearing, when the time differences are combined with intensity differences. The conclusion is that in natural hearing the intensity effect is much more important (9 to 10 times) than the time effect [5].

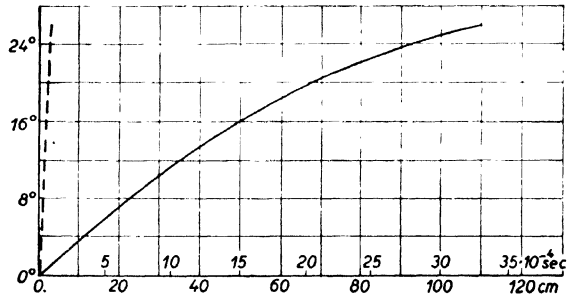


Fig. 245. Angular rotation of the 'sound image' away from the middle position when only time differences are applied. The broken line curve gives the angles in case of natural directional hearing, where the time difference is always accompanied by an 'added' intensity difference.

(v) *Localizing of a Sound Source, not Situated in a Horizontal Plane through the Ear Axis*

WALLACH [6] has shown that small movements of the head are conclusive for localizing a sound source in the vertical plane. If φ is the angle between the direction of the sound source and the ear axis and α the angle between the projection of this direction on the horizontal plane and the ear axis, then $\frac{d\varphi}{d\alpha}$ gives an indication of the elevation

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of the sound source. Small movements of the head around a vertical axis give information about $\frac{d\varphi}{d\alpha}$. For a sound source situated in the horizontal plane $\frac{d\varphi}{d\alpha} = 1$. For a direction perpendicular to this plane $\frac{d\varphi}{d\alpha} = 0$. Directions symmetrical to the horizontal plane have the same value of $\frac{d\varphi}{d\alpha}$, but a small movement of the head around another axis gives information to discriminate between these two possibilities.

§ 3. Stereophonic Sound Reproduction

Stereophonic sound reproduction can give a listener with the help of electromechanical devices, such as microphones and loudspeakers, the same impression of the position of sound sources (orchestra, speakers) as if he was listening in the studio or theatre to the original sound. Listening to a single loudspeaker, the sound of the different voices or of the different instruments comes from one point. With more sound channels, at least two, it is possible to give a more dimensional impression of reproduced sound. We will restrict the problem to the reproduction of different sound sources in a horizontal plane. For a good reproduction the relative angular bearings of the sound sources as observed by the listener must be proportional to the relative angular bearings of the original sources.

The first demonstration of this kind of sound reproduction was given by ADER in 1881 during the Paris Electrical Exhibition. ADER demonstrated with several carbon microphones and a telephone set for each listener.

In 1933 there was a sound transmission from Philadelphia to Washington. Three microphones were placed in front of the orchestra. Each of these microphones was connected to an amplifier and a loudspeaker [7].

In fact if we place a number of microphones before the different sound sources in the studio and the same number of loudspeakers (and amplifiers) in the listening room, the reproduced sound will give very good information about the position of the different sound sources.

Of course it is necessary to have the same configuration of micro-

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phones and loudspeakers. For the sake of economy it is desirable to diminish the number of these transmission channels. The minimum is two and we will discuss different solutions for the reproduction of a two channel system.

Using separate microphones, 'the time effect' as we discussed in § 2 is mainly responsible for the stereophonic effect. As we have mentioned before the intensity effect is predominant in normal directional hearing. Several experiments have shown that the 'definition' of the 'sound image', when we listen to a two microphone 'time effect' system, is not quite satisfactory. It is necessary to have a distance of 5–7 feet between the microphones.

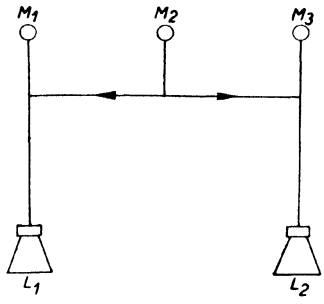


Fig. 246. Three microphones.

In order to avoid large changes in intensity when a sound source moves in front of the microphones from one side to the other, the distance of the sound source to the microphones must be at least 10–12 feet.

From an acoustical point of view this is not always permissible (influence of reverberation, large distance between the persons in a dialogue). A better solution is to use 3 microphones, connected as indicated in Fig. 246.

The results of experiments of a two microphone system [8] on the linearity between the angle φ of the sound source and the deviation u of the 'sound image' from the middle of the loudspeakers are given in Fig. 247 and Fig. 248.

We see that for a microphone distance $b < 40$ cm, the sound image stays between and never reaches the loudspeakers. For greater microphone distances the linear reproduction is restricted to small values of φ($b = 120$ cm, $2\varphi = 50°$, $b = 80$ cm, $2\varphi = 75°$, $b \,.\, 2\varphi_{max.} = 6000$).

A good solution of the microphone problem [8] is to put the microphones diametrically in a sphere with the dimensions of the human head (dummy head). This pickup system is based on the 'intensity effect' of the directional hearing, as we have seen the most predominant of the two. The 'sound image' has a good definition and it is possible to get a good reproduction with the different sound sources on a distance of 4–5 feet of the microphone combination.

References p. 435

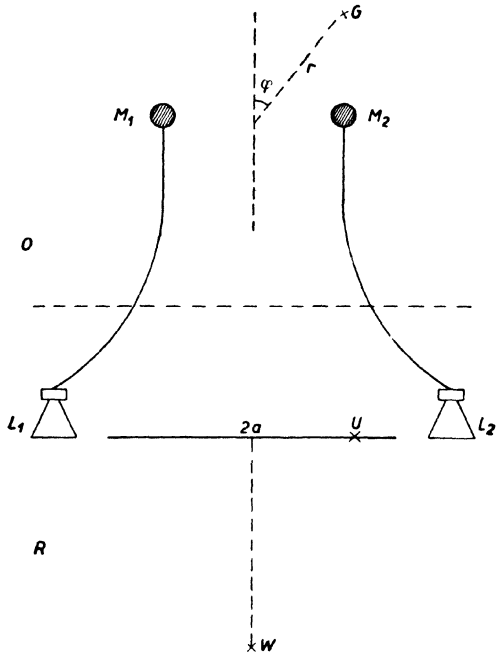


Fig. 247. In the recording room O two microphones (distance b) and a moveable source of sound G are set up. In the recording room R, the observer W observes the sound image of G between the loudspeakers L_1 and L_2 at a distance u from the midpoint.

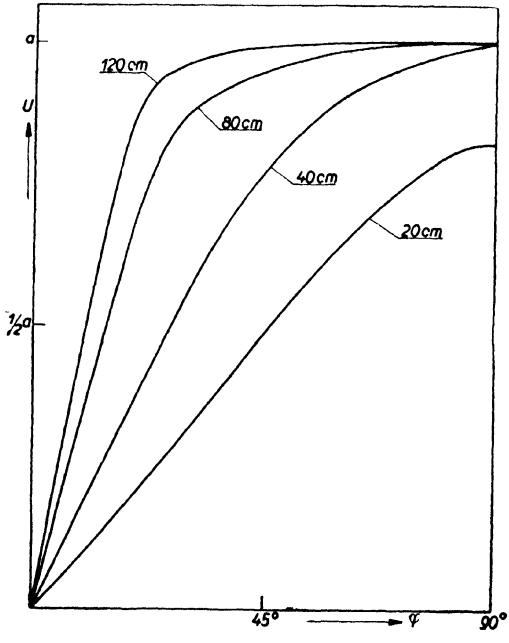


Fig. 248. In recording the sound of a source G, which is placed at an angle φ, with regard to the plane of symmetry, u represents the distance of the sound image from the centre observed by W in the reproduction room. The different curves refer to the distances recorded in the figure between the two microphones (b).

In Fig. 249 and Fig. 250 the results are given with this arrangement. For different sizes of the dummy head (diameter d) there is a good linearity between u and φ for wide angles ($d = 22$ cm, linearity to $2\varphi_{max.} = 90° : d = 17$ cm, $2\varphi_{max.} = 120°$, $d \cdot 2\varphi_{max.} \sim 2{,}000$).

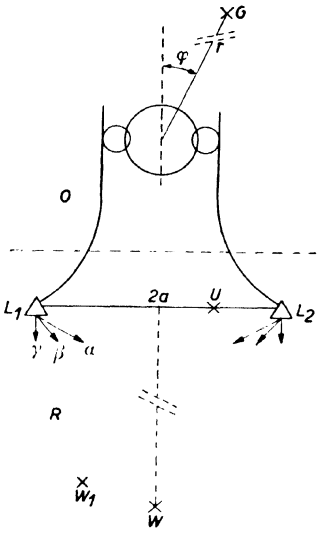


Fig. 249. In the recording room O an artificial head and a moveable source of sound G, are set up. In the recording room R, the observer W oberves the sound image of G between the two loudspeakers L_1 and L_2 at a distance u from the midpoint.

The conclusion is that a microphone arrangement in a dummy head (two channel system) gives excellent stereophonic reproduction.

Since Fig. 250 is valid for the observer W, it might be thought that an observer at any other point in the hall could localise the sound image at another point u. It would lead us too far afield to deal with this question in detail and we shall confine ourselves to a single aspect. When W moves along the line of symmetry the sound image scarcely moves. The displacement may, however, become somewhat greater, when W moves to one side to a point W_1 (see Fig. 249). At this point the influence of L_1 is dominant and the image shifts somewhat to the left. This could, however, be avoided by giving L_1 and L_2 a certain directional effect, *i.e.* by providing that the same sound energy is not radiated in the directions α, β, γ (Fig. 249), but that, as the lengths of the arrows indicate, there should be a decrease in the intensity radiated in the order given. In this way the dominating influence of L_1 (and the diminished influence of L_2) can be compensated, and an undistorted image can be obtained throughout practically the whole hall.

§ 4. Recording of two Sound Tracks for Stereophonic Reproduction

This can be done by a photographic or mechanical magnetic recording system[9]. It is important that the mutual distance of the reproducing heads (measured in the direction of the film or tape) must be the same

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as the mutual distance of the recording heads, because otherwise unwanted time differences occur and in consequence a displacement of the 'sound image'.

A difference of $2 \cdot 10^{-4}$ sec is the maximum value that is permitted.

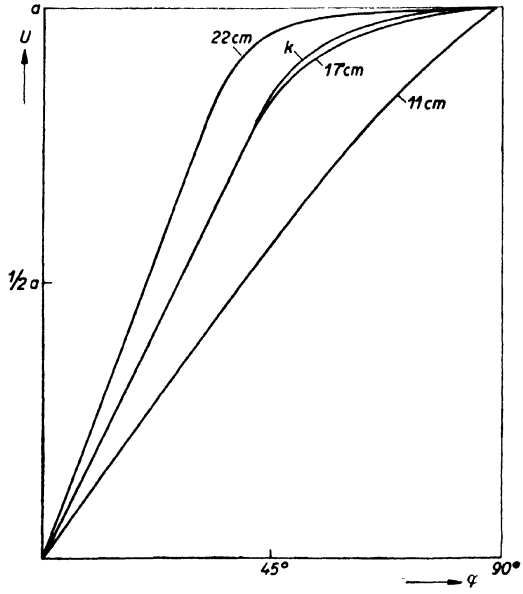


Fig. 250. In recording the sound from a source G placed at an angle φ with the plane of symmetry of the artificial head, u represents the deviation from the midpoint of the sound image as observed by W in the projection room. The various curves are for the different sizes of artificial head as indicated in the figure. k stands for a perfectly shaped head.

For a photographic recording (film speed 90'/min) it is necessary to adjust the relative position of the heads within a limit of 90 microns (4 mils). Magnetic recording (tape speed 15 "/sec) requires an accuracy of 75 microns (3 mils).

Cross talk of one channel into the other results in a diminishing of the intensity differences between the two channels, and thus causes a displacement of the sound image. The effect is not noticeable when cross talk is less than −25 db.

This is no problem at all for photographic and mechanical recording and in magnetic recording a distance of 1 mm between the two tracks is more than sufficient.

References p. 435

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DIVISION SIX

ANALYSIS AND SYNTHESIS OF SOUND; DESIGN AND PERFORMANCE OF MUSICAL INSTRUMENTS

CHAPTER 17

ANALYSIS OF SOUND

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§ 1. Analysis of Timbre or Wave-Form

The procedure that we adopt for analysing a complex note into its constituent 'tones' and their relative phases depends firstly on whether the note is held steady for sufficiently long to enable an analysing instrument to do its work on the source directly or whether the note is so ephemeral that a short history of the wave-form must be obtained on a record and portions of the latter submitted to subsequent analysis. Of course, this expedient of obtaining an oscillogram of the wave-form may be adopted in any case and often is used even when the note is quite steady.

Up to forty years ago, it was common to analyse a steady note, *e.g.* that of a musical instrument, by picking it up on a set of resonators, such as those constructed by HELMHOLTZ, tuned to the fundamental and harmonics of the note. This had the disadvantage that a retuning of all the resonators was necessary when the fundamental changed and that it did not record inharmonic overtones. Owing to these limitations this direct method of analysis is not used nowadays. It will also be noted that it could not deal with phase relationships though this is not important in acoustics when we are making our analysis from the aspect of the ear.

Other types of analyser proceed by frequency steps usually of an octave, recording everything within the band as an average. If the note does not fluctuate widely, it is possible to use this type on a sustained note which does not remain constant in timbre.

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These instruments, which are suited to a quick analysis of noise but not to precise delineation of steady quality, will be described first.

§ 2. Band Analysers

The basic principle of these instruments is that the sound, transduced into an amplified electric signal, is passed through a series of filters each of which has a fairly broad peak and sharp cut-off at each end. The width of the band determines the amount of detail in the analysis and also the time taken to perform the operation, these two factors being in inverse proportion. For the analysis of noise the bands may cover an octave each but a better arrangement makes the band-width proportional to the mean frequency in each. This is because in many instances the power diminishes in the upper part of the gamut; also such an arrangement imposes less stringent demands on the filter to attenuate the sound in the part which it rejects.

Another division is possible if one has regard to the subjective estimation of pitch (p. 262). Such an arrangement will split up our record into blocks of frequency which are of equal importance to the ear rather than into a purely arithmetic division.

An adaptation of this principle by FREYSTEDT [1] enables a rapid delineation of a sound into bands to be performed. He has filters spaced three to the octave to cover the range 50 to 4000 c/sec. A rotary switch brings these in turn into the circuit of the amplifier and the voltage passed by each is displayed on the cathode ray oscillograph. It is sufficient for the switch to halt for one tenth of a second at each stud for the response of each filter to be established and photographed on a moving film.

An improved version of this apparatus devised in the U.S.A. [2]

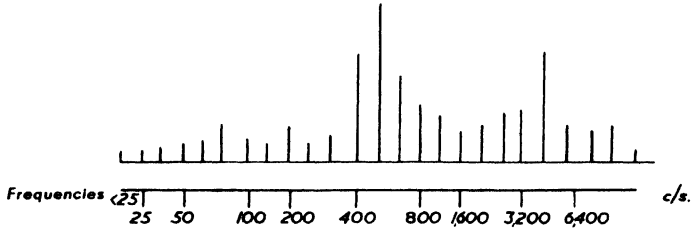


Fig. 251. Record on cathode ray screen of band analyser.

for underwater sound enables a spectrum to be scanned 60 times a second in steps, three to the octave.

This is very useful for rapidly varying sounds. The output of each filter is applied in succession to displace vertically the electron beam of a cathode ray oscillograph, while the usual horizontal sweep is in operation. This gives a display of vertical lines as shown in Fig. 251.

A variant of the band-filter analyser, not so often used, employs a set of tuned reeds in place of the electrical filter. The electric signal is fed to an electromagnet in front of which stands the array of reeds, which respond by vibrating more or less. These vibrations are recorded optically through mirrors mounted on the reeds. The apparatus thus links the early device of HARTMANN-KEMPF [3] for recording A.C. frequency with HELMHOLTZ and his set of resonators, to which reference has already been made. The reeds however suffer from the disadvantage that their natural rate of damping is small and that if mechanical damping is applied the frequency spreads over into that of neighbouring vibrators. This indeed is a concomitant of all filters, mechanical or electrical, and limits the speed of accurate analysis by their means.

§ 3. Heterodyne Analysers

This apparatus consists of a variable frequency oscillator whose output is mixed with the incoming electric signal at such a level that summation as well as difference tones are involved in the resultant (p. 4). The summation tone is picked out by a narrow band filter and the amplitude of its rectified response shown on a thermionic voltmeter or on an oscillograph screen. The filter often takes the form of a quartz crystal in electrostrictive or a nickel rod in magnetostrictive oscillation, say of 100 kc/sec. Thus if the oncoming signal has a component tone of 500 c/sec, the crystal will respond when the variable oscillator is tuned to 100,000 — 500 = 99,500 c/sec. Care has to be taken that the degree of non-linearity is not such that combination tones other than the simple sum and difference are involved. Normally the dial of the oscillator is rotated by hand, but for rapid analysis an electric motor may do this in synchronism with the sweep of the oscillograph electron beam, so that a record like that shown in Fig. 252 is obtained (to be compared with Fig. 251). (The upper curve exhibits 5-cycle band widths, the lower, octave bands.) The limitation of speed

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is again the damping of the filter, but in quartz this may be very small.

In many rapid analysers giving permanent records mechanical writing on paper may take the place of photographic recording with

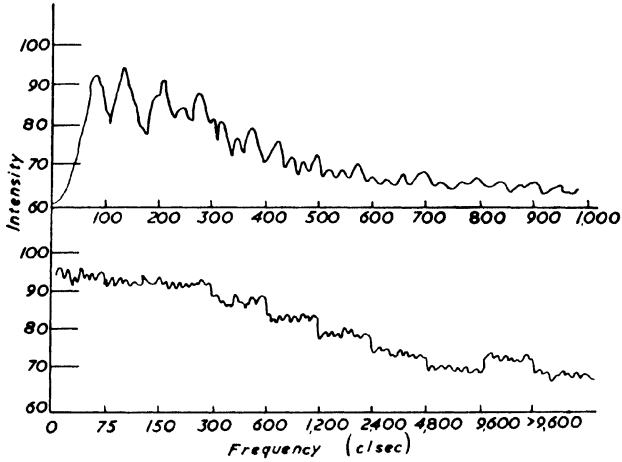


Fig. 252. Analysis of noise by filters.

a cathode ray oscillograph. Such a device as the Neumann writer may be incorporated in the output stage. In such an apparatus a pen moves step-wise over a piece of paper, which is moving at constant speed in a perpendicular direction. The pen, or the spot-light which may replace it when sensitised paper is used, may record either loudness or pitch. OBATA and KOBAYASHI [4] used two optical pens, one for each factor, on the same paper and so were able to analyse

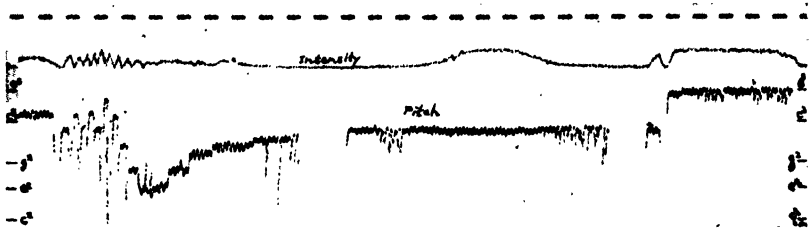


Fig. 253. Record of loudness and pitch during solo.

References p. 525–529

a solo of music as to its fundamental, frequency and intensity during the rendering (Fig. 253).

§ 4. Analysis of Steady Wave-Forms

It is often preferable to obtain a wave-form of the sound on a film and to analyse this at leisure. (In earlier times one adapted a gramophone record to this purpose.) There are three possibilities; (1) to make a graphical analysis by hand of the respective amplitudes of the lower harmonics, (2) to do the same on a machine, (3) to analyse the film optically.

The basis of FOURIER analysis is outlined in the Chapter 1. Graphical analysis is usually employed in simple cases where the overtones are harmonic and it is known that most of the power lies in the lower components. MILLER and DALE [5] have described methods of doing this which will not be reproduced here.

If one has many wave-forms to examine the above process is tedious and one employs for preference a mechanical or an optical machine.

D. C. MILLER made much use of the HENRICI [6] analyser in his classic work on the *Science of Musical Sounds*. This like most other machines requires a graphic or photographic enlargement of one complete period of the wave to be made, and placed with the time axis parallel to that of a traverser free to roll parallel to the displacement axis. A stylus beneath this carriage is moved manually along this trace. As this is done a number of spheres are caused to rotate by amounts corresponding to the sines and cosines in the required FOURIER series. A complete wave-length having been traversed the integrated components appear on dials.

A variant of such devices is the optical method for FOURIER analysis of MONTGOMERY [7], which acts on a small piece of sound film record of the variable-area type. On n small pieces of film of the same size and length as the specimen are recorded separately in variable density reproduction a harmonic series of cosines: $\cos x \ldots\ldots\ldots \cos nx$ the wave-length of the fundamental of this series being the same as that of the specimen.

If parallel light from a narrow slit is allowed to pass through the specimen with one of these films superposed, the amount transmitted

through the narrow slit should then be proportional to $f(x) \cos nx\, dx$. As any periodic function $f(x)$ can be represented as

$$a_0 + \Sigma a_n \cos (nx - \delta_n)$$

and the coefficients a_n are

$$\frac{1}{\pi} \int_0^{2\pi} f(x) \cos (nx - \delta_n)\, dx,$$

by the equations (p. 5), the value of the integral at its limits can be found by locating the slit positions at which maxima and minima of light are transmitted. The difference in transmission is a measure of a_n and the location of the maximum as a fraction of the wavelength (*i.e.* of one complete cycle of the wave trace under analysis) is a measure of δ_n. The light transmitted is measured on a calibrated photoelectric cell.

§ 5. Acoustic Grating

It is possible to construct a grating of equi-spaced vertical rods which shall perform for sound the same function which an optical grating does for light. MEYER [8] has constructed a model of a Rowland concave grating with a spacing of about 1 cm which he uses to analyse sounds in the upper sonic or ultrasonic range by the diffraction spectra which the grating produces on reflection. A ribbon microphone is traversed in front of the grating to catch the sound reflected from a nearly parallel beam directed on the grating from the source. Knowing the angle of diffraction and the spacing, the frequency of the component tones of the source may be calculated from the peaks of reflected energy, by the usual reflection grating formula. Fig. 254 shows three typical records of (1) two pure tones simultaneously sounded (2) a note, fundamental with overtones (3) a noise, obtained on this grating.

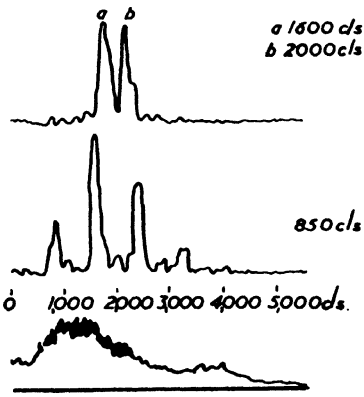


Fig. 254. Analysis by sound grating.

§ 6. Optical Analysers

The basis of this type is an apparatus due to DIETSCH and FRICKE [9]. It requires that the wave-form be made available as a hill-and-dale (variable *area*) type of film such as we have just been discussing but of larger size (3 or 4 inches wide) and of such a length that it can be wrapped round a glass cylinder with exact overlapping after one or several complete cycles. Along the axis of the cylinder (*C*, Fig. 255) is the filament of the special lamp *L*. By means of the constant-speed motor *E*, the glass cylinder is rotated and as this happens the lamp shining past the film and the slit *D* casts an image of a narrow section of the film on the photoelectric cell, *P*. The wave-form on the film is thus converted back into synchronous electric waves in the circuit containing the photoelectric cell and the amplifier, *A*.

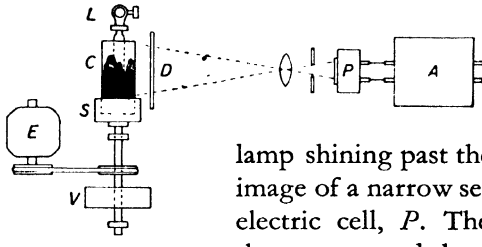


Fig. 255. Photoelectric analysis of varying sound.

After the current leaves the amplifier it may, of course, be immediately analysed in the wave analyser aforementioned, but the feature of DIETSCH and FRICKE'S device which is of particular value for inharmonic components was a vibrating reed to the electromagnet of which this current was fed. Varying the speed of rotation of the cylinder caused the reed to respond every time a component in the photoelectric current equalled in frequency the natural frequency of the reed.

