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Assessing the Acoustic Characteristics of Rooms: A Tutorial With Examples

Brad Rakerd

Eric J. Hunter

Mark Berardi

Department of Communicative Sciences and Disorders, Michigan State University
East Lansing, MI

Pasquale Bottalico

Department of Speech and Hearing Science, University of Illinois
Champaign, IL

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In both audiology and speech-language pathology, as well as in speech and hearing science research, the space where the work is done is an integral part of the function. Hence, for all of these endeavors, it can be important to measure the acoustics of a room. This article provides a tutorial regarding the measurement of room reverberation and background noise, both of which are important when evaluating a space's strengths and limitations for speech communication. As the privacy of patients and research participants is a primary concern, the tutorial also describes a method for measuring the amount of acoustical insulation provided by a room's barriers. Several room measurement data sets—all obtained from the assessment of clinical and research spaces within our own department—are presented here as examples.

Speech communication is achieved through a dynamic interaction among three components: (a) the *talker* who creates a spoken message, (b) the *transmission channel* over which that message travels, and (c) the *listener* who interprets the message as received. Physical changes to any of these may lead to compensatory changes in the others. One well-known example of this takes place when a transmission channel is disturbed by noise. In response, the talker may reflexively increase vocal effort to maintain a satisfactory signal-to-noise ratio (SNR) for the system as a whole (Lombard effect [Junqua, 1993; Lane & Tranel, 1971]). Talkers have also been shown to vary their vocal intensity and their speech rate across rooms that differ in size and/or reverberation time (Black, 1950). Changing a talker or listener's perceptions regarding one of the other components in the system can also affect performance. In a notable example, talkers have been shown to modify their articulation to speak more slowly and more clearly when informed that a listener has hearing loss (Ferguson & Kewley-Port, 2007; Picheny, Durlach, & Braida, 1985, 1986).

Speech-language pathologists and audiologists are particularly concerned with speech communication breakdowns and hence with any factors that may limit communication success. Whenever a conversation takes place indoors, there are two factors that can impose some limitations. The first is *room reverberation.* Reverberation arises whenever speech sound (or any other sound) reflects and re-reflects from a room's boundaries. This creates copies of the sound that linger in the room for some time. Room reverberation especially disturbs temporal speech cues because it "smears" the talker's message over time (Nabelek & Mason, 1981). The reflected speech may also mask the original (direct speech) sound to some degree (Nabelek, Letowski, & Tucker, 1989).

The second factor that can limit speech understanding indoors is the presence of any other sound within a room, including sounds that may have originated outside and then penetrated in through one or more of the room's boundaries. Interfering sound in a room is broadly referred to as *background noise*. Depending on its level relative to the speech, background noise may physically mask the target message in whole or in part. Furthermore, if the background noise is itself a speech sample, it may have the added effect of cognitively interfering with the listener's processing of the target message (Arbogast, Mason, & Kidd, 2002; Brungart, Simpson, Ericson, & Scott, 2001; Rakerd, Aaronson, & Hartmann, 2006).

Background noise and room reverberation can also have significant consequences in other domains. Both factors have been shown to reduce listener satisfaction when wearing hearing aids (Amlani, 2001; Amlani, Rakerd, & Punch, 2006; Boymans & Dreschler, 2000). Moreover, both can affect the accuracy of sound localization by both listeners with normal hearing and hearing impairment (Keidser, O'Brien, Hain, McLelland, & Yeend, 2009; Litovsky, Parkinson, & Arcaroli, 2009; Roberts, Koehnke, & Besing, 2003).

A final point about noise and reverberation is that their negative effects on speech perception can fall disproportionately on children of school age (Yacullo & Hawkins, 1987), especially if those children have hearing loss or learning disabilities (Crandell, 1992; Crandell & Smaldino, 2000; Nober & Nober, 1975) or if they are nonnative speakers of the primary language used for instruction (Crandell & Smaldino, 1996). Findings of this kind have led to a substantial revision of our national standards for classroom acoustics. The current standards (ANSI S12.60-2010) make two recommendations. First, the noise level in an unoccupied classroom should not exceed 35 dBA (see the Background Noise section below for a discussion of A-weighted sound level measurement). Second, the reverberation time should not exceed 0.6 s in smaller classrooms and 0.7 s in larger ones. Classrooms that meet these standards can be expected to provide increased opportunities for student success when listening to speech.

The acoustics of a room can also affect a vocalist. Whether the vocalist is a performer on stage or a teacher in a classroom, the room can have a notable impact on oral production. Room effects of this kind sometimes go unnoticed, especially when the conversations that take place in a room are occasional or brief. However, for talkers who must produce speech for extended periods, as teachers do daily, rooms can have a significant impact. A number of different room parameters have been found to be important in this regard. For example, there seems to be an optimal reverberation time (0.75–0.85 s) that teachers indicate that they prefer and that also requires reduced vocal effort (Bottalico & Astolfi, 2012). A parameter termed the *speech clarity factor* (discussed further in the Room Reverberation section) has been shown to quantify how the early reflections of sound in a room can be tuned to help a talker (Bottalico, Graetzer, & Hunter, 2015). Finally, Pelegrín-García, Smits, Brunskog, and Jeong (2011) have pointed up the importance of a parameter they call *room gain* for a room's overall level of voice support. A clear implication of findings such as these is that the talker's perspective should not be overlooked whenever designing a room.

Organization

This tutorial presents measurement data obtained in five different rooms as examples. In the Rooms and Equipment section below, we describe those rooms and the equipment used when making measurements within them.

