CHAPTER 1

Sound and Sound Sources

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Abstract: There is no difference in principle between the infrasonic and ultrasonic sounds which are inaudible to humans (or other animals) and the sounds that we can hear. In all cases, sound is a wave of pressure and particle oscillations propagating through an elastic medium, such as air. This chapter is about the physical laws that govern how animals produce sound signals and how physical principles determine the signals' frequency content and sound level, the nature of the sound field (sound pressure *versus* particle vibrations) as well as directional properties of the emitted signal. Many of these properties are dictated by simple physical relationships between the size of the sound emitter and the wavelength of emitted sound. The wavelengths of the signals need to be sufficiently short in relation to the size of the emitter to allow for the efficient production of propagating sound pressure waves. To produce directional sounds, even higher frequencies and shorter wavelengths are needed. In this context 'short' is measured relative to the size of the sound source. Some sound sources, such as dipoles and pistons, are inherently directional, whereas others, such as monopoles, are inherently omnidirectional.

Keywords: Bioacoustics, ka product, Sound production, Sound source.

1. INTRODUCTION

This chapter is about sound and how sound is produced. To start out, let us consider sound as everything we can hear, *i.e.* the entity that through the ears of normally functioning humans gives rise to our sense of hearing. Through our hearing we form an ever changing mental map of our environment consisting of a number of sound sources localized in space. It is common experience that if we

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move away from a stationary and constant sound source, its sounds become progressively fainter. Eventually we cannot hear them any longer, although somebody standing closer still can. This means that our hearing has a volume threshold, below which we cannot hear the sound. On the other hand, unfortunate individuals standing too close to an explosion will experience intense pain or may lose their sense of hearing (at least temporarily) because their eardrums are ruptured by the high sound volume. So, there is both a lower and an upper limit to the volumes of sound that our ears can analyze. Similarly, there are also limits to what pitch we can hear. High-pitched sound above our upper pitch limit is traditionally referred to as *ultra-sound* while sound below our lower pitch limit is called *infra-sound* (see Fig. **1**). On top of this, our hearing abilities deteriorate with age, so that especially our abilities to hear high-pitched sounds deteriorate as we get older – *i.e.* the limit to ultra-sound varies with individual age.

Such an anthropocentric grouping of sound into ultrasonic, sonic, and infrasonic sounds is far too limiting in bioacoustics. In principle there is no difference between, on one hand, those sounds that we can hear, and on the other hand the infrasonic components of the low-pitched rumble of elephants and the ultrasonic high-pitched cries of bats, none of which we can hear. For each hearing animal, there is a certain range of sound volumes and pitches that can be detected. So, for each species we can construct a diagram like that for humans in Fig. (**1**), with the meaning of ultra-sound and infra-sound differing from species to species. Therefore, in Chapter 1 and 2 of this book we will refrain from the use of the terms ultra- and infrasound. Instead, we will focus on the physical properties that really matters for sound production and propagation. Very often the most important property boils down to the ratio between the wavelength of the emitted sound and the size of the sound source or the range from the sound source to the receiver.

So far we have referred to sound in qualitative terms of subjective perception and used terms like *volume* and *pitch* to describe attributes to an auditory event perceived by humans. However, it is much more productive to define sound in objective terms of physics, since then we are not limited by arbitrary species dependent values. Here we use objective measurements to describe attributes to physical sound events, like the units on the axes in Fig. (**1**). Objective sound events outside an organism normally cause subjective (perceived) auditory events inside the organism (Blauert, 1997). The study of this phenomenon is known as *psychoacoustics* and this subject is dealt with in Chapter 10.

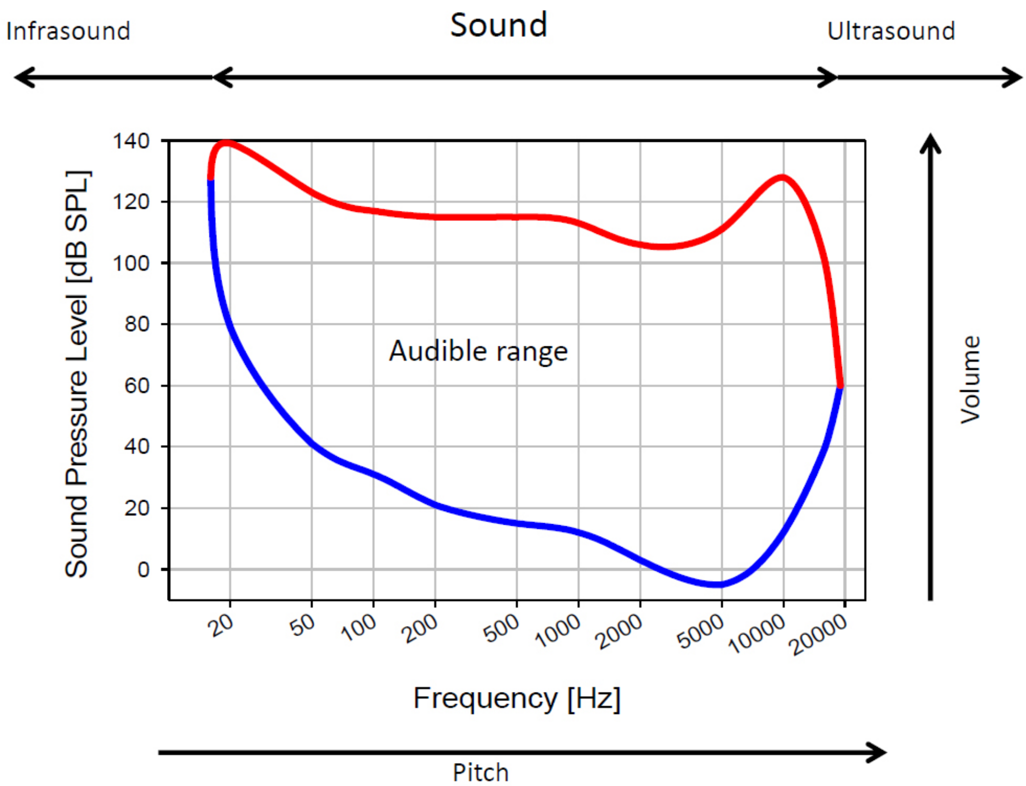


Fig. (1). An anthropocentric definition of infrasound, sound, and ultrasound is shown in relation to a map of sounds with pitch and volume perceptible to young and healthy humans. Perception by definition is subjective. The audible combinations of pitch and volume are delimited by the blue curve (faintest perceptible sounds of different frequencies) and the red curve (strongest perceptible sounds that may impair human hearing). A more accurate and objective description uses physical units of frequency measured in hertz (Hz) and sound pressure level measured in decibels (dB). Redrawn and modified from Wolfe *et al.* (2012).

In physical terms 'sound' can be defined, for instance, as '*mechanical vibrations and waves of an elastic medium*' (Blauert, 1997), or more generally as '*compressional oscillatory disturbances that propagate in a fluid*' (Jacobsen, 2007). What does it mean? Let us break up this statement in two components, the medium and the oscillatory disturbances.

First, in contrast to electromagnetic waves such as light that can propagate (or 'travel') in vacuum, sound needs an *elastic medium* such as air, water, or a solid for propagation. No sound can be heard when standing on the surface of the Moon because it lacks an atmosphere of gasses, through which sound can propagate. However, seismic sound waves can travel through the Moon, the Earth, and even through the Sun. In this chapter we will concentrate on sound in air, *i.e.* air is the elastic medium, through which sound waves propagate. Remember though that sound propagation is not limited to air. Biologically important sounds also propagate in fluids like water and as mechanical vibrations in elastic solids like wood and soil. Many marine and freshwater animal species produce sounds and listen to sounds under water. Below we shall point out where sound in air differs from that in water or in other important elastic materials. Seismic signals in the surface of the ground can be important for a number of species ranging from frogs to elephants, while plant material is the main communication channel for vibration signals emitted by small bugs (see *e.g.* Hill, 2009). Even though such signals are not considered being acoustic waves, they do interact closely with sound fields and generate sounds themselves. In this Chapter we will not treat them in depth, but can refer to textbooks such as Medwin & Clay (1998) and Cocroft *et al.* (2014).

The second component of the definition of sound above, the propagating oscillatory disturbances, concerns mechanical vibrations and waves that briefly disturb the state of an elastic medium when they pass a given position in space with the speed of sound, after which the particles of the elastic medium return to their previous state. There is no net movement of the single particles in the direction of sound propagation, only of the passing disturbance. The phenomenon is much like the 'wave' on a soccer stadium, where every person in the audience will only stand up and sit down again at the right time but the 'wave of activity' seems to move through the stands of the stadium with a certain velocity depending on how fast members of the audience move. In general, the propagation speed of sound depends only on the physical properties of the elastic medium.

The student of comparative bioacoustics typically handles sound signals in four major contexts: when recording, analyzing, synthesizing, and playing-back sound. In each context his or her understanding of the nature of sound, of the measures of sound, and of the basic laws of sound propagation in the field or in a laboratory setting are essential for characterizing animal vocalizations and for designing and interpreting bioacoustical experiments in a meaningful way.

The text below is meant to introduce novices (advanced undergraduate students, graduate students, and novice researchers) to some of the basic concepts of sound signals. Since many biologists are not at ease with too many and too complicated equations, we keep mathematical and physical notations to an absolute minimum, while trying to preserve sufficient rigor in the statements. For a rigorous mathematical treatment we refer interested readers to newer acoustics textbooks (*e.g.*, Blackstock, 2000; Fahy, 2003; Kinsler *et al.*, 1999; Kleiner, 2012; Pierce, 1989; Rossing, 2007; Urick, 1983) but also to well-illustrated broad descriptions like Heller (2013). Alternatively, readers may find extensive, easy to understand, and in most cases reliable information on websites like 'Wikipedia, The Free Encyclopedia' and 'the Discovery of Sound in the Sea'. Another very useful and reliable internet resource is the reference dictionary of sound and vibration terms in the library on the 'Brüel & Kjær homepage'. If you need to be absolutely certain that you use internationally recognized definitions and values, you should consult the 'International Standards Organization, ISO' and 'American National Standards Institute, ANSI', through their websites http://www.iso.org/ and http://www.ansi.org/, respectively.

2. BASIC MACHINERY

In order to establish a common ground we begin by reminding the reader about basic physical quantities and their derivatives in classical mechanics. We shall use SI-units throughout this chapter, *e.g.* meter (*m*) for length, *kilogram* (*kg*) for mass, and *seconds* (*s*) for time.

2.1. Displacement, Velocity, and Acceleration

If a solid body moves from position 1 (with coordinates x_1, y_1, z_1) to position 2 (x_2, y_2, z_2) in space, it is displaced an *average distance* (it could have made a detour) from 1 to 2 of $\Delta r = ((x_2-x_1)^2 + (y_2-y_1)^2 + (z_2-z_1)^2)^{½}$. The distance of ***displacement*** *(Δr)* is measured in the unit meter (*m*). While distance is a *scalar* with only a magnitude, displacement is a *vector* as it has both a magnitude and a direction. A vector is usually specified with an arrow above the parameter. Here, however, we are mainly interested in magnitudes. Therefore, we refrain from the more correct vector notation in what is to follow.

In the Newtonian macroscopic world any mechanical displacement is always a function of time, *r(t)*. If a body starts moving the distance Δr from position 1 at time t_1 and arrives at position 2 at a later time t_2, then on average it has moved at a *velocity* $v = \Delta r/(t_2-t_1) = \Delta r/\Delta t$. If we let Δr/Δt become infinitesimally small, we get $v(t) = dr/dt$ or the *instantaneous* velocity, which at times may be different from the average velocity, measured at any moment along the trajectory from position 1 to position 2. ***Velocity*** *(v)* then is a vector derived from displacement with respect to time and its magnitude is measured in $m\ s^{-1}$.

It is common experience that velocity of body movement also varies over time. The change of a body's velocity over time is its *acceleration*. Again we can talk about the velocity v_1 at time t_1 and velocity v_2 at a later time t_2. The velocity Δv has then changed by an average *acceleration* = *Δv/Δt*. With the relations above we get that ***acceleration*** is a vector derived from velocity with respect to time with the magnitude Δv/Δt, and measured in the unit $m\ s^{-2}$. Again for infinitesimally small time steps we can define an instantaneous acceleration and the relationship between the magnitudes of acceleration, velocity, and displacement becomes

$$a(t) = \frac{dv(t)}{dt} = \frac{d^2r(t)}{dt^2} \qquad \mathbf{(2.1)}$$

2.2. Force and Laws of Motion

In classical mechanics the physical quantities displacement, velocity, and acceleration characterize all body movements in space, be it of planets or very small volumes of air. But what *causes* movement of a body? Sir Isaac Newton stated in his *First Law of Motion* that neither a stationary non-automobile body of a certain ***Mass*** (*M*) nor such a body moving at a constant speed will start moving or change magnitude of direction and velocity, respectively, unless they receive some outside influence. This outside influence Newton termed a ***Force*** *(F)*. As change of velocity over time is acceleration, he stated in his *Second Law of Motion* that

Force (t) = Mass • acceleration (t),

or

$$F(t) = M \frac{dv(t)}{dt} \tag{2.2}$$

Except for the force of zero magnitude every force has a direction and thus is a vector. So $F(t)$ in (2.2) is the resulting force obtained by vectorially adding all forces acting on the body. From the section above it further follows that force is measured in the unit: $kg\ m\ s^{-2}$, which is usually denoted *1 newton* (or just *N*). While we can immediately relate to units such as 1 kg and 1 m, it may be difficult to get an intuitive feeling for 1 newton. It may help to think about an everyday object such as an apple. It typically has a mass of 100 g = 0.1 kg. The average gravitational acceleration from the Earth at sea level is about 9.81 m s^{-2}. So, according to (2.2) the weight of the apple (that is, the force with which the Earth is 'pulling' the apple) on the palm of your hand is approximately 1 N.

Certain bodies can be stretched or compressed by applying a force to them, for instance by pulling the ends of an expander spring (or sitting down on the springs of the mattress on your bed). It is common experience that the longer you stretch the expander spring, the more force you have to apply to its ends and also that it springs back into its former length once you let go. If a body moves back into its former shape once the external force is removed, we say that the body is ***elastic*** or has elastic properties. The force that restores the stretched or compressed dimension of the body, the ***restoring force***, F_r, is determined by the characteristic spring constant (*S*) of the body material and the displacement (*r*) of the body from its resting position. This is known as *Hooke's Law*:

$$F_r(t) = -S\, r(t) \tag{2.3}$$

where the spring constant S is measured with the unit $N\ m^{-1} = kg\ s^{-2}$. The minus indicates that the restoring force of the elastic body is directed opposite to that of the external force acting on the body. *Hooke's Law* applies to all elastic materials as a linear approximation and is usually valid up to a certain magnitude of displacement, beyond which the material may break or behave nonlinearly in other ways.

