Practice Guidance:

Acoustic Considerations for Audiology Clinics

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General foreword

This document presents Practice Guidance by the British Society of Audiology (BSA). This Practice Guidance represents, to the best knowledge of the BSA, the evidence-base and consensus on good practice given the stated methodology and scope of the document at the time of publication. Although care has been taken in preparing this information, the BSA does not and cannot guarantee the interpretation and application of it. The BSA cannot be held responsible for any errors or omissions, and the BSA accepts no liability whatsoever for any loss or damage howsoever arising. This document supersedes any previous recommended procedure by the BSA and stands until superseded or withdrawn by the BSA.

This document includes guidance on sound field measurements, so the previous Practice Guidance "The Acoustics of Sound Field Audiometry in Clinical Audiological Applications" (BSA, March 2019) is superseded by it.

This document will be reviewed by the date given on the front cover. However, should any individual or organisation feel that the content requires immediate update, review or revision, they should contact the BSA using the email bsa@thebsa.org.uk. Please add 'BSA document revision request' in the title. You will be asked to complete a short form with your reasons and this will be passed to the Professional Guidance Group for assessment. Comments on this document are welcomed and should be sent to:

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Shared Decision-Making

It is implied throughout this document that the service user should be involved in shared decision-making when undertaking audiological intervention, receiving subsequent information and understanding how it will impact on the personalisation of care. Individual preferences should be taken into account and the role of the clinician is to enable a person to make a meaningful and informed choice. Audiological interventions bring a variety of information for both the clinician and the patient which can be used for counselling and decision-making regarding technology and anticipated outcomes.

Contents

1. Abbreviations

dB	decibel (dimensionless)
dB(A)	A-weighted dB
dB(C)	C-weighted dB
dB(Z)	Unweighted dB
dB HL	dB Hearing Level
dB nHL	dB normal Hearing Level (or dB above normal Hearing Level)
dB SPL	dB Sound Pressure Level relative to 20 µPa (usually the exponential time-weighted SPL unless otherwise stated)
dB pSPL	dB peak SPL
dB b-p peSPL	dB baseline-to-peak peak equivalent SPL
dB p-p peSPL	dB peak-to-peak peak equivalent SPL
f	Frequency (Hz or s^{-1})
I	Sound intensity (W/m^2)
IL	Intensity level (dB) relative to 1 pW/m^2
L_{Aeq}	Equivalent continuous A-weighted sound pressure level (dB) relative to 20 µPa
$L_{Aeq,T}$	L_{Aeq} averaged over a time-period T
$L_{EP,d}$	Daily personal noise exposure level
$L_{S,max}$	Maximum slow time-weighted SPL detected during the measurement period
NBN	Narrow-band noise
p	Amplitude of the change in air pressure due to acoustic perturbation (Pa or Nm^{-2})
PSEM	Personal sound exposure meter
REM	Real-ear measurement
RETSPL	Reference Equivalent Threshold SPL (dB), measured in a coupler
RTSPL	Reference Threshold SPL (dB), measured in a sound field
rms	Root-mean-square
RT	Reverberation time (seconds)
SI	The international system of units (Système International)
SLM	Sound level meter
SNR	Signal-to-noise ratio (dB)
SPL	Sound Pressure Level (dB) relative to 20 µPa

2. Purpose and Scope of this Document

Acoustic considerations apply to all types of measurement conducted in audiology clinics. Guidelines for acoustic considerations have therefore been combined into this document which should be used in combination with other BSA Recommended Procedures and Practice Guidance. This document includes guidance on sound field measurements and therefore replaces the previous Practice Guidance "The Acoustics of Sound Field Audiometry in Clinical Audiological Applications" (BSA, March 2019).

The document is intended to give an overview of acoustic considerations that **all audiologists should understand and apply within their clinical roles**. Acoustic effects can undermine accurate measurement in many situations and clinicians shall remain alert to potential mistakes, variability and uncertainty. Although some elements of acoustics may appear technical to practising clinicians, it is important that these effects are taken into account. However, it is recognised that the guideline should be pragmatic and accessible so that its principles can be readily applied in clinic, so some of the guidance is simplified to avoid the need for complex calculations that may be impractical in a clinical setting.

This document is not intended to provide guidance on the design and construction of audiology test rooms and should not be used as such. However, some of the information presented (e.g. maximum ambient noise levels) could usefully be included in the specification for any new audiology facility.

BSA Practice Guidance documents are intended to provide evidence-based, practical guidance to clinicians. Explanations of some basic concepts have been included here for ease of understanding. Summary guidance is provided at the end of the document.

3. Basic considerations and definitions

This document contains brief descriptions of acoustic principles that should be understood by all registered clinicians in audiology. Further detail can be found in Kinsler et al (2000) which describes the key physical and engineering concepts. It is recommended that clinicians requiring a more accessible, non-specialist text should refer to Speaks (2018).

3.1 Sound propagation

Sound energy propagates as a vibration. In air, gas molecules are displaced towards and away from the source of sound, causing small variations in air pressure relative to the atmospheric pressure as the air is compressed and expanded at each point. We do not need to consider the molecular level further because there are about 4×10^{20} molecules in every cubic metre of air, which is far too small to be consequential at the wavelengths of audible sound (Speaks, 2018, p.12). It makes more sense to think of separate masses

of air jostling into each other along the direction of the wave propagation. Propagation of sound through air therefore depends on its elasticity and density, and these properties will be further dependent on other factors such as temperature, relative humidity and atmospheric pressure. In dry air, at normal atmospheric pressure, the speed of sound in air will increase from 331 ms^{-1} at 0°C, to 343 ms^{-1} at room temperature (20°C), to 352 ms^{-1} near body temperature (e.g. 35°C, in the ear canal).

Imagine an idealised sound source which radiates sound equally in all directions. In more technical terms, an ideal sound source is a "monopole" that is much smaller than the wavelength it radiates. As pressure changes propagate from a monopole, the sound wave can be thought of as a sphere increasing in size away from the source. Sound energy will be equally spread across the surface of the sphere, so the amount of energy at any one point can be calculated by dividing the total energy by the surface area of the sphere (= $4\pi r^2$, where r is the distance from the source at the centre of the sphere). This gives rise to the "inverse square law" (or the "inverse sound pressure distance law") for sound propagation, meaning that the intensity of sound reduces in proportion to r^2, the square of the distance from the source. In decibel terms (see Section 3.3) this means that both the sound intensity level (IL) and sound pressure level (SPL) decrease by 6 dB with every doubling of distance from the source. Sound propagation beyond the ideal situation of a "free field" is a much more complex engineering problem because of the effects of atmospheric absorption of sound, meteorology, the ground and other objects causing absorption, reflection and diffraction. Furthermore, many real sound sources will not be "ideal" and will have directional characteristics.

Sound travelling through air will come into contact with other objects. The vibration in air will excite vibration in these objects depending on their physical characteristics and the difference in impedance between mediums. This determines how much sound is absorbed by the object or reflected, and how much is transmitted through the object. Whilst we do not aim to detail these complex processes here, it is important to understand that these relationships are frequency dependent. Lower frequencies with larger wavelengths will transmit through solid objects much better than high frequencies. This is why, for example, you can hear the bass of music through a wall if your neighbours are having a party, but not make out the tune because higher frequencies are not audible. Likewise, lower frequencies will diffract around objects or through gaps in objects better than higher frequencies. This is because sound waves will pass around an object more easily when the wavelength is large in relation to the size of the object. We can demonstrate this by sticking our fingers in our ears and speaking. Bone-conducted sound normally reaches our ear canals and leaks out of the open canal. Sticking our fingers in our ears traps this sound and we notice a more "boomy" (low frequency) voice because the sounds that previously escaped, and have now been trapped, were predominantly low frequency. Likewise, this explains why low frequencies leak more from a poorly-fitting earmould or why high frequencies are more attenuated by ear wax in an ear canal.

Understanding the behaviour of sound in a real acoustic environment starts by considering its wavelength. There is a simple equation that relates the frequency f (Hz), wavelength λ (m) and speed c (ms^{-1}) of sound:

$$c = f \lambda$$

This should be well-understood by clinicians because it provides a ready tool to calculate where certain acoustic phenomena might impede or affect measurements in clinic, and some wavelengths are given in Table 1. For the range of tones used in normal audiometry, the wavelength of sound in air declines from approximately 2.7 metres at 125 Hz, to 4 cm at 8 000 Hz.

Table 1: the wavelength of sound at different frequencies travelling in dry air, at room temperature (20 °C) and normal atmospheric pressure.

Frequency (Hz)	Wavelength (m)
125	2.74
250	1.37
500	0.69
1 000	0.34
2 000	0.17
3 000	0.11
4 000	0.09
6 000	0.06
8 000	0.04

3.2 Room acoustics and resonances

As sound travels away from a source, it is relatively straightforward to estimate the decrease in sound level with distance in a free field, using the inverse square law. It is much harder to calculate the level of sound in an enclosed space because this introduces many other factors. When sound hits a surface it will be partly absorbed or reflected, varying with frequency. Sound may be reflected in simple ways, diffused or diffracted, depending on the wavelength.

A **free field** is defined where there are no objects impeding the progress of sound, or the surfaces of a room are 100% absorptive, and the sound level decreases by 6 dB each time the distance is doubled from a source.

A **diffuse field** is defined where the sound energy is equal at every point in a space, with the energy equally distributed in all directions.

Neither a free field nor a diffuse field usually exists within a typical room, unless in an anechoic chamber or reverberation room of a dedicated acoustics facility. In reality, rooms are somewhere between the two. A **quasi-free field** is defined where the acoustics behave like a free field within part of a room, usually near its centre. BS EN ISO 8253-2:2009 defines a quasi-free sound field more strictly (see Box 1). The deviations are so small that it is unlikely that any room in a typical clinical setting would meet this definition. In addition, the standard does not provide any guidance on the measurement uncertainties which would have to be taken into account when making the sound field measurements. Typically the deviation in sound level around the reference point would be greater than ± 2 dB (see Section 5).

Box 1: BS EN ISO 8253-2:2009 definition of a quasi-free sound field.

1. The loudspeaker is at least 1 m away from the reference point, at head height;
2. The reference point is the point where a patient's head would normally be (the mid-point between the ears);
3. All measurements are taken in the absence of the patient and the patient's chair, with everything else as it would normally be in clinic;
4. The sound does not deviate by more than ±2 dB from the theoretical value given by the inverse sound pressure distance law when measured 0.15 m to the left and right, above and below the reference point;
5. The sound does not deviate by more than ±1 dB from the theoretical value given by the inverse sound pressure distance law when measured 0.1 m to the front and back of the reference point;
6. The usable frequency range shall be those frequencies at which the above points are satisfied.

Another important aspect of room acoustics is the **reverberation time** (RT), defined as the time taken for a sound to decay by 60 dB from its peak level. RT will increase in proportion to the volume of a room and will decrease in proportion to the total amount of acoustic absorption. RT has an important relationship with speech intelligibility. In a room with many echoes and a long RT (e.g. a church) echoes of the same signal will arrive at many different times, making it much harder to understand. NHS Estates (1994) recommended that RT should be less than 0.25 seconds in audiology clinics, although this guidance was withdrawn and is now unavailable (NHS England, 2021). BS EN ISO 8253-2:2009 does not give guidance on an acceptable RT. However, in general, the shorter the RT the more speech intelligibility is improved (Yang & Hodgson, 2006; Puglisi et al, 2021).