Next, in the Room Reverberation section, we outline a series of measurements regarding room reverberation. All of those measurements have been found to be useful when characterizing a space's suitability for some aspect of speech communication, and all of them can be determined for a room by recording and then analyzing its *impulse response* (for a complete list of terminology and abbreviations, see Appendix A). With that in mind, we provide an account of the methods that we employed when making impulse response recordings and subsequent measurements for the example rooms. We also present data obtained in each room and consider their implications for optimal room use.

In the Background Noise section, we discuss methods that are commonly used when measuring the level of background noise present in a room and we report on noise measurements that we have made in several of the example rooms.

Finally, as the privacy of patients and research participants is always a primary concern, in the Acoustical Insulation section, we describe a method that we have used to measure the amount of acoustical insulation that is provided by room barriers and we consider the implications of measurements made in several clinical rooms within our department.

Rooms and Equipment

Faculty and students in the Department of Communicative Sciences and Disorders at Michigan State University do their daily work in a variety of clinical spaces, laboratories, recording booths, and classrooms. Acoustical measurements made in five of those spaces are examined throughout this tutorial. Below, we briefly describe those spaces, and the equipment and software that were used to make the measurements within them.

Clinical Rooms

First described are three rooms commonly seen in both communicative sciences and disorders departments and clinics: a small room suitable for one-on-one counseling or therapy, a somewhat larger room suitable for working with a family or small group, and a still larger room that can accommodate a substantial number of participants for group work. We refer to these as the small, midsized, and large clinical rooms, respectively. A description of each of them is given in Table 1.

Table 1. Descriptions of the three clinical rooms.

Small clinical room	Small room (64 ft^2) with carpeting on the floor and acoustical tiles halfway up the walls and on the ceiling; furnished with a small desk and a few small chairs.
Midsized clinical room	Medium-sized (169 ft^2) carpeted room with acoustical tiles on the ceiling only; limited furnishings and some open floor space; shares a wall and an observation window with the small room.
Large clinical room	A larger (285 ft^2) carpeted room with acoustical tiles on the ceiling only; can hold up to 20 occupants; tables, chairs, and computers present throughout.

Acoustically Special Research Rooms

Two acoustically special laboratory rooms—an anechoic chamber and a reverberant room—are described in Table 2. Both of these rooms have been used extensively for speech and hearing research. Such rooms can offer insight into extreme acoustic cases.

Table 2. Descriptions of the two research rooms.

Anechoic chamber	A 138-ft^2 room with soft, highly sound-absorbing surfaces throughout; used to make precise acoustic measurements that would otherwise be disrupted by the presence of reverberated sound.
Reverberant room	A 527-ft^2 room with hard, acoustically reflective surfaces throughout; used to measure things such as speech intelligibility or musical quality that may be strongly affected by room reverberation.

Equipment and Software

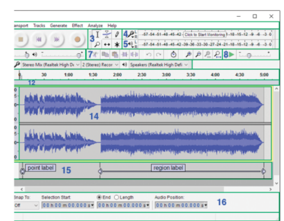
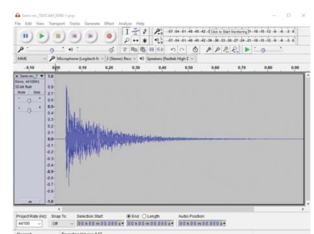
The equipment used when making audio recordings in the example rooms is listed, and briefly described, in Table 3. Also described are software programs/plug-ins used for the acoustical analysis in this study.[1]

Table 3. Equipment and software used when making and analyzing audio recordings for the example rooms.

Sound-level meter	NTI Class 1 sound-level meter NTI XL2 (NTI Audio, Liechtenstein, http://www.nti-audio.com)	
Omnidirectional microphone	Behringer ECM8000 omnidirectional condenser microphone with flat frequency response (http://www.behringer.com)	
Digital recorder	TASCAM DR-40 multichannel digital recorder (TASCAM, Japan, http://www.tascam.com)	

(continued)

[1]We report on this equipment and software, as well as its use, for example purposes only, not as an endorsement of these specific items. There are many commercial options available that can perform similar functions.

Impulse source	12-in. balloons, balloon pump, and pins	
Loudspeaker	Rokit5 G3 studio monitor speaker (KRK Systems, http://www.krksys.com)	
Audio editing software	Audacity multitrack audio editor and recorder (http://www.audacityteam.org)	
Acoustical analysis plug-ins	AURORA signal processing modules (plug-ins) for Audacity (and for Adobe Audition; Aurora Plugins, http://www.aurora-plugins.forumfree.it)	

Room Reverberation

The impulse response of a room captures a number of acoustic characteristics, including its reverberation. In this section, we describe three parameters of the impulse response that have been found to be useful when predicting a room's suitability for speech communication: (a) reverberation time, (b) early decay time (EDT), and (c) clarity factor.[2]

Reverberation Time

RT60. The reverberation time of a room is the time it takes for the sound pressure level of an impulsive sound to decay by 60 dB. When the full time course of the 60-dB decay can be measured directly from the impulse response, the resulting measurement is referred to as RT60. The reverberation time of a room generally varies with frequency, and it should therefore be calculated separately for different frequency bands. Most commonly, RT60 and other measures of reverberation time such as T20, which is described in the next section, are calculated at

[2]All three of these parameters are defined in standard ISO 3382-1 (2009), as are a number of the measurement procedures that we employed in this study.

third-octave center frequencies or at octave center frequencies. However, they are sometimes calculated at only a few frequencies in a range of interest, or they are calculated at multiple frequencies and then averaged before reporting. For comparison, here are some expected broadband RT60 values for several different types of rooms: conference room, 0.4–0.8 s; chamber music auditorium, 0.9–1.4 s; and orchestral hall or other large performance spaces, 1.5–2.4 s.