In the author's [10] variant of this apparatus, used by a number of his students in the past 20 years for analysing vocal and other musical sounds, the reed is replaced by a vibration galvanometer of 50 c/sec. A considerable length of film (covering many periods) is wrapped round a narrow perspex drum which can be rotated slowly through a geared electric motor. The speed of the drum at any instant can be calculated from the appearance of a stroboscopic disc rotating at constant speed and illuminated by a flashlamp lit intermittently by equi-spaced contacts on the drum engaging a fixed contact.

If now the drum is revolved with increasing speed the galvanometer will respond (at an amplitude which can be made a linear function of

the photoelectric current amplitude) whenever the number of the peaks of a particular component pass the slit at 50 c/sec. To obtain the complete spectrum the response (*i.e.* width of light beam) can be recorded on film alongside the frequency (*i.e.* speed of drum). The lamp is of course run off constant D.C. supply.

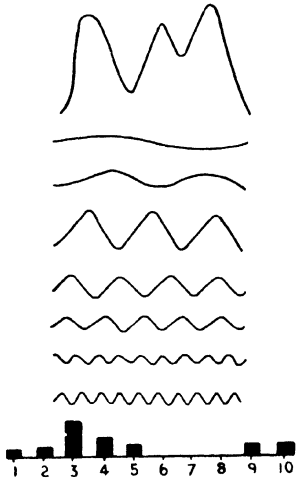


Fig. 256. Typical analysis of steady sound.

Fig. 256 shows an example of this analysis for a comparatively simple sound. The upper curve shows one complete period of the wave form as recorded by the oscillograph. Underneath are half a dozen simple tones into which it was analysed by the machine. These are also shown in relative magnitude by the height of the black rectangles on the 'acoustic spectrogram' at the bottom. In this case the components were true harmonics and comparatively low in the series. When the intensity of a musical instrument is raised or when we are dealing with complex systems like a bell or the human voice the components are no longer all harmonic to the fundamental, and moreover, they cover a greater portion of the musical scale, so that our analysis may extend to the detection of a score or more of partial tones within the frequency range of the apparatus.

It will be noted however that a normal oscillogram is not amenable to this machine as it appears as a black-on-white line. To get the hill-and-dale type of trace on the oscillograph which one needs for the machine, a high frequency signal (say 100kc/sec) should be modulated by the output signal and then sliced in two by half-way rectification. The oscillogram then appears somewhat as in Fig. 257c; the high frequency vibrations will probably smudge into a black mass when photographed on a slow-moving film, only the low frequency modulation of the wanted signal remaining sufficiently clear cut to interrupt the light.

§ 7. Unsteady Sound Analysers

It is possible to deal with certain classes of sound which have not a

steady state. The general case of casual and inharmonic unsteady sound such as some noises comprise defies FOURIER analysis, but there are two types of sound not uncommon in technical acoustics which these pieces of apparatus can deal with.

The first of these is the pulse. Inasmuch as a pulse resolves on analysis into an infinite FOURIER series, analysis by machine seems impossible but, in fact, the component intensities decrease as the frequency mounts, so that in practice a limited number of harmonics suffices. This is for a square wave. In echo-sounding we encounter another type of pulse whose envelope approximates to a castellated wave but whose fine structure is in fact a simple tone. Such, provided it lasts several periods, presents no difficulty (Fig. 257a, b).

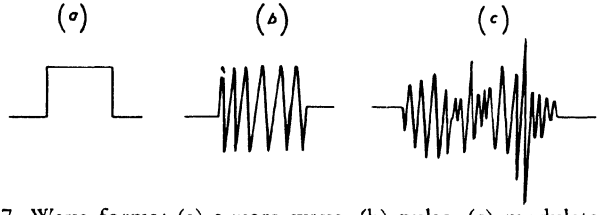


Fig. 257. Wave forms; (a) square wave, (b) pulse, (c) modulated signal.

We may at this point describe new methods for analysing fluctuating or transient sounds, since these are of the greatest use in connection with speech sounds. BROWN [11] uses the diffraction of light from a sound-film for this purpose. A general theory is given of the spectrum from such a generalised optical grating in motion through the spectrometer. The diffraction pattern may be regarded as an acoustic spectrum of the original sound averaged over a time T equal to L/V, where L is the width of the gate through which the film passes with speed V; to analyse transients, T must not be too large or too small. Unfortunately the dispersing power is necessarily of low order, and low dispersion cannot be counteracted by magnification owing to irregularity in the emulsion and the film surface generally.

IMAHORI [12] employs a photographic device, *i.e.* an 'analysing plate' on which a number of straight dark lines radiate out from the y axis in such a way that the light intensity in the shadow of the plate varies as $(a+b)$ sin $2\pi x/\lambda$ along the x axis, while the wave-length λ varies in linear fashion in the y direction. The incident light intensity is

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modulated in synchronism with the sound to be analysed and a photographic film moved at constant velocity along the x direction behind the analysing plate. If the source consists of a harmonic series of vibrations the film after development shows sharp lines the blackness of which varies sinusoidally over a uniformly brilliant background, one line corresponding to each component. These derive from stroboscopic interference between light fluctuations and motion of the analysing plate relative to the moving film. If the number n of lines inscribed on the plate is not large enough these lines broaden into bands whose overlapping limits the resolving power. Fluctuations can be traced by making n sufficiently small that the resolving power in the respect of frequency is sacrificed to that of time. An example is given (which we reproduce in Fig. 258) of the analysis of a 'square

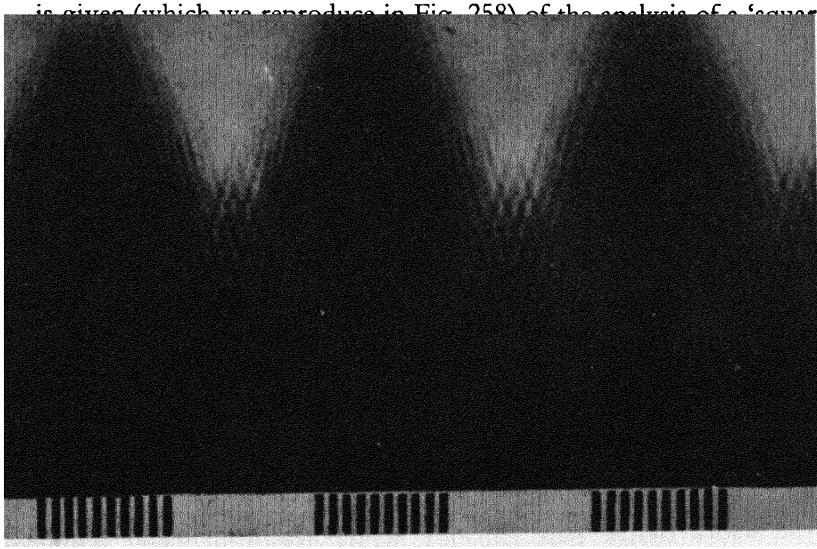


Fig. 258. Optical analysis of square wave.

wave' of 10 pulses succeeded by an intermission of 10 periods, then 10 pulses again, and so on as shown at the base of the photograph. The centre line corresponds to the fundamental frequency where the band spacing equals the wave-length of the pulses, whose sequence is illustrated by the pattern below the photograph.

The type of transient often encountered in the stopping and starting sounds of musical instruments is a modified copy of the full quality,

perhaps with one or two components of the steady timbre missing, perhaps with additions. This also the optical analyser can deal with step by step, if specimen portions of the film are selected and analysed during the transition stage. We shall meet examples of these later.

A useful development of the optical method is that devised at the Bell Telephone Labs. for the recording of speech sounds. The aim of the method is to produce visual patterns of the sounds of words or phrases which may be recognised as such and, in which, by comparison, the variants introduced by individual speakers may be studied. Sound from a microphone passes through a set of 12 filters each passing a band of 300 c/sec. The outputs of the filters are recorded photographically one above the other on a moving strip so that the blackness of a particular location represents the sound intensity in a certain band at a certain instant. The sound spectrograms thus record frequency, intensity and time on one strip (*cf.* p. 213).

By re-combining the components in their true magnitude and frequency we can, of course, remake the original sound of the source, or we may experiment with other combinations to initiate new tone qualities. These are the object of instruments which produce synthetic timbre by electrical means.

We shall describe some of these electrophonic synthesis instruments later after dealing with the more conventional mechanical types of musical instruments.

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CHAPTER 18

MECHANICAL MUSICAL INSTRUMENTS

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§ 1. The Basis of Musical Instruments

No attempt will be made in a text book of technical acoustics to write a history of musical instruments, but one must discuss in general terms their design and function.

(1) As a basis there is a vibrator, whose frequency is variable either by steps or continuously over a certain range of frequency. Further the step-wise variation may cover a whole region of the musical scale or be limited to the series of overtones of an invariable source; compare the orchestral horn with the post-horn.

(2) There is a means of exciting the vibrations of the primary source and this exciter may itself be a body with so many degrees of freedom, *e.g.* the reed-pipe.

(3) There is an amplifier or soundboard which often acts as a radiator for filling the auditorium with sound.

The primary vibrator may be:

(a) solid;- string, reed, membrane, plate, solid of revolution.

(b) fluid;- jet, fluid column or cavity,

(c) aetherial;- oscillating circuit.

All but the most primitive instruments then must be considered as coupled systems, of which the general theory has been discussed in Chapter 1. They can also be considered as lumped impedances, their natural frequencies being derived by equating the composite impedance to zero.

Considering the musical instrument as a transducer of energy we

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shall be interested in the polar curves of its output as well as the variation of output with frequency.

§ 2. The Formant

It was HELMHOLTZ who suggested after extensive studies of vocal sounds that specific vowels were characterised by a specific set of resonances, which HERMANN [13] called 'formants' (*cf.* Chap. 10, § 9). A vowel sound sung at a different pitch had all or some of these resonances in common, corresponding to the injection into the vocal resonators of a fundamental tone of varying pitch. The theory in its original application is an over simplification since a singer does not maintain his vocal resonators in the same form throughout the gamut as he sings, but the theory is obviously apposite to many musical instruments whose function is to produce a large number of distinct tones from the same resonator or, at most, to inject tones from different sources into a common sound-board for radiation purposes.

VIERLING [14] exemplified many of the characteristics of formants by devising electric circuits having fixed resonances, injecting into them notes produced by oscillators and observing the resulting wave-form which came out of the system. For example, if the formant is a resonance with broad peak at 1100 c/sec (Fig. 259b) and a note rich in

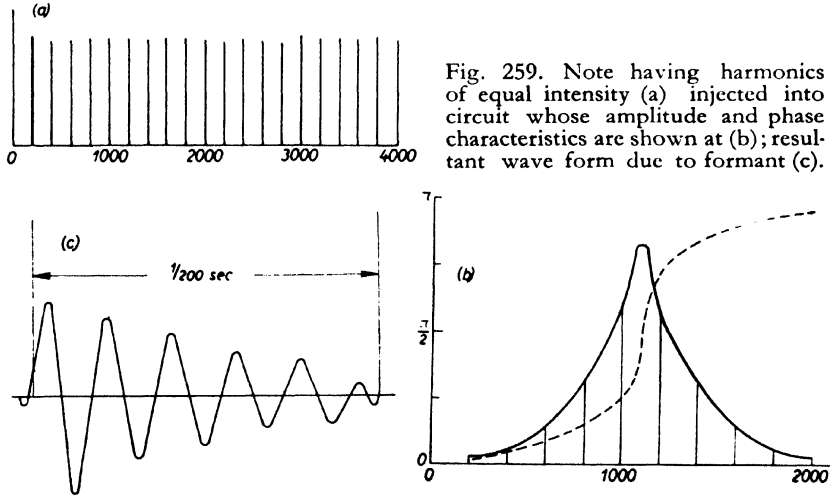


Fig. 259. Note having harmonics of equal intensity (a) injected into circuit whose amplitude and phase characteristics are shown at (b); resultant wave form due to formant (c).

harmonics, having a fundamental of 200 c/sec (as shown by the vertical lines Fig. 259a) is injected into it, different degrees of amplification and phase factors will be applied to these harmonics in transit through the formant circuit. If the harmonic series has equal intensity (an unlikely case in practical acoustics) the resultant wave-form is of the damped type (Fig. 259c), characteristic of relaxation oscillations and, in fact, reminiscent of the wave-form of the vowel '*ah*'.

VIERLING also tried the effect of injecting a short pulse of 200 c/sec into this resonant circuit, or suddenly starting and stopping a pure tone of 200 c/sec. The oscillations of the circuit at its natural frequency before and after the pure tone is established are important for an understanding of the transient sounds of all musical instruments and of the 'voiced consonants' of the human voice. Their theory is given in Chapter 1 (p. 1).

Naturally, these effects are more important as the formant lies higher in relation to the injected fundamental. VIERLING found no distortion of the 200 c/sec. sound when the formant was set to 300 c/sec, either for steady maintenance or pulses.

The formants in musical instruments may arise from the vibrations of the solid soundboards to which the primary vibrators are coupled, from the aerial cavities such as those in the bellies of string instruments and from the walls of the tubes in wind instruments. We shall have occasion to refer to many of these later.

§ 3. Primary Vibrators

Among solid bodies, classification may be made according to:

(A) whether it must be put under tension by an impressed force before it can vibrate, or

(B) whether the body possesses natural elasticity.

We can also group the different systems by the relative importance of the three principal dimensions involved, and the number of degrees of freedom.

(1) one dimension predominating; (a) strings, (b) rods.

(2) two dimensions predominating; (a) membranes, (b) plates.

(3) three dimensions comparable; solid spheres.

In most cases transverse vibrations will be usual but often longitudinal modes can exist and couple with the transverse modes.

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In fluid vibrations it is the bulk elasticity of the fluid which is involved and the motions are all longitudinal.

We distinguish;

(4) fluid columns when one dimension predominates as in many wind instruments.

(5) fluid cavities; all dimensions take part, as in the vibrations of bubbles, air cavities and liquid drops.

At sufficiently high frequencies, modes of vibration in which the diameter of a column intervenes must be considered. In other words, the distinction between classes (4) and (5) must be regarded in relation to the wave-length of the partial tone concerned.

§ 4. Stretched Strings

The elementary theory has been given in Chapter 1, § 9. Modifications of this have been made to deal with the practical complications of stiffness and yielding of the supports or bridges.

These two factors together with viscous damping cause the partials to be inharmonic. We shall refer to experiments with stringed instruments which demonstrate this.

The modes of excitation of a string comprise

(1) bowing,

(2) striking,

(3) plucking,

(4) wind action,

(5) electromagnetic attraction of steel wire.

§ 5. Analysis of Bowed String

The modes of vibration of the theoretical string according to the position of the point of excitation have been laid down in YOUNG's [15] law;- 'No overtone is present which would have a node at the point of excitation'. Thus if the string is moved at one point only x/l from one end, the xth mode and all multiples of this mode will be absent. Usually a number of modes round about the xth: $-x \pm 1, x \pm 2$, etc. will be removed in this way if the excitation occurs over a region instead of at a point.

The action of the bow is that of an escapement. The string is caught by the bow and dragged to one side until the mounting components

of string tension in a direction opposed to that of the bow overcome the static friction causing the string to jump back. The string is again caught up by the bow as these components are reversed. Two further theorems, beside that of YOUNG, have been put forward.

(1) HELMHOLTZ' law [16]; 'When a string is excited at an aliquot point $1/k$, the part of the string immediately under the bow moves to and fro with two constant velocities, whose ratio is equal to the ratio $1/(k-1)$ of the segments into which the string is divided by the point in question. The smaller of these two velocities has the same direction as that of the bow, and is equal to it'.

(2) KRIGAR-MENZEL's [17] law; 'When a string is bowed at any rational point p/q, where p and q are prime to each other, the part of the string immediately under the bow moves to and fro with two constant velocities whose ratio depends only on q and is $1/(q-1)$'.

RAMAN [18] considers the stationary vibration of the string as made up of two progressive waves travelling in opposite directions and reflected from the ends of the string. The transverse velocities in these waves as they cross the bowed point must satisfy the conditions there, *i.e.* that the resulting velocity of the string jumps between two finite and constant values U_1 and U_2. In consequence of the constant velocities at the bowed point, the acceleration of the string there is zero except for certain infinitesimal fractions of the period when the jumps occur; consequently the sum of the velocities of the string due to the two waves at the bowed point should remain constant except at the instants at which the jumps of velocity occur. It is possible to secure this condition if; on one side of the bowed point and distant l from it, the slope of the velocity wave due to the progressive wave alone were $\tan^{-1} j(U_1-U_2)/2l$ while that due to the retrogressive wave only, at the same distance l on the other side of the bowed point, were $\tan^{-1} j(U_2-U_1)/2l$ (j is an integer for the order of the mode).

The waves then have constant slope throughout, with discontinuities where the velocity suddenly drops. The waves are accordingly of zig-zag type.

When a given point is a node of one of the harmonics of the string the non-appearance of this harmonic in the complex note requires a number of discontinuities, at intervals equal to half the wave-length for this harmonic, to pass over the bowed point simultaneously in pairs, and so cancel each other's effects at this instant. All other dis-

continuities involve a jump in the velocity from U_1 to U_2 as they pass over the bowed point.

The superposition of these velocity waves causes the resultant velocity graph along the string to consist of j discontinuities between $(j + 1)$ straight lines inclined at an angle $\tan^{-1} 2j(U_1 - U_2)/2l$ to the x axis (Fig. 260), and the points where these inclines cut the x axis are the nodes of the j'th partial, for as two of them (the ends of the string) are points of continual zero velocity, so also must the others be. Thus A, B, C, D, etc. (Fig. 261a) are nodes of the stationary vibration; E, F, G, etc., are points where the jumps occur and thus form possible bowing points.

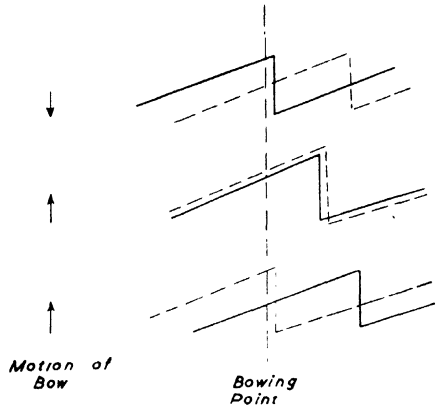


Fig. 260. Two progressive waves on string (- - - and ———) in relation to the bow's point of application and movement.

As

$$U_2/U_1 = \overline{EN}/\overline{EM},$$

$$\frac{U_1 + U_2}{U_2} = \frac{\overline{EB}}{CB} = \frac{jx_j}{l} \qquad \text{(VI, 1)}$$

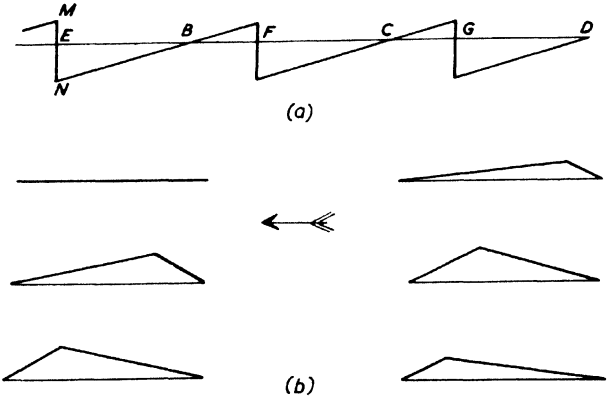


Fig. 261. Successive forms of bowed string.

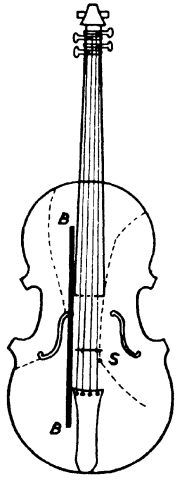
x_j being the distance of the bowed point from the nearest node of the j'th harmonic, and $CB = l/j$.

This is also the fraction of the period for which the string moves with the bow. From the velocity graph we can calculate the configuration of the string at each instant. The displacement for the single discontinuity case consists of the two straight lines meeting at the point which the velocity discontinuity has reached. The positive and negative wave forms superposed then give the instantaneous configuration of the string, as in Fig. 261. When the bow is applied at a node, Raman shows that the absence of several harmonics requires discontinuities separated by straight lines parallel to the x axis.

§ 6. Viols

When, as in the instruments of the violin family, the strings stretched over a 'bridge' are coupled to a wooden frame and an air cavity, the sound is (a) modified as to quality, (b) amplified, (c) radiated in a polar way, *i.e.* unequally round the source.

The first to make thorough scientific studies of the violin were GILTAY [19] and RAMAN. On the threshold of the electronic age BACKHAUS [20] did good work, while lately MEINEL [21] and also SAUNDERS [22] and his school have made a 'scientific search for the secret of Stradivarius', to quote the title of one of the latter's important series of papers.



The shape of the violin is sketched in Fig. 262. Prominent features are the 'f holes' through which the air vibrations communicate with the atmosphere, the 'bass bar' B designed to give strength to the upper wood to which it is glued and the 'sound post' S which lies under the bridge (usually under one foot) and prevents the case collapsing under the pressure of the bridge which in turn supports the pressure of the strings in tension.

Fig. 262. Shape of violin, showing possible nodal lines (– – –) and position of sound-post (S) and bass-bar (BB), which lie between the upper and lower wooden plates.

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§ 7. Effects of Pressure and Width of Bow

LIPPICH [23] asserts that increase of either bowing speed or pressure raises the general intensity, so that, for example, it is difficult to press lightly and play loudly or to press heavily and play softly at the same time. The effect of the finite width of the bow will be to remove some of the higher harmonics, for all those having a node under the bow will be absent. This will cause variations in the quality of the note produced as the bow is near or far from one end. With the bow of a violin for example close to the bridge, the note will be richer in the first half dozen harmonics than if it is applied at the usual place, *i.e.* at about one ninth of the length from the tail (*cf.* Fig. 262). Playing near the bridge of a violin is sometimes introduced by composers as a special effect (*sul ponticello*). The simple wave forms drawn in Fig. 261 will also not be exactly reproduced because only one point on the string can move with the bow; the other parts of the string under the bow will have a greater or less motion. This tends to smooth out the zig-zag wave-form and reduce the high harmonics.

§ 8. Resonant Effects

Each of the stringed instruments has one main resonance due to the air cavity and another, called the 'wolf note', ascribed to the body. In the violin these are respectively near 300 and 500 c/sec.

While these are apparent in the frequency spectra of both good and bad violins (Fig. 263), it is undesirable that they should be unduly prominent. Especially on a poor violin, when a tone near that natural to the belly is elicited, coupling between the two results in an uncontrollable wobbly note, which has earned this name of 'wolf'. The belly then executes a S.H.M. of large amplitude, whereas at frequencies a little above or below, its motion is quite complex. A fact which does not seem to be explained is that the wolf note is usually observed at the upper harmonics of the wooden system of the violin, and not at the fundamental.

RAMAN [18] supposes that at the wolf pitch the string starts sounding its fundamental, but resonant vibration of the wood drawing energy from it, the pressure between the bow and string falls, and the vibration changes to a type in which the octave predominates; on the consequent

cessation of the vibration in the wood, the fundamental reappears. These cyclic changes between octave and fundamental are apparent in the records of the movements of the string.

According to KESSLER [24], who has explored nodal patterns over the back of the violin, a mode with two nodal lines (shown as dashes on

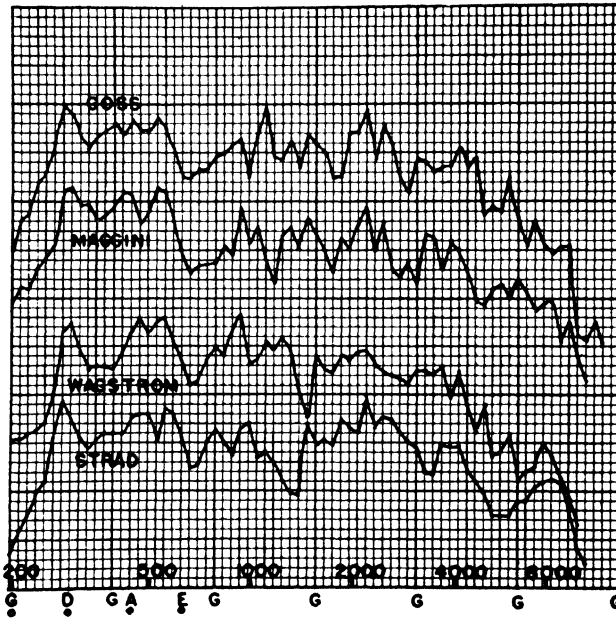


Fig. 263. Acoustic spectra of violins.