A critical aspect of sound measurement in enclosed spaces is the formation of **acoustic resonances** or **standing waves**. As pressure waves are reflected by surfaces, the sound field is a complex pattern of interference, with pressure cancelling and reinforcing itself at different points in space. Where a space has similar dimensions to the wavelength of sound, there may be a point at which sound waves are aligned and in phase, creating perfect reinforcement and cancellation of, what appears to be, a stationary wave.

There will be several modes of resonance possible. Consider a guitar string which is fixed at both ends (fig.1a). When the string is plucked, it will tend to vibrate in the resonant modes f_0, f_1, f_2, etc, where f_0 is referred to as the **fundamental frequency.** Because the string is fixed at each end, it cannot move, which restricts the modes of resonance to half a wavelength ($\lambda/2$), a full wavelength (λ), and one-and-half wavelengths ($3\lambda/2$), then 2λ, $5\lambda/2$, 3λ, $7\lambda/2$, etc. Guitars have strings of different thickness and tension, and it is by changing these physical properties that determines the frequency at which the dominant modes of vibration occur and the "note" of the string.

The same principle can be applied to sound waves in air. Sound waves in air are longitudinal, i.e. the masses of air are displaced by sound along an axis away from the centre of the sound source, but the displacement of the air mass can still be represented in the same way. Consider air in a tube, closed at both ends (fig.1b), which is analogous to the guitar string. The air cannot move (it has zero "particle velocity") at the ends because it is restricted by the wall of the tube. Resonant modes can be pictured in exactly the same way as for the guitar string, with points of maximum and minimum air particle velocity.

It should be noted that particle velocity is not the same as air pressure; in fact, maximum air pressure will be at the same point as the minimum particle velocity, and vice versa, so maximum pressure will occur at the fixed ends of the tube (fig.1b). This is because air molecules can still be compressed together at the end of the tube (a pressure maximum, or antinode), whereas they couldn't be compressed at an open end of a tube because this will always be the same as the pressure in the surrounding space outside the tube. If we were to move a microphone along the tube in fig.1b, we would measure zero pressure (equivalent to an infinitely negative SPL) at the **null points** or **nodes**, marked as "N" in fig.1b, and we would measure maximum pressure and SPL at the antinodes, marked "A". **The crucial point for clinicians to consider is that a measurement of SPL will be entirely dependent on where the microphone is placed for some frequencies of sound in an enclosed space.**

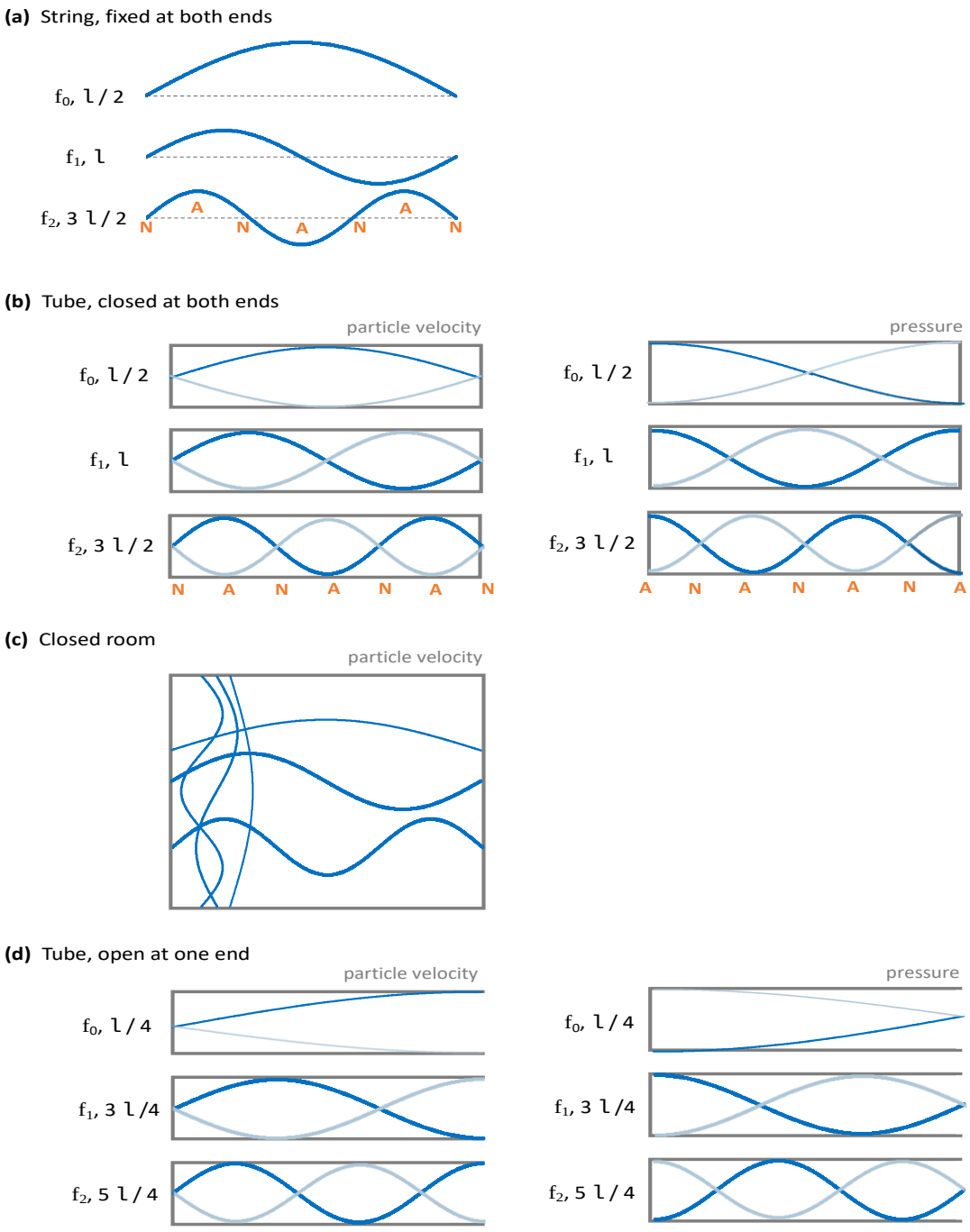

Figure 1: Description of resonant modes showing the variation in particle velocity (left) and pressure (right) with distance; A= Antinode, N= Node.

If we consider a room (fig.1c), there will be many resonances depending on the length, breadth and height of the room, as well as further resonances between objects, e.g. the gap between a sound booth and a wall. We can easily estimate resonant modes of a room from the equation above ($c = f \lambda$). If the room is 5 m long then, for the fundamental frequency f_0, $\lambda/2$ = 5m, so λ = 10 m. At room temperature, this is equivalent to a frequency of 34.3 Hz, with further resonant modes occurring at 69, 103, 137 Hz, etc. If the height of the room is 3 m, there will be other resonances at 57, 115, 172 Hz, etc. A 20 cm gap between a wall and cupboard would create resonances at 858, 1715, 2572 Hz, etc. It should be noted that, when we conduct sound field measurements in audiology, pure-tone stimuli are avoided. In these cases, narrow-band noise or warble tones are used to negate the worst effects of room resonances, but this does not mean that the effect of resonances can be ignored. Room resonances will be at the lower end of the frequency range used in audiology and will still affect warble tones and narrow-band noise. The placements of objects in the room (such as desks, cupboards and computer screens) will have an effect at higher frequencies.

One of the most critical areas in which resonances affect clinical audiology is during probe tube measurements in an ear canal. We can roughly approximate an ear canal as a tube that is 2.5 cm long, 1 cm wide, closed at one end and open at the other (fig.1d). The open end of a tube has to represent a minimum pressure point, or a node, of a standing wave because the air in the opening cannot be restricted. In this case the fundamental frequency f_0 occurs at $\lambda/4$, with further resonant modes at $3\lambda/4$, $5\lambda/4$, $7\lambda/4$, etc. If the air in the ear canal is warmed to 35°C, then resonances will occur at 3.5 kHz, 7 kHz, etc. For lower frequencies, a probe tube placed close to the eardrum will be closer to an antinode than a node. **At higher frequencies, there is a risk that a REM probe tube is placed at a node,** in which case no degree of gain added by a clinician will make a difference to the probe tube measurement. Of course, the sound field in a real ear canal is far more complicated than its representation by a simple tube: the ear canal is not a straight tube, the eardrum is not a rigid surface and absorbs sound, the presence of a standing wave affects how much sound is transferred to the ear canal, and there are other acoustic effects. Nevertheless, **clinicians shall understand that measuring sound levels at a single point in an enclosed space can be prone to significant error** and that this is especially important to remember whenever conducting probe tube measurements in an ear canal.

It should also be remembered that the pattern of resonances is altered when an ear canal is occluded because it then behaves more like a tube closed at both ends (fig.1b). The occluded ear will then have resonances starting at about 6 kHz and above. This also has important consequences during pure tone audiometry (PTA). Imagine an ear being occluded with headphones, which now creates a larger closed cavity than the ear canal alone, and the cavity size and shape will alter based on the position of the headphones. Let us approximate this by saying that the tube (fig.1b) is 5 cm long, which creates resonances at 3.5 kHz and above. Likewise, if we place inserts in the ear canal, we are shortening the closed tube and resonances will be at higher frequencies. Whilst these issues are, in principle, largely dealt with in the calibration process, **PTA with headphones can be variable at higher frequencies (4 kHz and**

above) because the exact strength and frequency of any resonance will depend on the placement of earphones and should, in any cases of doubt, be checked and altered.

In summary, **clinicians shall be aware that placing a single microphone, or probe tube, at a single point within an enclosed space does not fully represent the sound in that space**. Furthermore, **clinicians shall be aware that acoustic effects can lead to measurements being misleading in some situations**. These concepts should be part of the syllabus for training audiologists.

3.3 Description of sound levels

Measures of sound are typically described in terms of decibels (dB). Decibels are a dimensionless unit, i.e. they do not have an SI unit because they are a logarithm of a ratio based on a reference measure, and the ratio cancels out the unit measured. **When using decibels, it is therefore important to always state what reference value is being used.** A logarithmic scale is used for measurements of sound because of the enormous range of values and also because it is closer to the human perception of sound. For example, our perception of an equal increase in the level of sound is closer to a doubling of sound energy. For most listeners with normal hearing, a 0.5 to 1 dB change is just noticeable as the smallest perceptible change in an otherwise steady sound (p.151, Yost, 2007), whilst a 3 dB change should noticeable for most people and a 5 dB change would be obvious.