T20. A practical limitation often encountered in everyday rooms is that it can be impossible to directly measure the full 60 dB of decay due to the presence of some form of background noise. An alternative procedure often employed in these environments—and the one that we employed throughout in this study—is to directly measure the time needed for the impulsive sound pressure to decay by 20 dB and to then linearly extrapolate to estimate the 60-dB decay time. The resulting estimated value is then referred to as T20. It should be emphasized that T20 is an estimation of the 60-dB decay time by extrapolation, not a measurement of the 20-dB decay time.

To be confident that the observed decay is relatively constant throughout the T20 measurement (and that the extrapolation is valid), it is recommended that T20 not be measured during the initial portion of the impulse response, which tends to be very uneven. To avoid this, the T20 measurement should begin when the impulse response has decayed by 5 dB relative to its peak and conclude when it has decayed by an additional 20 dB beyond that point (i.e., the elapsed time is measured from −5 to −25 dB). An SNR of at least 35 dB is called for when measuring T20.

Early Decay Time

During the initial portion of the impulse response, contributions come from both the direct sound and the early reflections of that sound from nearby surfaces. These early reflections can produce sharp and uneven changes in the initial portion of the impulse response, which is why they are excluded from the T20 measurement. At the same time, early reflections have been found to improve speech recognition (e.g., Bradley, Sato, & Picard, 2003) and to be important for spatial hearing. A measurement termed *EDT* has been developed to assess them.

The EDT of a room is measured from a linear regression line fit to the initial portion of the impulse response. The time needed for the first 10 dB of impulse decay is measured directly from the regression line (elapsed time between 0 and −10 dB). Then, the time that would be required to decay a full 60 dB is estimated using the initial decay rate. Similar to RT60 and T20, EDT is typically computed separately for different frequency bands. EDT and T20 are often reported together to provide a more complete picture of a room's reverberant characteristics because two rates of decay usually do exist—an early energy decay and a later energy decay.

Clarity Factor

The clarity factor for speech (often referenced as *C50*) is the ratio of sound power present within the first 50 ms of the impulse response, when early reflections occur, to the sound power present thereafter. Measurement of C50 is motivated by the Haas effect for speech, whereby acoustical reflections that reach a listener within 50 ms of the direct sound have been shown to integrate with it and to have positive effects on the perception of the spoken message (Haas, 1972). C50 is expressed in decibels. It improves when strong early reflections are present in a room, and desirable values for C50 are +3 dB or greater (Marshall, 1995).

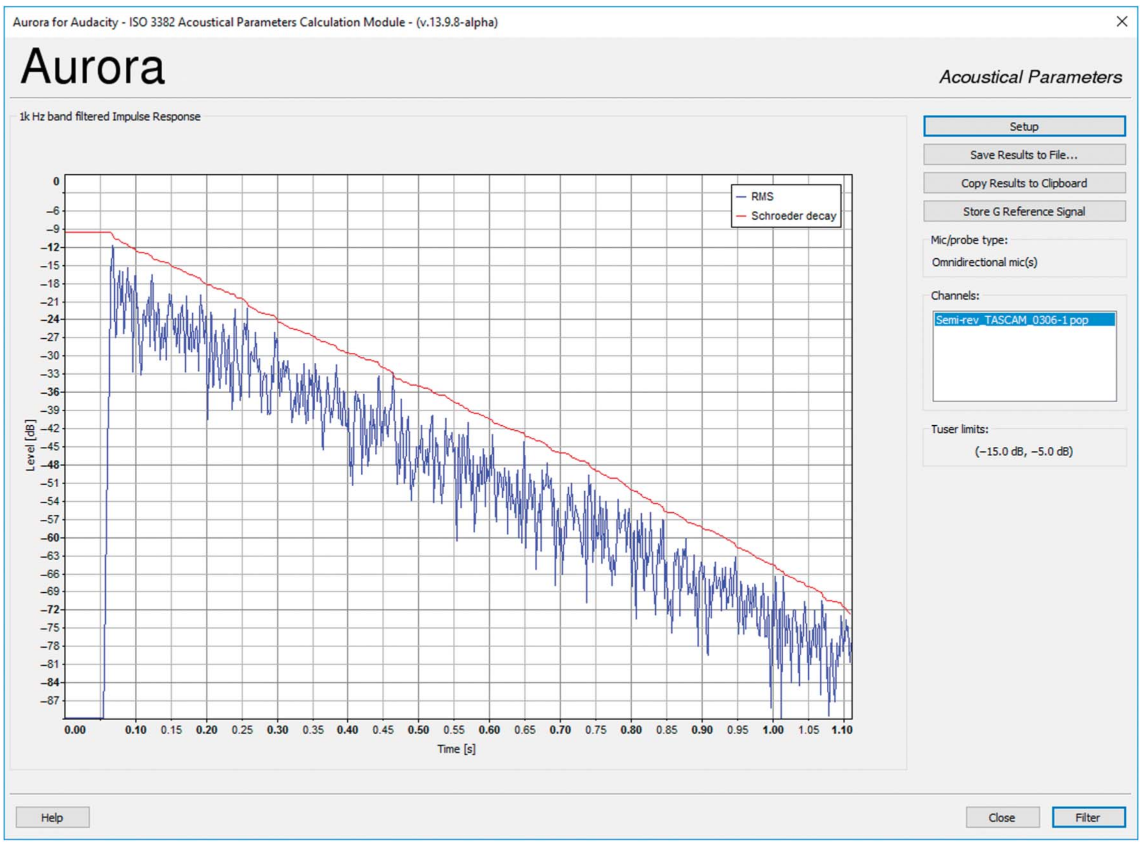
It should be noted that, when the signal of interest is music, it is a common practice to calculate a related but different clarity factor than the one used for speech. The music-related clarity factor has an 80-ms time window.

