CHAPTER

Cellphone acoustics

8

CHAPTER OUTLINE

PART XXVI: ACOUSTICAL TRANSDUCERS FOR CELLPHONES

The number of cellphones in the world is approaching 6 billion. These range from those that provide simple telephone services to those that serve as full business and entertainment centers. For all these, acoustical design is very important, but such design is involved because of limitations of space, ergonomics, and frequently the requirement of producing good-sounding music.

Shown in Fig. 8.1 is a cellphone with the characteristics of an entertainment center. The three main electroacoustic elements are the handsfree loudspeaker, the call loudspeaker and the microphone. The basic principles for the loudspeakers are the same as that for a loudspeaker in a closed baffle as presented in Chapter 7.

8.1 LOUDSPEAKER AND MICROPHONE

Basic considerations. The sound-producing part in all cellphones is a flat radiating surface, i.e., the diaphragm, which is analogous to the cone in a hi-fi loudspeaker. The diaphragm is set in motion by an attached moving coil located in a magnetic field. The diaphragm and the coil may be circular or square. In the cellphone of Fig. 8.1, there are two flat diaphragms and driving coils, one of which feeds into the handsfree loudspeaker opening and the other into the call receiver opening. The small fully enclosed space behind either diaphragm is analogous to an acoustic compliance. The combined mass of the diaphragm and coil is analogous to an acoustic mass. These elements at resonance determine the low-frequency cutoff of the radiated sound.

Handsfree loudspeaker. In the simplest cellphones, the ringtone is a buzz or a repeated ring resembling that of an old landline phone. In an entertainment center type, the ringtone is often an excerpt from a musical composition. Also, continuous music may be played through it. As shown, the

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FIG. 8.1 Cellphone Nokia model Lumia 800 showing the positions of the call loudspeaker, handsfree loudspeaker and microphone openings.

Courtesy of Nokia OY. Photograph by Enrico Pascucci.

handsfree loudspeaker opening is on the side of the cellphone so that the phone can be laid on a flat surface and thus is “handsfree.” If the surface is large enough, the energy output of the loudspeaker will be double that compared with the handheld position because the opening is radiating into half space. Like any small closed-box loudspeaker the trend in cellphones is toward the “acoustic suspension” concept, i.e., the stiffness of the suspension for the diaphragm is a small fraction of that of the airspace behind. The volume of the backspace is about 1 cm^3. This design is only practical if the diaphragm is fairly stable and the resonant frequencies are constant. Surprisingly, in the cellphone of Fig. 8.1, the user can carry on a telephone conversation with it on a surface, with the talker’s voice using the microphone next to the handsfree opening. This is possible because the output of the microphone has an “echo cancelling” feature in the digital circuitry and its output is also attenuated while the loudspeaker is operating. Current day handsfree loudspeakers are very loud, often producing as much as 105 phons at a distance of 10 cm.

Call loudspeaker. The call loudspeaker is similar to the handsfree loudspeaker. In the unit of Fig. 8.1, the volume of the space behind its diaphragm is larger and it contains the electronics and the keyboard. There is greater chance of leakage from this space and, if it should occur, the low frequency output will be diminished. The opening is always held close to the ear, which mitigates the effect of leakage from the rear space.

Microphone. The microphone is an electret or MEMS type, of which more will be said later. It is only actuated when voice transmission is required.

8.2 CIRCUIT DIAGRAM FOR A CELLPHONE LOUDSPEAKER

The circuit diagram for a cellphone loudspeaker system is given in Fig. 8.2. The elements are derived from the circuit of a loudspeaker in a closed box baffle as given in Fig. 7.6. The symbols are as follows:

$\tilde{e}_g$ is open-circuit voltage of the generator (audio amplifier) in volts (V).
R_g is generator resistance in electrical ohms (Ω).
R_E is resistance of voice coil in electrical ohms (Ω).
B is steady air-gap magnetic field or flux density in Tesla (T).
l is length of wire on the voice-coil winding in m.
$\tilde{i}$ is electric current through the voice-coil winding in amperes (A).
a is radius of diaphragm in m.
$S_D = \pi a^2$ is area of diaphragm in m^2.
$\tilde{P}_c \cdot S_D$ is force produced by the current in the coil in $Pa \cdot m^2$.
$\tilde{U}_c$ is volume velocity produced by the diaphragm in m^3/s.
M_{AD} is acoustic mass of the diaphragm and the voice coil in kg/m^4.
C_{AS} is total acoustic compliance of the suspensions in m^5/N.
R_{AS} is acoustic resistance of the suspensions in $N \cdot s/m^5$.
C_{AB} is total acoustic compliance of the back enclosure in m^5/N.
C_{AF} is acoustic compliance of the front cavity in m^5/N.
R_{AB} is acoustic resistance of the leak path through the enclosure (needed to relieve changes in atmospheric pressure) in $N \cdot s/m^5$.
M_{AH} is acoustic mass of the sound hole(s) in kg/m^4.
R_{AH} is acoustic resistance of the dust screen in $N \cdot s/m^5$.
Z_{AE} is radiation impedance (including the effect of proximity to the ear)