Sound energy expands outwards from a source of sound, as discussed in Section 3.1. If we add several sources of sound, it is the sound energy that adds up to increase the overall level of sound perceived by a listener. The measure of energy is the **sound intensity**, I, defined as the amount of sound energy transmitted per second over one square metre of space (W/m^2). This can be expressed as the sound **intensity level (IL)** by referencing it to a standard intensity of 10^{-12} W/m^2 which is 1 pW/m^2 i.e. 1 picowatt/square metre (BS EN 21683:1994):

$$IL = 10 \log_{10} (I / 10^{-12}) \text{ dB}$$

As sound propagates through air, it is more usual to think of it as perturbations in air pressure, p. Intensity is proportionate to the square of sound pressure, so we can reformulate the equation above as the **sound pressure level (SPL)** using the reference pressure of 0.00002 Pa (BS EN 21683:1994) which is close to the normal threshold of healthy human hearing:

$$SPL \text{ (dB)} = 20 \log_{10} (p/p_0) \text{ where } p_0 = 2 \times 10^{-5} \text{ Pa or } 20 \text{ } \mu\text{Pa}$$

An error is often made in utilising SPL, in which it is mistakenly presumed that adding two equal sound sources together will increase the SPL by 6 dB. Whilst it is correct to say that doubling the change in air pressure causes a 6 dB increase in SPL, this is *not* what happens when we add together sound sources. If we add a second loudspeaker to a room, emitting the same level of sound as the first, it is the sound intensity that adds together. There will be a complex interaction of pressure, depending on the phase relationships of the sounds, so that some pressure will cancel and some will add. Doubling the number of

sound sources will double the intensity only when the sources of sound are random (i.e. they are not correlated). Where two sound sources are not correlated and they both emit the same power, both IL and SPL will increase by 3 dB. For a more detailed description of this concept, refer to Speaks (2018, p.155). The important principle to understand is that **it is the sound intensity that should be used whenever we are adding sound sources or averaging sound exposure**. As is the case for many concepts in this document, these statements incorporate some simplification. Sound intensity is a vector, so assessing the net intensity at any point from multiple sound sources depends on the phase and direction of sounds.

When describing a level of sound it is important to also consider the amount of time over which it is being assessed. An instantaneous measure of the change in pressure in air generated by sound will vary over time, and sound levels are commonly described by the average change in pressure over a given time. It is therefore important to understand several different measures of SPL. These are defined below and greater detail can be found in Laukli & Burkard (2015). The concepts are summarised in fig.2.

Exponential time-weighted SPL

- This is the most commonly-used reading given by a sound level meter (SLM) and is frequently referred to simply as “SPL”, although technically it is defined as the unweighted SPL or dB(Z) (see Appendix).
- Sound pressure is averaged over a set time period. However, it should be noted that the sound pressure level cannot be determined from a simple average of the instantaneous values of the waveform. For example, averaging the positive and negative components of a sine wave would produce a result of zero. A root-mean-square (rms) approach is taken, where all of the instantaneous pressure values are squared (resulting in all positive values), averaged, and then the square root is taken of the average. For a sine wave, the rms value is always a factor of $\sqrt{2}$ less than the peak amplitude.
- Modern digital SLMs usually offer two exponential averaging times, known as time weighting 'F' and time weighting 'S'. These emulate the FAST or SLOW modes of older analogue SLMs, equivalent to a window of 0.125 seconds or 1 second respectively, and it usually takes about 4 time windows (0.5 to 4 seconds) for the SLM to reflect a steady sound. This is due to the exponential averaging of the time window measurements. This comprises a running average with the greatest weighting placed on the current measurements.

Peak SPL, pSPL

- This is the peak (maximum instantaneous positive or negative pressure) level of a signal.
- For a sine wave, pSPL will be 3 dB more than the exponential time-weighted SPL, reflecting the factor of $\sqrt{2}$ difference between the rms average and peak pressure value. This difference will vary between signals depending on the crest factor (the peak pressure divided by the rms pressure).
- Caution shall be exercised if using SLMs to measure peak pressure. The SLM will only record the correct value if its speed (integration time) is quick enough. The slower the SLM reacts, the smaller the peak amplitude will falsely appear. It is better to use a calibrated oscilloscope, or equivalent digital analyzer,

to assess peak amplitudes. Peak SPL will also be highly dependent on the frequency response of the measurement transducer.

- It should also be noted that the measure of peak SPL bears little resemblance to the sound that ultimately reaches the cochlea because energy will be spread out across the basilar membrane dependent on the nature of the signal, e.g. comparing clicks and chirps.
- For these reasons, peak equivalent measures are more commonly employed.
- It is important to note the difference between peak SPL and max SPL. Max SPL is the maximum SPL measured during a measurement run. It is sometimes implemented as "max hold" whereby the SLM only indicates the maximum SPL measured until a reset is carried out.

Peak equivalent SPL, peSPL

- This involves fitting an equivalent sine wave to the amplitude of a transient signal (fig.2b). It is important to recognize that there are two forms of peSPL and the format used should always be stated.
- The baseline-to-peak (b-p) peSPL fits a sine wave from its baseline to the peak (positive or negative) of the signal.
- The peak-to-peak (p-p) peSPL fits a sine wave between the maximum and minimum pressure of the signal. This is the measure more commonly used in audiology.
- It should be clear (fig.2) that the b-p peSPL can never be smaller than the p-p peSPL. If the signal is symmetrical (i.e. an impulse with equal positive and negative displacement, or a sine wave) then the b-p and p-p peSPL will be the same. In other words, p-p peSPL will be 0-6 dB less than b-p peSPL.
- peSPL is reported as the rms pressure of the equivalent sine wave, so peSPL is always 3 dB less than pSPL for a sine wave.

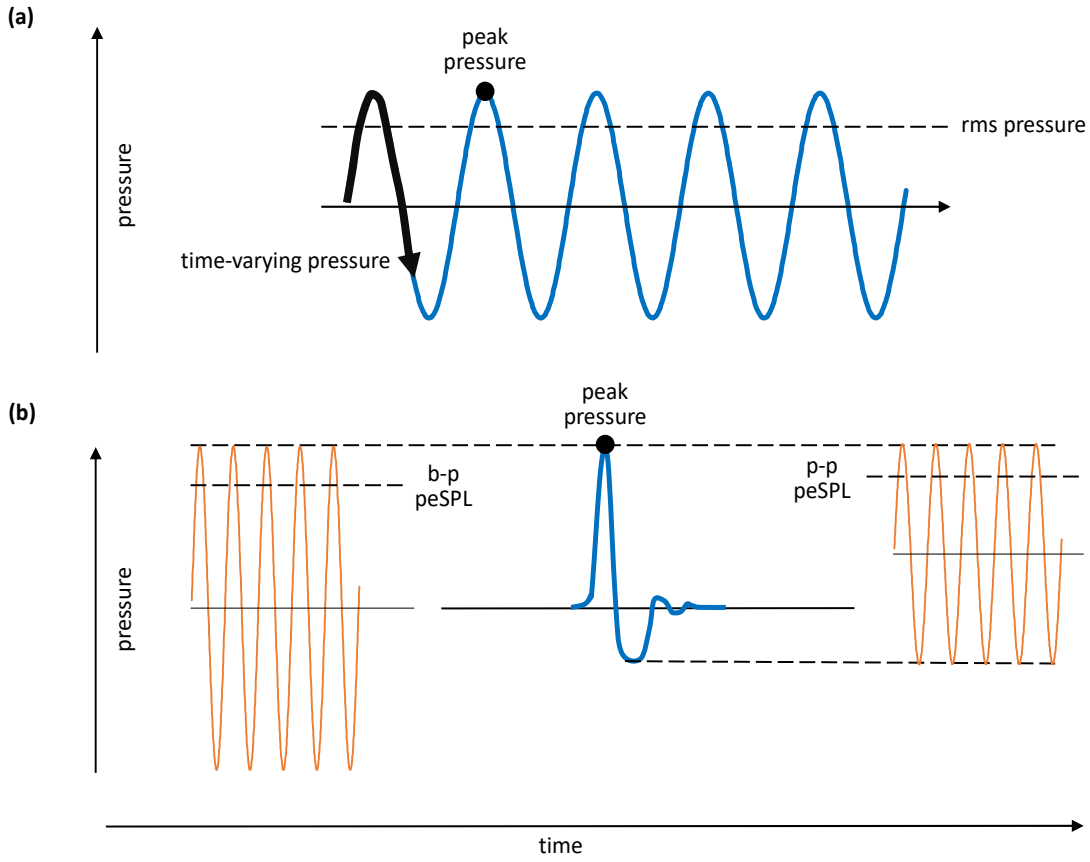

Figure 2: Definition of peak, peak equivalent (pe) and time-averaged rms pressure used to calculate SPL, peak SPL, baseline-to-peak (b-p) and peak-to-peak (p-p) peak equivalent SPL for (a) a sine wave and (b) a transient signal.

Reference Equivalent Threshold SPL, RETSPL

- The RETSPL is used when calibrating audiometric equipment and is defined in parts 1 to 9 of BS EN ISO 389 (1997-2019) for different transducers, sound fields and stimuli.
- The RETSPL is defined as the SPL measured in an appropriate ear simulator coupled to an earphone at the same level as the median average hearing threshold measured in a group of 18-25 year old otologically-normal persons.
- In other words, RETSPL is the SPL measured in an ear simulator at 0 dB HL, and this is different for each combination of stimulus, ear simulator and earphone.
- It is important that the correct RETSPL values are used for transducer calibration. Calibration providers should clearly document the source of RETSPL values so that audiology departments can ensure that

the appropriate figures are being used. If reference figures for the transducer are not available in the BS EN ISO 389 series (as might be the case with a newer transducer) then the RETSPL provided by the manufacturer should be utilised for calibration. It is important to be aware that manufacturer-provided reference figures will not have been subject to the full standardization process so some degree of caution may be required when comparing test results obtained with newer transducers to test results obtained from well-established transducers.

- For bone conduction calibration RETVFL are applicable, where VFL denotes the vibratory force level.
- The same principle applies to measurements taken in a sound field at 0 dB HL, but this is denoted by the Reference Threshold SPL (RTSPL).
- BS EN ISO 389-7:2019 provides RTSPL values for pure tones in a free-field environment and for third-octave noise bands in a diffuse-field environment (binaural listening in both cases). Although there is not a clinically significant difference between the two sets of figures (see Appendix for further details) it is recommended that the diffuse-field figures are used for calibration of both warble tone and NB noise stimuli. This is because the frequency spectrum of a warble tone signal is more comparable to a NB noise signal than a pure tone, as the warble tone contains a range of frequencies similar to the NB signal.
- The RTSPLs from BS EN ISO 389-7:2019 were derived from binaural listening conditions with the sound source in the frontal position (0° azimuth). Both these factors introduce a potential source of measurement error. Thresholds for binaural listening are about 2-3 dB more sensitive than for the monaural condition (Stream & Dirks, 1974) which means that if the two ears have asymmetrical hearing thresholds, then the result (presumed as applicable to the better ear) will appear worse by this amount. Correction factors are available from BS EN ISO 8253-2:2009 for loudspeakers placed at 45° or 90° azimuth (two loudspeakers at 90° is the recommended layout), but for normal clinical purposes use of these is not recommended. When sound is incident from a lateral direction there is a frequency-dependant increase in sound pressure at the ear closest to the loudspeaker. However, for assessment of minimal response levels where the better ear is not known then these should not be applied as there is an equal chance that the better ear is away from the loudspeaker and therefore subject to reduced sound level relative to the calibration. In principle, under certain specific conditions the correction factors can be taken into account (i.e. when the better ear is known, or when testing aided thresholds for a cochlear implant). However, it would be necessary to subtract about 3 dB as it would be a unilateral situation, making the overall correction very small at most frequencies.