Fig. 262) is characteristic of the wolf. The *f* holes have little effect on this but movement of the bass bar shifts the body frequency. It is better to reduce the wolf intensity by such adjustments rather than to try to kill it altogether as this would spoil the instrument's radiating properties. Fig. 264 exhibits changes produced in the response of a cello by successive shifts (*a*) (*b*) (*c*) of bridge and sound-post.

The various resonances should, of course, be fairly evenly spread over the playing range,—five octaves above the G string in the case of the violins.

Cheap violins lack intensity in the lower register, but this fault may often be corrected by the maker. The quality is affected by well-known adjustments;- the position of the sound post and the height,

weight and form of the bridge are those generally subject to variation. The right foot of the bridge which stands over the sound post hardly moves, but the bridge tilts laterally about this foot, thereby transmitting string vibrations to the belly through the bass bar. (It may also tilt longitudinally if unpleasant shrill longitudinal vibrations of the bridge are elicited but it is so positioned that it does not readily pass these on to the belly). The vibrations of the strings have to pass through a narrow 'waist' in the bridge to reach the feet and filing this waist has a profound effect on the acoustic spectrum, taking away any shrillness of the upper partials. Another dodge which the player can adopt for the same purpose is to slip a light paperclip on to the bridge under the strings from the high pitch side.

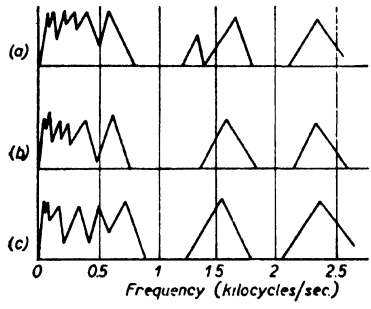


Fig. 264. Improvement of violoncello.

It is well known that an overall reduction of loudness and a special reduction of the intensity of the upper partials is affected by the player of any member of the string family putting a weighted clip over the strings. This, known as the 'mute', damps the motion of the bridge as it is pulled to and fro by the string. The load also alters the natural resonances of the wood and, not being rigid, itself undergoes elastic oscillations under the action of the string.

The total result is an alteration of quality towards mellowness—damping of higher partials—as well as an overall reduction of intensity, when the instrument is muted.

§ 9. Effect of Material and Varnish

It remains to consider the nature of the wood and the varnish. MEINEL [21] found that the latter had no effect on the position of the resonances, but could lower some of the peaks.

BACKHAUS [20] has recorded the acoustic and mechanical spectra of violins, mechanically played while preserving a constant bowing pressure. He varies the frequency by stopping the string and measures

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both the acoustic power at a fixed distance away and the amplitude of the belly. Provided the load on the bow exceeds 25 g weight, the quality is found to be independent of bowing pressure. The acoustic and mechanical spectra go hand in hand. Most interesting are his observations at frequencies near the wolf note. As the tuning of the string is varied in the neighbourhood of the wolf note, the system behaves like a heterodyne oscillator in which the condenser of one of the coupled circuits is tuned while the other is fixed. BACKHAUS uses a continuous bow something like a band-saw which excites—in the first instance—a string stretched over bridges mounted on a metal plate. He finds how the pressure of the bow on the string affects the duration of the starting noises characteristic of the sound of the bowed string. This time was a minimum for a force of 27 g on this skeleton violin.

The second part of his work deals with the coupling between the string and the belly of the violin. Beats characteristic of a system with two degrees of freedom are exhibited by the wave-form of the sound of the instrument as picked up on a microphone. A close coupling between the natural vibrations of the string and the forced vibrations of the wood is characteristic of a good violin; in other words, for these the material acts as a soundboard without those marked resonances such as—in extreme cases—give rise to a wolf note.

MEINEL with a similar bowing machine is interested principally in the effect of the thickness of the wood on the tone characteristics. If this is overdone, the spectrum is wanting in the lower frequencies (below 500 c/sec); very thin wood exaggerates the low at the expense of the high components and the tone lacks brilliance. Putting on the varnish made little difference, but moving the sound post redistributed the resonance peaks besides altering the efficiency of the apparatus, considered as a sound radiator. Some special shapes of violin (trapezoid, among others) failed to improve on the tone of the conventional shape and would be—one would imagine—awkward to play with the manually operated bow.

MINNAERT and VLAM [25] have endeavoured to establish the role of the bridge in coupling vibrations of string and belly in the violin. The vibrations of the bridge were observed by a beam of light reflected from a galvanometer mirror stuck on it with wax. Both flexural vibrations (both sides moving together in the direction of the string)

and torsional vibrations (one side moving towards the player while the other moves away) were noted, as well as simple pivotting to-and-fro without bending, as it were of a rigid body. All these were strongly dependent on the frequency of the note excited, but it seemed to be immaterial which string was bowed. Each of the three modes has a favoured pitch which was altered when a load in the form of a mute was put on the bridge, but not by changes in the resonance of the air cavity—accomplished here by the ingenious means of passing coal gas through the interior.

SAUNDERS[22] at Harvard University has carried out an independent investigation which to some extent overlaps the ones just described. He prefers hand bowing as it gives similar spectra to the mechanical bow and because musicians, not unnaturally, are inclined to discount the results of unconventional treatment. He considers that changes in bowing pressure alter the tone quality (presumably of the steady note) very little, change in bowing speed a little more, distance of bowing point (from the bridge) most of all. But the variations were not so great, if a medium distance were maintained, to prevent an average spectrum for each fundamental frequency being plotted. 'Every peak on such a response curve is the haunt of a wolf note which appears whenever the load on the bow is insufficient to 'kill' it. The kill results from the application of sufficient energy to wipe out the transient beats between vibrations of string and belly when left to themselves'. He has made comparative tests of a number of instruments including an aluminium copy of a Stradivarius, but confesses that much work is needed before the physicist can devise tests which will tell a good violin from a bad one.

Fifteen years ago a group under SAUNDERS began to study the mechanical behaviour of violins in the hope of finding out how the best ones differ in their vibration characteristics from those which are recognized as inferior. It was hoped that the results might lead to improvements in new violins.

The first publication included an account of an analyser developed for this work; the embarrassing variety of harmonic patterns disclosed in violin tones; the response against frequency curves obtained by means of the analyser from many new and old violins; the curves of total intensity against frequency; the mechanical efficiency of violins; the effects of variations of bowing, and of moisture, string tension,

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etc; the friction between bow and string; etc. This article raised many unanswered questions.

It was followed by a paper in which the response curves of several famous old violins were compared among themselves and with those of new violins; the filtering action of the bridge was discussed; a suggestion was made in regard to the meaning of the term 'carrying power' so often used by violinists; and an account was given of the results of a test of the ability of an audience to pick out the sound of a Stradivarius, when it and two new violins were played in succession behind a screen.

A brief discussion of the psychological effects involved was added in the attempt to explain why these judgments depended mainly on the order in which the two sets of instruments were played.

A note in the *Year Book of the American Physical Society* (1940) mentioned some experiments in which JASCHA HEIFETZ took part, which were designed to measure and compare, from high speed records of actual playing, the periods of growth and of decay, when an excellent and a bad violin were used in succession; hence to secure an indication of the value of the damping in the wood of these violins.

The latest publication was by WATSON, CUNNINGHAM and SAUNDERS [26] and gave an account of a new and superior method of obtaining response curves directly, without tone analyses, by exciting the violin electromagnetically with a force oscillating with a single frequency which could be varied over the whole range of the instrument. All response curves recently obtained have been done by this method, but this part of the experimental work came to an end in 1941. The new method included direct measurements of the decay of pure tones, from which the damping at a few selected frequencies could be obtained. One of these frequencies was chosen to coincide with that of the main vibration of the air inside the body; the others coincided with natural vibrations of the body, which were unaffected by the inside air.

The response curves obtained by the old and new methods differ in certain respects. The old curves are based on observations taken a semitone apart, while the new curves are continuous, and include all frequencies. Thus a narrow peak of response which might be missed by the first method is caught by the second. This is important in the high frequency region, where the overtones of the violin body are so

crowded that two or three may occur inside the interval of a semitone.

ABBOTT and PURCELL [27] have measured YOUNG'S modulus and decrement for various specimens of wood used in the manufacture of violins and of pianoforte soundboards. FUKADA [28] concludes that a suitable violin wood has a high YOUNG'S modulus and small damping.

ROHLOFF [29] has also obtained acoustic spectra of violins. In addition he has measured the rate of damping of notes produced in air and in vacuo. If the air is responsible for a high proportion of the total damping and the internal friction a small one, the apparatus is an efficient radiator of sound. This is the case with the old Italian violins examined. By comparing the decrement of beams, stressed and unstressed, in transverse vibration, this author concludes that the makers stressed the bellies of their instruments to produce this effect—but compare the remarks of SAUNDERS, above, on this point. The purpose of the varnish was to reduce the internal damping in the wood.

The physical nature of the differences between good and mediocre violins seem from the work of SAUNDERS and of MEINEL to be somewhat intangible. In fact, to many people, they do not exist, audiences of musical critics being unable to pick out a Stradivarius from other good modern violins if they can only hear and not see them. SAUNDERS maintains however that a famous player such as HEIFETZ can tell when he is playing on a good violin though unable to express the difference in words. In part it seems to be a matter of ease of tone production or 'singability' in the instrument, but the transients, which have not been much studied as yet, also seem to play a part. SKUDRZYK [30] says that Strads and their good modern copies radiate these transients (particularly in the low frequencies) much better than common violins. This makes them stand out well from the orchestra when playing a concerto.

CHESLOCK [31] has made a study of the *vibrato* on the violin. He finds that the player uses a repetition rate of about 6 cycles per second, the variation in pitch during a cycle amounting to about 3 per cent, corresponding to a quarter tone in the middle of the violin's range. If the range is greater than this, the vibrato is condemned by the musical listener as giving the impression of two notes played simultaneously.

GHOSH [32] has studied the acoustics of the 'stroh violin', a one-stringed fiddle in which the primary source is coupled through a lever

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at one end to the diaphragm of a sort of loud-speaker horn. The principle of sound production is the same as in the mechanical gramophone. He measured the bowing pressure required to maintain steady tone and found that it had to be increased along with radiating power in the neighbourhood of resonant frequencies of diaphragm or horn. Otherwise, a beating tone is heard, explained by RAMAN [33] in similar terms to those of the action of the mute on the violin (*vide supra*).

The vibrations of the diaphragm then decrease in amplitude, but as may be expected, they follow the change in the vibrational form of the string with a considerable time lag. The decrease in the vibration of the diaphragm results in a falling off of the rate of dissipation of energy, and when this is again below a critical limit the string regains its original form of vibration, passing successively through similar stages but in the reverse order. This is then followed by an increase in the vibration of the diaphragm, and the cycle repeats itself indefinitely.

§ 10. Struck Strings; The Pianoforte

When we apply the HELMHOLTZ-YOUNG laws to struck strings we observe that the choice of striking point and the width of the string over which the hammer acts play important parts in determining the quality. It is generally stated that the maker's choice of striking point—one seventh of the length from one end—is governed by the necessity of eliminating the seventh partial, but this has been questioned. In fact, the hammers are usually wide enough to cover at the same time the nodes of the 6th, 7th and 8th partials. Nevertheless some of these overtones may intrude, probably due to the yielding of the pads of the hammers.

The special features of this type of sound-producer lie in the mechanics of the impact and in the fact that the sound is not maintained. When the string is struck a wave passes in both directions, the one which is first reflected with reversal of phase at the nearer bridge may come back in time to help lift the hammer off the string in its recoil. The theory of KAUFMANN [34] which is based on these ideas has been checked experimentally by GEORGE [35], R. N. GHOSH [36] and DAS [37].

As the simplest, the original form of KAUFMANN will be outlined

here. Let M represent the mass of the system moved, *i.e.*, the mass of the hammer plus a mass equivalent to the string or part of the string moved, imagined to be collected at the struck point. Taking this as the origin of coordinates, we imagine the string to be bent into a form similar to that in Fig. 265. The short piece of string a, to the right of the struck point, is assumed to remain straight during the impact,

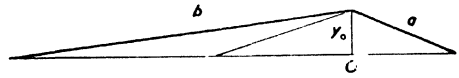


Fig. 265. Theory of struck string; O, striking point.

whereas the longer piece b is bent into some form of wave such that there is a definite displacement y_0 at O, and a definite slope at the same point on the negative side of O. Now the forces which tend to restore the string to its undisplaced position are the resolved tensions of F in b and a at O, *i.e.*

$$-F\frac{\delta y}{\delta x}\Big|_0 - F\frac{y_0}{\alpha} = M\frac{\delta^2 y}{\delta t^2}\Big|_0 \qquad \text{(VI, 2)}$$

If $y = f(x + Vt)$ on the left of O, $\dfrac{\partial y}{\partial x}\Big|_0 = \dfrac{1}{V}\dfrac{\partial y}{\partial t}\Big|_0$, so that the equation of motion of this point can be written;

$$M\frac{\delta^2 y}{\delta t^2}\Big|_0 + \frac{F}{V}\frac{\delta y}{\delta t}\Big|_0 + \frac{F}{a}y_0 = 0 \qquad \text{(VI, 3)}$$

This equation is of the form corresponding to damped motion and its solution (as long as the reflection of the wave has not reverted to O from the distant end) is

$$y_0 = \frac{U}{\sqrt{\left(\dfrac{F}{Ma} - \dfrac{F^2}{4M^2V^2}\right)}} \exp.(-Ft/2MV) \sin \sqrt{\left(\frac{F}{Ma} - \frac{F^2}{4M^2V^2}\,t\right)} \qquad \text{(VI, 4)}$$

where, at $t = 0$ $y_0 = 0$ and $\partial y_0/\partial t = U$, the striking velocity of the hammer. KAUFMANN put this result into a more practical form by putting in the fundamental time-period $(T = 2l/V)$
and mass of whole string, ml;

$$y_0 = \frac{UT}{\sqrt{\left\{\dfrac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}}} \exp.\left(-\frac{ml}{M}\cdot\frac{t}{T}\right) \sin\frac{t}{T}\sqrt{\left\{\frac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}} \qquad \text{(VI, 5)}$$

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assuming that the hammer is thrown back off the string at the instant the latter comes to rest, which occurs when the opposing force due to the string equals the pressure exerted by the hammer.

By differentiating the above and putting $\partial^2 y_0/\partial t^2 = 0$, we obtain the time of contact (τ) as

$$\frac{\tau}{T} = \left\{\frac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}^{-\frac{1}{2}} \tan^{-1}\left[\frac{\sqrt{\left\{\frac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}}}{\frac{ml}{M} - 2\frac{l}{a}}\right] \qquad \text{(VI, 6)}$$

When ml/M is small compared with l/a as KAUFMANN claims it is in practice, the angle in the above expression is approximately π radians, and

$$\frac{\tau}{T} = \frac{\pi}{\sqrt{\left\{\frac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}}}$$

$$y_0 = \frac{UT}{\sqrt{\left\{\frac{ml}{M}\left(4\frac{l}{a} - \frac{ml}{M}\right)\right\}}} \exp.\ -\frac{ml}{M}\cdot\frac{t}{l}\sin\frac{\pi t}{\tau} \qquad \text{(VI, 7)}$$

M. GHOSH [38] has also worked on the problem, treating the hammer first as rigid then as elastic. He shows that beside the main hump on the string produced by the impact, there are subsidiary kinks which he ascribes to waves on the short part of the string, neglected in KAUFMANN's theory, but satisfactorily explained in the revised theory. He uses HEAVISIDE's operational methods which had been introduced into pianoforte theory several years earlier by R. N. GHOSH [36] to deal with the general case where the striking point is remote from either end.

Previous analyses on the theoretical side had ignored the coupling between the vibrations of the two segments into which the string is divided at the point of impact, in other words, the effect of successive reflections of the waves from either end had been neglected and only the once-reflected wave considered. In the displacement: time curve for any point on the string there is superposed on the large hump caused by the initial blow, kinks due to disturbances that overtake the hammer from the remote end of the string. These are satisfactorily accounted for in GHOSH's theory.

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DAVY, LITTLEWOOD and MCCAY [39] have been able to measure the actual force between hammer and string from instant to instant during the short time of contact. A knowledge of the actual instantaneous acceleration of the hammer is of importance for the theory of the struck string, though it is usual among theorists to make an *ad hoc* assumption with regard to this aspect of the problem and to see how it checks with experimental observations on the subsequent motion of the string. A piezoelectric quartz pressure gauge was placed in the hammer head and the instantaneous force upon it observed in a cathode ray oscillograph with linear time-base. The areas under the oscillograms are proportional to the corresponding changes in momentum of the hammer so that their ordinates can be taken as proportional to the instantaneous values of the force of contact. The work falls into line with that of GHOSH just described, for the resulting force: time curve takes the shape of a hump with a number of overtones or kinks superposed as a ripple upon it.

As soon as we envisage elasticity in the hammer as well as in the string, we are dealing with a coupled system of two components each having natural frequencies which can react on each other. This suggests an alternative way of considering the mechanics of the system, treating it as KOCK [40] has done by analogy with the electric line

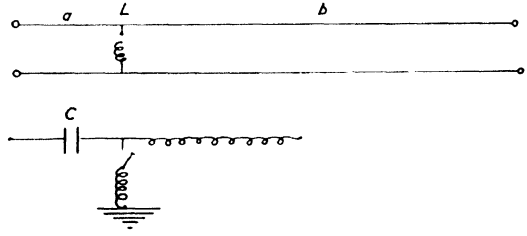


Fig. 266. Struck string as electric line.

(Fig. 266). The line having a length $a+b$ is connected to earth at each end through a condenser C, which represents the friction of the bridge on the sound board. The hammer is represented by an inductive switch L, near to one end and distant a from it. When the circuit is made, a 'potential' Li is introduced which sends waves to either end. The contact must be broken before the reflection of the principal wave from the remote end, otherwise energy will be returned to the exciter,

as to a transformer coupled to an electric line. If l and c represent the inertance and capacitance per unit length of the string, the wave velocity is $(lc)^{-\frac{1}{2}}$ and the time out-and-home of the wave is $2b(lc)^{-\frac{1}{2}}$. which can be set equal to the natural frequency of the interrupter and the nearby condenser, *i.e.* to $\pi\sqrt{LC}$.

The impedance of the short line can also be expressed, as for a cable, as $(l/c)^{\frac{1}{2}}$ cot $\pi a/x$; so that the capacity C which represents the roughness of the bridge becomes xc/π. tan $\pi a/x$.

Introducing this value of C into the equation for the time of contact, KOCK obtains

$$\frac{L}{\pi}\left(1-\frac{a}{x}\right)^2 \cot\frac{\pi x}{a} = \frac{L}{lx} = \frac{M}{mx} \qquad \text{(VI, 8)}$$

on inserting the mass of the hammer M and that of the unit length of the wire m, as the respective inertances. This equation gives the optimum point of contact as the ratio $a/(a+b)$ for given mass ratio and agrees well with the experiments of GEORGE. The function is plotted according to equation (8) on Fig. 267.

GEORGE'S [34] experimental results may be summarised as follows:-

(1) The amplitude of the fundamental is greater when the string is struck near one end and when the relative mass of the hammer is great.

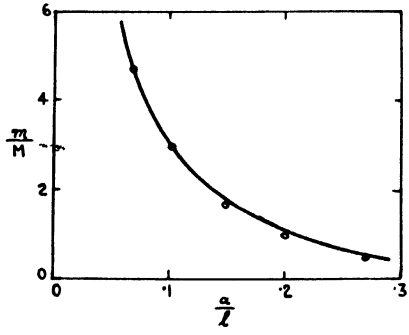


Fig. 267. Parameters of struck string.

(2) This amplitude varies in a discontinuous fashion, showing maxima and minima as the hammer is moved towards the centre, and the number of maxima and minima becomes less as the mass of the hammer is reduced, until there is only one maximum and two minima.

(3) Early workers in the subject thought it necessary to discriminate between a hard and a felt-covered hammer, but it appears that the distinction is merely one of the finite length of the string struck, similar to that between a narrow and a wide bow.

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§ 11. Timbre and Touch on Pianoforte

Dealing with the coupling between strings and soundboard on the pianoforte, VIERLING [41] finds that the coupling between an individual string and the board on which it is stretched is less important for the damping than that between the two or three strings which are simultaneously struck by an individual hammer and which are, of course, nominally though never quite actually, in unison. From the point of view of tone quality, it is not desirable that this 'trichord' should be exactly in tune, but the beating between them resulting from the mistuning, though it brightens the tone, enhances the rate of attenuation of intensity.

Two decades ago the Physical Society [42] held a discussion on that elusive factor in pianoforte technique called 'touch'. The opinion prevailing then among the disputants was that it was a fetish set up by the virtuosi for their public to worship. This view has received conclusive confirmation from the work of HART, FULLER and LUSBY [43] at the University of Pennsylvania. They made oscillograph records of the notes produced by two American virtuosi, and a set of records of the notes produced by a pendulum which on release hit the same piano key. In every case it was possible to get an exact match of one of the human records with one of the robot's. The inference is, in the words of a speaker at the 1913 discussion; 'The only factor that a player can vary is the speed with which the hammer hits the string.'

Even for a change of velocity of impact of the pianoforte hammer by 35 per cent, hardly any change in the tone quality could be detected.

R. N. GHOSH's [36] experimental results confirm this view and further lead to the conclusion that the vibration curve of the string is independent of the velocity of impact of the felt hammer. Whatever be the velocity of the impact, the time: displacement curve of any point of the string has been found to possess the same shape and form. The velocity of impact was about 100 cm/sec. or more and different velocities were arbitrarily imparted (very small velocities, however, being avoided), but the vibration curve was found to preserve in all cases its one definite characteristic shape and form.

The change of tone quality in aerial vibrations with loudness in the case of the pianoforte must therefore be ascribed to some other causes. If we remember that in the case of the pianoforte the free

vibrations of the soundboard play an important part in controlling the aerial tone quality, it is easy to explain these changes of the tone quality. The greater the velocity of impact the larger will be the amplitude of free vibrations of the soundboard. Other strings of the pianoforte remain in contact with the damper, hence they are not excited at all.

Finally we record the observations of SCHUCK and YOUNG [44] on the decay of amplitude and the frequencies of partial tones. After the initial peak of intensity at striking there is a fairly uniform rate of decrease, more rapid the higher the order of the partial, due to the increased air loading of string and soundboard at the higher frequencies. This leads to a simplification of the timbre as time goes on. The frequencies of these partial tones were determined by a vibration pick-up fixed to a rib of the soundboard and feeding a wave-analyser. The partials were found to divaricate from a true harmonic series to an extent roughly proportional to the square of the order – cf. Fig. 268 which shows this effect on the F strings, (F_1 to F_5).

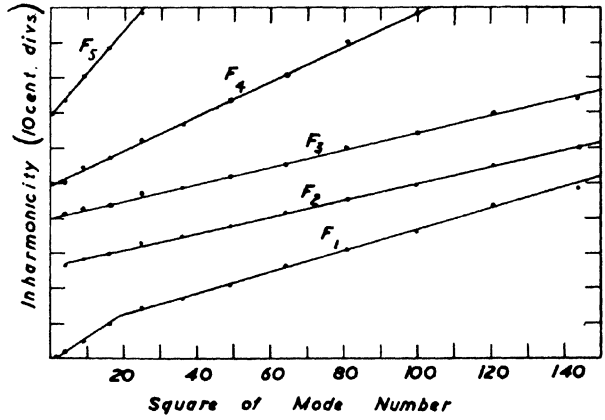


Fig. 268. Mistuning of piano strings.

§ 12. Pianoforte Tuning

Measurements by RAILSBACK [45] show that piano tuners tune progressively sharp in the upper registers, flat in the low.