The radiation impedance, Z_{AE} is difficult to specify because the impedance for the call loudspeaker opening is highly dependent on how the user uses the handset. Because both of the openings are small, the radiation impedance when not too near the ear will approximate that for a small diaphragm in the end of a tube. Possible means for assuring a known radiation impedance in the call loudspeaker

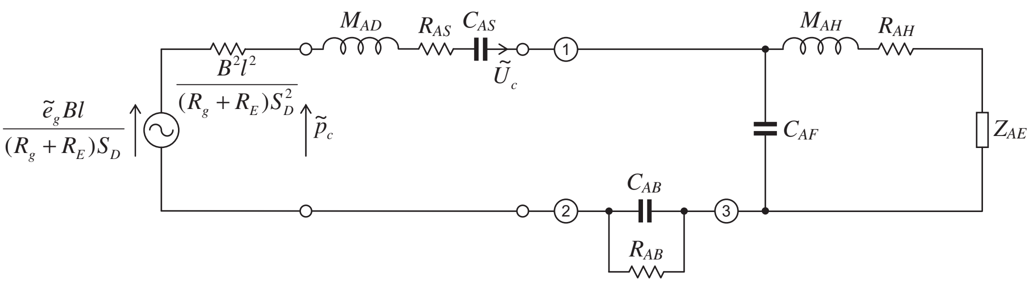

FIG. 8.2 Analogous circuit of call or handsfree loudspeaker in a cellphone.

All circuit elements are referred to the acoustical side. In the case of a call loudspeaker, Z_{AE} is the impedance of the ear including any leakage since it is unlikely to be sealed. In the case of a handsfree loudspeaker near a flat surface, Z_{AE} can be considered to be the radiation impedance of a piston in an infinite baffle.

opening is to deliberately build a degree of controlled leakage that is at least as great as the uncontrolled leakage that would occur in normal usage. The controlled leakage path may lead to the outside space as shown in Fig. 8.3a or to the internal space as shown in Fig. 8.3b. The more probable solution is that shown in Fig. 8.3a and for it a series acoustic resistance and acoustic mass must be connected between the circled "1" and "3" in Fig. 8.2. For the solution of Fig. 8.3b the mass and resistance should be connected between the circled "1" and "2". Obviously the controlled leakage addition will reduce the output strength. The solution of Fig. 8.3b in particular will cause loss of low frequencies due to the acoustic short-circuit between the front and back of the diaphragm unless the space inside the handset is very large ($> 60\ \text{cm}^3$).

Acoustic low-pass filter (Helmholtz resonator). In cellphones, for both the handsfree loudspeaker and the call loudspeaker, the compliance of the front cavity C_{AF} and the mass of the sound opening M_{AH} form a Helmholtz resonator. This is a 2nd-order low-pass filter. The angular resonance frequency is

$$\omega_0^2 = 1/(M_{AH}C_{AF}). \tag{8.1}$$

When listening to music the resonance frequency is normally set at the upper limit of the required frequency range and the Q of the resonance is controlled by the resistance R_{AH} of the dust screen. To calculate the dimensions of the resonator, we can either choose the radius a of the opening and

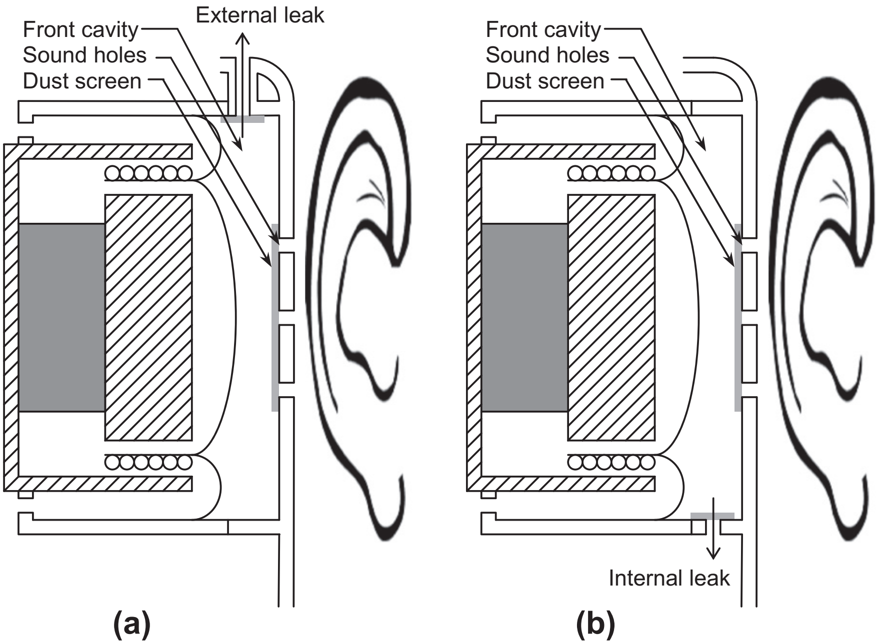

FIG. 8.3 Cross-section of leak-tolerant call loudspeaker in cellphone with (a) external leak and (b) internal leak [20].