Hearing Level (dB HL)

- dB HL is a relative scale that refers to an individual's hearing ability compared to normally-hearing young adults. If the hearing threshold is measured as 50 dB HL at any one frequency, then the individual cannot perceive the presence of a sound at any level lower than 50 dB HL.
- 0 dB HL represents the average hearing threshold in a population of otologically-normal 18-25 year-olds at each frequency.
- Accordingly, the conversion between dB HL and dB SPL will vary between transducers and is given by the RETSPL.

dB normal Hearing Level (dB nHL)

- Sometimes referred to as “dB above normal Hearing Level”.
- dB nHL is a similar concept to dB HL, in that it is referenced to the average threshold of normally-hearing young adults. However, dB nHL is usually employed for short duration stimuli such as clicks and tone bursts.
- As the duration of a sound becomes short (<< 500 ms), the stimulus requires equal energy within its duration to be similarly perceived. Accordingly, as the duration falls below 500 ms, the stimulus requires greater intensity to achieve the same total energy over the period. Threshold intensity becomes independent of the stimulus duration above 500 ms (Moore, 2012; Yost, 2007) which is why tone duration should always be more than 0.5 seconds during pure-tone audiometry.
- dB nHL is referenced to the threshold for each type of stimulus determined for a group of normally hearing subjects. Conversion factors should be quoted by manufacturers for each stimulus (this is discussed further in Section 6.3).

Adding and averaging sound levels

- It should be re-stated that dB is not a real physical property. The physical units usually used to describe sound are intensity and pressure. If we are averaging sound over time, or calculating the effect of multiple sound sources and adding them together, it must be the physical unit of intensity (proportional to the pressure squared) that is averaged or added. **Never directly add or average dB values** when averaging over time or calculating the combined level of multiple sound sources.
- An example is given in Box 2. It should be clear that higher sound levels will have a dominating effect on any average because, once converted back to intensity, lower sound levels are relatively inconsequential.

Box 2: example calculation of average SPL.

Example:
An individual spends 8 hours at work. During the day they sit at their desk for 6.5 hours in a steady background noise of 50 dB(A). They walk into a workshop several times, without hearing protection, and spend a total of 1.5 hours in a level of 90 dB(A).

Calculation:

Convert each SPL back to a "fractional exposure" e (proportional to intensity) using the equation:
$e = 10^{(SPL/10)}$

Average the exposure over time:
$\text{Average } E = (e_1 \times 6.5 \text{ hours} + e_2 \times 1.5 \text{ hours}) / 8 \text{ hours}$

Put the average exposure back into dB SPL using the equation:
$\text{dB SPL} = 10 \log_{10} (E)$

The average SPL over 8 hours, L_{Aeq} = 83 dB(A).

If the background noise for 6.5 hours fell to 30 dB(A), the noise dose would still be 83 dB(A). If the background noise for 6.5 hours increased to 70 dB(A), the noise dose would still be 83 dB(A).

Accordingly, the noise dose is largely determined by the higher sound level experienced over 1.5 hours. The average noise level, L_{Aeq}, of 83 dB(A) for the 8-hour shift in this case would result in a daily personal noise exposure level, $L_{EP,d}$, of 83 dB(A) for the complete 8-hour shift which is above the lower exposure action value of 80 dB(A) and therefore enough to mandate availability of hearing protection, but is below the upper exposure action value, and therefore hearing protection is not compulsory (see Section 6.2).

4. Ambient noise

4.1 General considerations and types of masking

It is important to consider all of the possible environmental effects that may alter test results within a clinical facility, not only the ambient noise. Acoustic considerations are only part of the design of a clinical facility, and test locations shall consider patient safety, comfort and privacy.

We normally define masking as "the process by which the threshold of hearing for one sound is raised by the presence of another" (ANSI, 2013). As part of audiological tests, clinicians normally consider the effect

of “energetic masking”, which occurs when the signal and noise occur at the same time and at similar frequencies (within an auditory critical band). Hearing a signal within an energetic masking noise is a matter of the relative strength of the signal and noise. It is important to understand that energetic masking is not the only form of masking, nor the only effect that impairs a patient’s responsiveness. For example, “informational masking” also occurs, where the target and masker are both audible, but the listener struggles to separate them, e.g. speech in a background of multi-talker babble. Within the category of informational masking, there are different effects that can draw the listener’s attention more than other masking signals, and increasing the complexity of the masking noise tends to further undermine audibility of the signal (Brungart et al, 2001; Culling & Stone, 2017). Temporal variation of a noise will also be more distracting (Culling & Stone, 2017) so, for example, a clicking sound at a lower level may have more impact than a constant sound. The ability to cope with masking becomes even more difficult as central auditory processing declines, especially as part of the ageing process (Kidd & Colburn, 2017) or may be more difficult for young children to cope with (Werner & Bargones, 1991). Any distractions within the environment may adversely affect the patient’s attention. In addition to environmental sounds, it should be remembered that individuals will also generate their own body noise during testing, especially apparent when they are wearing earphones, and that there will be a mixture of objective sound (e.g. blood flow, laboured breathing) and subjective sounds (e.g. tinnitus) that will also have a non-measurable masking effect. It is for this reason that **the ultimate arbiter of test reliability must be the professional conducting the audiological procedure**, ensuring that appropriate stimuli are employed within an appropriate environment.

Whilst this is a complex subject, it is possible to determine some clear guidance:

1. Sound level meters (SLM) can only measure ambient sound levels that contribute to energetic masking, so **it should be the professional responsibility of the clinician to determine if the test environment is sufficiently free of ambient noise or other distractions to allow testing to proceed**.
2. Tests shall be conducted in a separate, private, quiet room away from any sources of ambient noise and other distractions. **Audiological testing shall not, under any circumstances, be conducted in an open area where a patient has visibility of, or is visible to, individuals other than the tester** or the accompanying persons.

4.2 Ambient noise for sound field audiometry

BS EN ISO 8253-2:2009 provides maximum acceptable ambient noise levels for sound field audiometry where thresholds are measured down to 0 dB HL. Two sets of values are provided for a lowest test tone frequency of either 125 Hz or 250 Hz and Table 2, below, reproduces the figures for a testing range of 125 Hz to 8 kHz. A maximum amount of allowable threshold shift (uncertainty) of + 2 dB is quoted for the measurement of a minimum hearing level of 0 dB HL. The fact that the uncertainty is a positive value, and that there is no statement of an associated confidence level, leads to the conclusion that the figure should simply be subtracted from the measured threshold to determine the uncertainty range. So, for example, if the measured threshold is 5 dB then the true threshold could lie between 3 and 5 dB (because the effect

of the noise will always be to raise the threshold). It should also be noted that the uncertainty figure is a *maximum* at 0 dB HL. At levels above 0 dB HL the uncertainty figure will rapidly approach zero as the level increases. It is also stated that the allowable ambient noise levels can be increased by 8 dB if a maximum uncertainty of + 5 dB is permitted. **As most threshold test results will be obtained at levels above the minimum hearing level it is recommended that the + 8 dB is applied to the maximum ambient noise levels.** The associated increase in uncertainty from + 2 dB to + 5 dB will have a negligible effect on the overall uncertainty budget for sound field testing. It should also be noted that the values in Table 2 are for tonal stimuli, including pure tones and warble tones, and that narrow-band noise (NBN) stimuli require lower ambient noise values. However BS EN ISO 8253-2:2009 has no details of what the lower noise levels should be. When using narrow-band noise stimuli we therefore suggest, as a conservative measure, subtracting 8 dB from the maximum acceptable ambient noise level for tonal stimuli.

In most clinical practice, a minimum measurement level of 20 dB HL is sufficient for sound field assessment. Accordingly, in Table 2, we have provided recommended maximum acceptable ambient noise levels for tonal-stimuli measurements down to this and other thresholds. **Where NBN stimuli are used the maximum ambient noise level should be decreased by 8 dB** (see above).

Given the need for audiology services to operate in community locations, a degree of pragmatism must be applied to ambient noise limits. However, in all locations, it is important to understand the potential effects of noise and, as discussed in Section 4.5, this should be managed within the discretion of the clinician on a routine basis, prior to every test, and this should be part of standard operating practice and clinical training. It should be noted that the ability to hear a 20 dB HL tone in the presence of some ambient noise will indicate the ability to hear the same tone in quiet, so can be taken as an indication of normal hearing at that frequency. Conversely, patients with greater degrees of hearing loss will be unable to hear quiet sounds, so the clinician may judge that the hearing threshold is accurate in the presence of relatively quiet ambient noise. However, for patients with milder degrees of hearing loss, an apparently raised threshold may be an effect of ambient noise or a genuine hearing loss. In these cases, it is important that, if the test is undertaken with some noise present, this should be noted and the patient referred for a diagnostic hearing assessment in an appropriate environment.

Third-octave band centre frequency (Hz)	BS EN ISO 8253-2:2009 ($L_{S,max}$ dB)		Recommended maximum acceptable ambient noise ($L_{S,max}$ dB)		
	Measurement uncertainty due to ambient noise		Minimum measurement level		
	+ 2 dB	+ 5 dB	0 dB HL	10 dB HL	20 dB HL
31.5	52	60	60	70	80
40	44	52	52	62	72
50	38	46	46	56	66
63	32	40	40	50	60
80	27	35	35	45	55
100	22	30	30	40	50
125	17	25	25	35	45
160	14	22	22	32	42
200	12	20	20	30	40
250	10	18	18	28	38
315	8	16	16	26	36
400	6	14	14	24	34
500	5	13	13	23	33
630	5	13	13	23	33
800	4	12	12	22	32
1 000	4	12	12	22	32
1 250	4	12	12	22	32
1 600	5	13	13	23	33
2 000	5	13	13	23	33
2 500	3	11	11	21	31
3 150	1	9	9	19	29
4 000	-1	7	7	17	27
5 000	1	9	9	19	29
6 300	6	14	14	24	34
8 000	12	20	20	30	40

Table 2: Maximum permissible ambient noise levels ($L_{S,max}$) for sound field audiometry using tonal stimuli for a test tone frequency range of 125 Hz to 8 kHz.

4.3 Ambient noise for AC and BC audiometry

Placement of any earphone on an individual will offer some attenuation of ambient noise. Ambient noise at BS EN ISO 8253-2:2009 levels (Table 2) has no significant effect on the measured hearing threshold at 0dB HL, and higher levels of ambient noise can be allowed when earphones occlude the ears (BS EN ISO 8253-1:2010). In principle, the stated attenuation level for any earphone can be added to the allowable levels for sound field audiometry (Table 2) to give an equivalent allowable noise level when using earphones. BS EN ISO 8253-1:2010 provides a table for maximum acceptable ambient noise with typical earphones, but it is better to employ a specific limit based on the manufacturer's stated attenuation for any specific earphone. Inserts provide much more attenuation than supra-aural or circumaural headphones (Arlinger, 1986; Frank & Wright, 1990; Swanepoel de, et al, 2015). Bone conduction measurements offer no attenuation of noise for the patient so the maximum ambient levels should be taken as the same as the sound field values in Table 2. The maximum permissible ambient noise levels for bone conduction testing are also specified in BS EN ISO 8253-1:2010. If audiometry is conducted in the community, it is the responsibility of the clinician to assess the local conditions prior to each test, as described above (Section 4.2) and use appropriately attenuating headphones where necessary.