The process of piano tuning is carried out by 'laying the tempera-

ment' in an octave in the middle of the keyboard—usually the octave from F_3 to F_4—and then tuning up and down by octaves. To tune upward by an octave, one adjusts the tension of the upper string until its fundamental has the same frequency as the second mode of

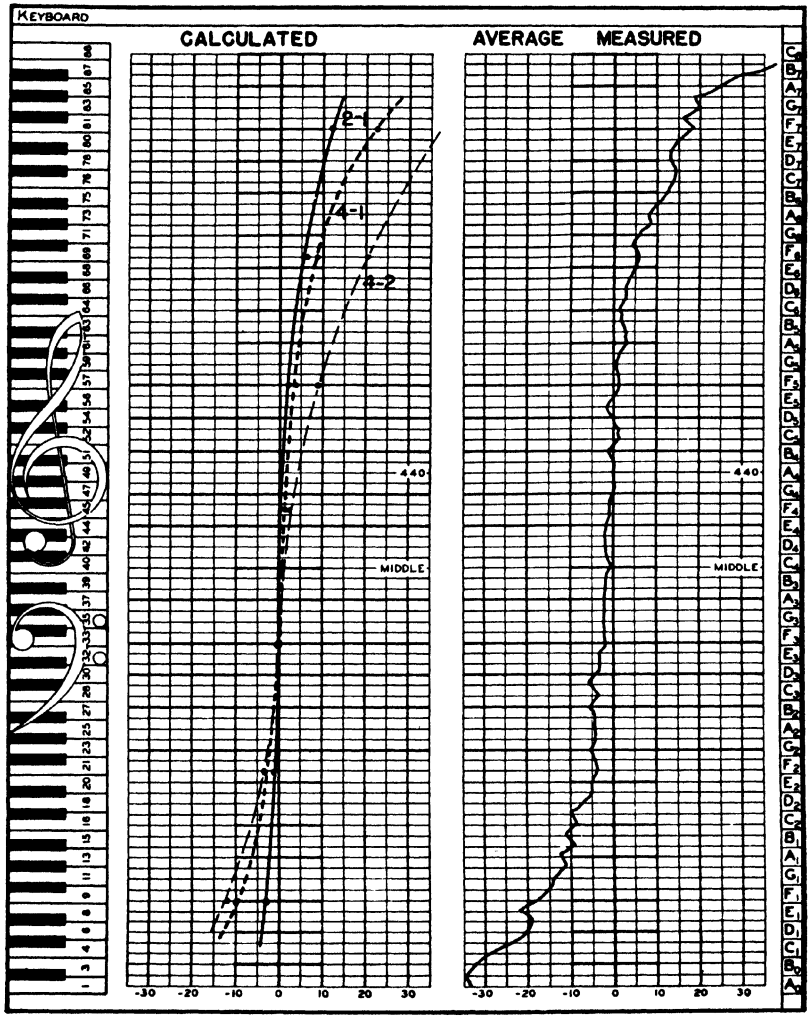


Fig. 269. Tuning of piano, theoretical to left, actual to right; abscissae, deviations in Cents (2–1 octave of one note of against fundamental of other, 4–1 double octave of one against fundamental of other; 4–2, double octave of one against octave of other).

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the lower string, as indicated by the absence of beats. If the second mode of the lower string is sharper than a true second harmonic of the fundamental, then the upper string's fundamental will be tuned sharp. The next octave jump will increase the sharpening still further, particularly since the inharmonicity increases toward the treble end, as shown in Fig. 269.

A tuner often checks his tuning by double octaves, that is, by the beating of the fourth partial of the lower tone against the fundamental of the upper tone. If this were done as a tuning procedure, the resulting tuning curve would be that labelled (4-1) in Fig. 269 which is seen to fall between the curves. Which of these methods of tuning is actually used and to which section of the keyboard it is applied appears to depend upon the individual tuner and the particular piano he is tuning.

In laying the temperament initially the tuning is also affected somewhat by the sharpening of the partials. The tuner adjusts the fundamental of one string to his tuning fork, then adjusts another string to the first by listening for beats between partials of the same approximate frequency. He adjusts the beats to be in accordance with a table calculated on the assumption that the partials are truly harmonic.

The measurements described here indicate that the 'stretched' tuning is a natural result of the inharmonicity of the partials of the piano and is perhaps necessary if the piano is to sound in tune with itself. Whether or not stretching is desirable may therefore depend upon the size and construction of the piano and whether it is to be used alone or with other instruments.

§ 13. Plucked String – Harp and Banjo

This, the simplest of all methods of producing sound from a string, has not been much studied by physicists. It is the simplest because in instruments like the clavichord, harp and banjo, the string starts its vibration from rest in the position to which it has been displaced; by the finger in most cases, by a piece of metal called 'tangent' on the clavichord, by a piece of horn called 'plectrum' on the banjo. Subsequent wave-form analyses prove to have the same characteristics as the bowed string, *i.e.*, the number and magnitude of the partials depends on the location and width of the displaced region.

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§ 14. Bars

Bars in transverse vibration are found rather sparingly among orchestral instruments probably because the overtones are inharmonic unless the bar is supported at two points (Chap. 1, § 10). As such we find them in the xylophone where small pieces of wood are lightly held by nails passing through two larger holes near one end. The inharmonic tones of the clamped bar can be discouraged by coupling it to a tuned resonator in the form of an air cavity which has harmonic overtones, so that the two components synchronise only in the fundamental. An example is the cembalo or celeste, a keyboard instrument of little bars mounted in front of resonators. But the same idea is adopted even with the supported bars of the xylophone to increase the intensity and similar instruments with gourds for resonators are found as music-makers among primitive tribes.

The reed which is so often found on wind instruments (p. 503) is acoustically a clamped bar whose vibrations are subject to an air column.

§ 15. Drums

A membrane stretched over a circular frame is the genesis of a number of instruments like the tambourine. More often the membrane is coupled to a resonant cavity as in the tympanum or two skins face each other across an intervening cylindrical air space as in the bass drum.

OBATA and TESIMA [46] have studied the modes of vibration of the latter type, pasting on the parchment strips of tinfoil which with a fixed metal plate formed a condenser microphone. So the relative amplitudes of different portions of the skin could be determined. On striking one skin the vibrations of both skins (coupled through the air inside the drum) were recorded on film.

Besides making records of the vibrations of the twin membranes of a cylindrical drum, they recorded the sound waves radiated by such an instrument. The point of greatest physical interest in their results concerns the relation between the vibrations of the skins (which should be tuned to the same pitch) at each end of the cylindrical column of air inside the drum. The air column couples the membranes in vibration; closely in a shallow drum; less tightly as the axis of

the cylinder is lengthened. The interesting phenomenon of the intermittent action of two equal coupled pendulums is found again here in the alternate rise and fall of the vibration of each membrane, the energy being bandied about between them so that first one and then the other is at its maximum amplitude. Lengthening the cylinder, besides reducing the coupling, pulls the fundamental of the skins down—possibly by increasing the damping—though it appears not to affect the overtones. No suggestion is made to account for this curious observation.

The choice of striking point effects the tone quality. It is the practice to hit the skin at a point one quarter to one third of the way across a diameter, as this encourages the fundamental of the skin. The pitch of the latter is however affected by the length of the cylindrical column, as one would expect from the analysis of the action already considered.

GHOSH [47] has drawn attention to certain Indian drums, in which, owing to progressive decrease of the mass per unit area of the membrane from circumference to centre, the partial tones are made to approximate to a harmonic series. RAYLEIGH [48] attributes to the drum proper another function besides resonance; it serves to spread its sound more uniformly, whereas an unassisted membrane would concentrate the sound in the direction of the vibration.

§ 16. Plates

Owing to their inharmonic overtones, plates are more often met as rhythm markers than as music makers. Such are the cymbals, gongs and the like. The commonest method of practical investigation of their modes of vibration is still the strewn sand of CHLADNI, but electrical pick-ups may also be used (*cf.* drums).

Some work has been done by WOOD and SMITH [49] on the natural frequencies of circular plates and sheets of various materials. The method of excitation was in most cases by direct contact with nickel rods of appropriate frequency, set in magnetostrictive oscillation, while to study the mode of vibration the CHLADNI method of sprinkled sand was applied. The small circular plates were tested at the natural frequency corresponding to one nodal circle. According to POISSON, the radius r of this circle should be related to that of a thin plate,

free at the edge, by the simple formula $r = ka$, where k is a constant of value 0.678. WOOD [50] does in fact find that this formula holds over quite a range of thickness and diameter. Only when the ratio of thickness to diameter reached one-fifth was there appreciable departure in the direction of decreasing k; at this stage the frequency also began to fall below that proper to a thin plate. This same method of excitation by contact with a magnetostrictive oscillator and making a CHLADNI plate of the substance has led to a determination of the velocity of transverse waves in a number of commercial materials, both alloys and insulators, to which the formula tested in the first research can be applied. The method is at its best in non-resonant materials such as cardboard, to which other ways cannot readily be devoted.

§ 17. Bells

Just as the tuning-fork may be regarded as a bar bent at the centre until the two blades are parallel, so a circular plate held by the rim while its centre is pushed in until it assumes a spheroidal shape may be regarded as the genesis of the bell.

Simple bells may consist of a hollow hemisphere of metal struck near the rim or, in the orchestral bell-tone producer or glockenspiel, of cylindrical tubes hung from a string near one end and hit with a clapper at the same place.

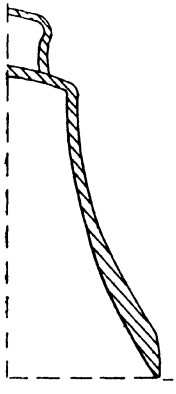


Fig. 270. Section of church bell.

In the form which by long association has been known as the church bell there are added complications; the section turns outward just above the rim; also the thickness is not constant, being a maximum at the inflection zone just mentioned—called the sound-bow—and tapering to an edge at the rim (Fig. 270).

The clapper strikes the bell at the sound-bow. Partial tones arise when the section has nodal lines of latitude and longitude set up on it. These would be inharmonic if the thickness were uniform, but one of the objects of the bell-founder in varying this thickness is to force them into an artificial harmonic series. Table 1 (after TABER

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JONES) shows how nearly this was accomplished in two chimes which he examined.

TABLE I

Partial	Nodal Meridians	Nodal Circles	Ideal Ratios	Actual Ratios	
				Carlile Chime	Harkness Chime
1	4	0	0.50	0.58	0.50
2	4	1	1.00	0.92	1.00
3	6	1	1.20	1.19	1.20
4	6	1	1.50	1.71	1.49
5	8	1	2.00 *	2.00 *	2.00 *
6	8	2	2.50	2.74	2.56
7	10	2	3.00	3.00	2.96
8	8		3.33		3.28
9	10		3.75	3.77	3.80
10	12		4.00	4.07	4.06

* Taken as reference standard.

As with the pianoforte the partial tones have different rates of attenuation (*vide infra*).

Further work on bells has been done by JONES and ALDERMAN [51] and by CURTISS and GIANNINI [52]. The former have determined the acoustic spectrum of a bell at various epochs after striking. Since the bell is theoretically a struck anisotropic plate we find analogies with the struck string in the rate of decay of partials. The variation in this respect is one of the characteristics of bell tone. The latter workers have succeeded in examining the nodal lines by the method of TYZZER [53], *viz.* excitation of each partial separately by means of a valve oscillator and loudspeaker and exploration of the vibrating surface with a sort of stethoscope. Whereas the fifth partial is the

TABLE II

Relative amplitudes of bell overtones at various epoches after striking (JONES AND ALDERMAN)

Order of partial	1	2	3	4	5	7
At strike	1	0.7	2.4	0.7	4	2.8
After ½ sec	1	0.6	1.8	—	0.8	0.75
„ 1 sec	1	0.6	1.8	—	0.1	0.4
„ 2 sec	1	0.5	1.1	—	0.1	—
„ 3 sec	1	0.4	1	—	—	—
„ 7 sec	1	0.4	0.8	—	—	—
„ 12 sec	1	0.2	0.4	—	—	—

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strongest immediately after striking, it practically disappears after 1 sec., leaving the third partial master of the field. In 12 sec. this too has nearly fallen below audibility, leaving the fundamental which shows the least attenuation of all.

This is for a tenor bell of 345 c/sec. In the treble the bells have a thicker waist (zone of minimum thickness of metal) and the attenuation of the partials is more nearly uniform. GIANNINI [54] thinks however that the large bells with finer waists which have the acoustic properties outlined above have a better and more characteristic bell tone, and that this pattern, with a thick sound-bow and thin waist, should be adopted throughout the peal. Carillons can further be improved by providing each bell with a damper, which is pulled off for a certain time as each key is pressed, like those of a piano. The time allowed for natural attenuation while the damper is off should not, however, be constant for all, but should be graded from treble to bass.

OBATA and TESIMA [54], have investigated the acoustics of Japanese hanging bells, whose tone quality differs materially from that of Western bells. They consist of a deep cylindrical shell ending in a hemispherical cap and are more uniform in thickness. While the inside is smooth, the outside is heavily embossed and includes striking points in the form of the sacred Buddhist symbol – a lotus flower. The external striker is a heavy piece of wood. The lower components have frequencies in the ratios 2 : 5 : 7 : 9 : 11, but many of them are not pure tones. For instance the 'fundamental' consisted of two tones 246 and 248 producing a very obvious short beat. The authors attempt to correlate the frequencies with the possible positions of nodal lines of latitude and longitude.

Another suggestive idea in the practice of carillon ringing is due to CURTISS and GIANNINI [52]. The main objections to large bells in a carillon are the cost of the bells themselves and the heavy tower to hang them in, together with non-uniformity of quality throughout the peal. The new idea is to use small bells, 3 to 15 in. high and to amplify them on a loudspeaker of some 200 watts. Tests with a number of small bells so amplified to form a peal are made possible owing to the low initial cost of the bells.

GIANNINI [55] continues his now extensive work on bells by directing attention to the importance of the position in a campanile at which the members of a carillon are hung. Contrary to the usual practice,

the smaller bells should be nearer the ground, as the directive characteristics of radiation from a source become more pronounced at higher frequencies and the polarity, in a vertical plane, of the radiation curve for a small bell hung high above the ground is such that the sound misses a listener at ground level. The general tendency to build tall towers should be checked and a return to the lower levels of the famous Low Country carillons should be made.

VAN HEUVEN [56] has reached the following conclusion from his experiments:-

'If the mass of the bells is plotted against the diameter of the sound-bow a large degree of uniformity is shown. This means that the partial frequencies of all bells are arranged relative to each other according to a standard pattern as follows; (1) Hum-note; (2) Fundamental (one octave above 1); (3) Minor Third (a minor third above 2); (4) Fifth (a fifth above 2); (5) Nominal (one octave above 2); (6) Harmonic Decime (one octave + a major third above 2), etc.

A graph of the hum-note frequency plotted against the diameter of the sound-bow shows larger deviations from the average relation than would be expected in accordance with the uniformity in profile shape. Thus small deviations of the profile shape cause large differences in the natural frequencies; it is therefore not possible to make well-tuned bells by casting only; the tuning must be completed by turning the bell on a lathe whilst controlling the partial frequencies.'

Steel bells show a systematically larger decrement (shorter reverberation time) than bronze bells of the same pitch. The internal damping of steel is the same or even less than that of bell-bronze. This discrepancy can be explained by the larger acoustical damping of steel bells, of which the dimensions are about 50% larger than for bronze bells of the same pitch. Attempts to imitate bell-tone by gongs made of steel wires bent into spirals and struck at their free ends have been made by clockmakers with a certain amount of success. Investigations have been made by POMP and ZAPP [57] to determine what is the best type of metal for the purpose. Carbon steel (0.63 per cent) tempered for 15 minutes at 300°C. gave the maximum duration of note.

§ 18. The Strike Note

The 'strike note' of a church bell is the tone which is most prominent

when several bells sound in succession, and it is this tone that is meant when the pitch of a bell is given. To the ear this prominent tone is about one octave below the fifth partial. In spite of study by various experimenters the question as to how the strike note is produced is not yet settled.

On several bells RAYLEIGH [58] found that the pitch ascribed to the bell was not given by any normal mode of vibration of the bell, and that this could not be reinforced by a resonator. In each case he found that the strike note was about an octave below the fifth partial tone of the bell, and on a bell which he was able to examine with some care he found that this fifth partial is heard when the bell is struck on the sound-bow, but soon falls off when the locality of the blow is varied, and in the upper three-fourths of the bell is very faint.

MEYER and KLAES [59] tested whether the strike note is of subjective origin – that is, whether the pitch of the strike note is determined by the fifth partial, but for some reason is judged to be in the lower octave. For this purpose the sound from the bell was picked up, amplified and sent to a loudspeaker by means of a system that was very free from distortion. Electric filters were employed to cut out chosen components of the sound. When only the first five partial tones from the bell were transmitted the strike note was not heard, but when the seventh was also transmitted the strike note was heard clearly. This experiment seems to show that the seventh partial in addition to the fifth plays an important part in the production of the strike note.

At one time they picked up the sound from the bell with a condenser transmitter. The response of the condenser transmitter was linear and the analysis showed no trace of a component at the pitch of the strike note. At another time they made use of a carbon microphone with a response curve somewhat similar to that of the human ear and in this case a tone of considerable intensity was found at about 533 cycles/sec – that is, at about the pitch of the strike note.

Moreover, it is well known that a simple tone may often be judged an octave higher or lower than the pitch given by its actual frequency. Thus it is possible for a difference tone from the fifth and seventh partials of a bell, introduced by the non-linear response of the human ear and lying in the neighbourhood of an octave below the fifth partial, to be judged as that of a tone an octave lower. This at least seems at present to be the best hypothesis as to the origin of the strike note.

References p. 525-529

ARTS [60] opines that the fifth partial suggests the strike note to the ear. He sometimes observes another subjective tone 'which has the peculiarity, (a) it is very modest in its appearance, (b) it does not react upon a tuning fork. It seems to be related to the third partial, since it has the same decay time'.

He also calls attention to a 'secondary strike note'. This is a tone which he localises at one fourth above the fifth partial tone of the bell and which starting very loudly performs a rapid diminuendo. Eight seconds was the maximum time of waning that he has noted. This tone appears to correspond to a mode of vibration which has a nodal circle about halfway between the rim and the shoulder of the bell.

§ 19. Wind Tones

Among primary sound producers unassisted by resonators we name:
(i) aeolian tones,
(ii) edge tones,
(iii) wind-maintained reeds.

(i) *Aeolian Tones*

The aeolian phenomenon is known to us in the 'singing' of telegraph wires, of the elastic cord stretched across the bars of a moving bicycle, and in the side movement of an oar drawn through water.

The first quantitative investigation of the dependence of pitch on wind-velocity was made by STROUHAL [61] in 1879. The apparatus that he used was designed to whirl a wire rapidly through the air in a direction at right-angles to its length; it was attached to the ends of two arms, the other ends of which were fixed to an axle, which could be turned (through gearing) by hand. It was possible so to turn the axle that the note remained constant long enough for the pitch of it to be estimated on a sonometer. The facts that emerged from the results were that the frequency f (1) was independent of the length or of the tension of the wire, (2) depended directly on the velocity V (3) depended inversely on the diameter of the wire d.

This paper remained the only contribution to the study of the subject until 1914, when these results were taken up again by KRÜGER and LAUTH [62], and analysed in the light of some discoveries of BÉNARD [63], who, examining the wake behind an obstacle placed in

running water, found two parallel rows of alternate and equally spaced vortices formed in the liquid, one each side of the obstacle. The production of these vortex sheets these authors considered to be the source of the sound, and they further pointed out that when the frequency of the vortex production coincides with the natural frequency of the body behind which they are formed, the latter, if free to move, will be set into forced vibration. RUBACH found from photographs of the wake of a cylinder in water that h/l was a constant, where h is the width between the two rows of vortices, and l the length between two successive ones in the same row. Also l was approximately proportional to, and greater than, d, so that $l = bd$, say; the velocity U of the vortex system relative to the (stationary) fluid was less than, but proportional to, V, *i.e.* $U = aV$. Consideration of these facts led KRÜGER and LAUTH to a theoretical basis for STROUHAL'S formula. The frequency of the vibration represents the number of vortices formed on one side of the wire in one second; $l/(V-U)$ is the time between the disengagement of two successive vortices from the same side of the body, *i.e.* the period of swing.

$$\frac{V}{fd} = \frac{V}{d} \cdot \frac{l}{V-U} = \frac{b}{1-a} \qquad \text{(VI, 9)}$$

which is a constant. From VON KÁRMÁN and RUBACH'S[64] results for a and b KRÜGER and LAUTH obtained the value 5.0 for this constant, which is in fair agreement with STROUHAL'S numbers, which lie between 6.3 and 4.9.

As mentioned earlier, STROUHAL and others have held that the tones given out are produced in the air, *i.e.*, by the mere formation of a regular series of vortices, but that when the frequency of their formation coincides with that of the wire, the latter 'chimes in' resonantly causing the sound to be amplified. The so-called aeolian tones are sounds produced by the wire itself, though the vortex-formation is the driving force which keeps the wire in vibration.

In a set of experiments by the author[65] the wires were mounted vertically in a wind-tunnel and their vibrations observed visually through a window. If thicker wires are used they must be loose to produce an audible tone, and then the tones given are weak, high harmonics, so that it is impossible to compare them with the sonometer.

References p. 525–52°

The STROUHAL parameter V/fd reaches a constant value of 5 at large values of d.

HOLLE [66] has made a painstaking research of aeolian tones, paying particular attention to the quality and intensity of the sound. The latter is, as we should expect, a linear function of the wind velocity, while spectrum analyses of the note given out show it to be remarkably free from overtones.

(ii) *Edge Tones*

When a fluid debouches from a slit into free space, surfaces of discontinuity as HELMHOLTZ called them are formed along the confines of the spreading jet where it mixes with erstwhile stagnant fluid. Although rotations and wave motions arise along these surfaces there is not the tendency to roll up into discrete vortices which the corresponding wake of an obstacle shows, unless a sharp edge is placed to bisect the emerging jet. In a way that is still rather imperfectly understood, but is believed to be connected with the secondary vortex system which is set up by friction along the wedge and the relationship of this to the system first-named, a steady formation of vortices takes place at the slit so that the wave-length of the system—distance between successive vortices on the same side—is equal to or is a small submultiple of the separation of slit to edge, l.

Taking the simplest case, a vortex A (Fig. 271a) leaves the outer wall of the orifice as the preceding one on the same side B strikes the edge.

There is a minimum distance l_0 for any given velocity of efflux V at which a tone can be produced. When the separation between the edge and slit is a little greater than l_0 the state of affairs is as shown in Fig. 271a. The vortex B has just struck the edge as the next one on the same side is emerging from the slit. The frequency of the 'edge tone' is given by the frequency with which the eddies strike the edge. If they move towards it with velocity U and λ = distance between the vortices in the same row, then $f = U/\lambda$ or since $\lambda = l_0$ and $U = aV$, where a is constant

$$V/fl_0 \text{ is constant} \qquad \text{(VI, 10)}.$$

When V is kept constant, *i.e.*, when constant pressure is maintained behind the slit, as l is increased beyond the minimum l_0, the pitch

of this 'edge tone' falls in accordance with this relation until at a value l approximately double of l_0, the system becomes unstable, and the tone, which is now the sub-octave of the original, may suddenly rise an octave to what it was at l_0.

Experiments in which smoke is mixed with the emerging air shows that the original spacing of the vortices is restored after the transition, but with twice as many between slit and edge. Fig. 271b shows their relative positions just before the transition. λ has doubled itself with l, and therefore the width of the system in accordance with the stability criteria is also double what it was in Fig. 271a. After the jump (Fig. 271c) λ resumes its original value so that $l = 2\lambda$, and, in agreement with the VON KÁRMÁN stability criterion h is halved, and the original narrow vortex street is recovered. The edge must bisect the two rows of vortices if the tone is to be elicited; consequently, in the shaded area

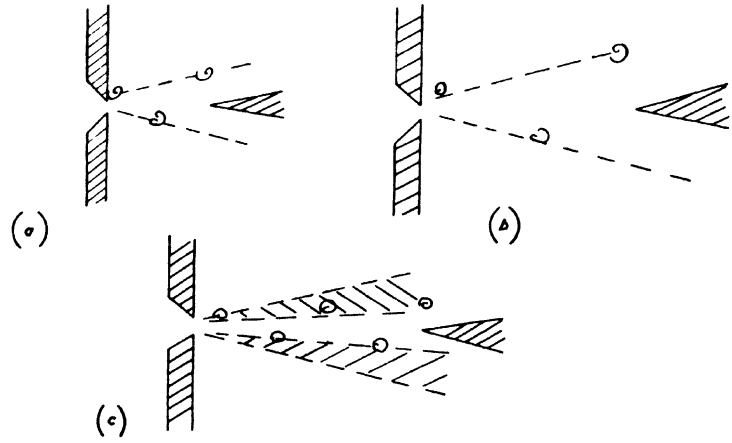


Fig. 271. Edge-tone formation.

of Fig. 271c, representing the space between the wide and the narrow street at the transition, only the graver tone can be elicited. In fact after the jump has taken place, the pre-transition tone can be brought back either (1) by moving the edge into the shaded region, or (2) by pushing an obstacle from the side partly into the path of the blast in Fig. 271c, and so deflecting the shaded portion of the stream on to the edge.