calculate the length l according to $l = n\,c^2 a^2/(4\pi f_0^2 V) - \zeta a$ or choose the length and calculate the radius according to

$$a = \frac{2\pi\zeta f_0^2 V}{nc^2}\left(1 + \sqrt{1 + \frac{nc^2 l}{\pi\zeta^2 f_0^2 V}}\right). \tag{8.2}$$

These are general formulas for a Helmoltz resonator, such as a bottle, which are derived from those given in Secs. 4.2 and 4.3 for an acoustic mass M_{AH} and acoustic compliance C_{AF} respectively.

The quantities are:

l is length of opening in m.
a is radius of opening in m.
ζ is end correction factor.
f_0 is resonance frequency in Hz.
V is volume of cavity in m^3.
c is speed of sound = 348.8 m/s at $P_0 = 10^5$ Pa and $T = 22°$C.

The end correction factor for the opening is given by

$$\zeta = \begin{cases} 1.28, & \text{unflanged at both ends} \\ 1.49, & \text{flanged at one end, unflanged at the other} \\ 1.7, & \text{flanged at both ends} \end{cases} \tag{8.3}$$

8.3 DESIGN CONSIDERATIONS

Dust screens. All of the openings, for the loudspeakers and microphone, must be covered by a dust screen, which not only protects against the ingress of magnetic dust, but also helps damp out the various Helmholtz resonances. Data on dust screens is given in Sec. 4.4.

Magnetic fields. High strength neodymium magnets have generally improved the performance of miniature loudspeakers; however, their proximity to sound holes means that magnetic dust can be sucked in and clog the coil gap. Magnetic fields can also affect magnetic strips on credit cards.

Acoustic shock. The location of a loudspeaker opening is very important because, if it is tightly held against the ear, damage to hearing may result. Loudspeaker openings are often located on the sides of a phone so that they cannot be sealed against the ear. Another frequent location is on a rear surface that is curved, which also prevents closure of the opening when laid on a table.

Protection against damage to the loudspeaker. It is vital that all seals be secure. If the back enclosure leaks, the stiff cushion of the air that limits the excursion of the diaphragm no longer exists and physical damage may result. At low frequencies the generated sound pressure is limited by diaphragm excursion and at high frequencies by power dissipation. The input voltage to a voice coil is often limited according to a DSP (digital signal processor) model [1] of the loudspeaker parameters and how they change according to conditions such as temperature. Also, audio peak clipping, called dynamic range compression, is used extensively to make the sound louder.

Turbulence. At low particle velocities, the flow of air in a tube is linear. In other words, the velocity increases with radial distance from the wall as discussed in Chapter 4. As the velocity is increased beyond a certain point, the flow becomes turbulent with the formation of chaotic eddies and vortices. The Reynolds number is given by

$$\mathrm{Re} = \frac{\rho_0 u d}{\mu} \tag{8.4}$$

where

ρ_0 is density of air in kg/m^3
u is particle velocity in m/s
d is diameter of tube in m
μ is viscosity coefficient in N·s/m^2 [see Eq. (4.23)]

The point at which the flow becomes turbulent varies considerably with both geometry and surface finishes. As a rule of thumb, the Reynolds number should be kept below 1000 and the particle velocity below 10 m/s. All surfaces should be smooth with no sharp edges.

Wind noise [2]. The location of the microphone on the handset is critical with respect to wind noise. The worst location is on a large flat area such as a front cover because there is greater boundary layer turbulence than on a small area. The bottom edge is better and the best position is away from the center of the bottom edge. The direction of the wind also makes a significant difference. When the microphone is on the surface facing the wind there is generally less noise than when it is facing away, because turbulence builds up as air flows past an obstacle. Trailing-edge vortex shedding also contributes to wind noise, but to a lesser degree than boundary layer turbulence. In general, all surfaces should be as smooth as possible with no sharp edges.

Handling noise. Handling noise is generated by friction when a handset is held. Smooth surfaces produce less noise than rough ones. The noise is amplified by structural vibrations within housing as well as normal modes within the internal space, although the latter will be reduced by the presence of electronic components. Therefore, the housing should be either rigid, well damped or both. Adding ribs to the inner surfaces of the housing increases its rigidity. These precautions will also help to reduce unwanted resonance peaks and dips in the handsfree loudspeaker and call loudspeaker frequency responses.

8.4 HEAD AND TORSO SIMULATOR

The most commonly used device for testing the call loudspeaker of a cellphone is a head and torso simulator or HATS which has a pair of "soft" ears (pinna) on a head with shoulders, as shown in Fig. 8.4. The HATS microphone is located at the eardrum position. Unfortunately, acoustical simulation of the head and torso simulator is not so simple because to date there is no reliable equivalent circuit available using lumped elements. Not only is the model complicated by leakage into open air, but the loading varies with force and handset position. Therefore finite element modeling would have to be used. However, once a finite element model has been made for the acoustic path between the call loudspeaker and microphone, which is mainly the ear plus the space surrounding it and the phone, the results can be converted into a 2-port model. The 2-port model is then imported into an equivalent circuit for the call loudspeaker and microphone.