4.4 Ambient noise in other situations

Ambient noise should be a consideration for all audiological procedures, e.g. speech testing, especially in the sound field, and hearing aid fitting.

For speech testing, the approach described in Section 4.3 should be used, based on the minimum speech stimulus level. However, speech signals are highly variable in sound level and the stimulus level is only representative of the average speech level, such that quieter elements of speech might be lost. It is not possible to provide evidence-based guidance for appropriate levels of ambient noise in this situation. On the basis of professional consensus, it is therefore suggested, in order to be prudent, that **10 dB shall be subtracted from the given speech stimulus level to calculate allowable ambient noise levels**. For example, to undertake a valid measurement with a speech stimulus given as 30 dB HL by calibrated audiometric equipment, the allowable ambient noise level should be the same as that for a pure-tone stimulus of 20 dB HL.

In the case of hearing aid fitting, the minimum speech stimulus used with the international speech test signal (ISTS) in real-ear measurements is usually 50 dB SPL. Given that this is denoted in dB SPL, it is suggested that the ambient noise should be at least 10 dB lower (i.e. less than 40 dB SPL) in each frequency band in the range of the test signal.

4.5 Measurement and management of ambient noise

Sound Level Meters

Sound Level Meters (SLMs) are governed by BS EN 61672:2013. This describes class 1 and class 2 SLMs where the class 2 device has a reduced specification compared to the class 1 SLM. It is important to note that BS EN ISO 8253, Parts 1 and 2, specify that a class 1 SLM must be used for measurement of ambient noise levels. A significant number of lower cost SLMs are available which claim to comply with BS EN 61672 and may sometimes be referred to as a "class 3" device which is not a recognised standard. However, unless there is specific evidence of type approval it should be assumed that the SLM does not comply with BS EN 61672 and will not be sufficiently accurate (Campbell & Williams, 2021). These devices can best be termed as "unclassified sound level indicators" to differentiate them from class 1 and 2 BS EN 61672 SLMs. **Audiology departments shall use their best efforts to ensure that any SLM used for the measurement of ambient noise to BS EN ISO 8253-2 is compliant with BS EN 61672 class 1.** The SLM must also have a third-octave band analysis function covering, as a minimum, the frequency range 31.5 Hz to 8 kHz. It is recommended that the calibration of a SLM be checked prior to and after use, using a sound level calibrator complying with BS EN 60942:2018. Audiology departments must also undertake periodic verification of all SLMs and sound level calibrators, including calibration by an independent provider. The period between calibrations may be defined by the manufacturer, but should be no more than two years as specified in UKAS guidance document LAB 23 (2023). Calibration shall provide some details of the low-level noise and linearity performance of the SLM.

It is important to determine the minimum sound levels measurable by a SLM. The manufacturer's specifications should be consulted, together with any low-level calibration results, to determine the minimum level that can be measured in *each* of the third-octave bands. Ideally the SLM should be able to measure 6 dB below the minimum level in each band but this figure could be reduced to 3 dB provided that the associated uncertainty of measurement is taken into account. As stated in BS EN ISO 8253-2:2009, few SLMs can measure as low as 5 dB SPL across the full frequency range, and many SLMs cannot measure accurately below 20 or 30 dB SPL. Audiology departments shall therefore consider whether valid ambient noise measurements are achievable using their own equipment, or whether an external agency should be engaged to undertake the measurements. The minimum measurable sound level should be checked as part of regular SLM calibration.

An A-weighted broadband noise measurement can be made but this cannot be used as a substitute for the third-octave noise measurements as it does not contain any frequency specific information. **Services shall not use a maximum dB(A) noise level which corresponds to the maximum third-octave levels in BS EN ISO 8253-2.** However regular A-weighted measurements can be useful in detecting a *change* in the ambient noise levels in a clinic. A class 2 BS EN 61672 SLM, or possibly an unclassified sound level indicator, should be suitable for this purpose providing that the low-level linearity allows appropriate measurements to be made and that this has been checked during calibration. For example, dB(A) measurements are

made in a clinic on a daily basis and the levels range between 30 and 40 dB(A). If a measurement of around 50 dB(A) is recorded on a number of consecutive days, this would indicate that there has been a significant increase in the ambient noise level that requires further investigation. The monitoring should be implemented in conjunction with regular full ambient noise tests to BS EN ISO 8253-2 which, as noted above, may be provided by an external agency.

Another type of device is often called a "noise dosimeter", but should be referred to as a "personal sound exposure meter" (PSEM) according to BS EN IEC 61252:1997. These devices generally provide a single broadband measure of noise dose relevant to an individual, usually over a range of no more than 80-130 dB(A). **PSEMs are not suitable for the assessment of ambient noise and shall not be used for this purpose**.

There is an increasing range of free apps on Smartphones that offer sound level monitoring. These are **not** appropriate for use as SLMs and can give a very wide range of readings when compared to a reference SLM. However, free smartphone applications used as frequency analysers might be employed to identify the frequency of any tonal noise in an environment, against which a clinician can make decisions, as long as they are not used to provide any indication of sound level.

Measurement of ambient noise

Audiology clinics exist within a range of facilities and it is recognised that there may be limitations to designing and altering infrastructure to meet the specific requirements of audiological testing. It is therefore important that **any measurement of ambient noise shall be conducted to accurately represent normal clinical conditions**. In other words, any measurement must take place on a typical clinical day, e.g. with heating, ventilation and lighting used as normal, with other clinics occurring as they normally would in adjacent rooms, with a typical number of patients in the waiting area.

The measurement of ambient noise is expressed as $L_{S,max}$. This is the maximum slow-weighted SPL detected during the measurement period. In order to make this measurement, the SLM used should have a third-octave analysis mode with a maximum hold facility. The time-weighting of the analyser must be set to "slow". The SLM microphone is positioned at the centre-point of a patient's head, without the patient present. Care should be taken with the mounting of the SLM to ensure that vibrations are not picked up by the microphone which could cause false results (a vibration isolating mount may be necessary). A measurement run is carried out over an appropriate period (BS EN ISO 8253-2 does not specify a duration for the test but this could typically be the length of time required to complete a hearing test). If any atypical transient noises occur during the test then the measurement run should be aborted and re-started. Once the run has been completed, the maximum SPLs in each third-octave band can be compared to the maximum acceptable ambient noise levels (Table 2). A measured level greater than the maximum level indicates a fail in that frequency band. Services should determine which ambient noise criteria (Table 2) apply to every room in order to determine the minimum threshold in dB HL that can be accurately measured in a room.

Management of ambient noise

Ambient noise should be considered as part of the design process for any facility and a new facility should only be approved for use after the appropriate ambient noise testing has been successfully completed. As detailed above, the appropriate ambient noise levels will depend on the testing requirements for the room, e.g. pure-tone audiometry with headphones, inserts or BC, sound field audiometry or screening. Prior to use, a baseline should be established for every clinical room. **Each test room shall clearly display details of the test types and the associated minimum hearing levels to which measurements can be accurately conducted in that room**. Once this baseline has been established, audiology departments shall monitor ambient noise on a regular basis for all test rooms. It is the responsibility of the audiology department and site management to ensure that ambient noise is assessed as part of quality management systems, and that appropriate responses are made to correct issues where they are reported by clinicians. As a minimum, assessments shall include ongoing assessment of noise interference by clinicians. A dB(A) measure to assess changes from the baseline may also be employed, as described above. If there is any indication that the ambient noise level in the room may have changed, other than for short-term temporary reasons, or if a room consistently fails to meet the required ambient noise levels, then it should be withdrawn from use. Noise-reduction measures should be considered, such as acoustic treatments or possibly changes to the building infrastructure, air conditioning or situation of noise sources. Specialist advice may be required where structural changes are indicated (e.g. from an acoustic consultant) to determine what options are available. A commonly implemented measure is to install some form of noise-reducing enclosure such as a self-contained sound booth.

It is recognised that the level of noise in any clinical environment will fluctuate and that some short-term noise (e.g. a door banging) may simply require testing to be briefly suspended. If a clinician is unable to conduct a test within a reasonable time (e.g. 20 minutes for pure tone audiometry) due to interruptions by external noise, then this shall be reported as a non-compliance. Any concerns about ambient noise that may affect a test, or significantly interrupt it, must be recorded in the clinical notes. It should also be remembered that ambient noise may be subjectively less than a patient's own "internal" noise, e.g. blood and breathing sounds or tinnitus when wearing headphones, which may indicate that a change to test procedure is more appropriate (e.g. use of warble tones) and this must also be noted. It is therefore recommended that a pragmatic approach is adopted. **Continuous subjective monitoring of ambient noise should be the responsibility of the clinician whilst preparing for and conducting tests**. It is **not** necessary to provide a SLM in every room or to continually measure ambient noise. Ambient noise shall be considered as part of the Stage A start-of-day tests carried out by the clinician for any audiology procedure (see Section 5.4). A clinician with normal hearing should be able to determine if the test stimuli are being presented at the appropriate levels in the test room. If there is a concern (or if the clinician has a hearing loss) it may be necessary to measure the ambient sound levels or to conduct a full hearing test with a known normal subject. If thresholds are obtained which are 10 dB or more above the subject's known thresholds then a full ambient noise level measurement should be carried out to determine if there has been an increase in the ambient noise level.

5. Sound field measurements and audiometry

5.1 Test conditions

This section refers to any audiometric procedure in which the stimulus is presented in a sound field, usually from a loudspeaker. This most obviously refers to sound field audiometry, as described in BS EN BS EN ISO 8253-2:2009, but also includes other procedures such as speech testing (BS EN BS EN ISO 8253-3:2022). Real-ear measurements (REMs) can also be considered a procedure conducted in the sound field. However, in most cases, REMs incorporate a reference microphone close to the ear of the listener, so sound levels are adjusted accordingly and the following considerations are largely overcome **as long as the room is not affected by standing waves**. It should be noted that equipment combining both audiometry, speech testing and REMs might use the reference mic in REM applications, but it is not employed in audiometry or speech testing modules. Some REM equipment may carry speech testing modules that can be directed via the loudspeaker, but the loudspeaker may not be designed for sound field testing. In these cases, audiology departments should assess this with their suppliers and the following considerations shall be employed when doing any sound field testing.

Ideally test rooms should meet the requirements of a quasi-free sound field. However, as detailed in Section 3.2, this is unlikely to be achievable in practice. It is therefore important that the maximum variation in sound level is assessed for each of the test procedures that the clinic carries out. Section 5.3 details how the sound field variability can be determined.

Ambient noise considerations should follow the guidance in Section 4 of this document. Room layout shall follow recommendations specific to the procedure being conducted.

Loudspeakers should be placed at the height of a patient's head, a minimum of 1 m away. **Loudspeakers should not be placed near a reflective surface (such as a table, wall, chair or sound booth) in order to avoid any unwanted acoustic effects.** Where test requirements necessitate placement of items close to the loudspeaker (e.g. a VRA cabinet) calibration should take place with these items in place and they shall not be subsequently moved without recalibration. Introduction of a reflective surface can increase the stimulus level at the patient's position, and this can be exacerbated by moving a loudspeaker towards the corner of a room. This may also introduce unpredictable resonance or cancellation effects in the test room (see Section 3.2). Loudspeakers should therefore **not** be mounted directly onto walls or ceilings.