The writer [67] (among others) has undertaken a determination of

the edge tone characteristics and so has G. B. BROWN [68]. BROWN has further repeated the standard experiments on the variation of frequency with velocity and distance from slit to edge. He denies that λ the interspacing is equal to or a sub-multiple of l, the perhaps too simple assumption that lies at the basis of most edge tone theories. Over a considerable range of λ, however, two simultaneous frequencies are produced and he proposes the following empirical formula:

$$l = 0.04j\,(V - 40)(\lambda^{-1} - 0.07) \qquad \text{(VI, 11)}$$

where $j = 1, 2.3, 3.8, 5.4$ for the first four stages of subdivision of the slit edge space into one or more wave-lengths.

In the second part of his work he examines the more plausible theories which have been put forward to explain the mechanism of edge tone production, *viz.* (1) the VON KÁRMÁN vortex street theory, which explains the disposition of the vortices in terms of the stability of the system, but does not properly account for the intervention of l in the expression for the frequency; (2) the sound pulse theory, by which each vortex on hitting the edge starts off another from the slit; this, incidentally, requires integral values of j; (3) the espacement theory, whereby any casual deviation of the jet from the forthright direction sets up a pressure increase in the surrounding air on one side of the wedge which, when it has grown to a sufficient magnitude, forces the jet back towards the other side, so making it pendulate. BROWN himself favours the third theory with some modifications, but admits that more must be learnt of the properties of fluid jets before one can hope to evolve a quantitative theory of edge tones.

Another investigation on jet tones, which can be regarded as an intermediate case between the aeolian and the edge tone, is to be found in a paper by CARRIÈRE [69], who placed a wire about 2 mm in diameter parallel to the slit from which an *ignited* gas jet was issuing. He found an approximate relation between the pitch of the tone produced, the velocity of efflux (or more precisely the square root of the gas pressure) and the distance between slit and wire, similar to that common to edge tones. The experiment is suggestive in respect of the escapement theory which BROWN endorses. If the wedge against which the pendulation is set up can be replaced by a thin wire without upsetting the mechanism of the edge tone, it is difficult to imagine that a difference of pressure could be set up between either side when

there is free communication between the two sides of the jet by way of the far side of the wire. CARRIÈRE whose work has been repeated by COOP [70] found that the wire itself vibrated if it happened to be in resonance with the vortex system. The matter is deserving of further research without the unnecessary complication of the flame, and some research on this point has been done by LENIHAN and the author [71].

TABER JONES [72] has more recently re-examined determinations of the constant. He would make BROWN's constant about twice as large.

(iii) *Reed Action*

An independent reed can be maintained by a blast of air and, as such, is common in reed organs, accordions and the like.

Fig. 272a shows the 'free' reed with (hatched) the tongue vibrating in and out of the slit. Fig. 272b shows the beating reed in which the motion of the tongue is stopped by the confines of the slit (known as

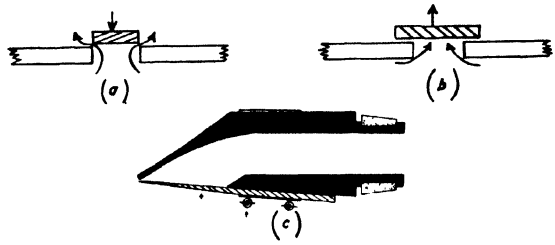


Fig. 272. Reed types/ (a) free; (b) beating; (c) clarinet.

the 'table') which it is too wide to enter. The accordion and harmonium usually have free reeds, the behaviour of which has been described by TABER JONES as follows:

When the tongue swings into the slot the inertia of the air prevents an abrupt cutting off of the flow. At the sides of the tongue the flow drops off very quickly, but just above it and just below it the surge continues temporarily, so that the pressure above the tongue is increased and that below is decreased. The resulting downward force comes at a time when the tongue is moving downward, and aids the downward motion.

When the tongue moves upward and opens the slot the inertia of

the air prevents the flow from being established immediately. During this interval of slower flow the reduction of pressure below the tongue is less than it is after the flow is well established. Thus the pressure effect which is superposed on the reduction of pressure below the tongue is now an increase of pressure below, and so aids the upward motion.

The 'beating reed' is used on a number of wind instruments, such as the clarinet (*q.v.*). McGINNIS and GALLAGHER [73] have fixed a clarinet embouchure (Fig. 272c) in a glass case in such a fashion that when it is blown by a stream of air passing through the case into the mouthpiece the motion of the reed induced by the air pressure can be projected onto a screen. They found that for about three-fourths of the natural period of the reed it adhered to the 'table' closing almost completely the air entry. During the remaining quarter period, the reed suddenly left the table to its maximum amplitude and as suddenly returned. It appears therefore that air passes the reed only for a fraction, less than one quarter of the cycle.

§ 20. Types of Air Columns, commonly coupled to Primary Sound Producers

By adopting ideas from alternating-current theory, the acoustician has been able to solve a number of analogous problems in sound wherever the acoustic elements could be grouped into 'inertances' and 'capacitances' and built up into circuits whose electrical counterparts had already been solved or were amenable to solution. In the case of an acoustic radiator whose frequency is to be varied over a certain range, the system is assumed to be driven at resonance and the damping may be neglected. In most wind instruments the variation is a step-by-step one, each step being a semitone.

The problem is then usually set in the form: What adjustments of impedance of the component parts are necessary for this variation? and of the alternatives, if any, which ones will allow the player to progress with the greatest facility from one note to another, For those who are familiar with loudspeaker practice, and particularly that of the exponential horn, it is important to realise that the function of a wind instrument is not primarily to radiate sound in a specified direction, but to produce a note of frequency determined by the

resonant state of the system of impedances of which it is made up. Though exponential horns have been employed on the brass-wind, they cannot be given too pronounced a flare, or their resonance peaks will overlap so much that the player is sore put to it to strike one note without warbling into another—a state of affairs which is closely approached on the modern French horn, to the anxiety of the tyro!

Though the notes on such an instrument as the horn are restricted to the harmonic series based on a few set speaking lengths, the wood-wind operate on a different principle. This is to make use of side holes, whose opening and closing varies the arrangement and value of the impedances so linked in the instruments. A simple wind instrument like the flageolet is, in fact, the acoustical analogue of a short artificial electric line, with the side-holes set as earthed branches to the main conduit. The 'conductivity' (or 'admittance', to use a more modern term) of these orifices can be nullified by covering with a finger or key – or reduced by half-covering, a practice adopted by flautists of the older schools. The flute then is the analogue of the cable (the air in the tube) loaded by inductances (the side-holes). Thus, if all holes are covered except one, the system may be treated as a cable (up to the site of the open hole) terminated by an inductance and another length of cable in parallel, and the net impedance equated to zero for resonance. Suitably placed vent-holes of smaller diameter than the note-holes assist the player to produce harmonics. When several holes, remote from the mouthpiece, are open, the contribution of those lower down may be neglected, but not that of those near the first open hole. Thus the note formed by covering the holes, 1, 2, 3 and 5 may be intermediate in pitch between that fingered 1, 2, 3, 4 and that of 1, 2, 3, 4, 5.

§ 21. Straight Tubes

We shall exemplify these applications of impedance principles in a few simple cases. The impedance of a column has already been calculated (p. 13). We shall write it thus:

$$Z_l = \frac{Z_0 - \frac{i\rho c}{S}\tan \beta l}{\frac{SZ_0}{i\rho c}\tan \beta l + 1} \qquad \text{(VI, 12)}$$

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An isolated orifice impedance Z'_0 follows if we make Z_l in (VI, 12) zero and l so short that $\tan \beta l \approx \beta l$:-

$$Z_0' = \frac{i\rho c}{S}\beta l = \frac{i\rho\omega}{\kappa} \qquad \text{(VI, 13)}$$

when κ is the 'conductivity' of the orifice.

In Fig. 273a a pipe closed or open at one end and partially stopped at the other is represented. The resonant frequencies are obtained by substituting (VI, 13) for Z_0 in (VI, 12) and setting Z_l (a) infinite or (b) zero, respectively. Then

$$\text{(a) } \tan \beta l = \frac{\kappa}{\beta S}; \qquad \text{(b) } \cot \beta l = -\frac{\kappa}{\beta S} \qquad \text{(VI, 14)}$$

These equations can be solved graphically by plotting $\tan \beta l$ (or $-\cot \beta l$) and $\kappa/\beta S$ severally against $(\beta l)^{-1}$. The intersections give the resonant frequencies for a given length of tube and conductivity of orifice.

§ 22. Tube with Side Holes

Suppose a volume displacement X_B arrives from the tube at the hole 2 (Fig. 273b) dividing there into X_2 through the hole and X_C down

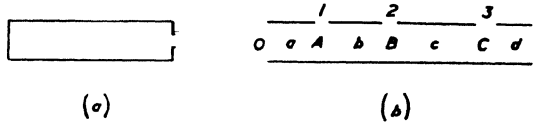


Fig. 273. Air columns with orifices (a) terminal; (b) side.

the remainder of the tube. We suppose with RAYLEIGH that the pressure amplitude at B acts equally on the impedances Z_B, Z_2 and Z_C of these respective components, so that

$$\frac{1}{Z_B} = \frac{1}{Z_2} + \frac{1}{Z_C}$$

similarly

$$\frac{1}{Z_A} = \frac{1}{Z_1} + \frac{1}{Z_B}$$

Finally at the entry of the tube the impedance Z_0 is given by

$$\frac{1}{Z_A} = \frac{1 + \dfrac{SZ_0}{i\rho c}\tan \beta a}{Z_0 - \dfrac{i\rho c}{S}\tan \beta a} \qquad \text{(VI, 15)}$$

For resonance the entry impedance will be made 0 (case c) or ∞ (case d) as before.

When an instrument has only one side hole open (or when more are open if we may ignore the effect of the open holes remote from the second opening, an approximation often sufficient in practice) the resonant solutions reduce to:

$$-\frac{\kappa}{\beta S} = \cot \beta b + \cot \beta a \qquad \text{(VI, 16)}$$

if the tube is effectively open at $x = 0$ (flute):

$$\text{or } \frac{\kappa}{\beta S} = \tan \beta b - \cot \beta a \qquad \text{(VI, 17)}$$

if it is effectively closed at $x = 0$ (clarinet).

These two cases can be solved graphically, as the author [74] has done, and compared with experiments on the notes of the two instruments. Fig. 274 shows how an equation like (VI,16) can be solved graphically for the case of bisection of the tube by the side hole ($a = b = l$). We plot $\cot(-\beta l)$ as a continuous line and three arms of $\cot \beta l$ as

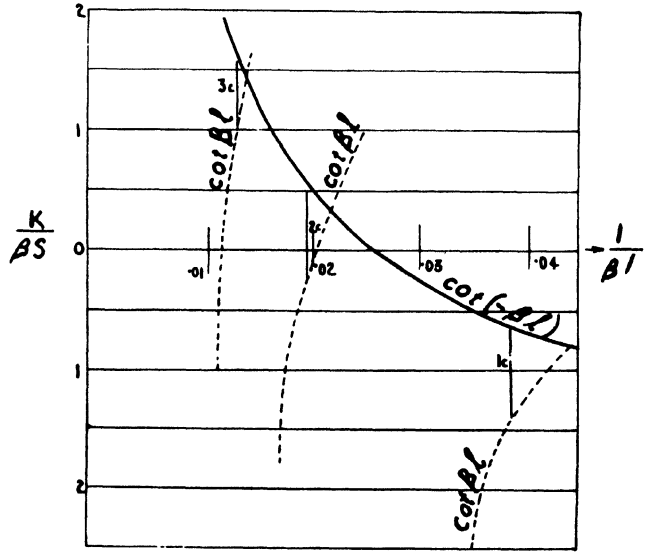


Fig. 274. Fingering of flute.

broken lines. Taking a typical orifice for which $\kappa = 1$ cm, $S = 1.35$ cm^2 we get $\kappa/S = 0.75$ cm^{-1} and set vertical intercepts of this value (in terms of the ordinate scale) between the two curves. These lines are marked c on the figure and we then have to look to see what values $(\beta l)^{-1}$ lie beneath them. Knowing l, the frequencies corresponding to these solutions of equation (VI, 16) are then found, of which that on the extreme right ($1c$) represents the fundamental and the other intercepts $2c$, $3c$ the overtones. When a and b are unequal, the larger should be plotted as l and the smaller as a fraction of l to preserve the correct ratio of a to b.

If several holes are open at once and are regarded as effective for determining frequency, the equations which have been developed must be solved by successive approximation.

When the tube is conical we take the formula e (p. 8) for spherical waves and combine sections of the pipe with orifice impedances.

The equation for the simple conical tube of length l (Fig. 276a, with side-hole closed) is

$$Z_0 = -\frac{i\rho c}{S}\left[\frac{Z_2 \sin\beta(l-\varepsilon_2)/\sin\beta\varepsilon_2 + \frac{i\rho c}{S}\sin\beta l}{Z_2 \sin\beta(l+\varepsilon_1-\varepsilon_2)/\sin\beta\varepsilon_1 \cdot \sin\beta\varepsilon_2 + \frac{i\rho c}{S}\sin\beta(l+\varepsilon)/\sin\beta\varepsilon_2}\right] \quad \text{(VI,18)}$$

where

$$\tan\beta\varepsilon_1 = \beta r_1 \text{ and } \tan\beta\varepsilon_2 = \beta r_2.$$

With hole open as shown ($Z' = -i\rho\omega/\kappa$) we find

$$\left.\begin{aligned}\frac{1}{Z_{l_1}} &= \frac{\kappa}{i\rho\omega} - \frac{S_1}{i\rho c}(\cot\beta l_1 + \cot\beta\varepsilon_1)\\ \frac{1}{Z_{l_2}} &= -\frac{\kappa}{i\rho\omega} - \frac{S_1}{i\rho c}(\cot\beta l_2 + \cot\beta\varepsilon_2)\end{aligned}\right] \quad \text{(VI, 19)}$$

When the tube is open at r_0 the resonance formula becomes

$$\frac{\kappa}{\beta S} = -\cot\beta(l_1+\delta) - \cot\beta l_2 \quad \text{(VI, 20)}$$

When the tube is stopped at r_0 (oboe)

$$\frac{\kappa}{\beta S} = -\cot\beta l_2 + \tan\beta(l_1+\delta) \quad \text{(VI, 21)}$$

where

$$\cot\beta\delta = -\beta r_0$$

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If $r_0 = 0$ or $\beta\varepsilon$ is so small that $\varepsilon_0 = r_0$, the equation reduces to that of the cylindrical tube open at the near end. Under these circumstances Fig. 274 which was drawn for application to the flute may equally well be applied to the oboe.

These applications of impedance principles to the fingering of wood-wind were first made by the author [74] and were later amplified by IRONS [75]. FOUCHÉ [76] has more recently extended the treatment to more complex cases.

§ 23. Compound Pipes

This is another important application of impedance principles.

Cases in which two pipes or a pipe and resonator are joined in series or parallel have been treated theoretically by IRONS [75]. Several of these cases are important in practice as they are used as organ stops, *e.g.* the BOYS' resonator which consists of a HELMHOLTZ resonator on the end of a pipe—the resonator may alternatively be let into the side of the pipe—and the flute-à-cheminée, in which a narrow pipe is a continuation upon a wider one. In the latter case, the resonant frequencies are given by a formula originally due to ALDIS [77] which BATE [78] has verified experimentally, and which we may write as the sum of two open pipe impedances in series. *i.e.*

$$Z = i\rho c\left[\frac{1}{\pi R^2}\tan \beta L + \frac{1}{\pi r^2}\tan \beta l\right] \qquad \text{(VI, 22)}$$

L being the length and R the radius of the wide pipe, l and r those of the narrow pipe.

Solutions of the equation just given show that the natural frequencies are not in harmonic relationship, so that the overtones will not be in resonance when the fundamental is.

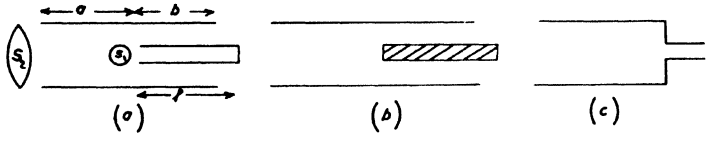


Fig. 275. Compound cylindrical pipes. (a) Haskell; (b) with co-axial rod; (c) with chimney.

TABER JONES [79] has treated the theory of the HASKELL organ pipe, an open pipe into which a shorter closed pipe is inserted (Fig. 275a). In terms of the symbols shown on the figure, he proves that the natural frequencies are given by

$$S_1 \cot \beta a + S_2 \cot \beta b = (S_1 + S_2) \tan \beta l \qquad \text{(VI, 23)}$$

The system of acoustic impedances is that of a stopped pipe with two open pipes in series branched upon the open end, and serves to produce bass notes in a comparatively restricted space, due to the method of folding one tube inside the other. To make this formula agree with the experimentally observed frequencies, it is necessary to make the usual assumptions with regard to end-corrections. The same formula will fit the case of simple pipe into which a coaxial solid rod is pushed (Fig. 275b), S_2 then being equal to the difference in cross-section of the rod and the pipe.

A rather more complicated case is that of a wide pipe with a constriction somewhere inside it. This we may regard as two pipes connected in series by an orifice (Fig. 275c). IRONS [75] has treated this case both theoretically and experimentally, using the method of KUNDT's dust figures to determine wave-lengths. The tube being virtually separated into two stopped pipes of length L and l by the constriction, the resonant frequencies will be given by

$$\frac{\kappa}{\beta S} = \cot \beta L + \cot \beta l \qquad \text{(VI, 24)}$$

TABER JONES [79] has examined organ pipes of two tubes joined by a cone. He obtains, in terms of the nomenclature of Fig. 276b.

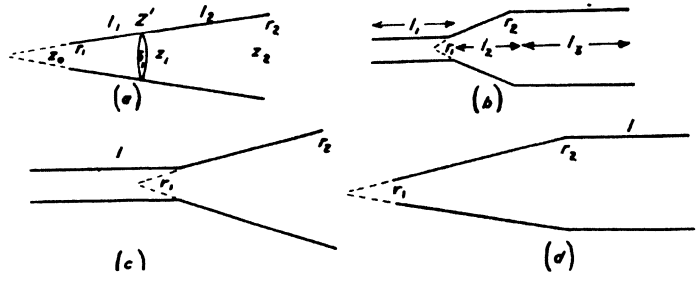


Fig. 276. Compound conical pipes.

$$\left(\frac{1}{\beta r_1}+\frac{\cot \beta l_1}{h}+\tan \beta r_1\right)\left(\tan \beta r_2\left[\frac{1}{\beta r_2}-\frac{\cot \beta l_2}{h}\right]-1\right)$$
$$=\left(\frac{1}{\beta r_2}-\frac{\cot \beta l_2}{h}+\tan \beta r_2\right)\left(\tan \beta r_1\left[\frac{1}{\beta r_1}+\frac{\cot \beta l_2}{h}\right]-1\right) \qquad \text{(VI, 25)}$$

h being the ratio of the area of the two cylinders. Experiment showed that the partial tones were inharmonic.

FOUCHÉ [76] gives these formulae for cylindro-conical combinations (Fig. 18/18c,d), S and l being the cross-section and length of the cylinder.

$$\frac{\tan \beta\,(l+e_1)}{\frac{S}{r_1}\tan\,\beta\,(l+e_1)-\beta S}=\frac{1}{\beta S}\tan \beta\,(l+e_2) \qquad \text{(VI, 26)}$$

$$\frac{\tan \beta\,(l+e_2)}{\frac{S}{r_1}\tan\,\beta\,(l+e_2)+\beta S}=-\frac{1}{\beta S}\tan \beta\,(l+e_1) \qquad \text{(VI, 27)}$$

e_1 and e_2 being the respective open-end corrections.

§ 24. End-Correction

It has been explained in Chapter 1, § 15 that a 'radiation resistance' is involved where the sound from a primary source debouches onto the atmosphere. This aspect is closely bound up with the mismatch between the sound waves within the pipe and those (of spherical wave-front) outside. Whereas the manufacturer of loudspeaker horns seeks the best match he can at all frequencies to avoid reflections into the horn and consequent resonances, the maker of a musical wind instrument must exploit such resonances for they lie at the heart of this method of sound production. Consequently we find that such instruments have cylindrical or slightly expanding tubes with, in some cases, a sudden flare at the bell mouth whose acoustical function is not important. Owing to the rapid expansion the bell cannot, in fact, much affect radiation, and in the wood-wind such little effect as it has is confined to the notes of the full-length tube.

At an open end, the reflected wave therefore returns from a point (which is also approximately the point from which the atmospheric radiation diverges) beyond the plane at which the walls of the tube terminate. This effective addition to the resounding length of the tube is known as the end-correction. RAYLEIGH [80] showed that to a

circular tube of radius r terminating in a hole in an infinite flange, the required addition was approximately 0.8 r. Experiments give 0.6 r for a flanged glass tube but it must be recognised that both corrections are, in fact, functions of frequency.

The actual problem was treated by analogy as that of the electrical resistance at the end of a cylindrical conductor terminating in an infinite medium. Actually RAYLEIGH used an approximation in which no provision was made for the continuity of the potential or its gradient on both sides of the end. KING [81] suggested an exact solution of the problem, making use of inversion theorems related to the theory of BESSEL functions, which leads to a value for the ratio of 0.821, though the difference between this and RAYLEIGH'S value will, as KING concedes, not amount to much in practice.

Our equation for the impedance of a pipe (VI, 12) can be regarded as giving the end-correction a at the partially stopped end—by equating the numerator to zero—in the form

$$\tan \beta a = \frac{\beta S}{\kappa} \qquad \text{(VI, 28)}$$

If the right hand side of this equation is small $a = S/\kappa$ but if the right-hand side is large we derive an end-correction in the gravest mode

$$a = \frac{\lambda}{4}\left[1 - \frac{\kappa\lambda}{\pi^2 S}\right] \qquad \text{(VI, 29)}$$

BATE'S [82] experiments on this matter led him to:

$$a = \frac{S}{2k}\sqrt{\frac{\pi}{S'}} \qquad \text{(VI, 30)}$$

S' standing for the area of the opening and k a factor depending on its shape (1 to 1.5). This is a special case of the equation $a = S/\kappa$, to which it resolves if κ is put equal to the diameter of the end and $k = 1$.

ROBINSON [83] has also measured the impedance of a hole of diameter d in a tube of diameter D. His results can be expressed by

$$\kappa = \frac{0.787d}{(1 - d/D)^{1.895}} \qquad \text{(IV, 31)}$$

if the hole forms a constriction in the tube and

$$\kappa = d(1 + d/D)^{1.19} \qquad \text{(VI, 32)}$$

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if a terminal. The latter is in agreement with another formula proposed by BATE.

TYTE [84] refers to an earlier paper in which he has calculated the end-correction for a conically flanged cylindrical pipe by analogy with corresponding problems in a.c. technology. He now shows that the experimental results of BATE and WILSON [85] fit this theory well and has extended the experiments to deal with cones of smaller apical angle, for which the cone is in the nature of an extension of the cylinder rather than a flange to it (*cf.* Fig. 276c). On the basis of this agreement he goes on to treat the classic case of the orifice in a diaphragm forming the open end of a pipe and suggests that the end-correction may be taken as inversely proportional to the area of the orifice. He tries out this idea on the experiments of IRONS, BATE and ROBINSON. The suggested relation between end-correction and reciprocal area holds whether the pipes are flanged or not.