Reverberation Measurement Examples

Measurements of room reverberation were obtained from impulse response recordings made in all of the rooms described in the Rooms and Equipment section. In each room, three arbitrary points were selected, all of them at least 1 m away from the room's surrounding surfaces. Six impulse response recordings were then made, with each of the three points serving as both a source location and a recording location. The height of the microphones above the floor was 1.2 m, corresponding to the ear height of average listeners seated in chairs.[3]

The impulse response of a room can be generated with a variety of impulsive sound sources (e.g., pistol shots, spark gap impulses, or balloon pops). In the examples reported here, the source was always a balloon pop.[4] Twelve-inch balloons were pumped until nearly the maximum size and then pricked with a pin. The pop impulse was then recorded using the omnidirectional microphone and digital recorder described in Table 3 (sample rate = 44.1 kHz, 16-bit digital resolution). After importing the recordings into a laptop, each impulse response was filtered into octave bands. A decay curve for each band was then calculated by a backward integration of its squared impulse response (Schroeder, 1965). A sample decay curve is shown in Figure 1.

Figure 1. Screenshot showing the decay curve of an octave-band filtered impulse response (band center frequency = 1000 Hz).



[3]Recording positions should always be selected to be representative of positions where listeners would normally be located within a room. They should also sample the space broadly, with each recording position about 2 m from the others and at least 1 m from the nearest reflecting surface, including the floor.
[4]Impulsive sounds (such as a "balloon pop") have a large peak amplitude and a very short duration. They also have a broad spectrum that is quite regular (Durrant & Lovrinic, 1984).

Measurements made for the full set of source and microphone combinations within a room (six in all) were pooled to obtain spatial average values of T20, EDT, and C50 for each octave band. Figure 2 shows these results for the three clinical rooms, and Figure 3 shows them for the two research rooms. Finally, for each room, the individual octave band results were averaged together to obtain more global indices to the room's reverberation. The resulting average values are reported in Table 4.

Figure 2. Reverberation measurements (T20, early decay time [EDT], and C50) for the three clinical rooms, plotted at octave-band center frequencies. Small room: dotted line; midsized room: dashed line; large room: solid line.

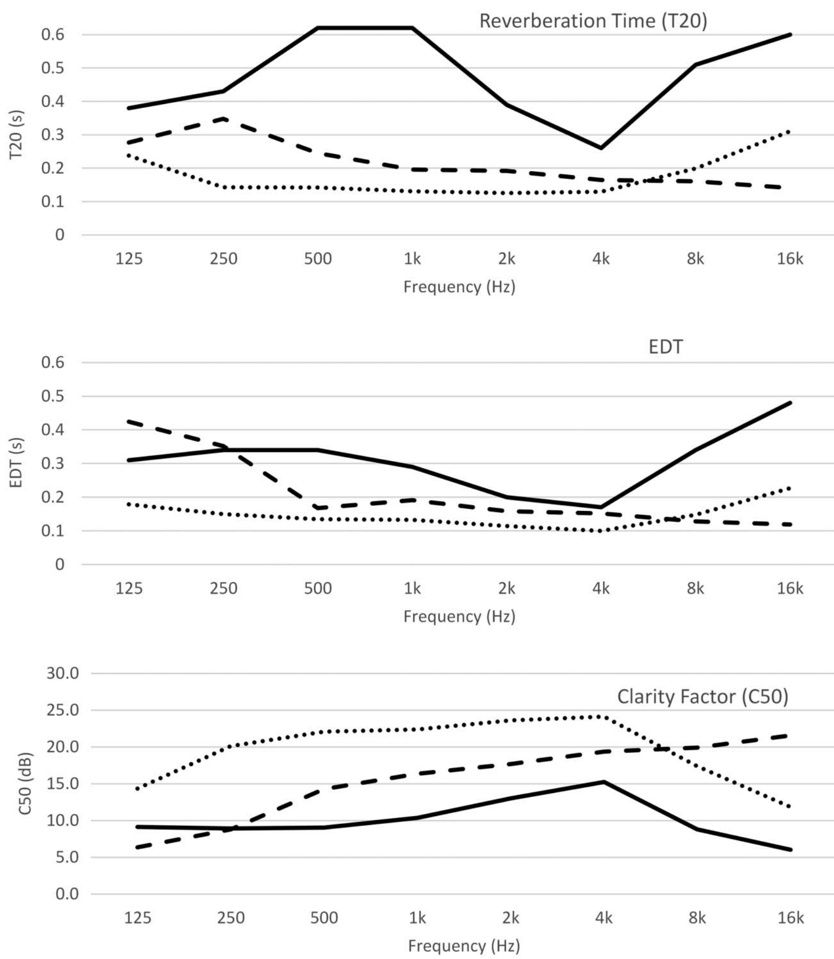


Figure 3. Reverberation measurements (T20, early decay time [EDT], and C50) for the two research rooms, plotted at octave-band center frequencies. Anechoic chamber: dashed line; reverberant room: solid line.