Hand-held sound generators (or "warblers") may be used but it shall be recognised that they introduce inherently greater measurement variability due to the lack of accurate positioning with respect to the patient. Measurement variability must be managed by determining the correct distance and position to hold the warbler relative to the patient, i.e. directly facing the patient. **Variability and the associated uncertainty of measurement must also be estimated** (see Section 5.3). It is preferable that tests are undertaken with fixed loudspeakers or headphones, but it is recognised that this is not always possible

for play audiometry or distraction testing, e.g. for children with sensory issues who may not wear earphones.
Sound field audiometry is more typically used to assess binaural hearing. However, monaural hearing may be assessed by occluding the non-test ear. Additional care should be taken to use the appropriate sound source (e.g. left or right) and that it has been calibrated for monaural testing. Binaural threshold measures generally provide information on the sensitivity of the better hearing ear only. Furthermore, if both ears are not the same then the results will indicate a 1- 3 dB exaggerated hearing loss, depending on frequency, because the reference thresholds available are for binaural listening (Stream & Dirks, 1974).

5.2 Test stimuli

BS EN ISO 8253-2:2009 defines three types of test signals that may be used in sound field audiometry: pure tones, frequency-modulated tones (often referred to as warble tones) and narrow-band noise. **In a quasi-free sound field, or similar conditions likely to be found in clinical rooms, only frequency-modulated tones and narrow-band noise (NBN) should be used**. Pure tones may cause marked standing waves resulting in unpredictable and unacceptable fluctuations in sound pressure level within the room. However, it should be noted that warble tones and narrow-band noise do not preclude the existence of standing waves, and this should be assessed.

FRESH™ noise is a narrow-band stimulus suitable for threshold estimation in sound field audiometry. It is calibrated in dB HL in a similar manner to the NBN stimulus defined in BS EN ISO 8253-2:2009. Although it is not currently covered by this standard, it is considered to be a valid test stimulus and it may provide better frequency specificity than NBN.

Likewise, ISO standards do not consider other test stimuli such as Ling sounds, speech or music. In these cases it is crucial to remember that a single dB SPL measure of the stimulus cannot fully represent the dynamic range of the stimulus and is only an average. It is possible that speech recognition or a response to music occurs closer to the peak SPL of a stimulus, which is louder than the average SPL. Similarly, speech intelligibility might be impaired by not hearing quiet elements of speech, below the average SPL for that stimulus. Caution shall be employed in reporting the level of responses to non-tonal or non-standard stimuli due to the difficulties in calibration and in knowing which element of the stimulus generated a response. **Responses to non-standard measures can only be, at best, indicative and it is recommended that descriptive word-based reporting is employed, rather than numerical values,** in order to avoid giving a false impression of test accuracy that might incorrectly indicate an equivalence to more reliable responses to calibrated tonal stimuli. Use of non-standard stimuli, such as music, should also consider the dynamic range compression of the stimulus as well as any frequency range limitation.

Where loud stimuli are employed for patients with severe or profound hearing loss, there shall be a consideration of other people in the room, including family members and clinicians. Stimulus levels and average exposure shall not exceed those described in Section 6.

5.3 Measurement variability and uncertainty

BS EN ISO 8253-2:2009 defines the criteria for a quasi-free sound field, measuring SPL at positions around the reference point. A similar approach can be used to establish the measurement variability in a test room's sound field and the associated contribution to the overall uncertainty of measurement for the hearing level determination. (N.B. ISO 8253-1:2010 Appendix A provides an analysis and sample uncertainty budget calculation for air conduction testing.) Together with the measurement at the reference point, six additional measurements are required at positions left, right, front, back, below and above the reference point. These measurements need to be repeated for each sound source (i.e. right and left loudspeaker), each test signal and each test frequency. A class 1 sound level meter (see Section 4.5), with third-octave analysis is required to make the measurements. It may not be practical for a department to undertake the required sound field measurements in-house, as it requires multiple measurements and the correct equipment, so an external agency may be employed to carry out the measurements. Where this has not occurred, an estimate can be used as an interim measure (see details below), but this should not preclude measurement of the sound field variability on at least one occasion.

The reference point measurement shall be undertaken with the microphone placed at the midpoint of the patient's head position with any patient chair removed, but with all other objects in the normal position. For example, in VRA, the play table should be left in its usual position between the patient and the clinician distracting because it represents a significant reflective surface. It is preferable to use a soft covering, such as a foam play mat, on a play table to minimise any acoustic effects, but this must be the same in normal clinics and during the measurement. Note that removal of a chair for measurements assumes that it would normally be covered by the patient. If a high chair is used with surfaces exposed beyond the patient's head, then this will have an effect on sound levels and is best avoided.

For adults, the sound level shall be measured at 20 cm to the left, right, front, back, above and below the reference point. Given that an adult may move their head within this range during testing, a contribution to the overall uncertainty budget for the hearing test should be calculated from the maximum variation in SPL using a rectangular weighting (i.e. dividing the maximum variation by √3). For example, if the maximum and minimum differences from the reference point SPL are found to be +3.8 dB and –3.2 dB, then the uncertainty contribution would be (3.8) / √3 = ± 2.7 dB. This process should be repeated for each test frequency.

For children, given the greater likelihood of movement, the sound level shall be measured at 30 cm to the left, right, front, back, above and below the reference point. Given that a child may move their head within this range during testing, a contribution to the overall uncertainty budget for the hearing test should be calculated from the maximum variation in SPL using a rectangular weighting (see example above). In some procedures, such as VRA, it is possible that children may move about more than this. To reflect this, the measurements can be repeated at points 50cm away from the midpoint to establish the potential measurement variability and the associated contribution to the uncertainty budget.

It has been shown that for a distance of 30 cm in all axes the maximum variation in SPL can typically be estimated as ±5 dB (Beynon & Munro, 1995; Shaw & Greenwood, 2012). In the absence of actual measurement data this figure can be utilised as an interim measure, provided that the test room meets the requirements set out in this document.

Where hand-held sound generators ("warblers") are used, a similar approach shall be employed. Sound levels 30 cm closer to and further away from the warbler shall be determined, and 30 cm either side of the midpoint. It is also recommended that measurements be undertaken in warbler positions that may occur in clinic, e.g. with the generator ± 30° from the centre-line to the midpoint. Variability in position should be minimised by clinical protocols and the contribution to the uncertainty budget should be calculated in a similar manner to that detailed above.

It should be remembered that introducing any object into the test room will have an unpredictable effect on the acoustics in the room. Clinicians shall be aware of this and shall ensure there are no obvious causes of significant alterations to the stimulus level at the measurement point, e.g. introduction of a reflective surface close to that point.

5.4 Calibration

BS EN ISO 8253-2:2009 section 12 contains details of the Stage A/B/C test system recommended to maintain the calibration and serviceability of the audiometric equipment.

Stage A checks should be conducted at the start of the day, before any patient testing is carried out, and should include:

- Examination of equipment, connections and cables;
- Output at low levels, e.g. audible at 20 dB HL for all sound sources used for testing;
- Listening at higher output levels (60-70 dB HL) to check for distortion and noise-free signal presentation;
- Listening for any unwanted sound in the test environment (e.g. from the audiometer, from tone interrupter operation, or from other equipment in the room);
- Correct operation of subject response and monitor circuits;
- A check that the patient reference position is correct and clearly identified;
- A check of ambient noise (as mandated above).

Wherever non-standard stimuli (e.g. Ling sounds, speech or music) are used and the calibration level can be easily altered, there is a risk that the set level may have changed. In addition to a listening test, it may be necessary to also monitor the output level using an appropriate SLM to ensure that the level set initially is being maintained.

Stage B calibration should be conducted annually. Calibration should be carried out to the appropriate parts of BS EN ISO 389 (reference levels) and BS EN 60645 (equipment specifications) together with any applicable recommendations (e.g. this and other BSA documents):

- Frequencies and characteristics of frequency-modulated tones;
- Sound pressure levels at the reference point;
- Attenuator steps (linearity);
- Harmonic distortion.

The design and commissioning of a test facility may be considered as Stage C calibration. This should include an assessment of the sound field variability and the measurement of the ambient noise in the test room. It might be considered desirable to undertake additional room measurements (such as reverberation time, speech intelligibility, standing wave/room mode analysis) but this would require specialist knowledge and equipment, so an external agency would most likely have to be employed for tests of this type. In line with the rest of this document, it is considered good practice to monitor the sound field variability (section 5.3) and ambient noise (section 4) and this is particularly important for sound-field measurements.

5.5 Presentation of Results

As discussed above, prior to any presentation of results, departments must consider measurement variability and the associated uncertainties, and **report in a manner which reflects the true accuracy and reliability of results**. For example, sound field audiometry using VRA or speech testing with calibrated equipment will have less uncertainty than other procedures which utilise live voice or non-standard stimuli, e.g. Ling sounds or music. In these cases, they should be described as indicative measures and reports should not suggest an absolute value without proper caution provided. Where accurate measures can be determined, Table 3 provides the symbols recommended for unaided and aided sound field audiometry. It should always be noted on the audiogram form whether symbols represent thresholds or minimum response levels. An example of results plotted on an audiogram form is given in Fig.3.

Situation	Symbol
Unaided sound field, binaural	S
Aided sound field, binaural	A

Table 3: recommended symbol for sound field audiometry.

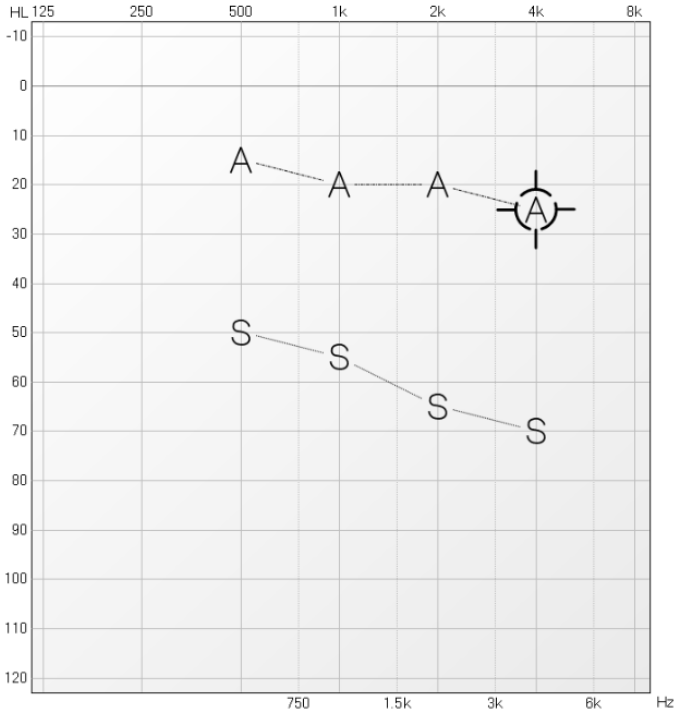

Figure 3: example audiogram for sound field audiometry.