The question of whether a flare at the end of a cylindrical tube amounts to an end-correction on the tube or makes the system a compound pipe has been examined experimentally by BATE and WILSON [85]. TYTE had dealt with this problem theoretically and obtained an expression

$$\sqrt{\pi}\, r \sqrt{\frac{1}{\varphi} - \frac{1}{4\pi}}$$

for the effective length of a cylindrical pipe (end-correction) of radius r to replace acoustically a conical extension of solid angle φ.

The experimental results, together with some more by HIGGS and TYTE [86], fit the theory fairly well for semi-vertical angles greater than 10°. For less pronounced flare the system behaves as a compound pipe, determined by eqn. (VI, 26).

WIRZ [87] suggests a formula for the end-correction of circular tubes based on the flange width b (= thickness of walls, in most cases):- $0.60 + 0.22 \exp(-kb/r.)$ which he confirms by experiment, using a value for the constant: $0.129 < k < 0.136$ to cover extreme variations. When $b/r \to 0$, the correction $\to 0.6$, a value which HIGGS and TYTE have established by experiment for the flangeless pipe.

INGERSLEV and NIELSON [88] have made experiments with a pipe of 35 cm length and 9.5 cm diameter, stopped at one end with a flanged open end (various values of b) and having a partially stopped open end at

the other. They found that even a small flange had a great effect on the end-correction. Beyond $b = 3r$, the effect was that of an infinite flange. The radiation resistance however depends on b/λ and not on r/λ.

TABER JONES [89] has studied experimentally the conductivity of a rectangular slit on one end of a pipe and also of holes on the side of a pipe. He gives a figure (Fig. 277, here) showing the ratio of the measured end-correction to that calculated from eqn. (VI, 28). Curve A is for a 'mouth' in the side, B for a symmetrical slit, C for a short rectangular opening in the end of the pipe. It seems, as perhaps is to be expected, that the theoretical equation is only valid for fairly symmetrical openings at the ends of pipes and not for slits passing right across or for the side-mouths and side-holes of pipes, save when the width is small.

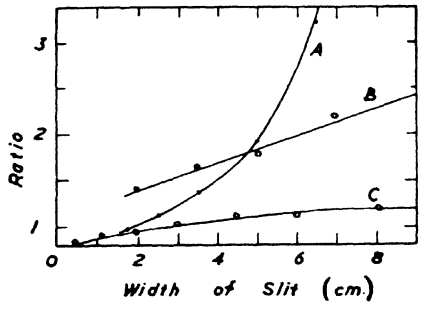


Fig. 277. Ratio of measured to theoretical end-corrections.

FOUCHÉ [76] gives yet another formula; to be compared with (VI, 31), (VI, 32):

$$\frac{1}{\kappa} = \frac{1}{d} - \frac{1}{D} + \frac{1}{\pi D\sqrt{(1 + \beta^2 D^2 4)}} \qquad \text{(VI, 33)}$$

(the first term representing the radiation resistance) but quotes no experiments to test it.

It is to be noted that all these theories neglect the friction both in the pipe and in the terminal. The combined effect of viscosity and inertia, as the author [90] has shown, is to cause a non-uniform particle velocity across the mouth of the tube.

§ 25. The Annular Effect

It has been assumed up to the present that the motion across the section of a pipe or orifice is uniform. It is obvious that in the immediate neighbourhood of the solid confines of a conduit there will be a drop in the displacement amplitude to zero, but before considering

this effect of viscous resistance, there is another effect, more important in practice, which invalidates our assumption. It was discovered by the author, in experiments with a hot-wire traversed across an orifice or conduit in which aerial oscillations were taking place, that the velocity amplitude was much greater in the outer annuli just before the wall was reached than at the centre of the tube. There is in fact a peak of alternating velocity whose magnitude (relative to the central velocity) increases with frequency, but whose distance from the wall decreases as this factor goes up. This circumstance, which gives the distribution of velocity across the tube or orifice quite a different trend to that of one-way air flow, immediately calls to mind the skin effect in electrical technology, shown by high-frequency alternating currents flowing in a magnetic conductor. Fig. 278 shows some typical results. This 'annular effect' [90] can be explained by considering the equations of motion of the air in the tube, and is due to a combination of inertia and viscosity. It appears from the theory that the distance (δ) of the peaks from the walls of the tube is given by the formula $\delta\sqrt{f} =$ a constant.

This law, that δ is proportional to $f^{-\frac{1}{2}}$ is borne out in practice; the position of the peaks is shown by the dotted line on Fig. 278. The existence of this annular effect has been confirmed by CARRIÈRE [91] who noted the distances traversed by smoke particles executing S.H.M.

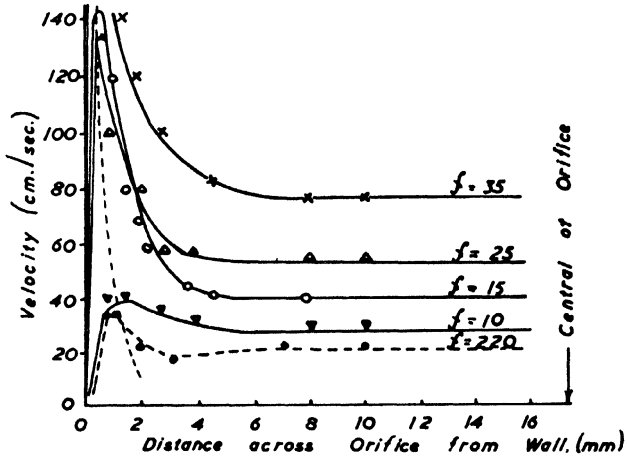


Fig. 278. Annular effect in pipes at various frequencies (f).

in a pipe. The amplitude was a maximum at points a short distance from the walls of the pipe.

These divagations from the assumed uniform particle velocity across a pipe near the mouth undoubtedly account for the frequency-dependent deviations of end-corrections from the simple HELMHOLTZ-RAYLEIGH formulae.

§ 26. Effects of Temperature on Pitch of Wind Instruments

Since the velocity of sound rises with temperature θ in accordance with the well-known formule $V_\theta^2 = V_0^2(1 + \theta/273)$, a corresponding rise of pitch with the temperature of the ambient air and the player's breath is to be expected and accounts for the need of the player to 'warm up' his instrument before he tunes with the strings and keyboard instruments in an orchestra.

POTTLE has done some experiments (reported by YOUNG [92]) on the dependence of tuning on temperature. He finds an average change of 1 cent per degree Centigrade in the orchestral wind instruments. From this the average internal temperature (a mean between that in the player's lungs and the atmosphere) can be inferred.

The organ, of course, remains at the local atmospheric pitch. Hence the undesirability of having the two types of wind instrument playing in consort.

§ 27. The Effect of Wall Materials on Quality

The role of the material of a pipe on the quality of the note it emits has long been a source of argument, some maintaining as the writer has done, that this factor has little effect, others that it exerts the major control over the formant. We shall assume in what follows that the walls are neither so thin and lissom as to give way under the compressions and rarefactions set up by sound waves in the column nor so strong and elastic that they are easily set in resonant (longitudinal or transverse) vibration with the edge tone or reed to which the column is coupled. (Wolf notes of the latter type are not unknown in ill-made metal flutes and clarinets.) Even in the absence of a definite 'wolf', LOTTERMOSER [93] disclosed a modulation of the upper partials of the column when they happened to lie near natural (longitudinal) tones

of the metal pipe. (The latter are usually too high in pitch to affect the fundamental tone of the column.)

BONER and NEWMAN [94] have made a very painstaking research on this point, in which they were careful to compare flue pipes of different wall material otherwise identical in construction, *i.e.* having the same mouth, the region most sensitive to slight adjustments. They used eight different materials, including shellacked paper and thin copper. The former pipe improved in speech as the shellac dried and, when the tube was grasped by the hand, the latter continued to 'ring' in one of its natural longitudinal frequenciesafter the wind was cut off from the pipe.

The general results may be thus summarised:

(1) the fundamental amplitudes were nearly all equal,

(2) the even harmonics of the paper pipe were weaker than those of the rest,

(3) the wooden cylinder is as strong in harmonic development as the metal cylinder, contrary to what many organ builders say.

The extent to which the walls confining a vibrating column of air may take part in the vibration has also been studied by KNAUSS and YEAGER [95], using a cornet having a crystal pick-up attached to it, but though metallic vibration was detectable it proved to be insignificant as a contribution to the total sound output of the instrument.

On the whole then, the effect of wall material on timbre is small.

§ 28. Flue Pipe and Flute

These are compound systems in which an edge tone is coupled to a column of air of fixed length (in the organ flue pipe) or of variable length (in the orchestral flute and piccolo). Fig. 279 is a typical diapason pipe. The air issues from the wind chest via the boot and the slit at the lower lip. It strikes the upper lip or languid and in doing so sets up an edge tone, the dynamics of which we have already discussed (p. 482). The distance between the two lips (so called 'height of mouth') is

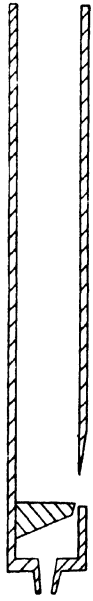


Fig. 279. Section of diapason organ pipe.

the invariable factor in the edge-tone formula. V is also set by the pressure in the wind-chest, but we are going to admit the possibility of varying it in what follows. In the orchestral flute on which the player produces the edge tone by blowing across a side hole set near one end, the blast velocity is completely at will and l—which becomes the distance between the player's lips and the opposite side of the hole—can be varied within certain limits as the note elicited is changed.

Fig. 280 indicates by the broken line the normal rise of edge-tone frequency with $V(\propto\sqrt{p})$ and the readjustments which take place when one of the jumps to an eddy system with additional vortices occurs (as on Fig. 271c). The thick horizontal lines represent the

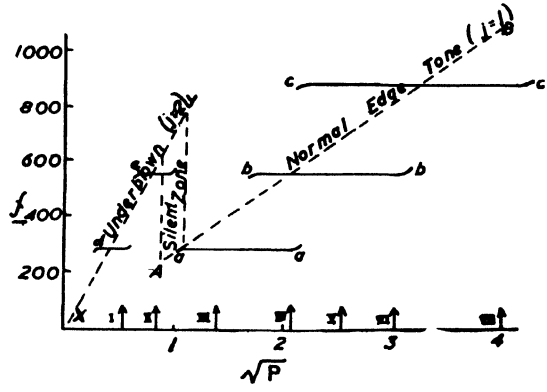


Fig. 280. Organ-pipe tones in relation to edge tones.

natural tones of a column of an open tube to which the edge tone system finds itself coupled in this type of organ-pipe (fundamental 250 c/sec.)

The column can pull the edge tone out of its natural frequency and sound its own fundamental or overtones over quite a range of blowing-pressure. Only when they are badly out of synchronism does the column tone first rise or fall a little and finally become silent.

The 'underblown tones' are feeble, owing to the small power output at low pressures, but may be heard when the wind is first turned onto the pipe and before normal pressure and speech have been attained. As such they are undesirable but Fig. 271 shows that they will not be heard if the edge lies a little to one side of the main stream

(The shaded area in this figure is a silent region for the underblown pipe.) Accordingly the maker adjusts the wind to blow a little outside the upper lip—at an angle of about 20° to the axis of the pipe according to KUHN [96]—though not so far out that the normal tone will fail.

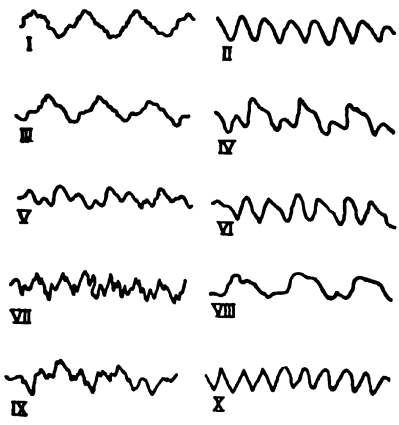


Fig. 281. Oscillograms of flue pipe at various pressures.

Oscillograms of the sound of the flute diapason pipe to which Fig. 280 refers, have been obtained by MOKHTAR [97] at the instigation of the author. These are shown on Fig. 281, the roman figures referring to the values of pressure marked as abscissae on Fig. 280. The fundamental (III) is produced best (but mixed with a high harmonic) at a pressure of 2 cm of water. When this pressure is raised above 4 cm the fundamental disappears (IV). At 9 cm the third harmonic is prominent (VI) and at 14 cm the fourth (VII).

When the pressure is brought below normal on this pipe, the underblown octave is feebly heard at 6.5 mm (II) and the underblown fundamental at 2.5 mm of water (I).

Oscillograms VIII, IX and X refer to an open pipe of about the same length at 1.45, 6 and 16 cm respectively. On overblowing, the third harmonic is prominent.

KUHN [96] has carried out a careful investigation of the effect of changes in the geometry of a typical organ pipe near the mouth on the tone-quality. In terms of our Fig. 279, he varies (1) height of mouth, (2) diameter of slit orifice, (3) horizontal position of upper lip, (4) breadth of mouth. He found that for each partial tone there was a setting of the upper lip both horizontally and vertically which gave the greatest amplitude to this partial in the complex note of the pipe. This effect is apparently connected with the angle at which the air debouches from the slit, since altering slit diameter had a similar effect. All partial amplitudes increased with the breadth of the mouth and with wind pressure.

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Other things being equal, a narrow scale encourages the upper partials at the expense of the lower. The organ builder constructs different stops of pipes on this principle; large-scale pipes for the mellow foundation tone, narrow-scale for string tone.

The radiation resistance at the open end is proportional to f^2 (p. 15) whereas the internal damping due to viscosity is proportional to $f^{\frac{1}{2}}$ (p. 20). Consequently the frequency at which the total resistance is a minimum rises as the diameter decreases and the higher partials (for a given length) are elicited with greater effect in the narrow tube. INGERSLEV and FROBENIUS [98] have also studied the relationship between terminal impedance and timbre, and MERCER [99] the science of 'voicing' a pipe. The modern flute (the design of which was due to BOEHM) has a scale of about 30 times its diameter. The tube is cylindrical except in the region of the blowing hole where it contracts. The contraction continues beyond this hole to the cork, an adjustable stopper by which the player tunes the full-length tube. Six side-holes to be covered by the fingers are available and others, normally closed, opened by depressing keys. As STEINHAUSEN [100] showed with a pressure probe and YOUNG and LOUGHRIDGE [101] with a hot-wire anemometer, when two neighbouring holes are open, the pressure amplitude continues beyond

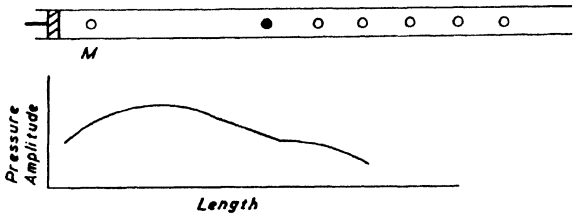


Fig. 282. Variation of pressure amplitude along flute tube.

the nearer hole (to the blowing hole) to reach zero somewhere in the neighbourhood of the second hole. In Fig. 282 is shown a skeleton flute with mouth hole *M*. No. 1 side-hole is closed and remaining ones open. The pressure amplitude is plotted on the graph below. The pertinence of our impedance theory for the side-holes (§ 22) should be apparent.

LÜPCKE [102] has recently made a careful examination of the timbre of the orchestral flute, varying the blowing pressure to get the best output. The fundamental was, of course, very prominent always and

the third partial was usually stronger than the second especially when cross-fingering was used. As the player runs up the scale the overtone structure increases with a jump to greater overtone emphasis at the first overblown note (*d*). The pressures used by players (measured by a manoometer in the mouth) were low compared with those on the organ (about 1″ water). There was a slight rise of pitch—keeping to one note—with blowing pressure.

When the player has ranged over the scale from the lowest note (D) upward by uncovering the holes in turn and using the keys and the normal edge tone, he overblows to the octave (and sometimes alters the effective 'height of mouth' with his lips), starting again with all holes closed and uncovering in turn. Normal fingering would for example give two adjacent notes on the scale by covering 1, 2, 3, 4; then 1, 2, 3; but a pitch intermediate between these could be reached by covering 1, 2, 3, 5. The last is an instance of 'cross-fingering' (*cf.* p. 487).

§ 29. The Reed Pipe

In this class of sound-producing systems a column of air is closed at the mouth by a 'beating reed' (*cf.* p. 486). Normally the reed stands clear of the orifice leading to the pipe, but when wind under pressure is introduced to the chamber surrounding the reed, it is deflected by the rush of air into the pipe, until it shuts up the pipe orifice; then, being under tension, it springs back and lets the air into the pipe again, and the cycle is repeated. We have already discussed the motion of the reed acting alone. Now we must consider its effect on the behaviour of a column of air to which it is coupled. Various possibilities may occur, depending on the form of the 'coupling'.

Firstly, let us consider the reed organ pipe. The air tube or pipe

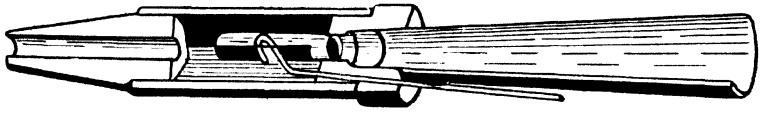


Fig. 283. Section of reed organ pipe.

proper is generally conical. The reed, generally of brass, is at the vertex and is contained at the end of a narrow tube, to which one end of it is clamped. The reed and this narrow tube are enclosed in a short supply pipe termed the 'boot'. The conical pipe and the reed

are tuned to the same frequency, the tuning being accomplished by a spring which presses on the reed near its clamped end, shortening it or lengthening it as desired (Fig. 283). The resulting note is due to the fundamental of the reed with the coincident fundamental of the pipe, together with some overtones of the latter. As the overtones of the reed are inharmonic, these are scarcely produced because they do not form part of the 'note' of the column of air. The vibrations of the reed are always simple harmonic although the air vibrations involve the overtones proper to the form of the air column.

The relationship between the pitch of sound emitted by a reed of

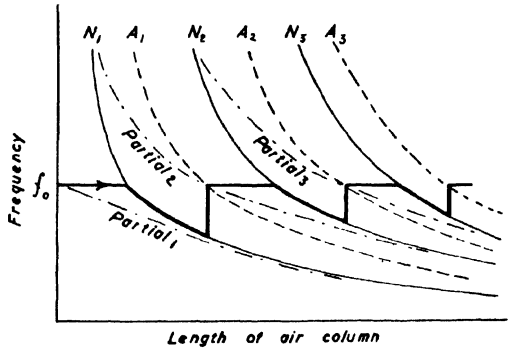


Fig. 284. Variation of frequency of reed pipe with length of column.

invariable length, as the column to which it is coupled changes in length, was examined over a century ago by WEBER [103]. Fig. 284 (after FOUCHÉ [76]) shows what happens. The hyperbolic broken curves show how the pitch of the partial tones of the column vary with its length, the signs N_1, A_1 etc. indicating the movement of the nodes and antinodes in the column. (These curves are given by $f\lambda$ = constant.)

The reed's natural frequency f_0 is in tune with each of these at a certain length. As the tube is lengthened these resonances are picked up in turn, but further lengthening drags the reed frequency down until it frees itself and continues at f_0 until the next resonance is picked up. The extent to which it makes these diversions from its natural bent depends upon its elasticity and damping. Thus the thick continuous line on the figure represents the behaviour of a hard metal organ reed; he chain line that of a soft bamboo reed, which accommodates itselft

more or less to one of the partial frequencies proper to the column.

These theoretical ideas have been well shown in the oscillograms of reed pipes made by MOKHTAR [104], whose work we shall now detail.

The actual mechanical motion of the reed of such a pipe was recorded on photographic film, using a very light, small mirror stuck on the reed itself and a beam of light from a pointolite lamp. The vibration of the air in the resonating column very near to its base was recorded also photographically by the aid of a manometric capsule, having a tiny light mirror stuck on the membrane, and a beam of light. The recording of both the mechanical motion of the reed and of the response of the manometric capsule was done simultaneously and on the same photographic plate to enable a study of the change of phase to be carried out.

The reed was a beating reed in a wind-supply chamber of the type commonly employed on organs. It had a natural frequency of 213 per sec. under a blowing pressure of 0.48 cm of water.

The mechanical vibrations of the reed are always simple harmonic (No. 1, Fig. 285) although the air vibrations involve the overtones proper to the form of the air column and depend on the conditions under which the pipe is excited.

If the mouthpiece containing the reed is excited when uncoupled to

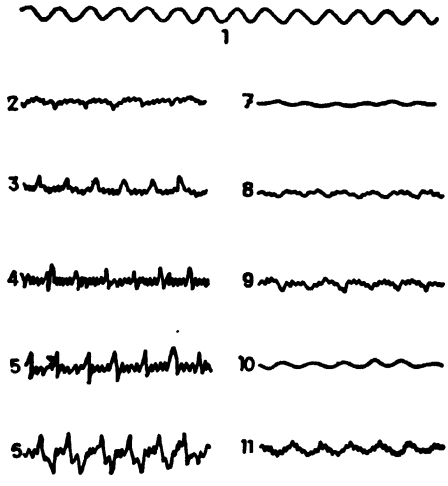


Fig. 285. Oscillograms of reed pipe with increasing pressure.

a resonator the character of the sound heard is usually of a raucous tone, unpleasant for the ear, especially when the pressure is high. This is due to the fact that the more sudden the discontinuity of any periodic motion, the greater the relative importance of the high upper partials into which that motion can be resolved. Thus higher pressure causes the closure of the air-duct to be more sudden, and therefore the higher upper partials become more obtrusive. This is illustrated clearly in the series of oscillograms 2, 3, 4, 5 and 6, which are all recorded for the same mouthpiece alone excited under successively increasing blowing pressures of 0.58, 0.73, 0.80, 0.93 and 3.15 cm of water respectively and without a resonator. The gradual appearance and growth of the upper partials of the reed are easily followed in the successive oscillograms. Even the first oscillogram (no. 2), which is recorded for a very low blowing pressure, gives a distinct indication of these overtones and as we proceed to the last of the series (no. 6) the overtones completely overcome the fundamental.

The reed is now coupled to a pipe resonator. In this case the coupling is loose and the vibration of the reed is predominant. Nevertheless, the quality of the resultant note is affected by the presence of the resonator, in that the upper partials are less distinct and therefore the tone of the pipe is purer. This is illustrated by the series of oscillograms 7, 8 and 9 which are recorded for the same mouthpiece of the first series when coupled to a resonator in the form of a brass cylinder 45 cm long and 3.6 cm in diameter (natural frequency 185.5 c/sec.) mounted in the usual fashion directly on the supply box. The corresponding blowing pressures were 0.56, 0.75 and 1.03 cm of water respectively. These are to be compared with the oscillograms 2, 3 and 5 respectively, which refer to nearly the same blowing pressure but with the reed uncoupled to a resonator .

To provide a further illustration to the effect of pressure the length of the resonator is now decreased to 39.5 cm (natural frequency 215 c/sec.) and the two oscillograms 10 and 11 recorded under a blowing pressure of 0.70 and 1.3 cm of water respectively. In this case the natural frequency of the resonator is very near to that of the reed, and that is why the upper partials which were clear in oscillograms 3 and 5, somewhat damped in 8 and 9, are now highly damped.

Records were next taken starting from a length of the resonating air column a little shorter than that corresponding to the first overtone

of the column to that which would give the fundamental. The whole series (Fig. 286) is recorded under one blowing pressure of 4.8 mm of water, this being slightly above the minimum pressure required to excite the pipe. The curves given by the manometric capsule are recorded below. Those of the reed (above) are always S.H.M. and of fundamental frequency 213 c/sec. We can examine the phase relationship from left to right between the motions by observing the lie of the peaks in a vertical direction. In the first three oscillograms 12, 13, 14 (Fig. 286), the lengths of the tube are 18.2, 19 and 20 cm respectively. Over this range two partials are prominent in the waveform of the system; one is in unison with the reed and the other the octave of this. In No. 15 (22 cm length) the octave has nearly and in No. 16 (25 cm) completely given way to a pure tone in unison with the reed. This state of affairs is continued at No. 17 (29 cm).

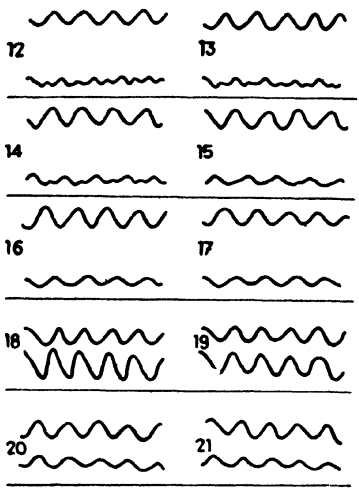


Fig. 286. Relative amplitude and phase of reed and air columns of different lengths.