2.3. Work, Energy, and Power

Everybody has experienced that it takes energy to perform physical work. In terms of physics ***work*** (W) is performed when a force applied to a body results in a displacement of the body. The larger the force or the longer the displacement, the larger the amount of work is performed:

$$W(t) = F(t)\, r(t) \qquad \textbf{(2.4)}$$

which is measured in *joule* = $J = N\,m$. So, if you lift an apple on the palm of your hand 1 m upwards (*i.e.* against gravity), then in physical terms you perform a work of 1 J.

Incidentally, by lifting the apple you have increased its ***potential energy*** by the same amount. In physical terms ***Energy*** *is defined as the capability to perform* ***Work***. Both energy and work are measured in *joule*. Whenever a body such as the apple is placed in a gravitational field, for instance on your desk, its *potential energy* E_p according to (2.2) and (2.4) depends on its mass (M), the acceleration due to the gravitational field (g), and the potential distance (Δr) that it can move relative to a reference point in the direction of the field, for instance to the floor:

$$E_p = M\, g\, \Delta r \qquad \textbf{(2.5)}$$

If you let go of the apple, it will fall to the floor with increasing velocity, because its potential energy (energy stored by the position of a mass in a gravitational field) is transformed into ***kinetic energy***, or movement energy, until it comes to a stop, now with less potential energy than before. In classical mechanics the *kinetic energy* E_k of a non-rotating object such as the falling apple depends on its mass and velocity:

$$E_k = \frac{M\,v^2}{2} \qquad \textbf{(2.6)}$$

The total energy of a body in a gravitational field is the sum of its potential energy and its kinetic energy

$$E_{total} = E_p + E_k \qquad \textbf{(2.7)}$$

The kinetic energy does not disappear when the apple comes to rest on the floor, but is transformed into an unordered type of energy, ***heat.*** In the macroscopic Newtonian world energy is always conserved but it will be transformed into different forms when bodies influenced by forces move about and bump into each other.

Work as defined by (2.4) can be performed slowly or rapidly. The *rate* at which work is performed is termed ***Power*** (P):

$$P(t) = \frac{W(t)}{\Delta t} \qquad \textbf{(2.8)}$$

The unit of power has its own name, *watt* = $W = J\,s^{-1}$. Note, that by combining (2.4) and (2.8) and using the relation between displacement *r* and velocity *v*, we find that power is also equal to force times velocity:

$$P(t) = \frac{F(t)\,r(t)}{\Delta t} = F(t)\,v(t) \qquad \textbf{(2.9)}$$

All definitions and considerations above apply not only to large solid bodies but also to small volumes of elastic media such as air or water and therefore to sound.

3. THE NATURE OF SOUND

3.1. Elasticity and Waves

Normally we do not think of air as being elastic. However, take a syringe (or a bicycle pump) and press the plunger while closing the outlet with a finger. It is obvious that the air resists the volume reduction when you press the piston, because the air particles are confined to a smaller and smaller space. The elastic air is *compressed*. If you let go of the piston, the restoring force (cf. (2.3)) of the compressed air will return the piston to its starting position. You have to exert a *force* to keep the piston in position. This force equals the restoring force exerted by the compressed air on the stationary piston (this is given by *Newton's Third Law of Motion*, stating that if a body exerts a force on a second body, then the second body exerts a force of equal magnitude but opposite direction on the first body). If you keep your finger on the outlet but instead pull the plunger, you will also feel an increased resistance as the air particles occupy an increasingly larger space. This *rarefaction* of the air means that the air particles in the syringe exert a force that becomes progressively smaller than that of the outside atmosphere. Again the plunger will return to the starting position, if you let it go. One characteristic of sound ('compressional oscillatory disturbances that propagate in a fluid') is such an alternation between local ***compressions*** and ***rarefactions*** of the elastic medium. In propagating sound waves the compression and rarefaction cycle is so short that although the temperature in a point varies during the cycle there is no heat exchange with the surroundings and the total heat energy stays constant, *i.e.* the process of sound propagation is ***adiabatic***. It will only be ***isothermal*** under very special circumstances, *e.g.* when sound is propagating in very narrow tubes such as the acoustic trachea of the cricket ear (Larsen & Michelsen, 1978).

So, fluids like air and water will resist a change in volume but they will not resist changes in shape. This means that they cannot transmit sheering forces (at least to a first approximation). Consequently, sounds in fluids can only propagate as compressional or *longitudinal waves* (vibrations of the particles are parallel with the direction of sound propagation). In contrast, solid elastic media such as wood can transmit sheering forces and therefore also *transverse waves*, vibrations perpendicular to the direction of sound propagation, and other types of vibrations, *e.g.* bending waves that transmit different frequencies with different speeds, *i.e.* they are *dispersive*.

Seismic signals in the surface of the ground can be important for a number of species ranging from frogs to elephants, or in plants where bugs and spiders may detect them (see *e.g.*, Michelsen *et al.*, 1982; Michelsen, 2014; Cocroft *et al.*, 2014). Waves propagating on a water surface can, just as seismic signals, be used for communication in some species of animals living on or close to the surface (Bleckmann *et al.*, 2014). All these types of signals, however, are usually not classified as acoustic waves. They are transverse rather than longitudinal and in the case of water surface waves the oscillation between potential and kinetic energy is quite different from what is observed in propagating acoustic waves (Clay & Medwin, 1977).

3.2. Sound Pressure

In the syringe-example above, the force exerted on the air trapped in the syringe by the plunger is distributed over the area (A) of the end of the plunger. ***Pressure (p)*** is defined as the force acting on a unit area and it is measured in the unit *pascal* = *Pa* = *N m*$^{-2}$.

$$p(t) = \frac{F(t)}{A} \tag{3.1}$$

Sound pressure is one of the most important characteristics of sound, since this is the quality that we measure with transducers like microphones (or hydrophones), which transduce air (or water) pressure variations into electrical signals. If the medium is air, the air particles are always compressed to a certain degree depending on the weight of the air column above the position in the Earth's atmosphere. (Note that in acoustics the ***medium particles*** are NOT the single molecules but small volumes of the elastic medium; so small that they can be regarded as units with constant acoustic variables (p, u, ρ) but so big that they retain fluid properties. Therefore, an air particle contains millions of molecules!). This constant (or only slowly changing) atmospheric compression determines the number of air molecules per unit volume or the ***density of the medium*** (ρ), which is measured in kg/m^3 ($\rho_{air\ 20°} \approx 1.2$ kg/m^3; $\rho_{sea\ water} \approx 1030$ kg/m^3 close to the water surface). The *standard static atmospheric pressure* is defined as 101,325 Pa or about 10^5 Pa. Sound pressure variations in the human hearing range are much smaller, from 10^{-5} Pa (about -6 dB SPL) at hearing threshold to 10^2 Pa (about 135 dB SPL) at intense pain in the ears (cf. Fig. **1**), *i.e.* the sound pressures in the human hearing range are 3-10 orders of magnitude below that of standard atmospheric pressure. Under water the static pressure is increased by about 10^5 Pa for every 10 m water depth. The changes in static pressure as marine organisms dive up and down through the water column are in general many orders of magnitude higher than the pressure fluctuations created by underwater acoustic signals.

In principle, *sound pressure is defined as the* ***instantaneous*** *difference between the total pressure and the standard static atmospheric (or underwater) pressure*. In practice, however, we usually refer to the time-averaged value of the squared sound pressure, the so-called ***Root Mean Square*** or ***RMS***-value that can either be obtained directly from a sound level meter or calculated from:

$$p_{RMS} = \left(\frac{1}{T_{av}} \int_0^{T_{av}} p^2(t)dt\right)^{½} \qquad \textbf{(3.2)}$$

where T_{av} is the time segment that is averaged. The square of the sound pressure is used in the RMS definition since sound pressures fluctuate between positive and negative values, and thus a simple time average of sound pressure will be very close to zero. Only in very extreme situations, such as during explosions, the time averaged pressure may be different from zero. The RMS pressure is popular for another reason: when multiplied by itself (*i.e.* squared) it is directly proportional to the acoustic intensity, as will become obvious below. In special situations other pressure values than the RMS may be used, *e.g.* peak-values.

In commercial sound level meters displaying a running RMS, T_{av} conventionally can be swapped between the values F = 125 ms or S = 1000 ms ('F' denoting 'Fast', and 'S' denoting 'Slow') but other values of T_{av} may also be used. Note that in the special case when the pressure signal is a sinusoid of amplitude Y, it can be shown from (3.2) that the RMS-value reduces to the simple expression $p_{RMS} = Y/\sqrt{2}$.

While force and displacement are always directional (a vector), pressure (*e.g.* atmospheric pressure) is non-directional (a scalar quantity) at any point of the sound field. In a given position in space you will therefore in principle measure the same sound pressure no matter, in which direction you point your microphone. Please note, however, that there are two types of precision microphones, *pressure microphones* and *free-field microphones*. The electrical output of a *pressure microphone* is directly proportional to the real (and direction-independent) pressure in the sound field, provided that the dimensions of its diaphragm are small relative to the wavelength of sound, *i.e.* you may point the microphone in any direction and get the same output. If the dimensions of the microphone are large relative to the wavelength, then the microphone will disturb the sound field. Because of build-up of sound pressure in front of the membrane it will measure a larger sound pressure when pointed towards a sound source than the actual pressure at this position when the microphone is not there (a half inch microphone, for instance, will start to disturb the sound field for frequencies above 4 kHz, where it typically measures about 1 dB too much and the error gets progressively larger with higher frequencies). In a *free-field microphone*, on the other hand, the manufacturer has compensated for this wavelength dependent diffraction to obtain the real sound pressure. So, a free-field microphone must be directed towards the sound source to measure the real sound pressure. Conversely, if a pressure microphone is used in the free field, it should be oriented *perpendicular* to the direction to the sound source. *Free-field microphones* are preferable, if you know the direction to the sound source, while *pressure microphones* are better if the sound source direction is unknown, for example when measuring ambient noise, moving sound sources, or when measuring in diffuse sound fields and in closed couplers.

3.3. Sound Intensity and Power

In many situations the bioacoustician only needs to consider pressure variations in order to measure and analyze sound. There are, however, a number of cases when it is important to consider the *energy* carried by sound, *e.g.* when sound passes from one medium to another (see Chapter 2) or in order to calculate the average sound pressure at a distance from the recording microphone.

According to (2.4) the energy in the form of work includes both force and displacement. As air is not a solid body we need to consider instead a volume of air carrying the sound or more precisely the propagating compressional oscillatory disturbance. This is usually done by considering work ($W = F\,r$) per unit time and per unit area, where r is the displacement of air particles during compression and rarefaction, and where the unit area is perpendicular to the propagation path of the sound wave. Since work per unit time equals ***power*** *(P)*, this defines another characteristic of sound, the sound ***intensity*** *(I)*.

$$I(t) = \frac{P(t)}{A} \qquad \textbf{(3.3)}$$

which is measured in the unit $J\,s^{-1}\,m^{-2} = W\,m^{-2}$ (cf. (2.8)).

It is important to note that *sound power (P)* (measured in W = J/s cf. (2.8)) is a property of the sound source and does not change with distance from the source. The same sound energy emitted from the source (assuming no absorption of sound in the medium) will pass through the surface of a sphere with the source in the center, no matter how large its radius is. However, if you at progressively longer distances from the source measure the energy passing a fixed area, *e.g.* the area of a microphone diaphragm, then you will see the energy per area decreases with distance as the same source energy becomes distributed over a progressively larger spherical surface area. *Sound intensity (I)* is a measure of exactly that as it is defined as *the time averaged energy that per time unit (s) passes through a unit area (1 m*^{2}*) perpendicular to the direction of a plane progressive wave.* In other words, no power flows in the direction perpendicular to the direction of particle trajectory. So, sound intensity will decrease with distance from the source (see Chapter 2 for further discussion).

Using (2.9), (3.1), and (3.3) we get:

$$I(t) = \frac{P(t)}{A} = \frac{F(t)u(t)}{A} = p(t)\,u(t) \qquad \textbf{(3.4)}$$

where $u(t)$, the ***particle velocity,*** is the magnitude of the directional movement velocity of the air particles during compression and rarefaction. Note that particle velocity is different from and usually much smaller than sound propagation velocity or the ***speed of sound*** *(c), i.e.* $u \ll c$. Particle velocity is proportional to source level, and it varies with distance to the sound source (see below).

3.4. The Speed of Sound

The speed of sound depends only on the properties of the medium and remains constant as long as the sound propagates in a medium with constant properties (such as density, humidity, and temperature). For all practical purposes the speed of sound in a standard atmosphere with 0.04% CO_2 can be reasonably well approximated by the formula

$$c = (331.45 + 0.607\, T_C)\, {}^{m}/_{s} \qquad \textbf{(3.5)}$$

where T_C is the temperature in degrees Celsius. The speed of sound in air mainly depends on temperature and is, for instance, 344 m/s at 20°C, 356 m/s at 40°C, but only 319 m/s at -20°C. A generalized algorithm for sound in humid air has been developed by Cramer (1993) and an interactive calculator using this algorithm can be found, for instance, on the website of the National Physical Laboratory, UK (http://resource.npl.co.uk/acoustics/techguides/speedair/). The importance of relative humidity (RH) is very limited. At 20°C and standard atmospheric pressure, for instance, c = 343.5 m/s at 10% RH and only increases to c = 344.6 m/s at 100% RH.