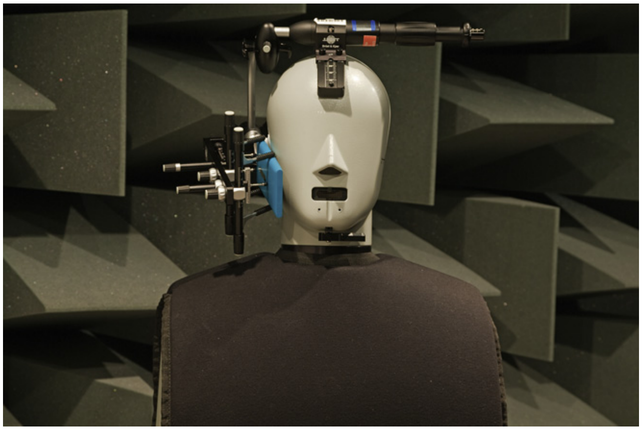

FIG. 8.4 B&K type 4128D head and torso simulator with cellphone Nokia model Lumia 800.

The simulator has a sound source inside the mouth and a microphone inside the ear.

Courtesy of Brüel & Kjær Sound & Vibration Measurement A/S and Nokia OY. Photograph by Enrico Pascucci.

It should be noted that the head and torso simulator is a multipurpose device. Even though the head beyond the immediate vicinity of the ear has little influence on the testing of a call loudspeaker, it is important for testing the microphone in a cellphone, in which case the sound source is in the mouth. Because the call loudspeaker opening is not in tight contact with the ear opening, it is probably sufficient to assume that the radiation impedance for it is similar to that for a diaphragm in the end of a tube. In testing, the head and torso simulator has largely replaced the older ear simulators which looked nothing like real human ears. They were originally used for testing of the receivers in a conventional phone handset. For readers who are interested to find out what the impedance of an ear or that of an older simulator looks like, the data sheets [3] for the B&K 4185 sealed ear simulator or B&K 4195 non-sealed ear simulator with IEC low-leak and high-leak couplers are informative.

8.5 MICROPHONES

Electret microphones. Until the electret condenser microphone (ECM) was invented by Sessler and West [4–7], all condenser microphones required a polarizing voltage supply. Such condenser microphones are treated in Sec. 5.5. The name electret literally means "**electr**ostatic magn**et**". It is a dielectric in which a permanent charge is embedded. In Sessler and West's invention, as shown in Fig. 8.5(a), the dielectric was a metalized Teflon foil which formed the diaphragm and the electrode was a stationary plate located behind it. In recent years, the charge storage capacity has been increased by using porous membranes [8]. Metalized electrets have a dipole charge because a charge of opposite polarity to that contained by the membrane is induced in the metallic coating. Although this gives a more stable charge, it means that, for a given membrane charge, the resulting field decays with increasing electrode separation, whereas in an externally polarized condenser microphone the field

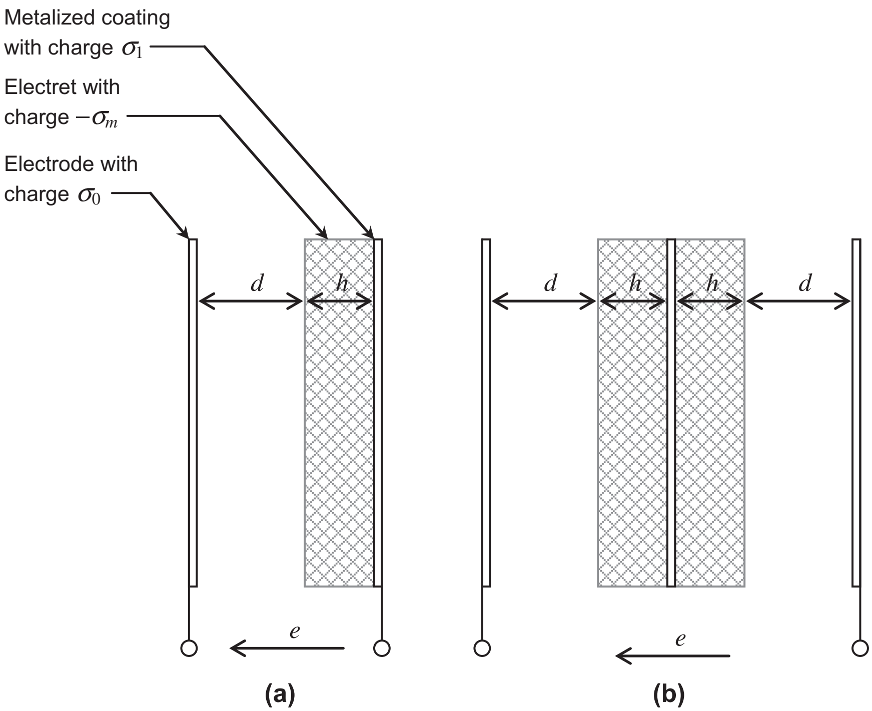

FIG. 8.5 Simplified cross-section of an electret transducer: (a) Single-ended (b) Push-pull.