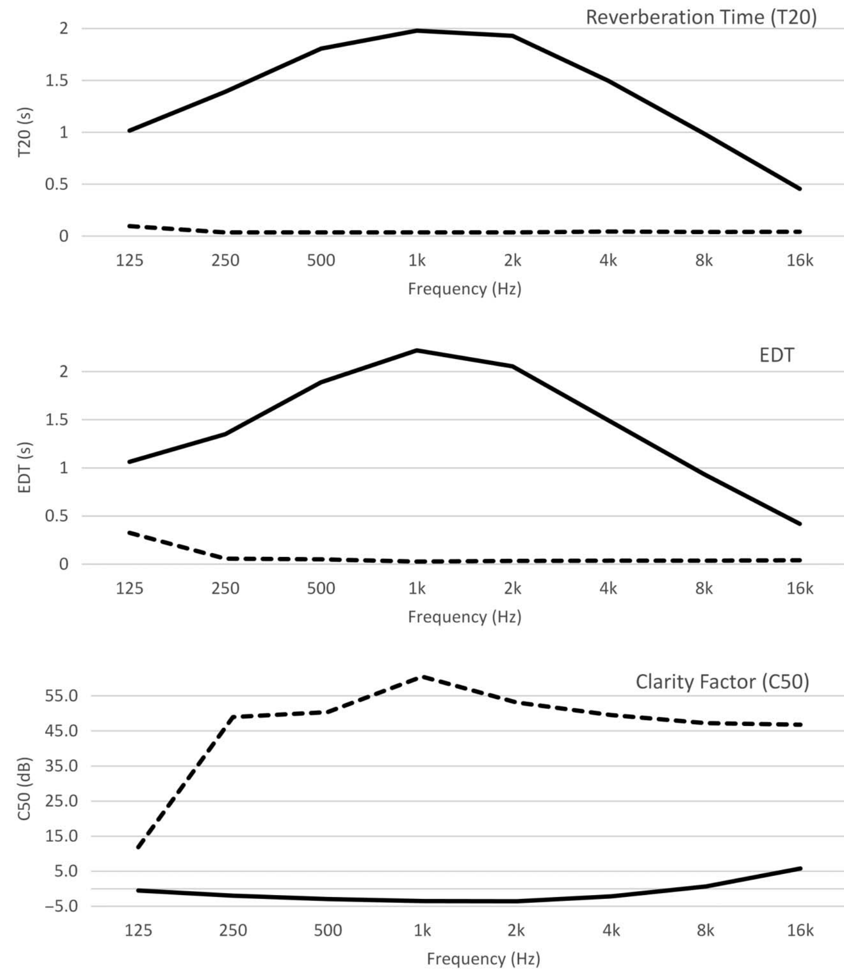


Table 4. Reverberation measurements of T20, EDT, and C50; reported values are averages over all frequency bands measured.

Room	**T20 (s)**	**EDT (s)**	**C50 (dB)**
Small clinical room	0.18	0.15	19.50
Midsized clinical room	0.22	0.21	15.50
Large clinical room	0.48	0.31	10.10
Anechoic chamber	0.05	0.08	46.10
Reverberant room	1.38	1.49	−1.00

Some Observations

Using Table 4 as a starting point, it can be seen that the anechoic chamber had a near-zero reverberation time as would be expected and that the reverberant room had the longest reverberation time of any room tested, also as expected. Measured values of T20 and EDT for the reverberation room were all in the 2-s range at vowel formant frequencies (see Figure 3). This indicates that this room would provide poor support for extended conversations, because speech reverberation of this magnitude is highly noticeable and distracting. It also substantially distorts and masks speech cues.

Reverberation time increased with room size for the three clinical rooms, with the average value of T20 equal to 0.18 s for the small room, 0.22 s for the midsized room, and 0.48 s for the large room (see Table 4). We sometimes use the large clinical room for classroom instruction. Average T20 and EDT values for this room (0.48 and 0.31 s, respectively) indicate that its reverberation time is quite acceptable for this use. However, Figure 2 shows that T20 in particular was quite variable across frequencies and, at some of them (500 and 1000 Hz), T20 approached the upper limit of what is recommended for classrooms (0.6–0.7 s).

Four of the five rooms had very similar EDT and T20 times. The large clinical room was an interesting exception. It had a much shorter EDT than T20, pointing up a clear difference between the early reverberation decay and the later decay in that space. Figure 2 shows that the small clinical room had notably higher values of C50 than the other two clinical rooms at almost every frequency. When counseling someone or having any other conversation where the intelligibility of speech is of the highest priority, a room like this one would be ideal.

Almost every measurement of T20, EDT, and C50 shown in Figure 3 was dramatically different for the anechoic chamber and the reverberant room. This clearly shows that the reverberation characteristics of a room can be well managed with proper engineering. Finally, it should be noted that the *y*-axes in Figure 3 all span a much larger range of values than the corresponding axes in Figure 2. This was needed to display the extreme values of reverberation (as indexed by both T20 and EDT) found in the reverberation room and the extreme values of clarity found in the anechoic chamber.

Background Noise

Noise is part of our environment, some as a result of our modern society and the rest as part of nature. Laboratory studies often employ physically well-defined noise sources such as white noise, which has an equal energy at all frequencies, or pink noise, which has an equal energy in all third-octave bands. The background noises encountered in rooms are generally much less regular. Background noise is any unwanted sound that may be present in a room, and it can have multiple sources, some, such as the noise from an overhead fan, coming from inside the room itself, and others such as traffic noise coming from outside.

An important thing to determine about background noise is its overall intensity or *sound level*. Sound level is measured and reported in decibels, and it should always be measured with a calibrated sound-level meter (SLM). SLMs typically have user selectable options for frequency weighting and time averaging (see below). To obtain accurate measurements, the SLM must be properly configured, and when making a measurement, it should always be positioned well away from any acoustically reflective surfaces within a room. Minimally, it should be at least 1 m away. Finally, if at all possible, the background noise level in a room should be measured at multiple locations and then averaged to get a representative estimate.