The speed of sound in sea water, c_o, is more difficult to calculate as it depends non-linearly on parameters such as temperature, static pressure (depending on depth), and salinity. Several empirical formulas have been developed by fitting to experimental data from several experiments. For instance, if in a sea water body the temperature is 13°C and the salinity 35ppt, then from the water surface and down to 100 m the speed of sound in seawater (c_o) can be reasonably well approximated by the formula:

$$c_o = (1500.16 + 0.0163\, z)\, {}^{m}/_{s} \qquad \textbf{(3.6)}$$

where z is the depth in meter (Blackstock, 2000). As a rule-of-thumb and for many practical purposes the speed of sound in water is approximately c_o = 1500 m/s. If the precise speed of sound in a certain body of water is required, you should consult international standards, which nowadays can be found on websites such as that of the National Physical Laboratory, UK (http://resource.npl.co.uk/acoustics/techguides/soundseawater/content.html), where interactive calculators are available for a number of different algorithms to calculate c_0. One of those, the international standard algorithm, the UNESCO algorithm, has been dealt with in detail by Wong and Zhu (1995). The speed of sound in pure water is generally somewhat lower and may be approximated by the formula (Lubbers and Graaff, 1998):

$$c_o = 1404.3 + 4.7T - 0.04T^2 \qquad \textbf{(3.7)}$$

where T is the temperature in °C.

4. THE SOUND FIELD: WAVES IN TIME AND SPACE

Once a sound source (for example the piston mentioned above) creates local fluctuations of compression and rarefaction of the medium, these fluctuations start to propagate away from the source. A propagating acoustic *wave* has been created. A propagating wave is a transport of energy. Waves in air or water are *longitudinal* as the maximum pressure fluctuations occur in the same direction as the waves propagate. A propagating sound wave can be quantified by a few parameters, some of which are determined by the elastic medium and others by the sound source. We have already seen that the speed of sound is determined by the elastic medium. If the piston, or as a more biologically relevant example a male cricket's wing, vibrates regularly, then the air particles will progressively move around their equilibrium positions away from and towards the vibrating wing. The propagating sound wave ('the disturbance of the medium') will be a simple sine wave propagating away from the wing as illustrated in Fig. (**2**).

The sound source such as the vibrating cricket wing dictates the ***amplitude*** of the sine wave (how large the fluctuations in pressure or particle motion are), the ***frequency*** of the wave (*f*, how many times per second pressure variations occur, measured in *hertz* and abbreviated $Hz = s^{-1}$), its ***period*** (*T,* measured in *seconds, s*), and its ***wavelength*** (λ, measured in *meters, m*). For any simple sine wave sound (pure tone) the latter parameters are related through the equation:

$$c = f\,\lambda = \frac{\lambda}{T} \tag{4.1}$$

Sine waves are important in sound analysis as any sound signal can be decomposed into a number of concomitant sine waves by a mathematical operation, the *Fourier Transform* (see Chapter 9).

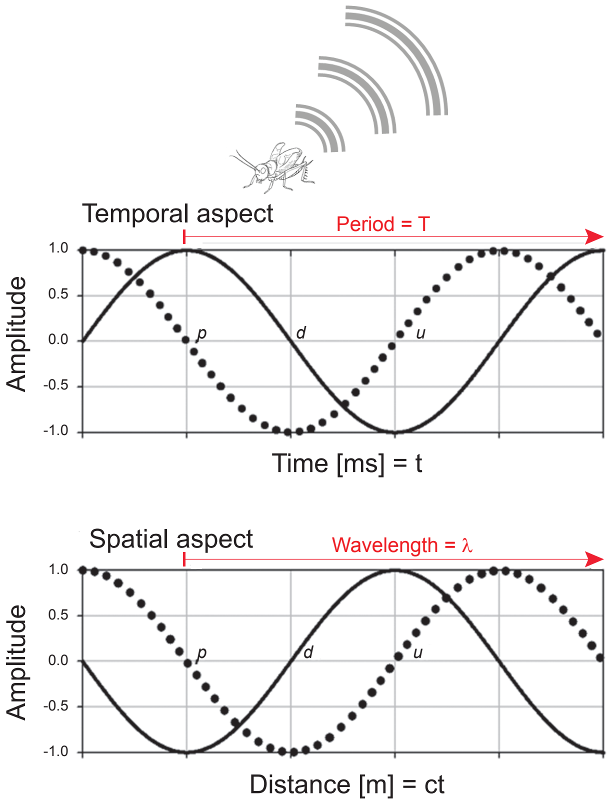


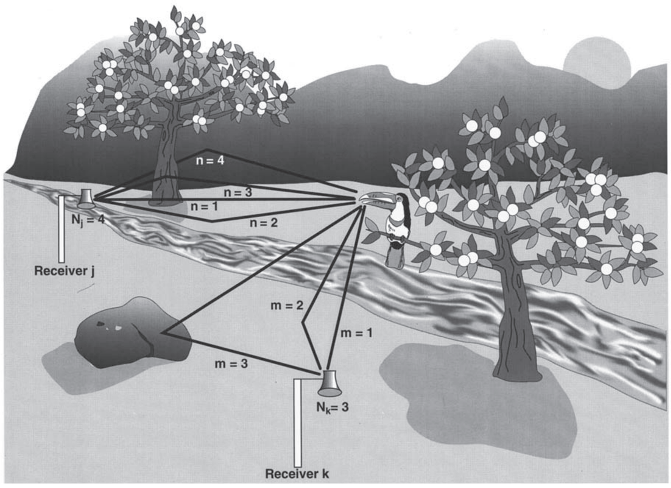
Fig. (2). Sound waves in time and space. A male field cricket emits an almost pure tone. An observer with a temporal aspect is located in one position far from the cricket and observes the variation in amplitude of pressure (*p*), particle displacement (*d*), and particle velocity (*u*) as time goes by. She notes that pressure and particle velocity are always in phase (therefore the curves are on top of each other) and can measure the period *T* and from that deduce the frequency $f = 1/T$. Another observer with a spatial aspect 'freezes' time and can move in space and measure the wavelength, λ. From this he can deduce $f = c/\lambda$, where c is the sound velocity. For crickets the pure tone is often about $f = 5{,}000$ Hz and $T = 0.2$ ms. During that time the wave will have propagated the distance $\lambda = 6.8$ cm.

At every point in time and space, where there is sound, there is a ***sound field*** characterized by both a pressure fluctuation relative to the ambient pressure and a local displacement of particles. Close to the sound source particle displacement will dominate, while far away from the source pressure fluctuations will dominate. The local particle displacement can be characterized either by the displacement distance or by its time derivatives velocity and acceleration (cf. (2.1)). Even though the wave is a simultaneous function of time and space, we usually look at its temporal and spatial properties separately.

When treating the *temporal* properties we envision ourselves standing at a certain location in space and observing the acoustic wave passing as time goes by Fig. (**2**). This is the perspective we take, when we record sounds with a microphone at a certain location in space and therefore it is easy to visualize. When investigating the *spatial* properties of the sound field, we envision time as frozen (Fig. **2**) and ourselves as being able to move in space to investigate pressure fluctuations along the propagation path. This is of course not possible in reality but may be partly realized if we use a receiver array to simultaneously measure or estimate the peak pressure or particle velocity for a number of locations in space along the propagation direction of the sound away from the source.

Keep in mind that temporal and spatial aspects are two different representations of the same overall acoustic wave. Also note that we often like swapping between not only temporal and spatial representations, but also between different ways of illustrating the sound field. For example, when describing the spatial sound field, we may depict the moving wave front as one or several ***rays*** propagating through the medium and being perpendicular to the wave front, which at distances far from the source can be considered plane. ***Ray tracing*** is great in illustrating how sound travels from a source to a receiver, or how sound is being reflected off a surface, as if they were small particles (Fig. **3A**; see also Chapter 2). On the other hand, to understand phenomena like ***diffraction*** (changes in sound pressure due to bending of sound waves around objects with dimensions comparable to the wavelength of sound), or to analyze the sound field through a complicated medium or close to a sound source, we often use ***gradient maps*** of some sort, representing peak amplitudes with colors or shades of grey (Fig. **3B**), or by contour plots. Presenting sound as rays can be viewed as interpreting the sound field as *particles*, whereas with gradient maps we envision it as *wave phenomena*. This is analogous to how physicists describe light both as waves and particles, depending on the situation. Both the wave and the particle interpretations of sound are valid, but their validity differs depending on the wavelength and the size of the sound source and distance from the sound source of interest.

(a)



(b)

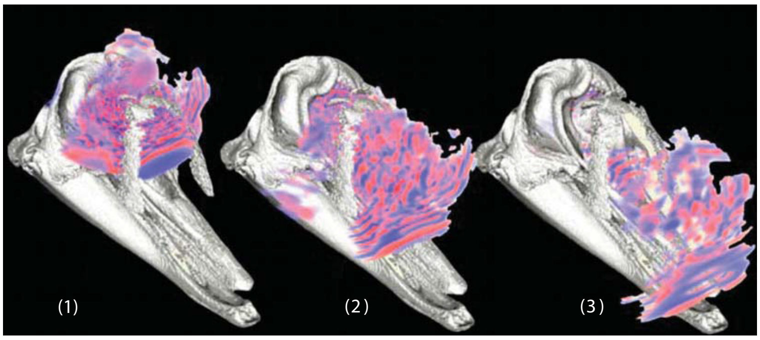


Fig. (3). Different representations of a sound wave. **(a)** Ray tracing representation of sound propagating from a bird to a set of receivers (Reproduced with permission from Spiesberger, 1998). m=1 and n=1 are direct paths from the bird to the receivers. n=2 and m=2 are paths reflected off the ground, whereas n=3, n=4 and m=3 are reflected at other objects (a tree and a stone) before reaching the receiver. **(b)** Gradient map representation of sound waves emitted from the head of a Cuvier´s beaked whale (*Ziphius cavirostris*) using a finite element modeling technique (Reproduced with permission from Cranford *et al.*, 2008). Red and blue indicate positive and negative sound pressure, respectively. The gradient map shows how the sound beam is transmitted through the skull of the Cuvier's beaked whale from time (1) to time (2) to time (3). The change over time in the gradient map is attributed to the bioacoustic properties of the tissue layers in the beak of the whale, and how the thicknesses of the layers change anatomically along the axis of the whale.

Illustrations of sound fields may have their origin in vastly different methodologies, all with their own weaknesses and strengths. Measurements can sample sound in time and space using microphones or hydrophones. Computer models can predict the sound field using a discrete version of the acoustic wave equation or numerically calculated ray tracings. Graphical techniques are especially useful in situations where the propagation environment is spatially very complicated; they usually start out from ***Huygens' Principle*** that the wave front at any point in space at a certain time is the summed contributions of spherical wavelets emanating from the wave front at different locations at a previous time. Great care must be taken when comparing illustrations of the acoustic field using different methodologies.

4.1. Pressure Fluctuations and Particle Motions

It is evident that the pressure and particle disturbances must be related somehow: the particles moving together is what causes the pressure to increase, and at the same time an increase in pressure will put a force on neighboring particles to move in a direction of lower pressure. Actually, by rewriting Newton's second law $\Delta F = M\frac{dv(t)}{dt}$ (2.2) for the difference in forces ΔF separated by Δr meters and acting on an area of A m^2 we get

$$\frac{\Delta F}{A\Delta r} = -\frac{M}{A\Delta r}\frac{dv(t)}{dt} \quad \textbf{(4.2)}$$

which by the introduction of ***the pressure difference*** $\Delta p = \Delta F/A$, ***the density*** $\rho = M/(A\ \Delta r)$ and by letting Δr become smaller and smaller at the limit is:

$$\frac{dp}{dr} = -\rho\,\frac{dv(t)}{dt} \quad \textbf{(4.3)}$$

The negative sign is needed to indicate that positive pressure gradient results in a force acting in the opposite direction. Thus, the spatial change in pressure, or rather the space derivative of sound pressure (***the pressure gradient***), causes the acceleration of the acoustic particles (or fluid elements) that are essential to wave motion. The higher density, the larger pressure fluctuations are created for a specific acceleration. In other words, it takes less acceleration to create a specific pressure disturbance in a medium of high density (such as sea water) than in one of low density (such as air) or put differently: it is easier to produce large sound pressures in water than in air.

A sound source like the vibrating cricket wing (Fig. **2**) creates particle motion in the elastic medium by 'pushing' (compressing) and 'pulling' (rarefacting) the particles. Very close to a sound source, the physical displacement of particles can be considerable and much larger than what is found further away where the particles only move around their steady state position. The ratio between acoustic pressure (p) and particle velocity (u) in a sound field is called the ***acoustic impedance*** (abbreviated Z and measured in $rayl = Pa\ m^{-1}s = N\ s\ m^{-3}$)

$$Z = \frac{p}{u} \Leftrightarrow p = Z\,u \qquad \textbf{(4.4)}$$

The acoustic impedance is a concept that students of bioacoustics often find hard to grasp. It is always useful to go back to the actual definition of the term: the impedance is telling us, how much pressure you get out of a certain amount of particle velocity. If you get a lot of pressure out of it, it means that the medium is resisting the density change caused by the particle motion. Such a medium has high acoustic impedance. Likewise, if you only get a small pressure out of moving the particles together, it means that the medium can sustain a larger density change and that the acoustic impedance is lower. So, the impedance can be viewed as a type of resistance, just like electric impedance refers to the resistance to alternating currents. Therefore, the relationship between pressure and particle velocity is sometimes called ***Ohm's Acoustic Law***, as it is directly analogous to the relationship between electrical potential and current, where the difference in potential energy (u, voltage) drives the electron flow (i, current) through electrical components that exert some resistance (Z, impedance) to the flow, *i.e.* Ohm's Electrical Law is: $u = Z\,i$.

For a plane progressive wave in the far field (see definition below) we often refer to the *characteristic impedance*, Z_0 of the medium. The characteristic impedance is given simply by $Z_0 = \rho_0\, c_0$, where ρ_0 and c_0, respectively, are the density and speed of sound in the medium, when no sound is propagating in it. It is only for plane waves, and only far from the source, that the impedance can be calculated that easily. In other situations, Z is a very complicated function of the medium properties (its sound velocity and density) and of source properties (the frequency and amplitude of the acoustic signal, and the distance and direction to the source). It is also important to realize that the particle motion and pressure components may or may not oscillate in phase. As Z_0 is a real number the pressure and particle motion components of a plane wave oscillate in phase as depicted in Fig. (**2**). Closer to a sound source, the impedance Z is a complex number, and here the pressure and particle motion components do not oscillate in phase.