6. Noise exposure and maximum stimulus levels

6.1 General approach

This section considers safe noise exposure levels for patients and clinicians. It specifically addresses the maximum stimulus levels that should be employed in audiometric tests, including audiometry, acoustic reflex testing, OAEs and electrophysiology, employed within paediatric or adult audiology and vestibular assessment. **Please be aware that the maximum exposure levels for children are lower than those for adults** (see Section 6.6).

It is the responsibility of all clinicians to be aware of safety and comfort for both the patient and for themselves. This section provides recommended maximum sound levels for various type of stimuli. It should be made clear that provision of a maximum level does not, in itself, justify using these levels. **Stimulus amplitudes shall not be increased beyond any level that causes discomfort for a patient. Clinicians should also avoid increasing stimulus levels beyond those that are likely to affect a patient's**

management plan and seek to minimise noise exposure at all times. For example, establishing a hearing loss beyond 100 dB HL may have no clinical significance if it does not determine an exact level of achievable hearing aid gain or further indicate a dead region. In patients with a severe-to-profound hearing loss there may be a narrow range between audibility and the discomfort or pain threshold, so caution should be exercised in increasing higher sound levels and a smaller step-size used, i.e. 5 dB. Louder levels also have the potential to exacerbate tinnitus, so additional care should be taken where this is reported by a patient.

6.2 Noise exposure legislation

Legislation for noise exposure at work (HSE, 2005) applies to all forms of occupational noise. The legislation sets two action levels:

- The lower exposure action values are a daily (or weekly) average noise exposure of $L_{EP,d}$ = 80 dB(A) or a peak sound pressure level of 135 dB(C). At this level, employers assess risk and provide access to hearing protection.
- The upper exposure action values are a daily (or weekly) average noise exposure of $L_{EP,d}$ = 85 dB(A) or a peak sound pressure level of 137 dB(C). At this level, hearing protection is mandatory.

In addition to the two action values, the legislation also sets values for an absolute limit: 87 dB(A) $L_{EP,d,}$ and peak sound pressure level of 140 dB(C).

The "daily personal noise exposure level" ($L_{EP,d}$) is based on the equivalent continuous A-weighted sound pressure level (L_{Aeq}) modified by the length of the working day:

$$L_{EP,d} = L_{Aeq} + 10 \log_{10} (T_e / T_0)$$

where T_e is the duration of an individual's working day and T_0 is 8 hours. $L_{EP,d}$ can therefore be interpreted as the average noise level over an eight-hour period. For the purposes of this document, **$L_{EP,d}$ shall not exceed the lower exposure level of 80 dB(A) for any patient, clinician or accompanying person**. It should be remembered that dB values cannot be averaged directly, but must be converted to the appropriate physical unit (intensity) before averaging and being converted back to a dB value (as described in Box 2). Details of dB HL, A and C weightings in relation to dB SPL are given in the Appendix. It should be noted that the legislated values are based on sound level measurements in a sound field at the position of an individual's head, in their absence. Sound levels measured in an ear canal or coupler will likely be greater, so the limits to stimulus levels suggested here can be taken to be conservative.

The noise exposure at work legislation is sometimes misinterpreted. The action levels for $L_{EP,d}$ represent an *average* exposure over a day and do *not* suggest that instantaneous SPLs of 80 or 85 dB(A) are unsafe. The relevant legislative figure relating to the peak, instantaneous level of sound for which action must be taken is 137 dB(C), although the lower action level of 135 dB(C) shall be used to remain cautious. It should be noted that the C-weighting is quite close to the unweighted dB SPL at most frequencies (see the

Appendix) so, for practical purposes, one can consider the maximum peak level of any sound used in audiometric testing to be 135 dB pSPL at most frequencies. It is noted that, internationally, a higher level of 140 dB pSPL is taken as the peak limit (Portnuff et al, 2017).

Accordingly the daily personal noise exposure level $L_{EP,d}$ does not always provide guidance regarding safe stimulus levels in audiometric tests because stimuli are typically of short duration. Consider an example, using the calculations described in Box 2. Most of us would think it unreasonable to expose an individual to 100 dB SPL continuously for 5 minutes at most frequencies. If we did so, their daily (8 hour) noise exposure would be L_{Aeq} = 80 dB SPL. Whilst an extended exposure to 100 dB SPL appears excessive, it does not contravene the noise exposure at work regulations and demonstrates its lack of applicability to safe stimulus levels. **The daily personal noise exposure level shall not be used to indicate the safety or risks associated with individual short-duration stimuli used in audiometric tests.**

6.3 Transient stimuli

The 135 dB pSPL limit should be used to determine the maximum peak level of any transient stimuli such as tone bursts, chirps and clicks, typically used in electrophysiology.

We define a transient stimulus as one in which each element of the stimulus lasts less than 0.5 seconds. **Clinicians shall take care to convert the 135 dB pSPL limit to the units used on measurement equipment**, paying particular attention to the different measures of SPL described in Section 3.3. It should be noted that this will be dependent on the manufacturer's conversion factors where normal levels (dB nHL) are used and that the manufacturers may quote the conversion factors to dB nHL against different measures (e.g. pSPL, b-p peSPL or p-p peSPL). In some cases the BSA provides specific, separate guidance for procedures that recommend standardised reference levels, e.g. in "Recommended stimulus reference levels for ABR systems", and those recommendations should be followed for those specific procedures. Please also note that masking noise used within electrophysiology should also be subject to the limits described in Section 6.4.

Box 3 provides an example of a maximum stimulus level calculation. A similar calculation shall be conducted for each piece of equipment delivering transient stimuli, for each type of stimulus, each transducer employed and each frequency. This shall incorporate all equipment conducting ABR, CAEP, ASSR, OAE, VEMP or any other transient electrophysiological stimulus.

This guidance does not utilise a time-averaged dose that limits individual exposure over multiple stimuli. Whilst this may be desirable, there is no strong evidence to support a specific safe level over a specific time. It also difficult to be sure what level of stimulus is applied to sections of the basilar membrane at any point in time because the phasing of different stimuli may spread or concentrate the stimulus across it; e.g. chirps attempt to counter the delay across frequency-specific elements of the basilar membrane and apply the same level to each element at the same time. Robler et al (2018) suggested that the average

level over one second should not exceed 132 dB SPL, but this can lead to higher exposures than suggested by the 135 dB peak SPL limit. Determining the energy delivered by a transient signal is a complex calculation for an audiology clinic, requiring the signal to be well-defined mathematically before calculating the rms pressure and converting back to SPL. Furthermore, we do not know what physiological mechanisms are causing damage to the ear and whether these are instant (traumatic) or cumulative. High peak amplitudes may cause traumatic damage, whereas degradation of the ear due to long-term exposure is concerned with multiple mechanisms, e.g. cumulative ionic overload in the transduction and synapse mechanisms of the inner ear. Given the paucity of evidence for instantaneous measurement levels that may cause either process, we consider it prudent to use peak amplitude as the limitation for test signals, in combination with a clinician's care to use no stimulus amplitude greater than can be justified by clinical need. The 135 dB pSPL limit for transient stimuli is preferred because it is simpler to employ in a clinical setting and sufficiently conservative, as long as it is understood that this is a maximum and clinicians should take every action to avoid unnecessary exposure to this level or levels close to it.

Box 3: example calculation of maximum stimulus levels.

Example:
A manufacturer provides VEMP equipment. Stimulus levels are given in units of dB nHL. The manufacturer provides a conversion table between dB nHL and dB peSPL. Conversions vary for different earpieces and frequencies. The conversion factor for a 2:1:2 cycle, at 500Hz using insert earphones is 23.5 dB.

Calculation:
Maximum peak SPL = 135 dB pSPL.
Convert to peSPL: 135 – 3 dB = 132 dB peSPL.
Convert to dB nHL: 132 – 23.5 dB = 108.5 dB nHL.
Round down to the nearest 5dB.
The maximum safe stimulus level for this situation is 105 dB nHL.

The same manufacturer provides a conversion factor of 25.5 dB for the headphones provided with the system, and 69.5 dB for the BC. Accordingly, for the same 500Hz stimulus and parameters, the maximum safe stimulus is 105 dB nHL with headphones and 60 dB nHL with BC.

Whilst 135 dB pSPL is given as a maximum, **clinicians shall not use higher amplitudes than necessary** and shall seek to minimise the noise exposure to a patient by limiting the number of sweeps, cycles and stimulus duration, whilst only attempting to gain valid results that will have some impact on the individual management plan (Robler, 2018). **Patient feedback shall always be sought and exposure ceased**

whenever a patient reports discomfort. High stimulus levels (e.g. those close to the suggested maximum, or above it) should not be used without a clearly defined and documented clinical need.

6.4 Intermediate duration stimuli

Whilst the 135 dB pSPL limit may be appropriate for transient stimuli with a very short duration, it is not appropriate for other tonal stimuli, such as pure tones used within audiometry, because they concentrate a greater amount of energy in a single area of the basilar membrane over longer periods. The limit of 135 dB pSPL is equivalent to a continuous sine wave of 132 dB SPL, so it should be clear that this does not provide sufficient safety guidance. Unfortunately, the $L_{EP,d}$ also remains an inappropriate measure. To illustrate this, consider the example of a constant exposure to 110 dB SPL; this would not exceed the 85 dB $L_{EP,d}$ mandatory limit until after 90 seconds of constant exposure, roughly equating to 45 or more pure tone stimuli at 110 dB SPL of any frequency during a hearing test. Tonal stimuli of intermediate length therefore need a different safety limit, but this area is less well evidenced. It is difficult to establish what level of sound starts to cause actual physical harm to an individual without subjecting experimental participants to unethical risk, so a cautious approach must be employed.
Some guidance can be derived from research into the safe limits for short duration pure tone stimuli used in acoustic reflex threshold (ART) tests, although the suggested limits vary (Schairer et al, 2007). Hunter et al (1999) concluded that there was a risk to residual hearing at levels of 120 dB SPL or more. Although the evidence base was not strong and does not address frequency-specific levels, a maximum presentation level of 115 dB SPL was recommended. Arriaga and Luxford (1993) suggested a limit for ART of 105 dB HL based on a small number of case reports. Given that the acoustic reflex is stimulated at approximately 20dB less for broadband stimuli, it was also suggested that broadband stimuli are limited to a maximum of 95 dB HL.

Given the lack of evidence for specific safe limits in other tests, such as pure tone audiometry, a conservative recommendation is made based on the available evidence which should apply in most clinical situations: **115 dB SPL should be the maximum limit for any intermediate duration, tonal stimulus employed within audiometric testing**. **This shall be converted to the units of any audiometric equipment (e.g. dB HL) for the transducers used and a further limit of 105 dB HL shall be imposed at any frequency.** It is recognised that there may be specific situations and tests that require higher levels than the suggested limit; e.g. profound hearing losses, particularly at low frequencies, and testing related to cochlear implantation that may need to establish whether an ear has minimal outstanding function or is effectively “dead”. In these cases, a clinical need must be clearly established and documented where the suggested maximum limit is to be exceeded.

Frequency (Hz)	dB HL maximum limit
125	70
250	80
500	100
750	105
1 000	105
1 500	105
2 000	105
3 000	105
4 000	105
6 000	100
8 000	100
Broadband	95

Table 4: maximum stimulus levels for intermediate-duration tonal and broadband stimuli presented via supra-aural headphones (see Table A2), based on a maximum exposure level of 115 dB SPL at each frequency. Rounded to the nearest 5 dB and capped at 105 dB HL.