During the whole of this series the peak of the air compression just inside the pipe has lagged more and more, up to a quarter period behind the peak of the reed's motion (complete covering of the wind orifice), and the length of the resonating column is becoming more and more out of tune with the reed until at last each hinders the vibration of the other. Consequently the pipe is silent, although at a higher blowing pressure the reed can be forced to vibrate again. Let us now keep to our former pressure and still lengthen the pipe. At a length of 39.0 cm the air column almost jumps into vibration, being pulled down to resound to the note of the reed. When the length reaches that corresponding to best resonance the amplitude of vibration is a maximum, as shown by No. 18 (40 cm). A further increase of the resonator length sets it out of tune again, although it continues to give forced resonance to that of the reed, but of course with gradually diminishing response

(No. 19 at 42 cm; No. 20 at 47 cm; No. 21 at 49 cm). During this stage the pressure in the air column is very slightly in advance of the reed.

The optimum length of tube will be that for which a compression is transmitted to the far end and returns in step with the next reed pulse. In the present instance this happens when the tube is about 18.5 cm long (No. 12). If the tube is lengthened the returning compression arrives progressively later, until it arrives too late for the reed to cope with it, and the system is reduced to silence. If the tube is further lengthened until the compression is nearly in step with the next-but-one pulse from the reed the system is again maintained. Quite a wide range of phase incompatibility is permissible without intermission of the sound.

The first overtone of the column can be excited under the same conditions, but is naturally in evidence only at the shorter lengths. It will be noticed that only the quality changes with lengthening tube (Nos. 12–17). There is none of the gradual lowering of pitch with length (*cf.* Fig. 284).

§ 30. Orchestral Reed Instruments

The oboe and a number of acoustically similar instruments have a conical tube and at the mouth a double reed which the player grips with his lips. Its 'tonality' is therefore like that of the flute; it has six side-holes (with keyed extras) and overblows to the octave. In order to assist this overblowing the instrument has, unlike the flute, a small vent-hole, covered for the lower octave, at a position judged to correspond to the average point in the pipe at which an additional antinode will be required in eliciting the octave of the various possible speaking lengths.

Its quality in the steady state is very like that of the violin. Only their characteristic formants of transients (*vide infra*) allow one to distinguish the sounds of these instruments in an orchestra.

The clarinet has been more exploited by scientists than all other orchestral wind instruments, for it presents several features of interest. We have already considered the characteristics of the reed's vibration (p. 485). Fig. 272c shows how it is gripped by the 'ligature' (a metal band) to the 'table', the bevelled end of the ebonite or wooden tube.

The player grips this mouthpiece between his lips folded back over his teeth. The tube is cylindrical about 2′ long and ½″diameter.* Like the flute, the modern clarinet tube is sometimes made of metal.

As the reed begins to close the aperture a condensation starts down

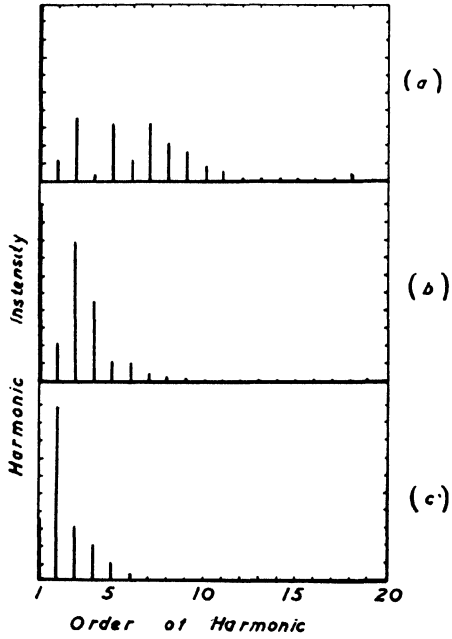


Fig. 287. Analysis of clarinet quality (a) low; (b) medium; (c) high register.

the tube but is reflected as a rarefaction from an open side-hole or from the distant end of the tube and on return finds the aperture closed by the reed. It is reflected therefore as a rarefaction for a second trip down the tube. Again reflected at the open end, this time as a condensation, it returns to push open the reed aperture. Thus in one complete cycle of the reed's vibrations the sound travels four times the length of the tube as in the stopped diapason organ pipe. Accordingly, even harmonics should be missing in the timbre. McGINNIS, HAWKINS and SHERA [105] have taken a large number of oscillograms

* The clarinet is a transposing instrument and is used in several lengths in the orchestra. The reason for this, which has nothing to do with science, will be found in text-books of music.

of the clarinet and while there is a considerable difference in waveform between notes quite close together on the scale, they are able to give three average spectrograms for the three 'registers', (a) low (b) medium and (c) high, which we reproduce (Fig. 287). The characteristic quality of the clarinet is ascribed to the strong odd harmonics, though it will be noted that the even ones are not entirely absent, as simple theory would indicate. In the high register the second harmonic, in fact, overpowers all the others but it is well known that the characteristic clarinet timbre cannot be distinguished in playing high notes.

TABLE III

	Register		
	Low	Medium	High
1. Energy up to 8th partial;			
(a) odd partials	68 %	70 %	40 %
(b) even partials	16 %	28 %	60 %
2. Energy of higher partials	6 %	2 %	0 %

The spectra naturally change with blowing pressure and with the effective length of the reed. The player can adjust this by the grip of his mouth according to the pitch of the note he wishes to elicit, and to a certain extent pull the column out of its natural frequency. Natural vibrations of the reed, though feeble, can be detected in the clarinet timbre (ASCHOFF [106]).

Since the column on the above analysis behaves like a stopped pipe, it overblows to the third harmonic. To complete the musical scale, some additional side-holes (compared to the flute and oboe) are necessary and this entails somewhat more complicated fingering. This 'bridge portion' between the lower and upper (overblown) register is the weakest feature of the instrument, both in intensity and quality. As on the oboe, overblowing is assisted by opening a vent-hole.

REDFIELD [107] raises some important points in connection with the theory of wind instruments. The most telling of these is that the embouchure is not a node of the column of air in the instrument, for this column is effectively continued more or less into the player's mouth, larynx and chest. This explains the discrepancy between the theoretical speaking lengths—plus end-correction at the flare—and the actual frequencies of such instruments. The mouthpiece end is

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in fact to be treated as approximating to an antinode, not as a node.

GHOSH [108] develops the suggestion—not substantiated by direct observation, as far as the writer knows—that the reed never entirely closes the mouthpiece, so that the acoustic admittance of the mouth never becomes zero at any epoch. It is further assumed that the pressure inside the player's mouth remains constant while the reed acts in the fashion of a valve to produce periodic variations of pressure in the embouchure of the instrument. When all side-holes are closed and an antinode lies at the open end, a formula for the frequency which has already been given (p. 488) is derived;

$$\tan \omega l/c = -\omega S/c\kappa \qquad \text{(VI, 34)}$$

(ω = pulsatance, c = velocity of sound, l = length, S = area of cross-section of pipe, and κ = conductivity of embouchure). After giving the first five solutions of this equation, which correspond roughly to the 1, 3, 5, 7, 9 members of the harmonic series, GHOSH obtains approximate equations for the motion of the reed, considered as a wedge-shaped bar clamped at one end. On account of the aerial vibrations, the pressure in the mouthpiece fluctuates cyclically about the value $P/(1+S\kappa l)$ where P is the pressure inside the player's mouth. The motion of the reed will impose further pressure variations on these. If the chink opens at an epoch of maximum pressure, more air will flow in and accentuate the peak of pressure, but if the opening occurs at a minimum there will be a tendency to level out the fluctuation, *i.e.* to make an antinode of the mouthpiece. Theory shows that the best condition for maintaining the coupled system will be that the flow of air into the mouthpiece should precede the phase of maximum pressure therein by a quarter period. Finally, the feeble even numbers in the spectrum are shown to be components in the reed's vibration forced upon the unwilling column of air.

Of the other common orchestral instruments, the saxophone is acoustically interesting as it combines a conical tube (nearly always metallic) with a clarinet reed. Deprived of its side-holes and key apparatus, its form bears a close relationship to the reed organ pipe but the reed is, of course, more pliable as it has to conform to a range of notes instead of resting on one. It overblows to the octave like the oboe.

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§ 31. Brass Wind Instruments

With the exception of some obsolete keyed bugles and the ophicleide, these differ from the wood-wind in that they have no side-holes. The possible notes were formerly limited to the harmonic series based on the fundamental of the tube (the fundamental itself not being obtainable with any worthwhile intensity on all but the smallest instruments). On modern instruments there are available by the operation of valves three or four extra lengths of tube to enable the player to cover the diatonic scale.

Exceptionally, the trombone with its telescopic slide allows of continuous variation of tube length in place of step-by-step but evenso the player eventually confines himself to a number of set positions of the slide.

These instruments are also distinguished by the fact the player's lips, held in a cup-form mouthpiece, take the place of reeds. For this reason, some acousticians refer to them as cup-mouthpiece instruments.

Among themselves they are distinguished by (1) the shape and size of mouthpiece, (2) the vertical angle of the quasi-conical tube.

The reader will appreciate that the player's object is to exploit the natural resonances of the tube, and that this becomes more difficult as the flare of the tube increases. A wide-angled horn has in fact, no resonances. This probably explains the difficulty of learning the French Horn, especially in the upper register where the harmonics almost overlap making it difficult to avoid 'warbling' from one to the next.

On the other hand, the trumpet and, still more, the cornet from the small vertical angle of the bore lend themselves to ease of exciting the natural tones of the column. In all cases the resonances will be relatively less sharp at the higher frequencies, because open ends will radiate more readily at high pitch.

In the latter instruments the writer has suggested that the shape of the mouthpiece aids tone-production in that at the correct blowing pressure an edge tone may be set up between the vibrating lips of the player and the sharp edge of the cup where it leads into the pipe.

The horn lacks this characteristic as the cup is shaped like an inverted cone. This feature added to the wide flare of the tube gives the

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horn flexibility without stability. One can't have it both ways; its the same problem as that of the fighter aircraft!

MARTIN [109] and HENDERSON [110] found that the vibrations of the lips were acoustically purer than those of the column (*cf.* the reed pipe in this respect, p. 505) and WOODWARD [111] that the player can force the column about 20 vibrations sharp or flat of its natural bent at any pitch. Further, LONG [112] has found that the velocity of sound (as product of wave-length and frequency) varies markedly in both the flaring bell and in the narrow part of the tube (beyond the mouthpiece). There is also a pressure antinode near the mouthpiece for all those tones which can be 'buzzed' with the lips, and these are as to wave-length the odd modes. Owing however to the dispersion of the velocity of sound in the instrument, they appear *as to pitch* to cover a full harmonic series (minus the fundamental).

WEBSTER [113] finds that the shape of the bell has small effect on the intonation of the whole instrument, *i.e.* on the pitch of all partials.

§ 32. Transients

We have thought it best to leave to the end of this chapter an account of the transients of musical instruments because of its general application, although most of the careful analysis has been done on the violin and the two prototypes of organ pipe.

We have already dismissed as origins of the formant many of the features of individual instruments formerly thought to be of primary importance, leaving the transients which precede and follow the establishment of the steady state as discriminatory factors, *e.g.*, between the violin, the orchestral oboe and the oboe-imitation stop on the organ.

THIENHAUS and FRANZ [114] were among the first to record the initial transients of organ pipes on their octave filters. During the 1939–45 war, NOLLE and BONER [115] in U.S.A., and COOK [116] in England examined oscillograms of organ pipes.

All agree that (1) the speech of flutes is relatively slow and the 'starting sounds' contain tones—some inharmonic—which disappear by the time that the steady state is set up, (2) that the reed pipe is comparatively quick in speech and a sample taken out of its starting oscillogram is usually a small copy of the final state.

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In view of what we have said (p. 500) about the underblown tones of flue pipes no one should be surprised at the first of these statements. Similar tones foreign to the fundamental of the resonator will appear on any instrument while the exciter—in this case, the edge tone—is being pulled into consonance with the resonator whose pitch is going to determine the steady state. Not only do the partials of the pipe pass through a range of changing intensities during the unsteady state as the wind pressure in the pipe builds up, but they vary also their relative phases as the wave-form changes by degrees to the final state.

Cook analysed his oscillograms of an open diapason to show the relative intensities of the first and second partials and also recorded the rise of air pressure. One notices the intrusion of even partials during the build up and their eventual disappearance. The order in which these partials appear usually depends on the *rate* of augmentation of pressure. When the air is turned on slowly the fundamental appears before the second harmonic and contrariwise. The transient states do not correspond to a series of possible permanent states under sustained force but are characteristic of non-linear systems.

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CHAPTER 19

ELECTROPHONIC MUSICAL INSTRUMENTS

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§ 1. Introduction

We may classify the available electrophonic apparatus thus:-
(1) Instruments giving single notes, in which the capacitance of an oscillator is varied to alter the pitch.
(2) Instruments containing a number of oscillators which are brought into action by depressing keys and in which the timbre is changed by interposing suitable filters.
(3) Instruments in which the tone-production is secured by shining beams of light through slits onto a photoelectric cell, interposing a film or disc with suitable black and white patterns inscribed thereon.
(4) Stringed instruments or vibrating reeds which initiate the notes through a number of pick-ups and suitable filters.

§ 2. Single Note Instruments

The prototype of these was the singing arc of DUDDELL [117]. The arc was shunted by a coil and condenser whose inductance and capacity determined the natural frequency of the circuit. By continuously variable or step-by-step changes in inductance it was possible to play tunes on the arc. Since then numerous instruments using valve oscillators (audions) have been patented but those with keyboards and one oscillator to each note have proved too expensive both in first cost and upkeep of their numerous valves to gain a footing in the market. There remain the solo instruments of which the Theremin and the

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Trautonium have reached commercial production. THEREMIN's instrument is a beat frequency oscillator in which the capacity controlling the frequency consists of a copper loop and a baton held in the hand (or sometimes the hand itself). The beat note when the hand is away from the loop is above the audible but descends through the musical gamut as the hand approaches the loop in virtue of the increasing capacity.

In the Trautonium[118], sponsored by the Telefunken Company of Germany, portamento playing is still possible or one may proceed directly from note to note in proper musical style. A grid glow tube is used as a variable frequency generator. The grid potential which determines the pitch is controlled by the length of resistance wire cut off by the player pressing some point on the wire onto a mark on a metal plate behind the wire, like the frets on a banjo. Another resistance under the plate is varied by the pressure of the finger on the plate and so alters the loudness of the sound. It is therefore possible to get as nice a variation in playing as a virtuoso may get from a violin by the mere gradation in the pressure of a single finger on the wire, including effects such as the vibrato and staccato playing.

§ 3. Oscillators

This type derives the timbre either electromagnetically or electrostatically. Fig. 288 shows the scheme of the Novachord or Hammond

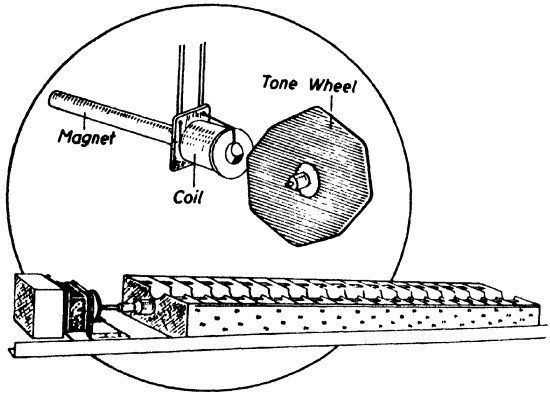


Fig. 288. Electromagnetic tone generator.

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organ [119]. A synchronous motor drives a series of ninety-one tone generators through gears and pinions. The tone wheel is a polygonal plate about the size of half a crown, rotating near a permanent magnet on which a coil is wound. As a high point on the wheel passes the magnet it induces a pulse of current in the coil. The speeds of rotation and the number of corners on each wheel are so calculated that each disc produces one of the ninety-one partials used in building up the combination of fundamentals and harmonics to give the required timbre. The latter is assured in the harmonic controller wherein the various frequencies are superimposed and flow as a single complex wave to the pre-amplifier located in the console.

In the electrostatic type, two electrodes are spaced a small distance apart in the air, one moving relatively to the other. The two members constitute two plates of a condenser, usually at a fixed distance apart. An undulating variation in capacity between the members is caused by rotating one member on which sinusoidal grooves are inscribed while the other is kept fixed. One of the stationary discs of insulating material, to the face of which are fixed eight concentric rings of conducting material cut into sinewaves, is shown on the left of Fig. 289.

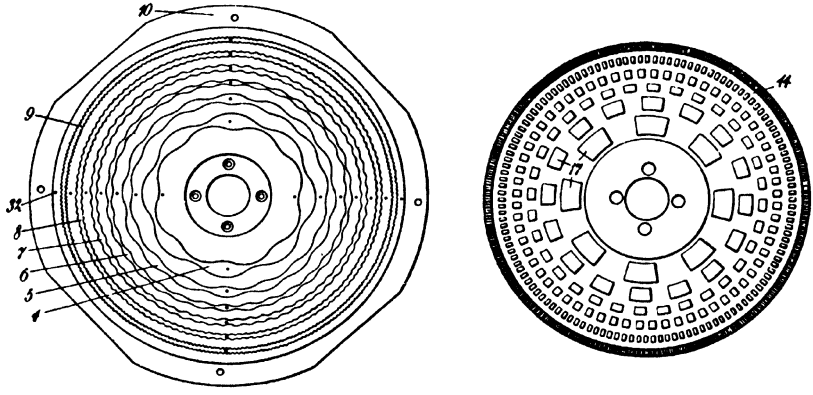


Fig. 289. Electrostatic tone generator.

The number of forms to each ring run in powers of two from the centre to the outer ring which has 256 undulations. A pair of these stationary discs face each other while between them another disc of bakelite rotates – seen on the right of Fig. 289. The rotating disc is

provided with eight concentric rings of apertures, each of which covers a half wave-length of the corresponding sine wave. These condensers of varying capacity are connected in the grid circuits of valves so as to produce the necessary fundamentals and overtones for covering the musical gamut.

§ 4. Photoelectric Synthesisers

(3) There are several varieties of the photoelectric organ. In one, of French origin [120], a number of sinusoidal patterns are painted in black on a disk (*D*, Fig. 290), which is rotated on an electric motor

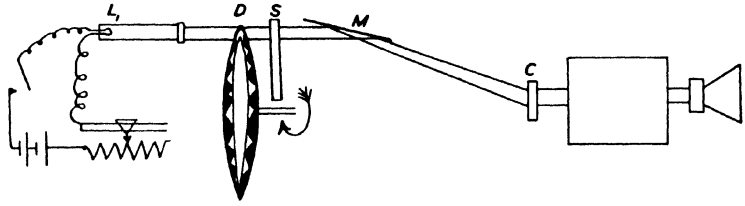


Fig. 290. Photoelectric tone generator.

of constant speed. Each pattern is provided with a slit *S* and a small lamp *L* behind it. The intensity of the light passing through the slit is thus varied periodically and the combined illumination, by the aid of mirrors (such as *M*) or prisms is brought onto the photoelectric cell *C*, from which the amplified current is fed to the loudspeaker. Thus a disk having a number of sinusoidal patterns from four upwards, when rotated at 50 revolutions per second gives a harmonic series of tones – based on a fundamental of 200 cycles per second. The timbre is varied at the will of the performer through the strength of the current supplied to each of the small lamps, one for each pattern. Alternatively patterns corresponding to preset timbres may be painted on the disc (*cf.* GREENLEE [121]).

Although the three organs described normally give tones of fixed intensity, it is not difficult, though of course it adds to the elaboration, to introduce circuits which will damp the sound of each note from its inception and so mimic pianoforte, harp or bell tone, but of course the artificial wave form does not vary during attenuation in the same way

that it does in some of the prototypes. It is not beyond the wit of man to devise circuits which will change the quality at the same time as they make the note die away. Some inventors indeed derive both constant and evanescent tones as required from the same damped source *e.g.* a struck reed. As soon as the sound has reached maximum intensity the pick-up is switched over to a circuit with a time constant which can be made to give long or short damping at will or to maintain the tone without dissipation as long as the key is held down, in spite of the damping in the acoustic vibrator, (this necessitates a negative resistance in the circuit).

§ 5. Electromechanical Instruments

Of types which use a conventional musical instrument such as a pianoforte, guitar or violin and pick-ups placed over the strings, the most famous are the Neo-Bechstein, a pianoforte without soundboard, and the Electric Guitar, both of which have pick-ups behind the strings. On all these instruments a magnetic pick-up may be used, that is to say, as the steel wire vibrates it changes the magnetic induction in a small coil nearby or it may work on the electrostatic principle in which case the vibrator moves to and from a plate behind the wire. The wire is then charged to a difference of potential over the plate so that the pair make up a condenser of varying capacity, which can be connected into the grid circuit of a valve amplifier.

§ 6. Circuit Details

(1) As already mentioned, the first generator, in the days before the thermionic valve, employed alternating currents.

The next stage of development was the employment of a gas discharge tube. Fig. 291 shows the circuit of the Trautonium of TRAUTWEIN in which the neon tube N in parallel with the condenser C is charged through the resistance R until it discharges. These voltage pulses applied to the valve V occur at a rate dependent on the product RC, but the fundamental is obscured in the final output by 'formants' introduced by applying the output voltage to resonant transformers such as T, so that the output wave-form is very complex. Latterly the neon-tube has been replaced by a thyratron (*cf.* KOCK [122]).

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Electronic organs having a keyboard can also derive their component tones from a twin triode, V_1 and V_2 in Fig. 292—which is the basis of the Hammond organ—connected to a frequency dividing

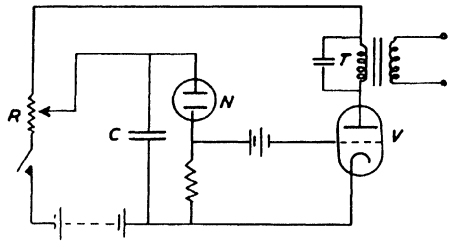


Fig. 291. Circuit of Trautonium.

valve V_3. The circuit is a multi-vibrator, with fundamental determined by L and C. The cathode of V_3 is so biased by the resistor R_3 that it feeds the control tube (a sharply cut-off pentode) with an appropriate

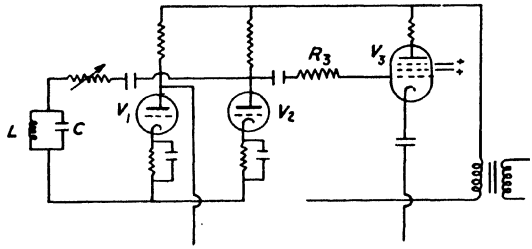


Fig. 292. Circuit of Hammond organ.

sub-harmonic of the primary oscillation. The operation of V_3 is controlled by the depression of the conventional pianoforte key. Adjustable keying of this valve (through appropriate delay resistances) adjusts the starting and stopping transients.

The third basic type of circuit is that used on the Theremin solo instrument. In this two radio-frequency oscillators produce a difference tone; normally itself ultrasonic, but when the capacity of the loop antenna in one of the oscillators is lowered by the approach of the hand or an earthed baton an audiofrequency beat tone results. In the original instrument, one tone alone was produced at a time, but

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subsequently the scheme was adapted to a manual of which each key when depressed, added a fixed capacity.

The other principal class of keyboard instrument, of which in Great Britain the Electrone is the archtype, uses electrostatic variation of the capacity in a valve oscillating circuit. One plate of the condenser may be a fixed pick-up (Fig. 293) and the other a disc or cylinder

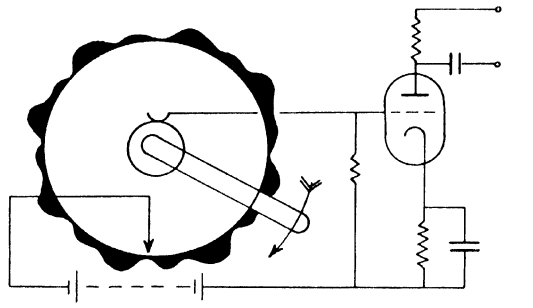


Fig. 293. Basic apparatus of Electrone.

shaped into a sinusoidal procession of hills and dales to generate a pure tone, or complex notes based on a suitable fundamental may be generated by the passage past it of a disc notched or provided with stepped excrescences (*cf.* Fig. 289). The components of the complex note must then be subsequently sorted out electrically.