4.2. The Far Field

If you are far away from an acoustic source (*i.e.* much farther than the linear dimensions of the source itself, for instance its radius, a) in a part of space dominated by the propagating sound wave and with no steep gradients in sound speed, then we say that you are in the ***Acoustic Far Field*** (Fig. **4**). Later on, in Section 7, we will see that there are two different criteria that have to be met to be in the far field. For now, it suffices to say that in the far field meeting both requirements, the sound wave has some very agreeable properties. In a limited part of space it can with good approximation be considered a plane progressive wave, in which the particle velocity *(u)* is in phase with (varies simultaneously and in the same way as) the sound pressure *(p)* (see Fig. **2**) and the acoustic impedance becomes greatly simplified as the characteristic impedance of the medium:

$$Z_0 = \rho_0 c_0 \;\Rightarrow\; p = Z_0 u = \rho_0 c_0 u \qquad \textbf{(4.5)}$$

In air at 20°C Z_0 is about 413 rayl (Breazeale and McPherson, 2007), which means that at a sound pressure of, for instance, 1 Pa the vibration velocity of the air particles is about 2.4 mm/s. In distilled water at 20°C and standard atmospheric pressure, the characteristic impedance Z_0 is 1,480,000 rayl, or almost 3600 times larger than in air (cf. Table **1**). Thus, water will resist the compression caused by a certain magnitude of particle motion much more than air. Because of this, for a certain magnitude of particle vibrations, the sound pressure will be 3600 times *higher* in water than in air. Likewise, for a certain sound pressure the vibrations generated in an acoustic field are 3600 times *smaller* in water than in air. For a sound pressure of 1 Pa in water, for instance, the vibration velocity is only about 0.7µm/s.

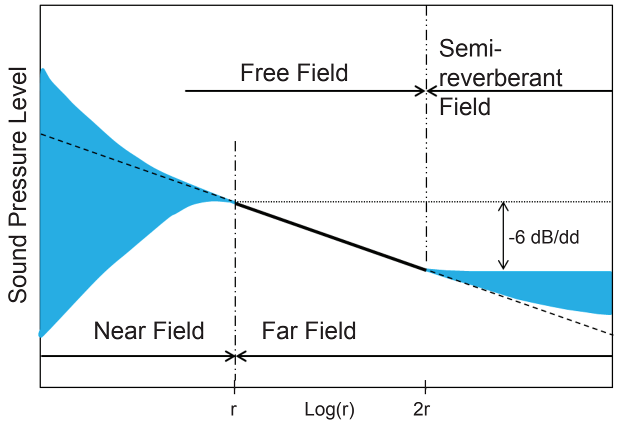


Fig. (4). Different acoustic fields. The Near Field is closest to the source. Here the sound pressure amplitude is difficult to predict and can be higher or much lower (blue areas) than expected from extrapolation from the Far Field. In the Far Field the sound pressure amplitude is well defined and decreases by 6 dB with doubling of distance (if medium absorption is negligible), if it is also a Free Field, *i.e.* without acoustically significant objects. Also, in the Far Field, the acoustic pressure is given by particle velocity (u) times the characteristic acoustic impedance (Z_0). Further away (usually very much further away than 2r, typically 15-100r) where larger surfaces (soil surface or seabed) and objects interfere with the direct sound field, it becomes a Semi-reverberant Field. This must be taken into consideration especially when measuring in laboratory settings.

Table 1. Basic metrics of sources and medium in air and under water (Kinsler *et al.* 1999).

Magnitude	Unit	Air	Water	Water/Air
Speed of sound	m/s	330	1500	4.5
Free-field acoustic impedance	Pa m/s (rayl)	413	$1.48 \cdot 10^{6}$	3584
Acoustic free-field intensity at 1 Pa	W/m^2	$2.4 \cdot 10^{-3}$	$6.8 \cdot 10^{-7}$	$2.8 \cdot 10^{-4}$

We can also say that the Acoustic Far Field is the part of space away from sound sources where the ratio between sound pressure and particle velocity is determined only by the characteristic impedance Z_0 and where sound pressure levels obey the inverse-distance law (see Chapter 2).

Using (3.4), (4.4) and (4.5) we can now get a useful equation to calculate ***Sound Intensity(I)*** in the ***Acoustic Far Field***:

$$I = pu = \frac{(p_{RMS})^2}{Z_0} = \frac{(p_{RMS})^2}{\rho_0 c_0} \quad \textbf{(4.6)}$$

Equation (4.6) is used in everyday calculations of sound intensity when you are dealing with far field conditions and measure sound pressure with a calibrated microphone or (in water) hydrophone. ***Calibration*** means that it is known by stimulation with a well-defined standard (*e.g.*, a so-called *pistonphone*, available for both microphones and hydrophones) how the transducer's electrical output measured in, for instance, mV relates to the sound pressure at the microphone diaphragm measured in Pa. Intensity can also be measured directly in air with a special *intensity probe* consisting of two opposing microphones (or in water with two small hydrophones) separated by a known distance, which enables us to measure both the sound pressure and the sound pressure gradient. The latter can then be converted to acceleration from Equation (4.3) and eventually to particle velocity by integration (Fahy 2003).

On top of this, in bioacoustics we also like to deal with the time-integrated intensity, which is the acoustic energy passing through a unit area. We can call this the ***acoustic energy per area*** *(E)* and measure it in J/m^2. When expressed in dB, *E* is called the ***sound exposure level*** *(SEL)*. Among bioacousticians ***E*** is sometimes called *energy*, *energy density*, and quite often even *energy flux density*. The latter two names, however, are confusing, as in other branches of acoustics energy density denote the total amount of acoustic energy within a certain volume, and energy flux density denotes intensity. Also, please do not confuse *acoustic energy per area (E)* measured in J/m^2 with *power (P)*, which is measured in *J/s*. Fig. (**5**) illustrates the relationship between pressure amplitude, RMS-value, and accumulated energy per area as calculated for a real-world sound signal.

4.2.1. Acoustic Energy

The bioacoustic term '*acoustic energy per area'* measured in J/m^2 should not be confused with the term '*acoustic energy density*' measured in J/m^3 used by 'real' acousticians. The latter is the total acoustic energy in the volume encompassed by the propagating sound wave. It is present in two forms, the kinetic energy density (E_k) of the oscillating air particles and potential energy (E_p) of pressure fluctuations (this is however a different type of potential energy than the one introduced in (2.5) and (2.7)). In a small volume of air, an acoustic particle, where u and p are constant, the *instantaneous total energy density* $\boldsymbol{E_i}$ is given by

$$E_i = E_k + E_p = \frac{\rho u^2}{2} + \frac{p^2}{2\rho\, c^2} \qquad \textbf{(4.7)}$$

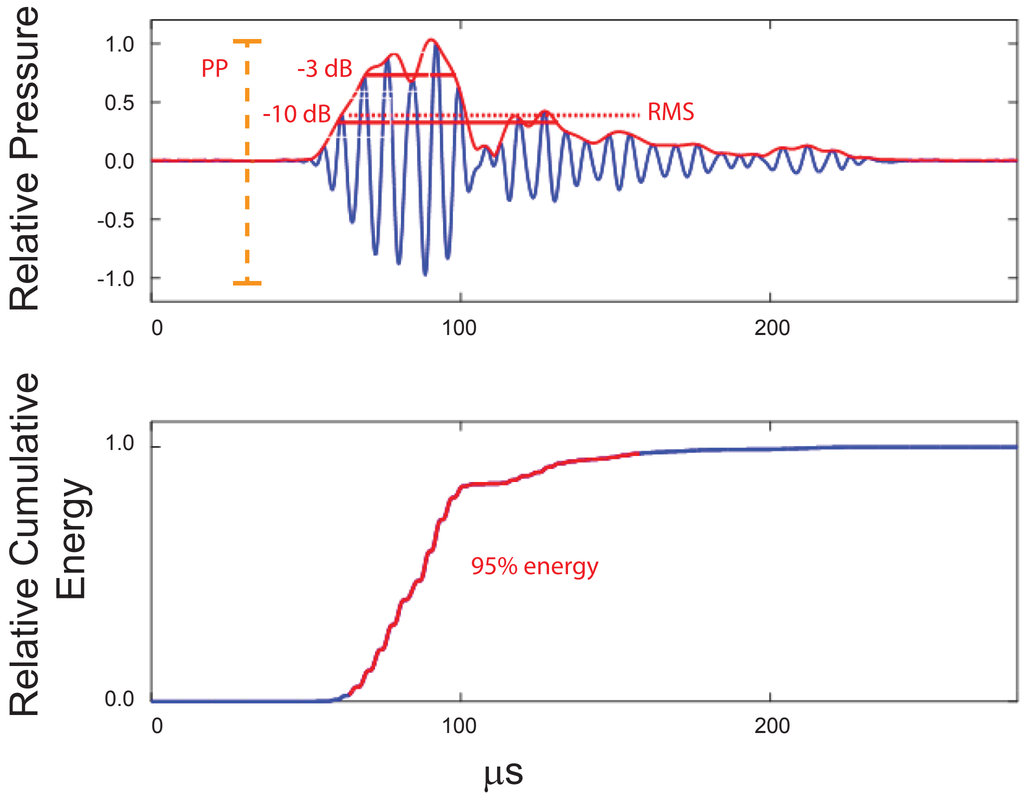


Fig. (5). The relationship between the pressure amplitude (instantaneous and peak-to-peak, pp), the RMS-value, and relative accumulated energy in a harbor porpoise (*Phocoena phocoena*) signal. In the upper panel, the sound pressure signal and its envelope are shown, and in the lower panel its accumulated energy as a function of time. Also, the duration of the signal, measured as the -3 dB and -10 dB relative the envelope peak, as well as the 95% energy interval, are shown (Redrawn and modified from Wahlberg & Surlykke, 2014).

For a plane, progressive sound field E_k and E_p are always equal and just like with sound pressure the instantaneous value is rarely interesting but the time average energy density is. So, the **acoustic energy density of a plane progressive wave** is given by

$$E_i = \frac{(p_{RMS})^2}{\rho c^2} \tag{4.8}$$

which is important as most sound fields with good approximation can be regarded as plane fields or a superposition of plane sound fields. Note the similarity between (4.6) and (4.8): in a plane progressive field the intensity, $I = E_i\, c$.

4.3. Other Sound Fields

In addition to the 'agreeable' or relatively simple acoustic *Far Field* there are other types of sound fields. An ***Acoustic Free Field*** Fig. (**4**) is a sound field in a part of space with no surfaces that diffract or reflect sound in the frequency region of interest and also often are 'far' from the source such that they obey simple equations.

If such objects are present and diffraction and reflection do occur it is a ***Semi-reverberant Field*** (Fig. **4**). As a 'rule-of-thumb' objects in a sound field will diffract sounds with wavelengths progressively smaller than about 1/2π of object dimensions (cf. (6.3) below and see Chapter 2 for diffraction and reflection). So, to be on the safe side in a laboratory setup involving the measurement of sound (for instance, when stimulating an animal preparation with sound and measuring its response) the diameter of rods and other objects placed in the sound field should be below 10% the wavelength of the highest frequency used in the experiment. In addition, the boundaries of the room, in which experiments are performed, should effectively absorb all incident sound in the frequency range of interest, thus creating an '*anechoic room*' and approaching a free sound field in the space of interest, *e.g.* that normally occupied by the experimental animal's head.

A ***Diffuse Sound Field*** is approximated for technical purposes in a so-called '*hard room*' and is roughly defined as a region of space with a very large set of statistically uncorrelated plane waves propagating from all directions with a uniform probability distribution such that the sound pressure is the same everywhere (Fahy, 2003). This type of field is hardly ever present in bioacoustic research.

A ***Closed Sound Field*** is a small enclosure or pressure chamber (for instance an *acoustic coupler*) sometimes used in setups in the lab to produce well-defined sound stimuli or to calibrate transducers.

Finally, there are the problematic ***Near Fields***, which are much more complicated to describe and to map. Unfortunately, many bioacoustic measurements, especially in laboratory settings, take place in the less well-defined acoustic near field and should therefore be given special consideration (see section 7 below).

5. SOUND AMPLITUDE

5.1. Logarithmic Machinery

The most common way to report the amplitude of an acoustic signal is not on a linear scale, but on a logarithmic one. This makes sense for many reasons: it simplifies calculations (believe it or not!) and the results are more directly related to how we and other animals perceive sound. However, they are a large source of confusion, often caused by a lack of understanding for their basic mathematical properties. Therefore, take some time to dwell on the following.

Let y, a, and x be any number. By definition: If $y = a^x$ then

$$\log_a(\mathrm{y}) = x \tag{5.1}$$

We call '*log*' the '*logarithm of y*', and '*a*' the '*base*' of the logarithm. The base can be any number, but popular choices are 2, 10, or the irrational number *e* (which is about 2.71). When you read about logarithms or calculate them, take great care! The notation can mean many different things. You really have to make sure, which base is implied – else the numbers may make no sense at all. It is quite unusual to see the base written out explicitly, such as '$\log_{10}$'. It is often assumed that '*ln*' means natural logarithms (with base *e*), and '*log*' or 'lg' commonly mean logarithms of base 10. This is the notation we will use in what follows. Some calculators, programs, and texts use a different notation; natural logarithms can be written as *log* and base 10-logarithms as *log10*. Confused? Never mind – you should be!

Now, consider the three logarithmic laws:

$$\log(x\,z) = \log(x) + \log(z) \quad \textbf{(5.2)}$$

$$\log\left(\frac{x}{z}\right) = \log(x) - \log(z) \quad \textbf{(5.3)}$$

$$\log(a^x) = x\,\log(a) \quad \textbf{(5.4)}$$

These laws can easily be verified from the definition of logarithms above. It is useful to memorize that $\log_a(a) = 1$ and $\log_a(1) = 0$ (which both follows directly from (5.1)). With the help of (5.1) and (5.4) you can transfer a logarithm from base *a* to another base *b*:

$$log_b\,(y) = log_b\,(a^x) = x log_b\,(a)\; log_a\,(y) \quad \textbf{(5.5)}$$

Thus, for instance, you can transform a logarithm from base 2 to base 10 by the formula $log_{10}(y) = log_{10}(2)\, log_2(y) = 0.30\, log_2(y)$.

5.1.1. dB Basics

The decibel scale, dB, used in acoustics is defined using logarithms of base 10. The 'Bel' is in honor of the Scottish-American inventor Alexander Graham Bell and the 'deci' means 1/10.