Frequency Weighting

An SLM measurement that treats all frequency bands equally has traditionally been referred to as one made with *flat* weighting, but more recently, this has been called *Z-weighting* (or zero weighting). A more commonly used SLM setting is A-weighting (dBA), which employs a filter that mimics the frequency response of the ear for lower amplitude sounds. (The dBA filter approximates the 40-phon equal-loudness curve.) B-weighting (rarely used) and C-weighting filters are designed to approximate the ear's responses to progressively higher amplitude sounds. There is a D-weighting filter that is sometimes used in outdoor conditions, usually in assessing airport/aircraft noise. Noise tables, standards, and noise ordinances for municipalities are commonly written in dBA. As a reference, the sound level of normal conversational speech at 1 m is between 50 and 60 dBA, and that of a lawnmower at the same distance is between 85 and 100 dBA.

Time Averaging

An SLM will generally offer options to integrate its input over a short time window (125 ms) or over a longer window (1000 ms). One or the other of these settings is generally used when measuring the sound level of a background noise that is continuous and relatively constant over time. However, in many cases, the background noise present in a room will include some sources that are nonsteady (fluctuating), such as the noise coming from passing cars or from conversations taking place in an adjacent room. For nonsteady noises, a longer period of time averaging is needed, and to make these measurements properly, they should be made with an SLM that offers a time setting of Leq. The *Leq* for a nonsteady noise is the sound level of a continuous noise that would have equal power when integrated over the duration of the Leq measurement. The decision to use either Leq or a slow or fast time setting usually depends on the regulations being followed; for example, many occupational noise regulations stipulate that Leq should be used.

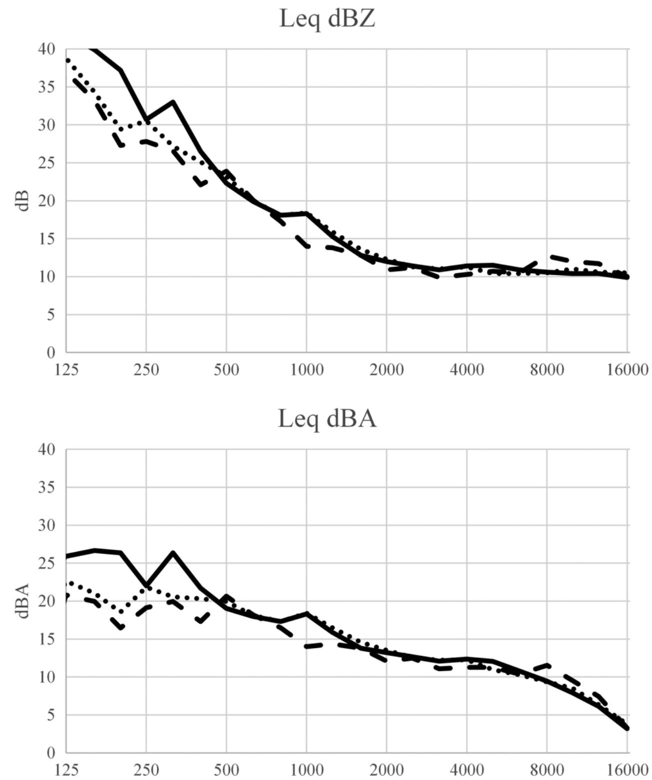
SNR

SNRs are always an important consideration when evaluating speech transmission for a space. For example, a classroom designed to meet current recommendations will have a background noise level of 35 dBA or less. This standard was chosen because speech levels within the classroom will generally exceed it, which means that conversations there will generally unfold in conditions where the SNR is positive.

Noise Measurement Examples

In each of the three clinical rooms, we measured the unweighted Leq (dBZ) and the A-weighted Leq (dBA) for every one-third octave frequency band from 125 to 16,000 Hz. The measurements were made from 30-s background noise samples taken with the SLM described in Table 3. (The SLM was mounted on a tripod and placed at least 1 m from any reflective surfaces in each room.) The upper panel of Figure 4 shows the Leq dBZ results, and the lower panel shows the Leq dBA results.

Figure 4. Flat-weighted (Leq dBZ) and A-weighted (Leq dBA) sound level measurements of the background noise level present in each of the clinical rooms. Results are plotted at one-third octave-band center frequencies. Small room: dotted line; midsized room: dashed line; large room: solid line.



The first thing to note in Figure 4 is that the background noises present in the three clinical rooms were all very similar. The only noticeable difference was a slightly increased power in the lowest frequencies for the noise in the large clinical room. The second thing to note is that the background noises all had most of their power in the lower frequencies, which is consistent with a heating, ventilation, air condition system (HVAC) being the primary source of noise in all cases. Finally, a comparison of the upper and lower panels of Figure 4 shows that there are some marked differences between the unweighted and A-weighted sound level measurements. A-weighting greatly reduced the measured noise power at low frequencies (below about 500 Hz) and at very high frequencies (above 8000 Hz) to reflect the fact that listeners are relatively insensitive to these frequency components when listening to sounds (such as these background noises) that have a relatively low intensity overall.

Table 5 reports average values of Leq dBZ and Leq dBA for each of the three clinical rooms, where the averages were computed over the full set of frequency bands (125–16,000 Hz). For comparison, average Leq values are also given for measurements made in the anechoic chamber and in the reverberant room. It can be seen that the background noise levels in all of the rooms more than meet the current standard for classroom use (noise level ≤ 35 dBA). It can also be seen that both of the research spaces, which were expressly engineered to have low background noise levels, did in fact have measured levels well below those of the clinical rooms.

Table 5. Leq sound level measurements made with flat frequency weighting (dBZ) and A-weighting (dBA); reported values are averages over all frequency bands measured.