remains constant. However, this is not a major issue in a microphone where the electrode spacing is usually very small. The capacitance of the electret membrane is given by

$$C_M = \frac{\varepsilon_0 \varepsilon_r S}{h} \tag{8.5}$$

where ε_0 is the permittivity of air, ε_r is the relative permittivity of the dielectric, S is the area of the membrane and h is the thickness of the electret membrane. This voltage then also appears across the air gap when the microphone terminals are at the same potential. The capacitance of the air gap is given by

$$C_G = \frac{\varepsilon_0 S}{d} \tag{8.6}$$

where d is the width of the gap. Then the total capacitance across the input terminals is

$$C_{E0} = \frac{C_M C_G}{C_M + C_G} \tag{8.7}$$

The negative membrane charge is usually expressed as a charge per unit area $-\sigma_m$, or charge density, so that the total charge is $-\sigma_m S$. The charge induced in the electrode is given by

$$\sigma_0 = \frac{C_G \sigma_m}{2(C_M + C_G)}. \tag{8.8}$$

The charge induced in the membrane coating is given by

$$\sigma_1 = \frac{(2C_M + C_G)\sigma_m}{2(C_M + C_G)}. \tag{8.9}$$

so that when $d = \infty$, there is zero charge on the electrode ($\sigma_0 = 0$) and the charge on the coating is equal and opposite to the electret charge, so that $\sigma_1 = \sigma_m$. In general, the total induced charge on the electrode and coating is equal and opposite to the electret charge. When $d = 0$, the induced charge is shared equally so that $\sigma_0 = \sigma_1 = \sigma_m/2$. Although the charge may be distributed throughout the electret, we may model it as a concentrated layer somewhere near the middle, depending on how symmetrical the charge distribution is. Inevitably some charge is lost both near the outer surface and near the coating, where there will be some recombination of positive and negative charge. Hence we may treat the concentrated charge layer and coating as electrodes of a capacitor across which there is a polarizing voltage E, where the dielectric thickness is $h/2$, so that

$$E = \frac{S\sigma_m}{2(C_M + C_G)}. \tag{8.10}$$

Hence we can use the same equivalent circuits as those for an externally polarized electrostatic microphone shown in Fig. 5.20.

The voltage sensitivity versus average diaphragm displacement $\Delta\delta$ is given by [9]

$$\Delta e_{in} = -\frac{E}{d}\Delta\delta = -\frac{h\sigma_m}{2\varepsilon_0(\varepsilon_r d + h)}\Delta\delta \tag{8.11}$$

It is well known that single-ended electrostatic transducers are nonlinear because the charge varies with displacement. This is not such a problem with microphones at moderate sound pressures because the displacement is very small and hence almost linear. However, for microphones at high sound pressures or loudspeakers, which have to displace a significant volume of air, a linear transducer may be created using the constant-charge push-pull principle. All conventional electrostatic loudspeakers with external polarizing supplies use this principle, which was first proposed by Frederick Hunt and then commercialized by Peter Walker [10–12]. An electret equivalent of this is shown in Fig. 8.5b, which has been proven to be linear both theoretically [9] and experimentally [13]. Figure 8.6 shows a cutaway view of an electret microphone.

So far we have only considered the foil electret microphone. In fact there are three types:

Foil-type or diaphragm-type. The diaphragm itself is made of an electret dielectric. However, the electret may not be strong enough to maintain tension over a long period, especially if it is porous, and may therefore require support from an additional membrane.

Back electret. The electret film is adhered to the back plate of the microphone capsule, which forms an electrode, and the diaphragm is made of a metalized but uncharged material.

Front electret. This is a newer design, which is essentially the reverse of the back electret. The back plate is eliminated from the design, and the condenser is formed by the metalized diaphragm and the inside surface of the capsule. The electret film is adhered to the inside front cover, which is perforated to let sound through.

The circuit for an electret microphone shown in Fig. 8.7 is much simpler than that for an externally polarized microphone, shown in Fig. 5.18, because the electret provides both the polarization for the capsule and the bias for the FET.