The decibel scale is defined by

$$dB = 10 log_{10}\,(x) \quad \textbf{(5.6)}$$

The factor 10 is introduced to give the decibel numbers a convenient magnitude. Also, the factor 10 makes the decibels biologically relevant: in bioacoustics, 1 dB roughly corresponds to the smallest sound intensity ratio that a human, or most animals for that matter, can discern. The number x is a ***ratio*** between a certain *measurement value* and a *reference value*. For sound intensity levels, the reference is denoted I_0. The ***dB-value*** is thus a ***relative measure*** and therefore it is ***meaningless without stating the reference*** – just like a %-value is meaningless unless you know '% of what'. The only situation where a reference is not needed is when *e.g.*, reporting 'a decrease of 20 dB'. Here the reference is not explicitly written out, but what we are really saying is 'a decrease of 20 dB relative the initial level' or similar. Also here, the notation is analogous to percentages: 'A decrease of 90%' really means 'A decrease of 90% as compared to the initial level'.

We define

$$Sound\ Intensity\ Level: \qquad L_I = 10\ log_{10}\left(\frac{I}{I_0}\right)\ dB \qquad \textbf{(5.7)}$$

Sound Intensity Level is sometimes abbreviated *SIL*, which however is not an officially recognized term. In (5.7) *I* is the linear intensity actually measured or calculated (in W/m^2) and I_0 in air is 10^{-12} W/m^2 = 1 pW/m^2. However, intensity probes are expensive (as they need to measure both pressure and particle velocity c.f. (3.4)), so we rarely measure sound intensity directly. Instead we use pressure sensors (microphones or hydrophones) to measure the sound pressure *p* and then calculate *I* using (4.6) and assuming we are in the acoustic far field. The sound power level, L_W, and the sound energy level, L_E, are defined similar to (5.7) with the reference values 1 pW and 1 pJ/m^2, respectively. Note that the very small *pico-units* of the reference are used always to get positive dB-values of intensity and energy levels.

When using (4.6) to substitute *I* and I_0 we get with the help of (5.4):

$$Sound\ Pressure\ Level: \quad L_p = 10\ log_{10}\left(\frac{p^2}{p_0^2}\right)\ dB = 20\ log_{10}\left(\frac{p}{p_0}\right) dB \qquad \textbf{(5.8)}$$

where both *p* and p_0 are RMS-values. In air the reference value p_0 = 20 µPa (which corresponds to an intensity of roughly I_0 = *1 pW/m*^{2} and is close to the human hearing threshold at 1-2 kHz). In water (and other media) p_0 = 1 µPa RMS, which has been selected without any special bioacoustic observation. (Note that pressure amplitude reference values are always in *micro-units*). In the bioacoustic literature *Sound Pressure Level* is often abbreviated *SPL*, but this abbreviation is not included in the ISO and ANSI definitions and should therefore be used with care.

Except when indicated, all the reference values above are defined by the ***International Standards Organization, ISO***, as 'Preferred reference values for acoustical and vibratory levels' in the document 'ISO 1683:2008'. If you really need to know precise and internationally recognized values in acoustics you should consult the ISO homepage (http://www.iso.org/) or ANSI (www.ansi.org). To give you an impression of what different sound pressure levels means we have collected some examples in Table **2**.

Table 2. Some examples of common sounds in air and their amplitudes in dB SPL.

dB SPL	Everyday sounds	Animal source levels
0	Human hearing threshold at 1-2 kHz	
10	Rustling leaves	
20		Human soft whisper
30	Bedroom night	Purring cat
40	Living room	
50	Office	
60	Normal conversation	
70		Barking dog
80	City street (no truck)	Blackbird song
90	Heavy truck	Roaring lion
100		Echolocating brown bat
110	Thunder, construction site	
120	Jet taking off nearby	
130	Pain in the ears	

5.2. Decibel Arithmetic

It is important to note that when expressed on a decibel scale, sound intensities are *not* linearly related. This means that you cannot use normal rules of addition and subtraction for dB-values. For instance, you will NOT obtain an average of *e.g.* 5

dB-values by first adding them together and then dividing the sum by 5. You need to transform the values back to linear values, calculate the average and then transform the average back to a dB-value. To appreciate this, consider how to calculate the pressure from a certain dB-level. That is, if

$$L_p = 20\ log_{10}\left(\frac{p}{p_0}\right)\ dB \qquad \textbf{(5.9)}$$

then

$$p = p_0\ 10^{\left(\frac{Lp}{20}\right)} \qquad \textbf{(5.10)}$$

If you want to add a sound of 94 dB relative to (re.) 20 µPa with another sound of 94 dB re. 20 µPa, then the result is NOT 188 dB re. 20 µPa. The result is actually 100 dB re. 20µPa because 94 dB re. 20 µPa corresponds to 1 Pa, and 2 Pa is the same as 100 dB re. 20 µPa – try to calculate this by yourself with (5.10), or by using (5.2-5.4).

You should note, however, that this addition only works for *correlated sound sources*, *e.g.* for two loudspeakers playing the same sounds and for any point in space where the distances to the two sources are equal. For locations where the difference in distance is half a wavelength, the two waves will be out of sync and the resulting sound pressure here will be 0. If the sound sources are *uncorrelated* such as two blackbirds each singing at 94 dB then the resulting level is about 97 dB or an increase of only about 3 dB, not 6 dB. Please also note another rule-o--thumb: if there is a difference of more than 10 dB between two nearby sound sources, then the resulting sound pressure is determined only by the louder of the two sources.

There are some good 'tricks' that should be learned by heart about decibels and ratios. You can easily verify the following:

2 times higher sound pressure corresponds to adding: 6 dB = 20 log(2)
3 times higher sound pressure corresponds to adding: 10 dB = 20 log(3)
5 times higher sound pressure corresponds to adding: 14 dB = 20 log(5)
10 times higher sound pressure corresponds to adding: 20 dB = 20 log(10)

½ the sound pressure corresponds to adding: -6 dB = 20 log(0.5)

If for instance the sound pressure amplitude is 15 times larger than a reference amplitude, then it is 3 x 5 times larger or according to the values above about (10 + 14) dB = 24 dB larger. This quick calculation is within 1 dB of the precise value (23.5 dB). With the help of these few values and (5.2-5.4) you can quickly transfer from linear units to decibels without using a calculator. You can easily find dB-calculators on the internet that let you add many dB-values but without understanding the nature of the dB-scale you could get in trouble.

5.3. Comparison of Sound Amplitudes in Air and Water

Some animals, for instance seals, can produce and hear sounds both in air and under water. It is therefore important for bioacousticians to be able to compare sound amplitudes in the two media - but what is the right measure? For example, what does an L_p-value of 60 dB SPL re. 20 µPa measured in air correspond to in water?

First there is the difference in reference values: 1 µPa *versus* 20 µPa. A factor 20 (= 2 x 10) corresponds to 26 dB. So, a sound pressure level comparison between air and water differs by 26 dB (see Fig. **6**). To account for the difference in reference value you should add 26 dB to the 60 'air-dB' to get 86 'water-dB'. Some scientists argue that since eardrums respond to *sound pressure*, then 26 dB is the right conversion number.

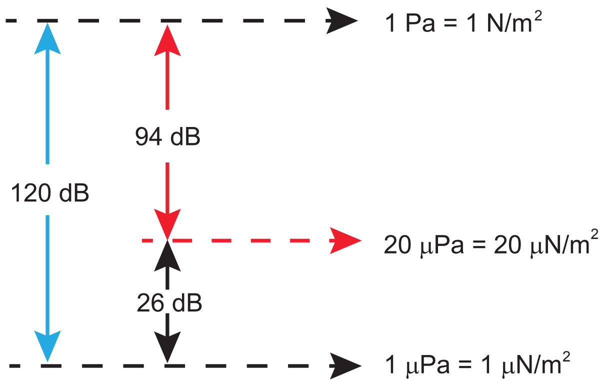


Fig. (6). dB-values. The relationship between sound pressures in air (red) and water (blue) is shown relative to their respective reference values.

So, by using (5.2) we get

$$L_{p\,water} = 20\log(\tfrac{p}{10^{-6}}) = 20\log(\tfrac{p*20}{20*10^{-6}}) = 20\log(\tfrac{p}{20*10^{-6}}) + 20\log(20)$$
$$= L_{p\,air} + 26\,dB \quad \text{or} L_{p\,water} = L_{p\,air} + 26\,dB \qquad \textbf{(5.11)}$$

Other bioacousticians maintain that most ears are energy detectors and that also the much larger *Z* of water should be taken into consideration. Therefore, the *intensities* in air and water should be compared instead. As noted above, the specific acoustic impedance of air at room temperature is about Z_{air} = 413 rayl, while in water it is about Z_{water} = 1.48 10^6 rayl. Using (4.6) and if the acoustic pressure is the same in both media, we then get that:

$$\frac{I_{water}}{I_{air}} = \frac{Z_{air}}{Z_{water}} = \frac{413}{1.48\,10^6} \approx \frac{1}{3600} \qquad \textbf{(5.12)}$$

Thus, if the sound pressure is the same in both media, then the resulting acoustic intensity is much smaller (36 dB, by taking 10 log(1/3600)) in water than in air.

Now, assuming that the acoustic intensity is the same in both media (and denoted *I*) and using (4.6) and (5.8), which is the initial definition of decibels, we get:

$$L_{p\,water} = 10\log\frac{I}{I_{0\,water}} = 10\log\frac{I}{\frac{(10^{-6})^2}{Z_{water}}} = 10\log\frac{I\cdot 20^2}{\frac{(20\cdot 10^{-6})^2 Z_{air}}{Z_{water}\,Z_{air}}}$$
$$= 10\log\left(\frac{I}{I_{0air}}\right) + 20\log 20 + 10\log\frac{Z_{water}}{Z_{air}} = L_{p\,air} + 26\,dB + 36\,dB \qquad \textbf{(5.13)}$$
$$L_{p\,water} = L_{p\,air} + 62\,dB$$

This means that a sound of for instance 60 dB re. 20 μPa in air has the same intensity as 60 + 62 = 122 dB re. 1μPa in water. In general we have that:

$$L_{I\,water} = L_{I\,air} + 62\,dB \qquad \textbf{(5.14)}$$

6. SOUND SOURCES

6.1. 'Small' and 'Large' in Acoustics

When studying the physics of waves in general and acoustics in particular, one usually ends up with complicated mathematical equations. These equations are usually greatly simplified by statements such as *'when an object is small'* or *'when a distance to a sound source is very long'*. It is important to appreciate that expressions such as *long distance* and *small object* are usually related to the *wavelength* of the sound in question.

Consider sitting in your boat fishing while rocked by the waves. Small waves, created by the splashes of the fish being brought up to the surface, are propagating on the water surface, and are completely reflected against the hull of the boat, without causing the boat to move. The long waves generated by yesterday's gale, on the other hand, are *not* reflected but pass the boat without being attenuated, while the boat gently rolls in the water. Thus, the boat interacts differently with different waves, depending on *the wavelength in relation to the size of the boat*. The same holds true for acoustic waves: a sound field is relatively easy to describe when the wavelength is very small or very large relative to objects, with which it interacts.

On the other hand, the interaction between the boat and waves of a wavelength comparable to the boat size, such as those generated by a gentle breeze, are very difficult to analyze. Similarly, it is a huge challenge in acoustics to describe the interaction between a sound field and an object of *similar size* to the emitted wavelength. This phenomenon is known as *diffraction*.

Analogous to the boat example, when we talk about *small*, *large, close* and *far* (away) in acoustics, this is usually done relative to the wavelength, λ, of the emitted sound. A sphere of radius a, for example, may be said to be large compared to the wavelength if $a >> \lambda$. In many cases it is not the radius but the circumference (= $2\pi a$) of the sphere, which is relevant; then the sphere is considered large, if $2\pi a >> \lambda$ or $2\pi a / \lambda >> 1$. We usually simplify the notation through the introduction of the ***wave number (k):***

$$k = \frac{2\pi}{\lambda} = \frac{2\pi f}{c} \qquad \textbf{(6.1)}$$

Thus, the sphere with radius *a* is considered acoustically large if

$$k\,a >> 1 \qquad \textbf{(6.2)}$$

and acoustically small, if $ka << 1$. For other shapes than spheres the definition of 'large' is more complicated, but a good rule-of-thumb is using (6.2) with *a* being the largest dimension of the object.

Similarly, a distance *r* away from a sound source or a reflecting object can usually be considered large, if $r >> \lambda/(2\pi)$, or if $r/\lambda >> 1/(2\pi)$. Or, if we were to use wave number 'lingo': the *distance is acoustically large* when

$$k\,r >> 1 \qquad \textbf{(6.3)}$$

and the distance is short, if $kr << 1$.

6.2. Monopoles, Dipoles, and Pistons

Sound is produced by a sound source compressing and rarefacting an elastic medium, so that the disturbance is radiated away from the source at the speed of sound. Even though sound production can be incredibly complex, especially in many biological systems, there are some simple models that approximate many biological sound sources quite well.

Most notable are the ***elementary sound sources***, such as the *monopole*, the *dipole*, and the *piston* sound sources. The ***acoustic monopole*** is a pulsating sphere of radius *a*, radiating sound of equal intensity in all directions (Fig. **7a**). Compared to the geometry of a vocalizing animal it is extremely simple - but actually all (or at least many) sound emitters can be modelled by an assembly of monopoles (cf. ***Huygens' Principle*** for the construction of wave fronts). The radiated sound has two components, the propagating sound wave and the local flow of particles in the boundary layer displaced radially by pulsations of the sphere. The local flow rapidly diminishes with distance from the monopole (see Section 7 below).

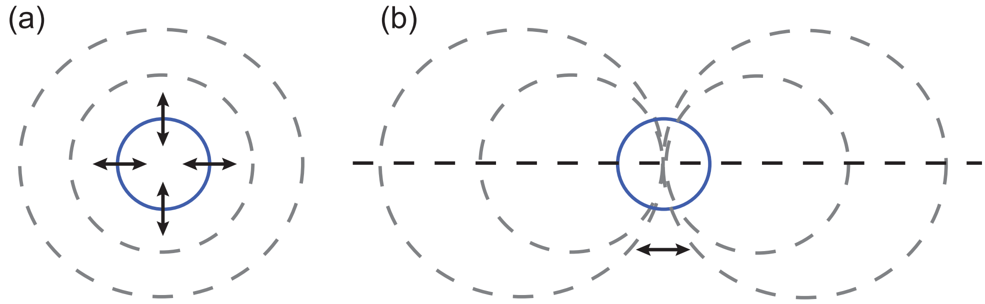


Fig. (7). Monopole and dipole sound production (drawn in 2 dimensions; in the real world it is a 3-dimensional development of the emitted sound). **(a)** The monopole is a sphere expanding and contracting, generating a homogenous omnidirectional sound field. **(b)** The dipole is a sphere moved from side to side generating a bidirectional sound field (the horizontal dashed line indicates the acoustic axis of the dipole sound source).