Room	**Leq dBZ**	**Leq dBA**
Small clinical room	31.1	26.8
Midsized clinical room	29.8	25.9
Large clinical room	34.2	27.8
Anechoic chamber	22.6	14.6
Reverberant room	24.6	20.1

Acoustical Insulation

All three of the clinical rooms described in the Rooms and Equipment section are entered from a corridor. A first question that we asked about these rooms is how well each of them was acoustically insulated from its corridor. Two of the clinical rooms—the small room and the midsized room—are separated by a common wall with a large observation window at its center, and for these rooms, we also asked how well that wall acoustically insulated these rooms from one another.

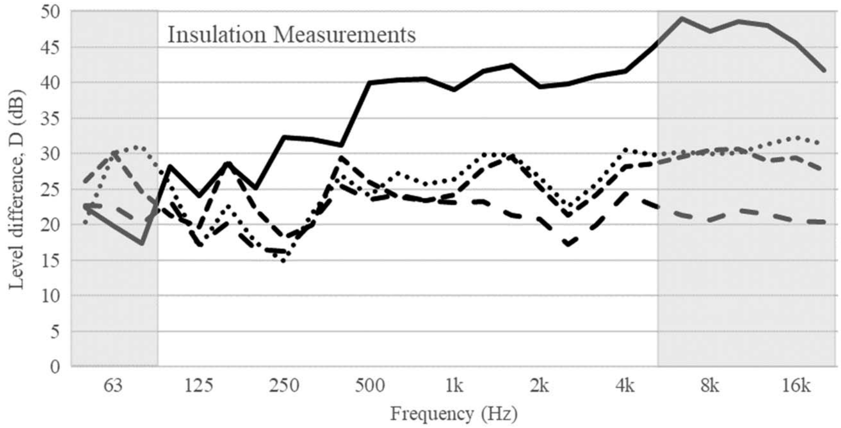
We measured acoustical insulation with a method described in ISO standard 140-4 (1998). Specifically, we measured the insulation parameter that the standard refers to as the *level difference*, symbolized as D. To find D, a sustained sound is generated in one room (termed the *source room*) and its space–time average sound pressure level is then measured both inside that room (L1) and in a comparison room (the *receiving room*) from which it is separated (L2). Parameter D then equals the decibel difference between these two levels (D = L1–L2).

Measurement Examples

Pink noise (which has equal power per one-third octave band) was used as the sound source for these measurements, and it was presented from two arbitrarily chosen locations in each source room. Ten-second samples of all noise presentations were recorded at three different locations within the source room and at three locations within a comparison receiving room.[5] Spatial average values of D were then found for every one-third octave frequency band from 63 to 16,000 Hz. The results are plotted in Figure 5. A subset of the frequency bands are highlighted in white in the figure (125–4000 Hz). These highlighted frequencies are especially important for speech communication.

[5]These locations were identified by applying the same decision rules that were described above for the multilocation recording of impulse responses.

Figure 5. Insulation measurements (level difference, D) for the clinical rooms, plotted at one-third octave-band center frequencies. Corridor > small room: dotted line; corridor > midsized room: short dashed line; corridor > large room: long dashed line; small room > midsized room: solid line.



The results for the three clinical rooms clearly indicate that they had similar levels of acoustical insulation from their respective corridors (compare the dotted line and short and long dashed line plots in Figure 5). Given the contemporaneous construction of these rooms, this is not unexpected. Upon examination, it was found that the most likely transmission path for sound exchange between each of the clinical rooms and its corridor was from underneath the door where there was a small air gap.

The corridor insulation values for all three clinical rooms were modest at speech frequencies (20–25 dB on average). This means that conversations taking place within any of these rooms could be audible in its corridor and that conversations in a corridor could disturb a clinical session. Although it is unlikely that any of these conversations would be intelligible to a listener given the level of insulation, it would be a best practice to keep the corridors vacated whenever a clinical session is taking place.

Acoustical insulation between the small clinical room and the midsized clinical room (solid line plot in Figure 5) was significantly better than the insulation of either of these rooms from its corridor. This was true at all frequencies above about 200 Hz, including the speech frequencies from 200 to 4000 Hz. Accordingly, it is expected that any conversation taking place in one of these rooms would be, at most, barely audible in the other. Certainly, it would not be intelligible there.

Summary

Speech communication is achieved through a dynamic interaction among a talker, a transmission channel, and a listener. Rooms are an important component of the transmission channel, and their acoustics can either promote or limit communication success. This tutorial reviewed key acoustical parameters regarding room reverberation, background noise level, and acoustical insulation and outlined best practice procedures to be followed when measuring those parameters. Measurement data from the example rooms used for both clinical and research purposes were presented.

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References

American National Standards Institute. (2010). *Acoustical performance criteria, design requirements, and guidelines for schools, Part 1: Permanent schools* (ANSI S12.60-2010). Retrieved from https://webstore.ansi.org/RecordDetail.aspx?sku=ANSI%2fASA+S12.60-2010%2fPart+1

Amlani, A. M. (2001). Efficacy of directional microphone hearing aids: A meta-analytic perspective. *Journal of the American Academy of Audiology, 12*(4), 202–214.