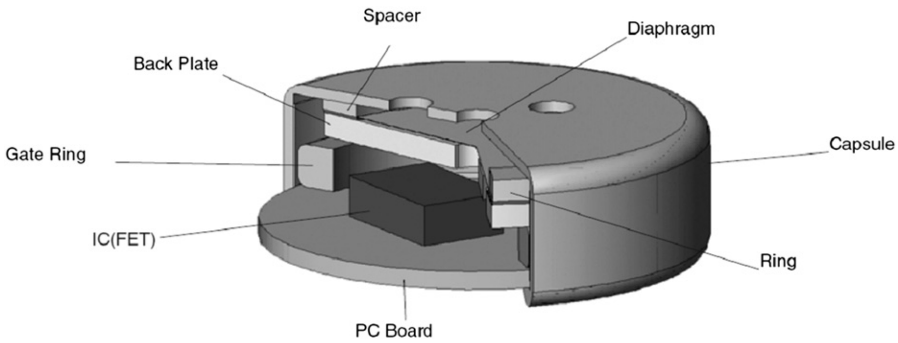

FIG. 8.6 Cutaway view of an electret microphone.

Courtesy of Hosiden.

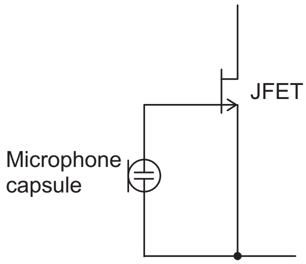

FIG. 8.7 FET circuit for an electret microphone.

When electret condenser microphones were first introduced, they were only expected to last for the lifetimes of the products in which they were used. However, with improved materials and processes, the ability to store charge over long periods has steadily increased. The loss of charge is compensated for by slackening of the diaphragm. Hence the sensitivity change over 28 years [14] has been found to be less than 1 dB.

MEMS microphones. At the time of writing, most low-cost cellphones use analogue electret microphones while mid-market models use both analogue micro-electro-mechanical-system (MEMS) microphones and digital electret microphones, where the latter are analogue electret microphones with onboard analogue-to-digital converters. This has the advantage of immunity to electrical noise pickup by the connections between the microphone and baseband chip. Many top-end models now use digital MEMS microphones because these have better signal-to-noise ratios (typically 69 dB). Also, MEMS microphones are smaller and can be surface mounted, which is an advantage in products that may have stereo microphones for recording as well as multiple microphones for noise-cancellation. The sensitivity of a MEMS microphone is typically −38 to −42 dBV/Pa and the acoustic performance is very stable.

A cross-section of a MEMS microphone [15, 16] is shown in Fig. 8.8. Although its operation is the same as that of a conventional capacitor microphone, as described in Sec. 5.5, the fabrication is somewhat different and is more akin to that of an integrated circuit. The substrate is made of silicon.

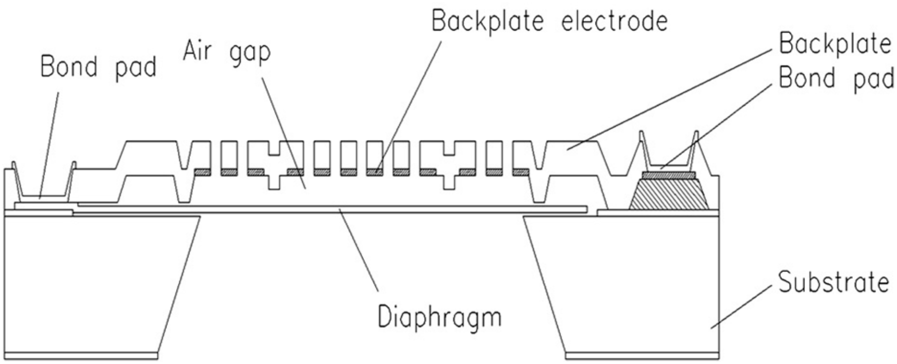

FIG. 8.8 Cross-section of a SiSonic™ MEMS microphone.

Courtesy of Knowles Electronics, LLC.

On this rests a "free" diaphragm of poly-silicon which is 1 μm thick. The patented free diaphragm design avoids film stress which would otherwise stiffen it. The backplate has a 0.5 μm layer of poly-silicon coated with a 1.5 μm layer of low-stress silicon nitride. The various layers are deposited and etched to form the entire structure. The gap between the diaphragm and substrate is created by means of a sacrificial layer of phosphorous doped glass, which is removed using an HF acid release process. The same method is also used to create the gap between the diaphragm and backplate. A detailed view of the backplate and diaphragm is shown in Fig. 8.9.

Because the cost of silicon wafer is fixed, the price of each MEMS microphone depends upon how many can be made from a single wafer. Hence, it is necessary for the diaphragm to be small for economic reasons as well as for miniaturization. The diaphragm is typically 0.6 mm in diameter with

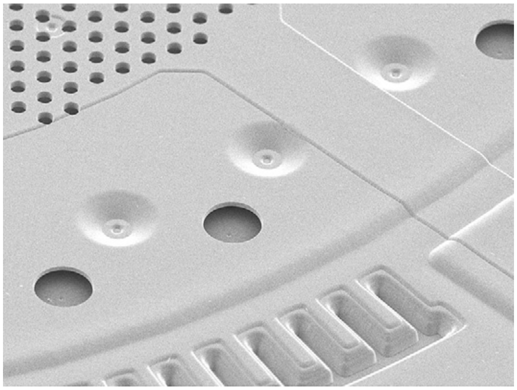

FIG. 8.9 Detailed SEM photograph of a SiSonic™ MEMS microphone.