The ***acoustic dipole*** is a rigid sphere that vibrates forth and back and thereby radiates sound in two opposite directions (Fig. **7b**) like the cricket wing in Fig. (**2**). The dipole is equivalent to two separated monopoles pulsating totally out of sync (180° out of phase). At equal distances from the centers of the two monopoles sound pressures cancel and sound therefore radiates in a 'figure-o--eight' pattern. It is intuitively obvious that if the vibration happens very slowly (*i.e.* T is large, f small, and λ is long compared to a, cf. (4.1)), then sound emission is 'short-circuited' by local flow because 'there is time for the particles to move from the expanding sphere to the contracting sphere' and the sound emission away from the dipole is small and inefficient. As λ becomes smaller relative to a (meaning that the frequency of vibration increases) there will be progressively less time for short-circuiting local flow. Therefore, sound emission efficiency (defined as the ratio of energy in the propagating acoustic wave to the energy used for vibrating the dipole) will increase to a maximum as a function of frequency (or as a function of $1/\lambda$, or as a function of the wave number k) and then remain constant when nearly all vibration energy is radiated as sound. On the so-called acoustic axis (the direction where sound production is maximal indicated with the dashed line in Fig. (**7b**)) the dipole will start to resemble a monopole. However, its directionality pattern is maintained so that oblique to the acoustic axis the acoustic pressure remains zero. Also, there is a 180° phase shift for the sound emitted on one side of the dipole as compared to the other side (*e.g.*, when one goes right the other goes left).

A third simple sound source often used for modelling vocalizing animals is the ***circular piston*** placed in a very large wall. In many situations calculations using this model give a reasonable estimate of the 'bioacoustic' sound field.

6.3. Sound Source Efficiency

Let us again define sound source efficiency as the ratio between the acoustic energy in a travelling acoustic wave and the energy that the sound source used for producing the wave. The sound source efficiency can be modified by the source in many ways. However, there are some physical limitations to what is possible, and a lot of these are dictated by the size of the source compared to the emitted wavelength: mice cannot roar with a pitch as low as the one from elephants and small fish cannot make the extremely low-frequency calls of blue whales.

The intuitive understanding of the *relationship between,* on one hand, *the size of the sound source relative to the wavelength and,* on the other, *the efficiency of sound emission* is quantified in Fig. (**8**). This figure has rather complicated units on the axes. It depicts the efficiency of sound production as a function of source radius and the emitted frequency (= *ka*) from two kinds of sound sources. The *free vibrating disc* corresponds to the dipole source described above (approximated *e.g.* by the field cricket's wing in Fig. (**2**)). The *disc with 'baffle'* is a dipole source placed in a 'wall' that prevents the short-circuiting local flow and increases the efficiency of sound emission, just like a piston or a loudspeaker membrane mounted in a closed cabinet. Such a baffled sound source has properties resembling a monopole at low frequencies. Tree-crickets, for example, make use of such a baffle, a leaf much larger than their vibrating wings, to increase the efficiency of their sound emission (see *e.g.* Prozesky-Schulze *et al.*, 1975). In a laboratory setting it may sometimes be desirable to extend the frequency range of a small speaker towards lower frequencies. This can be easily achieved by attaching a circular baffle with sufficient diameter to the rim of the speaker. Even a piece of cardboard will do.

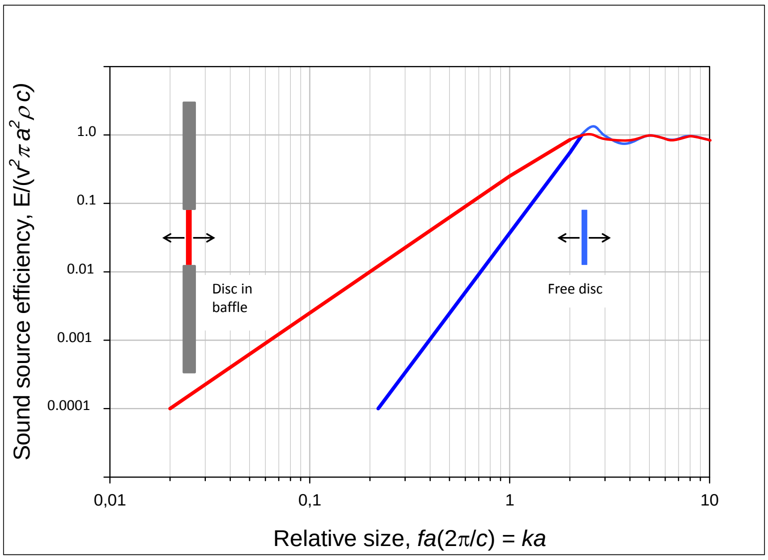


Fig. (8). Sound source efficiency as a function of wavelength-to-emitter size, *ka* (redrawn and modified from Michelsen 1992). Efficiency of a vibrating disc is shown both with and without a baffle (a very large 'wall'). For further explanations see text.

On the y-axis in Fig. (**8**), the emitted sound energy (here denoted *E)* is divided by the square of the RMS velocity of motion of the disc (v), the disc radius (a), a factor π, and the free field acoustic impedance, $Z = \rho c$ (cf. (4.5)). All in all, the denominator $v^2\pi a^2\rho c$ expresses the total kinetic energy, E_K, of the part of the medium put into motion by the sound source. E_K is divided into one part leading to the propagating acoustic wave (with the sound energy E), and one part resulting in the back-and-forth local flow of the medium close to the sound source. The ratio of E to $v^2\pi a^2\rho c$ therefore is a measure of how much of the kinetic energy results in the propagating acoustic wave. This may be taken as a measure of the efficiency of sound production by the sound source, cf. our initial definition above.

The cryptic notation on the x-axis of Fig. (**8**) $fa\ (2\pi/c)$, where a is the radius of the disc and f is the emitted frequency, can be reduced to the ka product: $f\,a\ (2\pi/c) = 2\pi\ (f/c)\ a = (2\pi\,/\lambda)\ a = ka$. Thus, what Fig. (**8**) shows is the acoustic efficiency as a function of the ka product, or the source-size-to-wavelength ratio for a baffled and an unbaffled vibrating disk. Note that the two curves behave very differently in the regions $ka >> 1$ and $ka << 1$.

Fig. (**8**) can be read in many different ways and contains a lot of useful information (once you get used to it!). It is well known from the acoustic literature (see *e.g.* Beranek, 1983) but was first introduced for bioacoustics applications by Axel Michelsen in the 1980s (*e.g.* Michelsen and Larsen, 1985).

Let's for instance keep v and a constant, and see what happens when the emitted wave number (or emitted frequency) is changed. At low frequencies, when $ka << 1$, sound production is very inefficient, and the efficiency is decreasing rapidly with increasing wavelength, or decreasing disc size. A disc mounted in a baffle (which is close to 'monopole kind' of sound source) is by far more efficient (10-100 times) than a free disc (a dipole) for small values of ka. This is because the above-mentioned short-circuiting between the two sides of the dipole leading to a large local flow but a very small propagating acoustic wave. Above ka-values of about 2 (indicated by the star on Fig. (**8**)) the sound production efficiency is stabilized close to 1, both for the baffled and unbaffled disc. Note that E for both sources is measured in the direction of maximum sound production (the so-called acoustic axis, see Fig. (**7b**)). As discussed below, the sound field from both discs will be highly directional at higher frequencies (high ka). Also, for the unbaffled disc, in the direction perpendicular to the acoustic axis there is no propagating sound wave for any choice of disk size and frequency.

A similar size-relationship of sound production efficiency as the ones found in Fig. (**8**) is found in sound sources with all sorts of geometries. This fact answers the question why mice cannot rumble like elephants, and why blue and fin whales as well as elephants are capable of producing the loudest infra-sounds in nature. If the sound source size is too small compared to the wavelength, most of the energy put into sound production is causing the medium to vibrate back and forth, rather than creating a propagating sound pressure wave.

Our analogy using a rocking boat to illustrate how objects interact with a sound field can be modified and used to give an intuitive feeling for these conclusions. Let's say that you wish to generate surface waves in a bath tub by moving a half-submerged small plate back and forth. If you move it very slowly (meaning that *ka* is small) then you will create some local flow, but not much of a propagating surface wave. As you move the plate faster and faster, the *ka*-product increases, and all of a sudden propagating surface waves start to emerge (try it!). In principle, Fig. (**8**) does not preclude large sound sources from generate high-frequency sounds. However, there are other physical constraints in the mechanical properties of the sound production organs making it difficult for a large structure to emit a high-pitched sound. (Just think of how difficult it is for you to move the plate in the bathtub at high frequencies).

Combining these constraints with Fig. (**8**) explains the observed double logarithmic relationship between animal size and frequency emphases in their vocalizations in air (Fig. **9**).

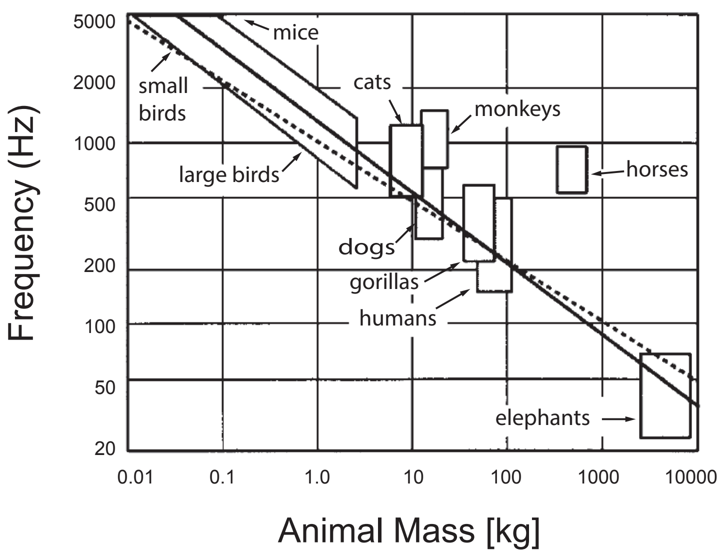


Fig. (9). Body mass correlates with vocalization frequency (boxes indicate frequency ranges) on a double logarithmic scale for the animals shown. The main communication frequency is inversely proportional to about the 0.4 power of the animal's body mass (solid line). Reproduced with permission from Fletcher (2004). Copyright 2004, Acoustical Society of America.

You can also use Fig. (**8**) to address some other enigmas: where on the curve would you expect to find an aroused male haddock calling its mating song to the females? Haddock vocalizations are centered around or below 200 Hz (Hawkins and Rasmussen, 1978) and thus have a wavelength in sea water longer than about

$\lambda = c_0/f = 1500/200 = 7.5$ m. The sounds are made by drumming on the swim bladder, which for a decent sized haddock may have a radius of, say 3 cm ($a = 0.03$ m). The *ka* product is therefore $2\pi a/\lambda = 2\pi\ 0.03 / 7.5 = 0.025 << 1$. Thus, the sound production efficiency of haddock mating song falls far down on the left-hand slope of Fig. (**8**), *i.e.* this fish does not seem to be very efficient in producing sound. From this you may conclude that the sound production system is maladapted in this species. However, this may not be so. Think about what happens to the rest of the vibration energy for $ka << 1$, *i.e.* the part that does not become a propagating sound wave. This kinetic energy E_k is induced into the water and creates a back-and-forth water flow, the local or *hydrodynamic flow*, around the sound source. The otolith and lateral line organs of fish are, depending on the nature of the generated vibrations, sensitive to this local flow of the medium (Popper *et al.*, 2003). So, maybe the haddock sound production is not maladapted after all, but the fish is producing a powerful flow field to communicate with nearby conspecifics rather than using its energy to emit a propagating sound pressure wave that may be picked up by predators.

Our discussion of sound production inevitably leads us into a discussion of the so-called near fields around a sound source. These are concepts much plagued by misconceptions, misunderstandings, and a non-standardized nomenclature. Many animals can detect particle motion (*e.g.* by sensory hairs on insects or otolith organs in fishes) or propagating sound pressure waves (*e.g.* by tympanate ears in birds, mammals, some insects, and fish with a swim bladder). Some animals can detect only the particle motion, whereas others are specialized for detecting pressure fluctuations – and still others, for instance some insect and fish species, can detect both components. Even though there are huge variations in sensitivity and frequency range of detection, no animals (as far as we know) are completely unable to detect acoustic signals.

There are no receptors *per se* for near fields or far fields, only for the vibration and pressure components of the sound field, which are present both at short and long range from the source. The discussion below on near and far fields may therefore seem to be only of academic interest for bioacousticians. However, in many situations it is important that you learn to judge whether the local flow component should be taken into consideration when measuring sound from a given source and whether your measurements are made sufficiently far from the source to judge the source level correctly. This may be achieved by using some simple rules-of-thumb that are derived below while discussing near and far fields.

7. NEAR FIELDS AROUND A SOUND SOURCE

If you wish to get thoroughly confused when learning about acoustics, you should start reading about *near* and *far* fields. As we have discussed above, the general idea with these terms is that there is a certain distance from the surface of the sound source, within which the characteristics of the sound field differ dramatically from the characteristics beyond this distance. You will soon notice, however, that the nature of these fields can be described in many ways. Also the transition from near to the far field characteristics of the sound field is not abrupt like a well-defined border but smooth and extended. Therefore, there are many ways to define a distance separating near and far fields.

There are two types of near field of concern for bioacousticians: one deals with the *local flow* and the other deals with *interference* of sound produced by different parts of the source. Most books and scientific papers on bioacoustics are *not* doing a good job explaining to the reader, which type of near field they are addressing. As it turns out, depending on which animal you work with, different near fields may have different significance. As will become obvious in what follows, for some 'bioacoustic sub-culture' of *e.g.* that of fish bioacoustics, the near-field of concern is always the one describing the magnitude of the local flow component of the sound field. For another 'sub-culture', *e.g.* that of bat scientists, the range of the interference near field can be of major concern, while the extent of the flow field is so small that it is unimportant. For bird bioacousticians neither type of near field seem to be important.