To cover the whole of the musical gamut, it is usual to have one for each semi-tone in an octave of the equi-tempered scale and let there be sufficient harmonic divisions of the surface that all the notes of one denomination are provided for. Thus all the *C*'s will derive from one disc, all the *C#*'s from a disc rotating $2\frac{1}{12}$ faster, all the *D*'s from a disc rotating $2\frac{1}{6}$ faster than the first one, etc. For further detail, we quote from a paper by BOURN [123], the designer of the Electrone.

The Electrone has certain unique features. First the generator; this is, in essence, nothing more than a variable capacitor, rotated by a motor, in which the electrodes are shaped to a given wave-form. If the capacitance is polarised, an alternating e.m.f. is generated, corresponding in form to the shape of the electrodes. A radial system of conductors scans the wave-electrodes and frequencies are generated

corresponding to the speed of rotation and the integral numbers of waves arranged in circles. Twelve such generators are used, each identical in construction, but run at speeds to give the corresponding twelve semi-tones of an octave. In this manner it is possible to generate all the frequencies required for the whole instrument. The output from the twelve units is divided into bass and treble channels, amplified and taken to the loud-speakers.

Now for the keying; as the generators are inoperative except when polarised, no keying functions are performed in the actual electrical sound circuits, but instead they are in d.c. circuits and external to them. The fact that d.c. is used enables simple resistance-capacitor networks to be employed in order to modify the build-up and delay time of the notes. The keying circuits are not quite straightforward, as it must be remembered that this is a potential-controlled device and by doubling the voltage the output watts will be quadrupled. Should it happen that two keys require the same note each at the same strength when sounded singly, and if both are held together, the volts at the electrode must rise $\sqrt{2}$ times in order to give the desired 3 db increase of level. A potentiometer network is used for this purpose which produces correct addition to within 1 db for about twelve key positions. This however, is more than is necessary in practice. The stop circuits have a similar network and give approximately the desired increment even when stops contain similar ingredient tones. Swell pedals merely serve to vary the voltage applied to the manual divisions. One great advantage of d.c. keying is that cables need not be screened, adequate insulation only being necessary without any danger of cross-talk.

Regarding the tonal scheme; the use of independent harmonics gives almost complete choice of tone-colour, provided that a sufficient number are used and that they are pure, genuine harmonic frequencies. Naturally, harmonics derived from the nearest chromatic note cannot be carried very far, and even then the dissonant beats cause a roughness of tone which is most disagreeable. If true harmonics are generated, say, up to the 32nd, great complication of both generators and key circuits is involved. In pre-war Electrones, sixteen harmonics were used. The harmonics which nearly coincided with a semi-tonal interval were borrowed *e.g.* the 3,6,9 and 12, and, of course, the octaves. The 5, 7, 10, 11 etc., were separately generated.

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(2)This type of instrument starts with a vibrating string as prime-mover and picks up, electromagnetically or electrostatically, a current of complex wave-form (usually). The separate harmonics or various tone-qualities are sorted out electrically from these.

In many cases the acoustic prime-mover executes damped oscillations, in some it is maintained.

The quality picked-up will depend partly on the method of maintenance of the string or reed, as we have already discussed *passim*, and partly on the position of the pick-up, and its size in relation to the whole string. In this respect, the pick-up may be regarded as playing the reverse role of the bow (for example) on the violin. In particular, it will not pick-up any harmonic which has a node opposite to it on the string (*cf.* p. 453).

A pick-up can be electromagnetic if the vibrator is made of magnetic material or electrostatic, if the vibrator is conducting. They have been applied to conventional pianofortes, guitars, harmoniums. Instruments with mechanically plucked or struck reeds have also been devised and others in which the reed is excited magnetically on depressing the appropriate key.

In many cases then the sound of the acoustic vibrator decays, but that of the output need not. If circuits are incorporated which maintain the electronic current at constant level until the key is released the organ can be imitated, starting from a pianoforte, or indeed any required wave-envelope copied, including transients at both ends.

Fig. 294 (after MIESSNER [124]) shows some possible mechano-electric systems.

In (a) the magnetic field of the magnet threading the coil is modulated by vibrations of the steel string, thus producing a voltage vibration in the coil; if the string is magnetized the coil core need not be. In (b) the string vibration modulates its capacity to the pick-up conductor, and thus modulates either a steady voltage or a high frequency current. In (c) the string bears down with some pressure on a resistance or piezoelectric device and, as it vibrates, this pressure is modulated, thus varying the resistance, or generating a voltage in the piezocrystal. In (d) the string is a conductor vibrating in a magnetic field and thus has voltage generated in itself. In (e) the string is magnetized and its longitudinal extensions so disturb its magnetic field that a voltage is generated in the coil.

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Lack of rigidity of the pick-up device may introduce spurious, unwanted frequencies or noises, if itself can vibrate, or if it is attached to some part of the instrument which vibrates differently to the string-bridge system.

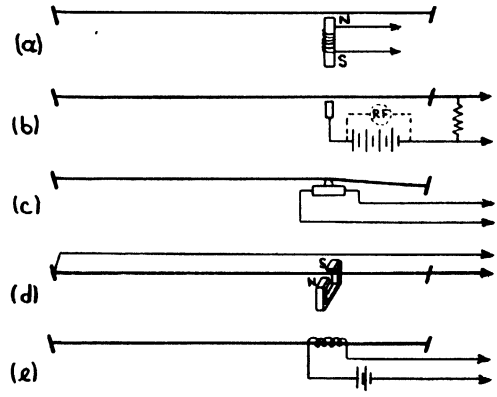


Fig. 294. Electromechanical tone generators employing strings.

MIESSNER finds that, for general purposes, a pick-up conductor, consisting of flat, one sixteenth inch, copper braid, glued to the narrow side of a strip of bakelite one eight inch thick by one half wide and securely fastened (conductor side up) to each of the wooden ribs, is a satisfactory type and mounting. This selection has, however, been rather largely influenced by the desire to secure, among other types of tone, one closely resembling that of the piano.

Three pick-ups on the Electrochord are used. One of these is very close to the back end of the string, at about one-sixteenth string length, where the high partials are prominent; another is placed at about the one-sixth-from-back position; a third is placed in a position about one third from the front end. To compensate to some degree for the irregular frequency response characteristics of loudspeakers, the strings are provided with individual pick-up spacing adjustment screws; these are mounted on the bridge and raise or lower the individual strings with respect to the pick-up strip. This adjustment operates much more effectively for the pick-up nearest the bridge than for the other two. A more perfect though more complicated

arrangement would provide fixed position strings and adjustable pick-ups for each string on each bar.

By suitable mixing of the output of these three pick-ups, as an artist uses his palette, any desired quality of the synthesised note from this one string may be derived.

Changing the damping factor to which we have referred is secured by a number of ingenious devices:- (a) the struck string may have another one in resonance alongside it and the pick-up be stationed on the latter. The damping of the resonant string is pre-adjusted mechanically; (b) the transmission efficiency of the (electrostatic) pick-up may be delayed by a changing polarisation voltage, so that it more effectively responds after the wire has been struck; (c) the wire may have a flat section (like a narrow strip) and be struck in such a direction that it tends to rotate in vibration, presenting its flat more-effective side to the pick-up after some delay.

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APPENDIX TO CHAPTER IV

*Absorption coefficients of some European building materials**

The materials have been classified in three classes, viz.:

I Acoustical plaster and other materials for plastic applications.

II Acoustical blankets, felts, loose material etc.

III Prefabricated units, tiles, fibre boards, panels etc.

The abbreviations used in the column 'Authority' stand for:

A et A	Ecole Nationale Supérieure d'Architecture et des Arts Décoratifs, Brussels, Belgium.
A et M	Conservatoire National des Arts et Métiers, Laboratoire d'Essais, Paris, France.
Berlin	Physikalisch-Technische Reichsanstalt, Berlin–Charlottenburg, Germany.
Berlin T.H.	Institut für Schwingungsforschung an der Technischen Hochschule, Berlin, Germany.
Göttingen	III Phys. Institut, Universität Göttingen, Germany.
L.L. Copenhagen	Lydteknisk Laboratorium, Copenhagen, Denmark.
N.P.L.	National Physical Laboratory, Teddington, Middlesex, England.
Stuttgart	Institut für Technische Physik, Stuttgart, Germany.
T.P.D.	Technisch Physische Dienst T.N.O. en T.H., Delft, Netherlands.
U.S.B.	United States Bureau of Standards, Washington, U.S.A.
reverb.	Reverberation method.
int.	Interferometer method.

* Completeness is not aimed at. American materials have not been tabulated, since they can be found in almost every textbook of American origin. See, for instance, V. O. KNUDSEN and C. M. HARRIS, *Acoustical Designing in Architecture*, p. 405, John Wiley & Sons, Inc., New York, 1950. Moreover, see: *Bulletin XIII*, Acoustical Materials Association, 59 East 55th Street, New York, N.Y., 1951, and *Handbook of Chemistry and Physics*, Chemical Rubber Publishing Co., Cleveland, Ohio.

Manufacturer or importer	Material	Class	Description
AUSTRIA			
Oesterreichisch-Amerikanische Magnesit Aktiengesellschaft	Heracustik-platten	III	Woodwool, magnesite binder
DENMARK			
International Isolations Kompagni A/S Ikas	Ikas A –17[3] Ø	III	Combination of hard and soft or hard and highly porous fibre boards: perforated
	Ikas A –17/A[3] Ø	III	perforated
	Ikas A –17/A[5] Ø	III	perforated
	Ikas AS–17/A	III	slotted
	Ikas E –19/A[3] Ø	III	perforated
	Ikas E –19/A[5] Ø	III	perforated
	Ikas E –54/A[5] Ø	III	perforated
Snedkermestrenes Træ- og Finerskæreri A/S	STF Akustipanel type 0	III	Combination of plywood, fibreboard and rockwood: unperforated
	STF Akustipanel type 1	III	unperforated
	STF Akustipanel type 2	III	perforated
	STF Akustipanel type 3	III	perforated
	STF Akustipanel type 4	III	slotted
	STF Akustipanel type 5	III	perforated
	STF Akustipanel type 6	III	perforated
	STF Akustipanel type 6b	III	perforated
	STF Akustipanel type 7	III	slitpanel of studs 44 mm; 6mm slits between studs
	STF Akustipanel type 8	III	slotted soft fibreboard
	STF Akustipanel type 9	III	slotted soft fibreboard
FRANCE			
Comptoir de Matériaux Acoustiques	Dalle Everson	III	Asbestos and mineral wool between asbestospaper, front surface perforated
Soc. Isover St. Gobain	Isover Superfine	II	Glasswool
	Isover F C Triple	II	Glasswool felt
	Isoson plâtre	III	Gypsum tiles, perforated with 8 mm holes at 20 mm centres
Soc. Française Knock-Out	Staff Verre K.O.	III	Hollow perforated gypsum tiles filled with glasswool
Stillite Française see Great Britain			
La Vermiculite	Vermex	I	Plaster, expanded mica with cement as binding agent
Vonex	Vonex	III	Perforated woodboard

Absorption Coefficients						Mounting	Thick-ness in mm	Authority	Weight	
125 c/s	250 c/s	500 c/s	1000 c/s	2000 c/s	4000 c/s					
15	23	23	51	73	75	rigid backing	25	Stuttgart and Berlin	11.5	kg/m²
23	29	32	73	74	77		35		14.5	kg/m²
29	32	37	75	74	79		50		19	kg/m²
13	46	43	53	59	39	22 mm air backing	17		6.2	kg/m²
12	33	52	58	72	47		17		5.6	kg/m²
3	12	56	70	90	67		17	L.L.	5.6	kg/m²
3	26	47	60	62	59		17	Copenhagen	5.6	kg/m²
9	19	70	69	43	24		19	reverb.	3.8	kg/m²
4	12	56	97	81	67		19		3.8	kg/m²
21	71	81	62	83	59	58 mm air backing	54		5.4	kg/m²
91	32	15	8	7	9	rigid backing	51		14	kg/m²
11	42	15	11	9	8		24		6.75	kg/m²
24	51	31	38	43	64		12		4	kg/m²
18	32	66	77	88	70		24	L.L	6.3	kg/m²
7	21	55	61	65	46		24	Copenhagen	6.6	kg/m²
21	45	71	52	69	80		23	reverb.	7.2	kg/m²
7	18	74	66	59	33		24		6.6	kg/m²
5	14	50	89	78	63		19			
21	52	50	52	52	36		35		11.15	kg/m²
20	80	45	61	83	70		25			
20	47	32	35	61	81		13			
	60	79	83	65	79	rigid backing	20	A et M reverb.		
	22	26	49	65	69	rigid backing	15	A et M	3	kg/m²
	37	43	65	66	67	rigid backing	30	reverb.	3	kg/m²
	58	62	69	86	90	rigid backing	40	A et M	1.3	kg/m²
	49	53	60	62	65	25 mm Isover felt backing	14	reverb.		
	37	71	70	57	65		75	A et M reverb.	44	kg/m²
	25	69	82						55–125	kg/m³
	60	70	90	55	36	25 mm Isover felt backing	6	A et M	5	kg/m²

Manufacturer or importer	Material	Class	Description
GERMANY			
E. Dyckerhoff	Dyckerhoff Akustik Platten	III	Woodwool with mineral binder
	Tekton Akustik Platten	III	Same as above
W. Genest	Hallonit Akustikplatten	II	Mineral wool slab
	Mando Akustikplatten	III	Slotted fibreboard
Grünzweig und Hartmann	Sillan	II	Long fibre rockwool
	Sillan Filtzplatten	II	Rockwool felt
	Novolan	I	Small fibre rockwool plaster
	Iporka Normal	II	Plastic foam 11–15 kg/m³
	Iporka Druckfest	II	Plastic foam 22–25 kg/m³
Kempchen R.	Resit		
Luneburger Faserwerke	Xylocal	II	Woodfibres
GREAT BRITAIN			
Burgess Prod. Cy. Ltd.	Burgess Acoustic Tiles 'A'	III	Pressed perforated metal tray, holes 2.4 mm, porous filling material
	Burgess Acoustic Tiles 'B'	III	Ditto, holes 3.2 mm
Cafferata and Co. Ltd.	Thistle plaster	I	Light-weight porous aggregate with gypsum binder containing a foaming agent
Fibreglass Ltd.	Superfine Fibre	II	Glasswool
	Resin Bonded Fibre	II	Bonded glasswool
Gyproc Products Ltd.	Dekoosto Acoustic Plaster	I	Porous and light-weight aggregates with gypsum plaster
	Acoustic Gyproc	III	Perforated gypsum plasterboard panel
	Gypklith	III	Compressed wood wool

Absorption Coefficients						Mounting	Thickness in mm	Authority	Weight	
125 c/s	250 c/s	500 c/s	1000 c/s	2000 c/s	4000 c/s					
11	25	50	65	58	58	rigid backing	25		9	kg/m²
19	48	59	58	57	56	50 mm air backing	25	Berlin T.H.		
22	34	60	55	54	64	rigid backing	50	reverb.		
24	57	60	54	57	69	50 mm air backing	50			
29	80	99	100	100	100	rigid backing	50	Stuttgart reverb.	200	kg/m³
32	60	55	57	68	86	50 mm air backing	20	Genest reverb.	6	kg/m²
	40	105	106	108	108	30% perforated hard fibreboard facing	50	Stuttgart reverb.	60	kg/m³
	15	23	53	88	94	rigid backing	30	Göttingen	110	kg/m³
	18	48	95	96	99	rigid backing	55	int.		
6	8	25	51	73	76	rigid backing	10		400	kg/m³
10	28	55	61	82	91	rigid backing	20			
8	12	23	55	66	62	rigid backing	30	Stuttgart	11	kg/m³
6	6	18	45	48	29	rigid backing	30	reverb.	22	kg/m³
30	50	65	70	80	81	rigid backing	40	reverb.	60	kg/m³
11	23	42	62	86	97	rigid backing	20	Stuttgart reverb.	100	kg/m³
15	50	75	80	75	75		45	N.P.L. reverb.	6.1	kg/m²
15	45	70	75	80	85		45		6.1	kg/m²
15	20	35	60	60	50	rigid backing	12	N.P.L. reverb.	54	kg/m³
10	30	60	80	85	65	covered with 10% perforated metal; rigid backing	25	N.P.L. reverb.	9.6	kg/m³
10	60	80	85	80	70		50			
10	35	55	65	75	80	unfaced; rigid backing	25		48	kg/m³
20	50	70	80	75	80		50			
	average coefficient 25									
15	70	90	70	45	30	backed by 25 mm rockwool and 25 mm air	10	N.P.L. reverb.	76	kg/m²
	45	75	80	65	85		25	N.P.L.	60	kg/m²

Manufacturer or importer	Material	Class	Description
Holoplast Ltd.	Holoplast 1″	III	Double panel, one side perforated, filled with kapok strips
	Holoplast $1^5/_8$″	III	Ditto, filled with 'Fibreglass' strips
	Holoplast Acoustic Tile	III	Ditto with 'Fibreglass' matt behind 1.6 mm lining perforated with slots
Jones and Broadbent	J & B Loose mineral wool	II	Inorganic fibres
	J & B Felted mineral	II	Felted and bonded inorganic fibres
	J & B Acoustic Panels	II	Loose mineral wool between wire netting and one side muslin cloth
May Acoustics Ltd.	Sabinite	I	Gypsum plaster
	'Maycoustic' Stone	III	Similar in composition as Sabinite
	'Maycoustic' Asbestonite	II	Composed of asbestos fibres
	'Cabot's Quilt'	II	Mineral wool
	'Maycoustic' Perforated Tile	III	Perforated asbestos facings with asbestos fibre backing
Newalls Insulation Co. Ltd.	Paxtiles	III	Asbestos fibres covered with plaster, perforated
	Paxfelt	II-III	Asbestos sheets
	Paxfelt Blanket	II	Asbestos blankets
	Asbestos Spray	I	Sprayed asbestos fibres
J. W. Roberts Ltd.	Limpet	I	Sprayed asbestos fibre
Stillite Ltd.	Stillite Scrim Quilt	II	Fibrised calcium magnesia in scrim cloth
	Semi-rigid Slab Acoustic Quality	II	Ditto felt
	Stilbat	II	Ditto
Thermacoustic Ltd.	Standard Thermacoustic	III	Selected woodwool with a mineral cementing agent
	Heavy Duty Thermacoustic	III	Ditto

Absorption Coefficients						Mounting	Thickness in mm	Authority	Weight
125 c/s	250 c/s	500 c/s	1000 c/s	2000 c/s	4000 c/s				
30	30	55	90	70	50	rigid backing	25	N.P.L. reverb.	9.8 kg/m² (without filling)
25	45	80	80	35	20	rigid backing	41	N.P.L. reverb.	11.3 kg/m² (without filling)
15	55	80	80	50	45	rigid backing	25	N.P.L. reverb.	
30	80	90	95	90	95		50	N.P.L. reverb.	160–190 kg/m³
35	75	95	95	95	95		50	N.P.L. reverb.	
30	80	90	95	90	95		50	N.P.L. reverb.	16 kg/m²
	20	37	60	61		rigid backing	13	U.S.B.	
	15	25	45	40		3 mm air backing	13	N.P.L. reverb.	
	55	85	80	70		covered with muslin, rigid backing	13	N.P.L. reverb.	
	42	74	77	69		38 mm air backing, canvas cover 25 mm distant		N.P.L. reverb.	
	70	65	65	75	70	rigid backing	30	N.P.L. reverb.	
	55	75	85	80		19 mm air backing	24	N.P.L. reverb.	7.5 kg/m³
	50	55	65	70	75	25 mm air backing	25	N.P.L. reverb.	144 kg/m³
	20	60	65	70		16 mm air backing	9.5	N.P.L. reverb.	144 kg/m³
	30	35	50	60			12.5	N.P.L.	
	60	65	60	60			25	reverb.	
25	25	45	75	75	70	asbestos sprayed on scratch coat	13		
30	40	65	80	75	75	of 13 mm asbestos cement;	25	N.P.L. reverb.	160–192 kg/m³
40	55	75	75	75	70	89 mm air backing	38		
50	70	90	90	85	80	rigid backing	50	N.P.L. reverb.	10 kg/m²
20	45	80	85	85	95	50 mm air backing	25	N.P.L. reverb.	200 kg/m³
15	55	70	80	85	85	muslin covered	50	N.P.L.	72 kg/m³
15	20	45	80	65	70	rigid backing	38	N.P.L. reverb.	19 kg/m²
20	30	80	75	75	75	rigid backing	50	N.P.L. reverb.	30 kg/m²

Manufacturer or importer	Material	Class	Description
NETHERLANDS			
Th. J. v. d. Berg	Atonex Tile	III	Perforated woodfibreboard
Combinex	Softex board 1 B.V. 2	III	Slotted woodfibreboard
	„ 2 B.V. 2	III	Ditto
N.V. Dekker's Houthandel	Prefocel Standard Tile	III	Perforated woodfibreboard
	Prefocel Extra Tile	III	Perforated woodfibreboard with ditto hardboard facing
Geldens N.V.	Pyrok foam	I	Mortar in which sand is replaced by light expanded mineral
Haarlemse Bouwstoffenhandel	Homaton	III	Perforated woodboard tile
Inverbo N.V.	Ankarboard	III	Perforated woodfibreboard
Loka C.V.	Kramfors	II-III	Bonded woodfibres
N.V. Nederlandse Durisol	Durisol (Mevriet)	III	Shredded wood, cement binder
N.V. Ned. Steenwolfabriek	Steenwolstuc	I	Mineral wool plaster
Nivam	Navitex	III	Strawfibre tile
Proost en Brandt	Acousti Celotex	III	Perforated woodfibre tile
	Acousti Tronal	III	Perforated woodfibre tile
Treetex N.V.	Superac I	III	Glasswool in hardboard enclosure; one surface perforated
	Treeperac	III	Perforated woodfibre tile
	Treetac	III	Slotted woodfibre tile
N.V. Utrechtse Asphaltfabriek	Acboma	I	Woodfibres with cement and lime binder
	Acboma (new composition)	I	Woodfibres with diatomite

Absorption Coefficients						Mounting	Thick-ness in mm	Authority	Weight
125 c/s	250 c/s	500 c/s	1000 c/s	2000 c/s	4000 c/s				
42	46	42	56	66	72	25 mm air backing	16	T.P.D. reverb.	
15	42	44	58	72	73	25 mm air backing	12.5	T.P.D.	
23	51	54	75	83	78	25 mm air backing	24	reverb.	
28	64	66	79	88	84	25 mm air backing	19	T.P.D. reverb.	4 kg/m²
28	37	42	66	60	66	25 mm air backing	22	T.P.D. reverb.	7 kg/m²
	12	34	48	39		sprayed on hard, rigid backing	24	T.P.D. int.	
	16	30	43	47		rigid backing	19	T.P.D.	
	50	46	35	41		25 mm air backing	19	int.	
20	48	53	68	71	64	25 mm air backing	19	T.P.D. reverb.	
22	48	98	100	88	70	rigid backing	40	T.P.D. reverb.	2.4 kg/m²
24	43	79	68	72		33 mm air backing	50	T.P.D. reverb.	
14	19	34	50	59		on 15 mm ceiling;	8	A et A	
26	30	44	65	84		60 mm air backing	15	reverb.	
11	50	50	58	66	60	25 mm air backing	12.5		
20	52	52	62	68	64	25 mm air backing	19	T.P.D.	
25	60	65	72	74	62	25 mm air backing	25	reverb.	
38	62	70	76	68	54	25 mm air backing	31		
	12	36	71	39		rigid backing	20	T.P.D.	
	20	53	68	37		rigid backing	30	reverb.	
	13	38	57	51		rigid backing	20	T.P.D.	
	24	50	66	54		rigid backing	30	reverb.	
31	44	56	53	37	55	rigid backing	28	T.P.D. reverb.	
8	60	83	92	65	92	rigid backing	25	T.P.D. reverb.	
15	66	67	88	78	56	rigid backing	19	T.P.D. reverb.	
		4	6.5	4.5		rigid backing	15	T.P.D. int.	
		21	18.5	21		rigid backing	15	T.P.D. int.	

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