The diaphragm is visible through the holes in the backplate. The small holes provides acoustic damping due to viscous flow losses.

Courtesy of Knowles Electronics, LLC.

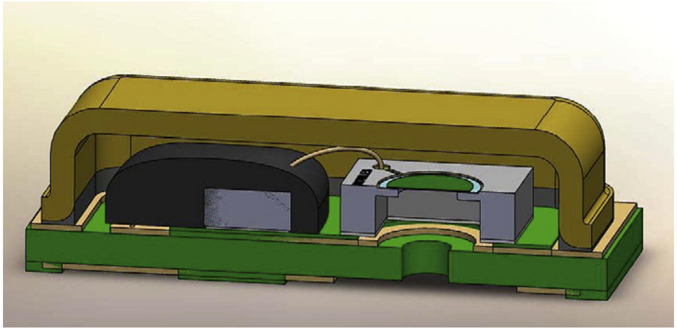

FIG. 8.10 Cross-section of a SiSonic™ MEMS microphone inside its package which provides an environmental and interference shield.

The integrated circuit, which provides the polarizing voltage and amplification, is on the left. Courtesy of Knowles Electronics, LLC.

a gap of about 4 μm. The polarizing voltage of 11 V is developed by a charge pump within a separate integrated circuit (see Fig. 8.10), which also contains the amplifier and analogue-to-digital converter, if there is one.

PART XXVII: TYPE APPROVAL TESTING OF CELLPHONES

The most commonly used standard for the acoustical testing of cellphones today is the 3GPP Technical Specification [17,21], which is part of the overall set of specifications for the 3rd generation network, although many network operators have their own type approval requirements which are usually variations of or additions to 3GPP. However, this only provides the acceptable limits for the test results. The test set-ups and methods for calculating the results are specified in other documents [18, 19].

8.6 MEASUREMENTS FOR TYPE APPROVAL

Since there are many measurements in type approval testing, we shall only discuss the basic acoustic ones.

Frequency response. Here we shall confine ourselves to broadband telephony (100 Hz to 8 kHz) as opposed to narrowband (100 Hz to 4 kHz). In order to measure the send frequency response, the sound source is calibrated by means of a microphone holder attached to the head and torso simulator (see Fig. 8.4). Then the calibration microphone is attached to the microphone holder with the diaphragm at the mouth reference point (MRP), which is 25 mm in front of the lip plane. The sound source is equalized to give a constant sound pressure of typically −4.7 dB Pa at all test frequencies. The frequency response is measured by connecting an analyzer to the point of interconnection (POI). In a land-line phone, this is an analogue connection to the public switched telephone network (PSTN), but in a cellphone it is usually the digital interface point (formerly digital air interface). The measured

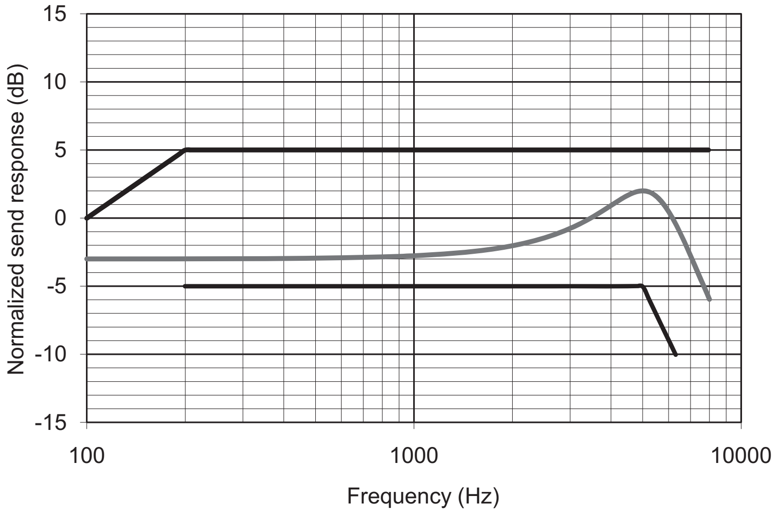

FIG. 8.11 Typical send frequency response (gray) with mask (black).

frequency response is plotted between upper and lower limits known as a mask. Thus in order to meet the requirements of type approval, the plot must fall within this mask. A typical frequency response is shown in Fig. 8.11.

In order to measure the receive frequency response, a signal is applied to the POI, and the microphone at the DRP within the HATS is connected to an analyzer. A typical frequency response is shown in Fig. 8.12.