7.1. The 'Flow' (Reactive) Near Field

We already noted that a vibrating sound source such as the acoustic monopole creates both a propagating acoustic wave and simultaneously a non-propagating back-and-forth motion of the medium. Close to the sound source the particle velocity therefore consists of two components. One is associated and in phase with the propagating pressure wave (cf. Fig. **2**). The other, the *local flow* (or in water the *hydrodynamic flow*) component is 90° out of phase with the acoustic pressure component. The propagating part of the sound field is called the *active part*, whereas the part that does not contribute to the net energy flow is called the *reactive part*.

For elementary and other simple sound sources, the near field effects can be treated analytically. It is easiest to start out with a monopole vibrating continuously with a frequency f (or a wave number $k = 2\pi f/c$). It can be shown (Kinsler *et al.*, 1999) that the analytical expression for the particle velocity u at a distance r from a monopole is given by

$$u = \frac{p}{\rho c}\left(1 + \frac{1}{j\,k\,r}\right) \qquad \textbf{(7.1)}$$

where p is the sound pressure, j is the complex number $j = (-1)^{1/2}$, which is used to signify the 90° phase shift between pressure and the reactive particle velocity u, and r is the distance or range from the source. Equation (7.1) is calculated by integrating the particle acceleration, which is given by (4.3) with $p = e^{\,j(\omega t - kr)}$, where $\boldsymbol{\omega}$ is the ***angular frequency***, $\omega = 2\pi f$.

The two terms in the parenthesis of (7.1) correspond to the far field and the near field contribution to particle velocity, respectively. Far from the source, where $kr >> 1$, the second term can be neglected, because it is proportional to 1/r. Here, in the so-called *far field*, the particle velocity (u) is proportional to the pressure fluctuations (p) with the proportionality factor $(\rho\ c)^{-1} = Z_0^{-1}$, where Z_0 is the characteristic acoustic impedance of the medium, as we saw in Section 4 (cf. (4.5)). The far-field contribution to particle velocity corresponds to the blue line in Fig. (**10**). As we approach the sound source, the contribution of the second term, the red line in Fig. (**10**), increases in importance. This contribution is 90° out of phase with the pressure fluctuations, as signified by the introduction of the complex number j in (7.1). It is caused by the local hydrodynamic flow, as the sound source is pushing the medium particles back and forth. Very close to the sound source in the *near field*, this term is completely dominating. For very small values of r, the parenthesis of (7.1) is reduced to $1/(j\ kr)$, and the acoustic impedance is now given by $Z = j\ k\ r\rho\ c$. This is quite different from the simple Z_0

= ρ c we encountered in the far field. The practical importance of this is that the particle velocities here are larger than you would expect from sound pressure measurements, and therefore can stimulate motion sensitive receptors more easily.

Another way to express the difference between the near and the far field is to say that the first term in (7.1), characterizing the far field, deals with the *elastic* properties of the medium, where the particle velocity is originating from the medium being compressed and rarefacted by the propagating pressure fluctuations. The second term of (7.1), which dominates in the near field, deals with the *inelastic* properties of the medium, where the medium is pushed back and forth very much like a stiff body. Between the near and far fields there is a transition zone, where both terms contribute significantly to the description of the particle motion.

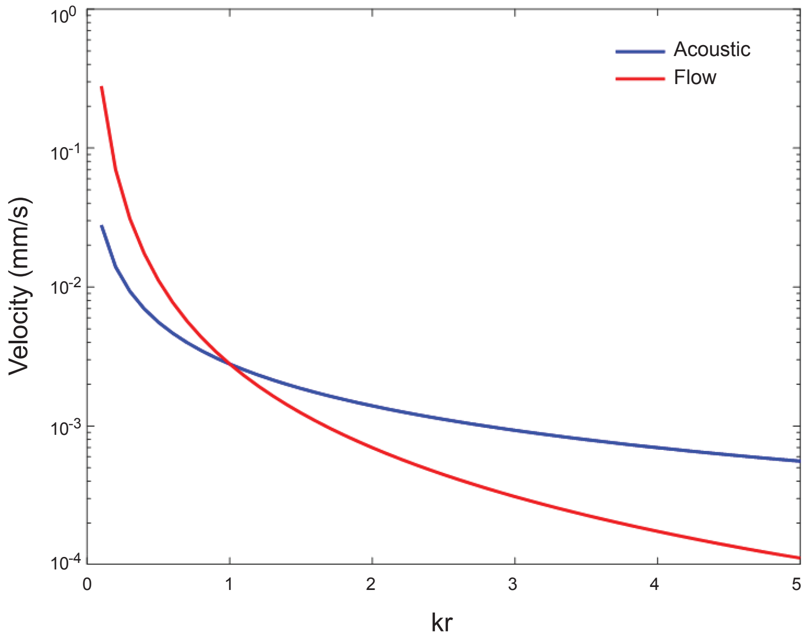


Fig. (10). The Flow near field around an underwater monopole sound source (source level 120 dB re 1 µPa @ 1m). The *blue line* shows the particle velocity generated by the propagating acoustic pressure wave (as calculated from $u = p/(\rho c)$, where p is the pressure and ρ is the density of the medium). The *red line* is the particle velocity *(u)* generated by the local medium flow around the sound source. The two components have equal magnitude at a distance corresponding to $kr = 1$. k is the wave number ($k = 2\pi/\lambda$, where λ is the emitted wavelength) and r is the range to the source (redrawn from Wahlberg and Westerberg, 2005).

The near and far fields we discuss here, we will call the ***Flow near field*** and the ***Flow far field***, respectively. They are also known as the *reactive near field* and the *active far field*. This is to discern them from another type of near and far fields of importance in bioacoustics and discussed below.

For a monopole sound source, if $kr << 1$ we say that we are well within the flow near field, whereas for $kr >>1$ we are in the flow far field. For $kr = 1$ the contributions of the flow and pressure fields are of equal size – this is evident both from Fig. (**10**) and by inserting $kr = 1$ into (7.1). For other sound sources than monopoles, the relationship between the two parts of particle motion is not as simple as for the monopole, even though it can still sometimes be calculated. For example, for a dipole the extension of the near field is not only a function of kr but also of the direction to the sound source: in the axial direction of the dipole, the flow near field extends to several kr, whereas in the plane perpendicular to the dipole the flow near field extends infinitely far, as there is no pressure component at all – only flow (cf. Fig. **7b**).

The concept of the flow near and far fields is very hard to grasp, so you may need some time to appreciate its virtues. For the time being it suffices to realize that the distance from the monopole sound source where the local flow near field has a major effect is dependent on the wave number, k, *i.e.* the effects are most pronounced at low frequencies and stronger in water (four times as large wavelengths for a given frequency) than in air. For example, for a monopole oscillating at 100 Hz in water the flow component will dominate up to 2.5 m away from the source, whereas in air the distance would be about 60 cm.

Why is all this important? For example, if you are interested in studying low-frequency fish hearing, a thorough understanding of the local flow near field is absolutely necessary. Fish can detect the local flow around a loudspeaker. When stimulating fish with sound you must therefore control not only the pressure component of the sound field, but also the particle velocity component. While this may easily be done in the far field it may be very tricky to accomplish within the Flow near field (Sand & Karlsen, 1986; Zeddies *et al.*, 2012). If you cannot control the sound field you must at least measure the particle velocity component as well as the sound pressure. This can be easily done by measuring the pressure gradient with two microphones or hydrophones, as described above.

7.2. The Interference Near Field

There is another type of near field that creates additional hassles to bioacousticians, the interference near field. Any real sound source has measurable physical dimensions and sound waves produced by different parts of the source add to each other by linear superposition (interference; cf. again *Huygens' Principle*). Far from the sound source, the sound rays are approximately parallel, and the interference pattern is a simple function of the *direction* to the source (see Section 8 below). Close to the sound source, however, the interference pattern is a more complicated function of *both direction and distance* to the sound source. For electromagnetic waves (*e.g.* radiation from an antenna), the corresponding phenomena are known as the *Fraunhofer far field* and the *Fresnel near field*, respectively. In the acoustic literature, this type of near field is rarely given any other name than just *near field*. Here we will call it the ***Interference near field*** to distinguish it from the *flow near field* described above.

In Chapter 2 we will show that in a free acoustic field the sound pressure decreases with $1/r$, where r is the range from source to observer. There is, naturally, a limit to how close to a sound source you can get while this relationship still holds true as the sound pressure cannot become infinitely high, when r becomes infinitely small. What happens is that when you move closer and closer to the sound source, the distances to various parts of the source become different and the source can no longer be considered a point source (with all sound rays being emitted from a single point). Instead, the source may be described as subdivided into a number of smaller sources, the contributions of which create a more complicated interference pattern.

This phenomenon can be illustrated considering only two sound sources (monopoles) spaced by a distance a (as in Fig. (**11a**)). Let us imagine we are standing at a distance r perpendicular to a line connecting the two sound sources, each of which emits a signal of the same amplitude and wavelength λ. At any point in the sound field the two waves will be added by ***linear superposition***. This means that in locations where they are completely in phase the sound pressure will be doubled (+ 6 dB), whereas in locations where they are completely out of phase the sound pressure will be reduced to zero. To obtain a complete negative interference the path length difference (Δ) from the two sound sources should be an odd multiple of $\lambda/2$ ($N\ \lambda/2$, $N = 1, 3, 5$ ...). *Note that the closer you are to the sources, the larger is the path length difference (Δ)*. At very large range, Δ becomes so small that it is not possible to fit half a wavelength into it. Thus the maximum range with negative interference must be the one where the path difference $\Delta = \lambda/2$. Beyond this range there can be no completely negative interference between the two sounds rays. From Fig. (**11a**) we see that this range may be expressed as $(s - r) = \lambda/2$ or by using the Pythagorean Theorem:

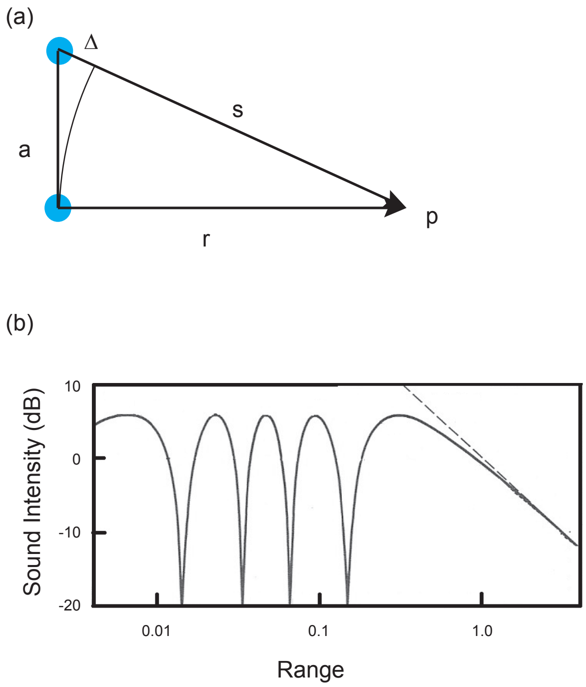


Fig. (11). The interference near field. **(a)** Two monopoles create a complicated interference pattern, which at close range depends on both the direction and the distance to the sound sources (for further explanation see text). **(b)** The Interference near field in front of a piston (ka = 30) transducer (*solid line*). Within the near field, there are large spatial variations in the sound intensity due to interference between sounds generated at different parts of the source. In the far field, path differences from sound generated at different parts of the source to the receiver become negligible, and the sound level usually follows a predictable geometric spreading law added up with absorption loss (in this case spherical spreading, *dashed line*). The range is given in units of a (the radius of the transducer). The decibels are dB re. source level (back-calculated sound intensity at 1 m). Note that in the near field, the sound level can be up to 6 dB higher than the source level. Redrawn and modified from Blackstock (2000).

$$\sqrt{a^2 + r^2} - r = \frac{\lambda}{2} \qquad \textbf{(7.2)}$$

This relationship is only exactly valid when there is a right angle between *r* and *a*, but it is approximately valid also for other angles. Assuming that *r* >> *a*, we can divide both sides of (7.2) by *r*. As a/r is very small, the left side of the equation can then be simplified and the equation becomes

$$\sqrt{\frac{a^2}{r^2} + 1} - 1 \approx \frac{a^2}{r^2} + 1 - 1 = \frac{a^2}{r^2} = \frac{\lambda}{2r} \quad or \quad r = \frac{2a^2}{\lambda} \qquad \textbf{(7.3)}$$

We can thus deduce that no interference minima can be found at ranges greater than about

$$r > \frac{2a^2}{\lambda} \qquad \textbf{(7.4)}$$

Most sound sources cannot be modelled by only two monopoles, but rather by a 'continuum of monopoles' along the physical structure that vibrates, for instance a piston (or the cricket wing in Fig. (**2**)). Surprisingly enough, in most situations (7.4) still remains a good rule-of-thumb for estimating the range, beyond which no destructive interference can occur from rays originating from different locations on the sound source. Thus, the limiting range is given by both the size of the sound source, but also by the emitted wavelength. The extent of the interference near-field is therefore correlated to the size of the transmitter and the inverse of the wavelength. Fig. (**11b**) depicts the interference near field and the far field for a spherical piston source. The interference near field part can look quite different for different types of sound sources and in different directions to the sound source.

Referring to Fig. (**11b**) we could also say that many biological sound sources may be approximated by a vibrating circular piston in a baffle and acousticians like to calculate the sound field created by a piston source. The complicated equations resulting from such calculations can be interpreted to show (but do not take it too literally) that the piston radiation starts out like a collimated plane-wave beam that propagates out to a certain distance, the so-called *Rayleigh distance*, $R_0 = ka^2/2$.

From here the beam spreads out spherically and roughly marks the end of the interference near field with its maxima and minima as explained above, but true far field conditions do not exist until a distance of at least $2R_0 = ka^2$ (Blackstock, 2000). So, a circular piston model predicts far field conditions for ranges where:

$$r > \frac{2\pi a^2}{\lambda} \qquad \textbf{(7.5)}$$

According to this interpretation, to be sure to measure in the far field of your laboratory setup when the piston (*e.g.* loudspeaker diaphragm) radius is, for instance, a = 5 cm and the lowest sound frequency you use is 2 kHz (λ = 17 cm), you should be at least 9 cm or more than λ/2 away from the source - but it is always better to be well over on the safe side, so as a rule of thumb a distance of λ is better.