We define "intermediate duration" as a stimulus lasting 0.5 to 5 seconds. As audiometric equipment generally displays units of dB HL, maximum stimulus levels by frequency are given for headphones in Table 4. Where equipment is calibrated to neither dB SPL or dB HL, an audiology department shall ensure that steps are taken to ensure these exposure levels are not exceeded and that clear guidance is provided with the equipment before it is used. Table 4 provides data for tonal and broadband stimuli. "Broadband" refers to a signal with energy across the audible frequency spectrum, such a white noise or pink noise which may be sometimes employed as a stimulus or as a masking sound. Where other stimuli are used, including narrow-band noise, narrow-band masking and warble tones, the same dB HL limit shall be employed as that for pure tones. **Clinicians shall also restrict the patient's exposure as far as possible, using only stimuli that do not cause a patient discomfort and that will lead to useful diagnostic information that is likely to have an impact on the patient's individual management plan.**

6.5 Considerations for conductive hearing loss

Noise-induced hearing loss caused by exposure to excessively loud sounds is often thought of purely in terms of inner ear damage, i.e. a sensorineural hearing loss. It might be considered possible to increase stimuli to higher levels in cases of a conductive hearing loss because of the attenuation caused by the conductive element, which protects the inner ear. However, it is worth remembering that acoustic trauma can also cause conductive losses via tympanic membrane perforation and disruption of the ossicular chain

(Esquivel et al, 2018). Whilst this is largely studied in terms of blast noise, we do not have evidence for what level of stimulus might cause similar injury. This might be considered unlikely, but it is recommended that clinicians should not exceed the recommended maximum stimulus levels given above in any case, unless a specific clinical need can be identified. There remains a risk that the boundary between a hearing threshold and the threshold of pain might be crossed, so increasing levels above the stated maxima is not recommended.

6.6 Considerations for children

Children have smaller ear canals than adults, resulting in the same stimulus creating higher sound levels. Rodriguez et al (2018) reviewed previous findings that showed that children aged 1 month to 5 years averaged 4 dB greater SPL in their ear canals compared to adults, and that this value decreased with age. Clinicians should take extra care when testing children and there should be few clinical situations that require results using such high intensity stimuli, except in some forms of electrophysiology. Nevertheless, for all pre-teen children, **maximum stimulus levels should be restricted to 5 dB lower than those proposed for adults for all types of stimulus**. Where other correction factors are employed within electrophysiological guidelines, separate safe limits should be specified.

7. Further Research

The recommendations in this document are largely based on well-established physical principles from the science of acoustics. However, some elements of practical audiology are not well supported by evidence:

- Non-standard stimuli (e.g. speech, music, Ling sounds) used in the sound field could be standardised to enable more accurate reporting. This would require a specified dynamic and frequency range for each stimulus, as well as established normative ranges.
- A method for assessing the measurement uncertainty for pure-tone audiometry is given in BS EN ISO 8253-1:2010 but it has not yet been extended to other core procedures. There is also a need for guidance on variability, how it can be assessed and how it relates to uncertainty of measurement.
- The recommendations for noise exposure given here are pragmatic. Provision of automated noise dose calculations within test equipment would be beneficial.

8. Summary Guidance

General:

- All audiologists should understand, and be able to apply, the principles and recommendations stated in this document within their clinical roles.
- These principles and recommendations should be part of the syllabus for training audiologists, and should be actioned by Higher Education Institutions and other training providers.

Acoustic principles:

- Clinicians should be aware of the effects of resonances during any audiological measurement. These will particularly affect sound field measurements, real ear measurements (REMs) and pure tone audiometry (PTA) at higher frequencies, leading to potential error.
- Clinicians should be aware that a measurement of sound pressure level (SPL) will be entirely dependent on where a microphone is placed for some frequencies of sound in an enclosed space, such as an ear canal. Clinicians should understand that this may particularly affect probe tube measurements.
- Clinicians should understand that sound intensity should be used when calculating the addition of sound sources or average noise exposure, not the sound pressure. This can then be converted back to a sound pressure level.
- Clinicians should be aware of the different type of sound level measures because these are easily confused and can lead to errors in calibration or estimated noise exposure.
- The correct units, frequency weighting and, where applicable, time averaging of any measure of sound shall be specified in all cases.

Ambient noise:

- It should be the professional responsibility of a clinician to determine that any environment is sufficiently free of ambient noise and other distractions.
- Tests shall be conducted in a separate, private, quiet room away from any sources of ambient noise or other distractions. Audiological testing shall not, under any circumstances, be conducted in an open area where a patient has visibility of, or is visible to, other individuals other than the tester or accompanying persons.
- In most clinical practice, a minimum measurement level of 20 dB HL is sufficient for sound field assessment.
- Higher levels of ambient noise can be allowed when the ears are occluded by earphones, but the allowable levels for bone conduction measurements should be considered to be the same as for sound field audiometry.
- 10 dB shall be subtracted from a calibrated speech stimulus level given by audiometric equipment to calculate allowable ambient noise levels, due to the variability inherent in speech signals.

- Any measurement of ambient noise shall be conducted to accurately represent normal clinical conditions.
- It is not considered practical to measure ambient noise on a continuous basis in clinic with a SLM. Continuous subjective monitoring of ambient noise should be the responsibility of qualified clinicians whilst preparing for and conducting tests.

Sound level meters (SLM):

- Audiology departments shall ensure that SLMs are compliant with class 1 standards.
- The minimum sound level measurable by a SLM in each frequency band must be clearly stated.
- Single broadband measures of ambient noise can be useful for detecting potential changes in test room background noise but should not be used to establish the level of background noise in a room.
- Personal sound exposure meters (PSEM), so-called “type 3” meters and smartphone apps are insufficient for assessment of ambient noise and shall not be used for this purpose.

Sound field audiometry

- Test rooms should ideally meet the requirements of a quasi-free sound field but, as a minimum, audiology departments shall determine the measurement variability within each room. Minimum test levels should also be included in the reporting of results on an audiogram.
- Hand-held sound generators (or “warblers”) may be used but it shall be recognised that they introduce greater measurement variability, so the uncertainty of measurement shall be established.
- In a quasi-free sound field, or similar conditions likely to be found in clinical rooms, frequency-specific signals should be narrowband, such as modulated tones and narrow-band noise.
- FRESH^{TM} noise is a valid test stimulus and preferable to narrow-band noise.
- Results shall be reported in a manner which reflects the true accuracy and reliability of results.
- Responses to non-standard stimuli (e.g. Ling sounds, speech and music) can only be, at best, indicative and it is recommended that descriptive word-based reporting is employed, rather than numerical values.

Noise exposure and maximum stimulus levels:

- Stimulus amplitudes shall not be increased beyond any level that causes discomfort for a patient.
- Clinicians must restrict a patient’s noise exposure as far as possible, using only stimuli that do not cause a patient discomfort and that will lead to useful diagnostic information that is likely to have an impact on the patient’s individual management plan.
- The daily personal noise exposure level ($L_{EP,d}$) shall not exceed the lower exposure level of 80 dB(A) for any patient, clinician or accompanying person.
- However, the average daily noise exposure level shall not be used to indicate the safety or risks associated with individual short-duration stimuli used in audiometric tests.

- A 135 dB pSPL limit should be used to determine the maximum peak level of transient stimuli (< 0.5 seconds) such as tone bursts, chirps and clicks, typically used in electrophysiology. Clinicians shall take care to convert the 135 dB pSPL limit to the units used on measurement equipment. High stimulus levels (e.g. those close to, or above, the suggested maximum) should not be used without a clearly defined and noted clinical rationale.
- 115 dB SPL shall be the maximum limit for any intermediate duration (0.5 to 5 seconds), tonal stimulus employed within audiometric testing. This shall be converted to the units of audiometric equipment (e.g. dB HL) for the transducers used and a further limit of 105 dB HL shall be imposed at any frequency. These levels shall not be exceeded except where a clear clinical need has been defined and documented.
- Maximum exposure levels for children are lower than those for adults. Maximum stimulus levels should be restricted to 5 dB less than those proposed for adults.

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Appendix:
Weighting and RETSPL Tables

Z, A and C weightings are specified in BS EN 61672-1 and Table A1 provides the figures for the audiometric frequencies. The Z (or zero) weighting is also commonly referred to as an unweighted or flat response. It is possible to calculate the different weighted levels of pure tones by using this table. For example, a 125 Hz tone with a measured SPL of 73.9 dB(A) would be equivalent to 89.8 dB(C) and 90 dB(Z). It is important to note that measurements of noise and other broad-band signals cannot be converted in this way as they contain multiple frequency components.

Table A1: Z, A and C frequency weightings at audiometric frequencies.

Frequency (Hz)	Z	A	C
125	0	-16.1	-0.2
250	0	-8.6	0.0
500	0	-3.2	0.0
750	0	-0.8	0.0
1 000	0	0.0	0.0
1 500	0	+1.0	-0.1
2 000	0	+1.2	-0.2
3 000	0	+1.2	-0.5
4 000	0	+1.0	-0.8
6 000	0	-0.1	-2.0
8 000	0	-1.1	-3.0

dB HL cannot be simply determined from SPL by a generic conversion factor. The selection of the RETSPL (RETVFL for BC) depends on the test signal, the transducer and the ear simulator employed for calibration and parts 1 to 8 of BS EN ISO 389 (1997-2019) provide current reference figures. BS EN ISO 389-7:2019 provides the sound field RTSPL for binaural listening in a free-field or a diffuse-field environment. Table A2 provides the sound field figures together with example figures for supra-aural and insert earphones. The difference between the free-field and diffuse-field RETSPL figures is less than 2 dB in the frequency range typically employed for sound field audiometry (≤4 kHz) and this would not usually be considered as a clinically significant difference. However, the BSA recommendation is to employ the diffuse-field figures

for calibration of warble-tone and NB noise stimuli (see section 3.3). It is important that the correct RETSPL/RETVFL are selected for calibration as there can be significant differences in the levels for different transducers. The BS EN ISO 389 series does not contain figures for all transducers currently in use, so it may be necessary to consult manufacturer's documentation to determine the appropriate reference figures for a particular transducer.

Table A2: Example RETSPLs for 0 dB HL.

Frequency (Hz)	RETSPL (dB)			
	Free sound field, binaural listening, sound source in front of listener	Diffuse sound field, binaural listening*	Generic RETSPL for supra-aural earphones (not model-specific), monaural listening to pure tones, measured in an IEC 60318-1 artificial ear	RETSPL for ER-3A inserts, monaural listening to pure tones, measured in an IEC 60318-4 occluded-ear stimulator
125	22.1	22.1	45.0	28.0
250	11.4	11.4	27.0	17.5
500	4.4	3.8	13.5	9.5
750	2.4	1.2	9.0	6.0
1 000	2.4	0.8	7.5	5.5
1 500	2.4	1.0	7.5	9.5
2 000	-1.3	-1.5	9.0	11.5
3 000	-5.8	-4.0	11.5	13.0
4 000	-5.4	-3.8	12.0	15.0
6 000	4.3	1.4	16.0	16.0
8 000	12.6	6.8	15.5	15.5

* BSA recommended figures for calibration of warble and NB noise stimuli.