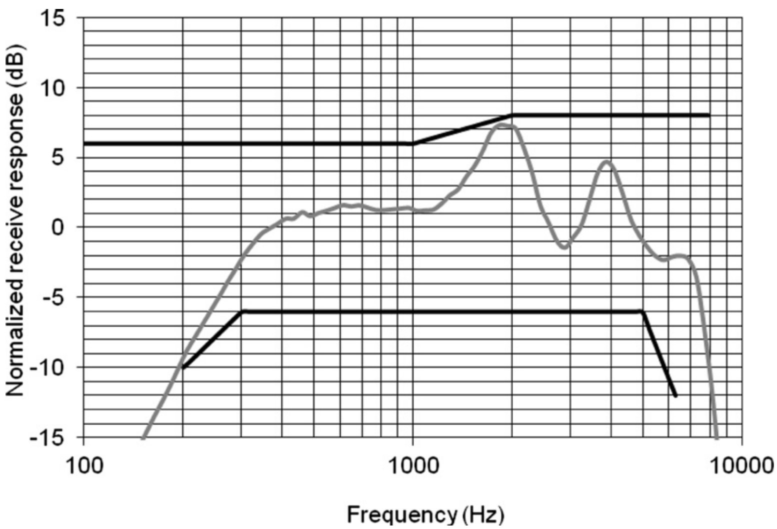

FIG. 8.12 Typical receive frequency response (gray) with mask (black).

Loudness rating. The loudness rating LR is calculated from the measured response over a set of frequencies which are spaced 1/3 octave apart according to the following formula:

$$LR = -\frac{10}{m}\log_{10}\sum_{i=N_1}^{N_2} 10^{0.1m(S_i - W_i)} \tag{8.12}$$

where

m = 0.175 is a constant
S_i are the sensitivities at frequency f_i of the send or receive electro-acoustic path
W_i are the weighting factors given in Table 8.1.

Notice the minus sign in Eq. (8.12), which means that the larger the loudness rating, the quieter the phone.

However, care has to be taken in evaluating the sensitivities from the digital POI of a cellphone. They are expressed in dB re V/Pa in the send direction and dB re Pa/V in the receive direction, which are analogue quantities. This is fairly straightforward in a land-line phone where we are simply concerned with the analogue voltage across a 600 Ω termination. In a digital system, it is convenient to use the maximum "full scale" level before clipping as the reference point, or 0 dBFS. This has been defined as equivalent to the voltage which is required to deliver 2.06 mW of power into a 600 Ω load. Hence

$$0\text{ dBFS} = 3.14\text{ dBm0} = 0.922\text{ dBV (or 1.112 Vrms) for send and receive} \tag{8.13}$$

TABLE 8.1 Weights for wideband SLR and RLR calculations.

***Frequency* (Hz)**	**W_{Si} (SLR)**	**W_{Ri} (RLR)**
100	103.0	115.4
125	75.3	87.5
160	60.2	72.3
200	59.5	72.1
250	52.9	67.2
315	59.4	75.8
400	45.4	63.6
500	56.6	74.6
630	53.5	70.4
800	53.8	69.9
1000	55.9	70.9
1250	64.2	78.4
1600	60.6	74.9
2000	73.7	85.2
2500	70.4	81.6
3150	87.1	95.4
4000	68.2	77.0
5000	84.5	91.7
6300	86.5	92.4
8000	71.0	89.0

For type approval, the send and receive loudness ratings must be:

$$SLR = 8 \pm 3 \text{ dB}$$

$$RLR = 2 \pm 3 \text{ dB}$$

Sidetone. When we talk, we normally hear our own voices via the external acoustic path between our mouths and ears. It is unnatural to have one ear obscured by a cellphone, so the acoustic path is replaced by an electrical one between the microphone and call loudspeaker. This is known as sidetone. There are two kinds, one being a deliberate path within the handset and the other the result of echo from an imperfect analogue interface, or hybrid, to a two-wire line within the network. The latter is minimized through use of a digital echo canceller because delayed sidetone is extremely irritating to the caller. Other sidetone paths may be carried by the mechanical structure of the phone or the space within it.

The sidetone rating is calculated using Eq. (8.12) as before except that the sensitivities S_i are for the path between the MRP and DRP and are thus dimensionless because pressure is measured (in Pa) at both points. Also, $m = 0.225$ and the weights are given in Table 8.2.

There are two sidetone ratings. One is the sidetone masking rating (STMR) and the other is the listener sidetone rating (LSTR). In the case of the STMR, the test signal comes from the mouth simulator, whereas in the case of the LSTR it is an external diffuse field. Hence the latter is a characterization of the room noise picked up via the electrical sidetone path. However it is often

Table 8.2 Weights for Wideband STMR and LSTR Calculations

Band No.	Mid-frequency (Hz)	W_i
1	100	115.4
2	125	87.5
3	160	72.3
4	200	72.1
5	250	67.2
6	315	75.8
7	400	63.6
8	500	74.6
9	630	70.4
10	800	69.9
11	1000	70.9
12	1250	78.4
13	1600	74.9
14	2000	85.2
15	2500	81.6
16	3150	95.4
17	4000	77.0
18	5000	91.7
19	6300	92.4
20	8000	89.0

more useful to evaluate the listener's sidetone performance of a handset indirectly by the difference:

$$D = LSTR - STMR \tag{8.14}$$

where D is a parameter of the phone which is *independent* of the network.

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