7.3. Bioacoustic Examples

Table **3** summarizes some examples of near-field ranges for different types of biological sound sources. Note that in general, lowering the frequency of the signal expands the extent of the flow near field while it decreases the interference near field, whereas the opposite is true when increasing the frequency content of the signal. It should now be clear why fish bioacousticians only worry about the flow near field and never about the interference near field: the ratio between source size and emitted wave length is never large enough for fish to cause any destructive interference pattern at ranges of biological importance from the calling fish. You can also explain why bat bioacousticians only consider the interference near field: at ultrasonic wavelengths, the extension of the flow near field will not have any biological importance. The extension of the interference near field can be considerable, on the other hand, due to the relatively large size of the sound source compared to the emitted wave length.

The *ka* products derived in Table **3** can also be used in Fig. (**8**) to understand how efficient the animal is in producing a propagating acoustic pressure wave. Some of the animals have huge *ka* products, which does not really seem necessary for maximizing sound production efficiency. Instead, these high *ka* products are used to obtain a highly directional signal, as discussed below.

Table 3. Some examples of sound sources and their approximate near-field ranges. *a* is the approximate effective radius of the sound emitter aperture (*e.g.*, for the blackbird and the bat it is the size of the mouth opening). *f* and *λ* are the characteristic vocalization frequency and wavelength of the source.

Sound source	Sound production mechanism	Source type	f (Hz)	λ (m)	a (m)	Interference near field (m)*	Flow near field (m)**
Cricket	Wing	Dipole	5,000	0.068	0.003	0.0008	0.04
Blackbird	Syringeal labia	Piston	2,000	0.17	0.015	0.008	0.1
Bat	Vocal folds	Piston	50,000	0.007	0.005	0.02	0.004
Tweeter	Membrane	Piston	2,000 20,000	0.17 0.017	0.03	0.03 0.3	0.1 0.01
Woofer	Membrane	Piston	200 2000	1.7 0.17	0.08	0.02 0.2	1.1 0.1
Cod	Swim bladder	Monopole	200	7.5	0.01	0.00008	5
Sperm whale	Sphincter	Piston	15,000	0.1	0.4	10	0.06
Harbor porpoise	Sphincter	Piston	135,000	0.011	0.02	0.2	0.007
Underwater loudspeaker	Piezo-electric ceramic	Piston	500 5,000	3 0.3	0.05	0.005 0.05	2 0.2

*The extension of the Interference near field is calculated as $2\pi a^2/\lambda$.

**The extension of the flow near field is estimated as the range, at which $kr = 4$.

8. DIRECTIVITY

For many animals it is desirable to be able to direct their sound signals to an intended receiver (Larsen & Dabelsteen, 1990) or to use it for echolocation (*e.g.* Jakobsen *et al.*, 2013). Directional sound production may be accomplished using a dipole sound source. The intensity will be maximal on the acoustic axis of the dipole, while no sound escapes in the plane normal to the center of the acoustic axis of the dipole (cf. Fig. **7b**).

Instead of using a dipole we may construct directional sound from an array of elementary monopole sound sources. In the *interference far field*, so-called *Fraunhofer interference* between the monopole sound sources creates positive and negative interference in various directions to the sound source due to variations in the path lengths from different monopoles and thereby phase-shifting of the signals. This is exactly what also happened *within* the *interference near field* discussed above for sound components originating from different locations on the sound source. The only difference is that the interference pattern created by Frauenhofer interference is *not* dependent on the distance (range) *but only on the direction* to the sound source. This can be seen if you consider the simple example of two monopole sound sources, situated the distance of a from each other and seen from a point P in space located at a distance of r and s from the two monopoles (Fig. **12a**). The cosine theorem tells us that the difference in distance between two paths r and s to the two sound sources is given by

$$\Delta = |r - s| \approx a\cos(\theta) \qquad \textbf{(8.1)}$$

which means that Δ only depends on the direction θ when $a/r << 1$. Frauenhofer interference thus defines the far field *directional characteristics* of the source. Compare this to the Interference near field zone treated above and in Fig. (**11b**), where the interference pattern highly depends on the *range* from the source.

To obtain completely destructive interference, which is a prerequisite to get a directional sound source, we want the path length difference to be $\Delta = N\,\lambda/2$, where N is an odd integer number. Another way to say this is that we may only get far-field negative interference, if we have $\Delta \geq \pi/2$. With the approximation made above this implies that $a\cos(\theta) > \lambda/2$; for modest cosine angles, where $\cos(\theta) \approx 1$, we thus want $a \leq \lambda/2$. Usually though, we like to use the wave number k instead of λ, and once again we can see that it is the ka product, which determines if a sound source is directional or not: $a > \lambda/2$ is the same as saying that $ka > \pi$. Even though this relationship was derived using two monopole sound sources, it is similar for many types of sound sources having a radius of a.

In Fig. (**12b**) you can see what the directional beam pattern looks like from piston transducers with different values of ka. Note that the $ka > \pi$ is not an absolute limit, but rather a smooth transitional region: *e.g.* even the sound source with $ka = 2$ shows some directional properties. Highly directional beams of, for instance, $ka = 10$ are used by echolocating animals like bats and dolphins (*e.g.* Jakobsen *et al.*, 2013). This helps the echolocator to determine the direction to any target of interest and also reduces the amount of unwanted echoes, so-called clutter. For songbirds and seals, on the other hand, the beams are closer to $ka = 2$ (*e.g.* Larsen and Dabelsteen, 1990). These signals are used for communication with conspecifics located in unknown directions from the caller and therefore it is desirable to make the signal as omnidirectional as possible.

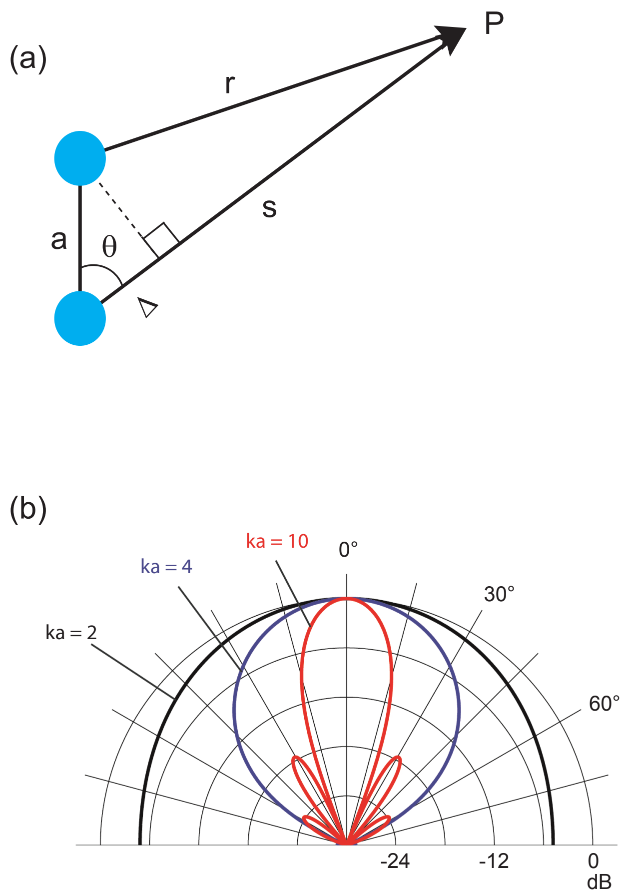


Fig. (12). Directional sound production. **(a)** Two monopoles emitting signals of identical frequency, amplitude, and phase are separated by distance *a* and their combined contributions to the sound field are measured at position P, where the pressure depends on the difference in path length Δ, which is approximately (see text) given by acos(θ). **(b)** The directionality of a piston calculated with different *ka*-values and varying from almost omnidirectional to very directional. 0° is on-axis. Note how side lobes occur for large values of *ka* (Courtesy of Lasse Jakobsen).

For continuous signals, beam patterns tend to contain many maxima and minima in off-axis directions, due to destructive interference of sound emitted from various parts of the source (Fraunhofer interference). For transient signals, the beam pattern tends to vary more smoothly off axis, as the signal duration is not long enough to develop completely destructive interference patterns.

You may boost your directionality using different ‘tricks’, for example by using an ***acoustic horn***. Horn-like structures are common in animals: for example, mole crickets create horn-shaped burrows, from which they emit their songs (Bennet-Clark, 1970), and the vocal tracts of mammals are usually shaped like horns. We know horns best from musical instruments, where the final piece of brass instruments is ‘folded out’ towards the listener. This helps transforming the sound source from a small size (with a small *ka* product and therefore little directionality) to a much higher *ka* product (keeping the frequency constant but increasing ‘*a*’ at the end of the horn) resulting in a much more directional signal with the horn in place than without it.

To further characterize the different directional patterns, the amplitude ***directivity factor (Q)*** is sometimes used. It is defined as the pressure at any angle θ relative to that (at the same range r) at $\theta = 0$, for which the peak radiation occurs (Blackstock, 2000). A quantitative measure of the directionality of the beam pattern is the ***half-power beamwidth***. It is $2\theta_{3dB}$, where θ_{3dB} denotes the angle where the pressure at a given distance has been reduced by 3 dB relative to the peak pressure at $\theta = 0$. For a circular piston of radius a the half-power beamwidth is $2\arcsin(1.616/ka)$, which for large values of ka approaches $3.232/ka$ (Blackstock, 2000). If for instance $ka = 10$, then the half-power beamwidth is 0.323 radians or 19°, see Fig. (**12b**). Another often used measure in modeling is the ***directivity index****(DI)*, which simply equals $20\log(ka)$ for a circular piston.

A highly directional signal is useful for echolocating animals, as it prohibits unwanted echoes from returning to the bat or toothed whale. From Table **2** we can calculate the *ka* product of echolocators, such as bats ($ka = (2\pi/0.007)\ 0.005 = 4.5$) or sperm whales ($ka = (2\pi\ /0.1)\ 0.4 = 25$), which makes a high directionality achievable. On the other hand, communication sounds such as the song of crickets ($ka = (2\pi\ /0.068)\ 0.003 = 0.3$) and blackbirds ($ka = (2\pi\ /0.17)\ 0.015 = 0.6$) are suitable for generating omnidirectional signals of high efficiency (cf. Figs. (**8** and **12b**).

8.1. Source Level

Since sound pressure level depends on the distance to the source, the concept of ***source level*** *(SL)* is introduced as it can be used to characterize the relative strength of different sound sources. *SL* is the sound pressure level measured at a distance (far field) in the direction of the highest source level ('on axis' cf. Fig. (**12b**) at $\theta = 0°$) and back-calculated to a distance of 1 m from the sound emitter, which is modelled as a point source. For a sound source like the cricket singing at 5 kHz (Fig. **2**) the *SL* can be directly measured at a distance of 1 m from the source. In most cases *SL* is an abstraction as it differs from the actual sound pressure level 1 m from the source, which may be within the near field – or for instance deep inside the head of a blue whale. Note that source levels in bats are normally back-calculated to a distance of 10 cm from the source. So, to compare published bat source levels with source levels of all other animals you need to subtract 20 dB, which makes bat source levels less impressive.

Precise measurements of source levels and of directionality of animal signals are not that easily performed, but that should not deter you from performing the measurements. You need an array of calibrated and synchronized receivers, and you usually need some way to reassure, in which direction the animal is oriented. Still, going through such ordeals can render some surprising findings. For example, the directional pattern of sage grouse vocalizations is much more complex than expected (Dantzker *et al.*, 1999). This is caused by the action of two sound sources, situated next to each other in the bird. Bats have recently been shown being able to modify the directionality of their sound emissions, so that they can use narrow, powerful clicks at long range from the prey and then 'open up' for the sound beam as they approach the prey, optimizing their chances of keeping track of and catching their food (Jakobsen and Surlykke, 2010).

Some animals, such as bats and toothed whales, are extremely loud. Actually, their sound signals approach the limit where sound production becomes inefficient due to non-linear effects. In air, beyond some 150 dB re. 20 µPa, the sound level is sufficient to cause excessive attenuation and distortion of the travelling wave. In water, this happens above about 240-250 dB re. 1 µPa. However, already around 220 dB re. 1 µPa, underwater sounds produced close to the surface can start becoming heavily distorted: 220 dB re. 1 μPa corresponds to a pressure of 100 kPa, which is about the ambient atmospheric pressure. For the negative components of the signal, it is not possible for the water close to the surface to generate a pressure that is below the atmospheric pressure, or 220 dB re.1 μPa. In this situation the water approaches pressures sufficiently low to generate bubbles, which collapse with a loud transient sound as soon as the pressure wave has travelled by and the ambient pressure is restored. This phenomenon is called ***cavitation*** and is of interest both in studies of propeller sounds from boats, but also in some peculiar cases of animal sound production (*e.g.*, snapping shrimps as shown by Versluis *et al.*, 2000).

CONCLUSION

Bioacoustic studies demand a keen knowledge of the underlying physical principles of sound sources and the sound fields they create. This knowledge can also be used to derive useful and testable hypotheses about sound production and reception. For example, the observation of bats changing the directionality of their calls immediately poses the interesting question: how is this accomplished? As directionality is dictated by the *ka* product, the bat could either change *k* (that is the wavelength or the frequency content of its signals) or *a* (the size of the sound source). Lowering the frequency emphasis of the call would decrease *k* and therefore also the directionality of the emitted signal. Likewise, increasing *a*, for example by opening the mouth, would create a signal with higher directionality.

PRACTICAL ADVICE

Besides understanding the physical limitations of sound production, bio-acousticians need to take great care when designing their experiments and analyzing the signals. Measurements of source levels and directionality can only be made if the receivers have been carefully *calibrated*, *i.e.* the settings have been adjusted so that the equipment correctly measures relative to standardized reference values. For trustworthy results check of *calibration* should be made before and after measurements to reassure the equipment has been working properly.

It is difficult to navigate through theoretical (and often mathematically) oriented books about acoustics explaining the concepts of sound sources and sound fields. There are many examples of bioacousticians creating fruitful alliances with engineers or physicists who can help them navigating through these problems. For many engineers, bioacoustics can become both an exciting and exotic diversion from their normal fields of study. Also, for constructing custom-built gear, which is often necessary to perform good measurements, collaborating with a skillful electro technician who is interested in your topic can be extremely rewarding.

SOFTWARE NOTES

For custom-built recording software we often rely on LabVIEW. We use commercial sound analysis packages, such as Adobe Audition, for the initial analysis of acoustic data. There are also some great software packages, such as Raven, Avisoft-SASLab Pro, and BatSound, specially created for bioacoustics applications. For more specialized analysis usually custom-made programming routines have to be created in *e.g.*, Matlab.

CONFLICT OF INTEREST

The authors declare they have no conflict of interests for this publication.

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