Amlani, A. M., Rakerd, B., & Punch, J. L. (2006). Speech-clarity judgments of hearing-aid-processed speech in noise: Differing polar patterns and acoustic environments. *International Journal of Audiology, 45*(6), 319–330. https://doi.org/10.1080/14992020600582109

Arbogast, T. L., Mason, C. R., & Kidd, G. (2002). The effect of spatial separation on informational and energetic masking of speech. *The Journal of the Acoustical Society of America, 112*(5), 2086–2098. https://doi.org/10.1121/1.1510141

Black, J. W. (1950). The effect of room characteristics upon vocal intensity and rate. *The Journal of the Acoustical Society of America, 22*(2), 174–176. https://doi.org/10.1121/1.1906585

Bottalico, P., & Astolfi, A. (2012). Investigations into vocal doses and parameters pertaining to primary school teachers in classrooms. *The Journal of the Acoustical Society of America, 131*(4), 2817–2827. https://doi.org/10.1121/1.3689549

Bottalico, P., Graetzer, S., & Hunter, E. J. (2015). Effects of voice style, noise level, and acoustic feedback on objective and subjective voice evaluations. *The Journal of the Acoustical Society of America, 138*(6), EL498–EL503. https://doi.org/10.1121/1.4936643

Boymans, M., & Dreschler, W. A. (2000). Field trials using a digital hearing aid with active noise reduction and dual-microphone directionality [Estudios de campo utilizando un audifono digital con reduccion activa del ruido y micrófono de direccionalidad dual]. *International Journal of Audiology, 39*(5), 260–268. https://doi.org/10.3109/00206090009073090

Bradley, J. S., Sato, H., & Picard, M. (2003). On the importance of early reflections for speech in rooms. *The Journal of the Acoustical Society of America, 113*(6), 3233–3244. https://doi.org/10.1121/1.1570439

Brungart, D. S., Simpson, B. D., Ericson, M. A., & Scott, K. R. (2001). Informational and energetic masking effects in the perception of multiple simultaneous talkers. *The Journal of the Acoustical Society of America, 110*(5), 2527–2538. https://doi.org/10.1121/1.1408946

Crandell, C. C. (1992). Classroom acoustics for hearing-impaired children. *The Journal of the Acoustical Society of America, 92*(4), 2470–2481. https://doi.org/10.1121/1.404481

Crandell, C. C., & Smaldino, J. J. (1996). Speech perception in noise by children for whom English is a second language. *American Journal of Audiology, 5*(3), 47–51. https://doi.org/10.1044/1059-0889.0503.47

Crandell, C. C., & Smaldino, J. J. (2000). Classroom acoustics for children with normal hearing and with hearing impairment. *Language, Speech, and Hearing Services in Schools, 31*(4), 362–370. https://doi.org/10.1044/0161-1461.3104.362

Durrant, J. D., & Lovrinic, J. H. (1984). *Bases of hearing science* (2nd ed.). Baltimore, MD: Williams & Wilkins.

Ferguson, S. H., & Kewley-Port, D. (2007). Talker differences in clear and conversational speech: Acoustic characteristics of vowels. *Journal of Speech, Language, and Hearing Research, 50*(5), 1241–1255. https://doi.org/10.1044/1092-4388(2007/087)

Haas, H. (1972). The influence of a single echo on the audibility of speech. *Journal of the Audio Engineering Society, 20*(2), 146–159.

International Organization for Standardization. (1998). *Acoustics – measurement of sound insulation in buildings and of building elements – part 4: Field measurements of airborne sound insulation between rooms* (ISO Standard No. 140-4). Retrieved from https://www.iso.org/standard/2210.html

International Organization for Standardization. (2009). *Measurement of room acoustic parameters–part 1: Performance spaces* (ISO Standard No. 3382-1). Retrieved from https://www.iso.org/standard/40979.html

Junqua, J. (1993). The Lombard reflex and its role on human listeners and automatic speech recognizers. *The Journal of the Acoustical Society of America, 93*(1), 510–524. https://doi.org/10.1121/1.405631

Keidser, G., O'Brien, A., Hain, J., McLelland, M., & Yeend, I. (2009). The effect of frequency-dependent microphone directionality on horizontal localization performance in hearing-aid users. *International Journal of Audiology, 48,* 789–803. https://doi.org/10.1080/14992020903036357

Lane, H., & Tranel, B. (1971). The Lombard sign and the role of hearing in speech. *Journal of Speech and Hearing Research, 14*(4), 677–709. https://doi.org/10.1044/jshr.1404.677

Litovsky, R. Y., Parkinson, A., & Arcaroli, J. (2009). Spatial hearing and speech intelligibility in bilateral cochlear implant users. *Ear and Hearing, 30*(4), 419–431. https://doi.org/10.1097/aud.0b013e3181a165be

Marshall, L. G. (1995). Speech intelligibility prediction from calculated C50 values. *The Journal of the Acoustical Society of America, 98*(5), 2845–2847. https://doi.org/10.1121/1.413184

Nabelek, A. K., Letowski, T. R., & Tucker, F. M. (1989). Reverberant overlap- and self-masking in consonant identification. *The Journal of the Acoustical Society of America, 86*(4), 1259–1265. https://doi.org/10.1121/1.398740

Nabelek, A. K., & Mason, D. (1981). Effect of noise and reverberation on binaural and monaural word identification by subjects with various audiograms. *Journal of Speech and Hearing Research, 24*(3), 375–383. https://doi.org/10.1044/jshr.2403.375

Nober, L. W., & Nober, E. H. (1975). Auditory discrimination of learning disabled children in quiet and classroom noise. *Journal of Learning Disabilities, 8*(10), 656–659. https://doi.org/10.1177/002221947500801010

Pelegrín-García, D., Smits, B., Brunskog, J., & Jeong, C. (2011). Vocal effort with changing talker-to-listener distance in different acoustic environments. *The Journal of the Acoustical Society of America, 129*(4), 1981–1990. https://doi.org/10.1121/1.3552881

Picheny, M. A., Durlach, N. I., & Braida, L. D. (1985). Speaking clearly for the hard of hearing I. *Journal of Speech and Hearing Research, 28*(1), 96–103. https://doi.org/10.1044/jshr.2801.96

Picheny, M. A., Durlach, N. I., & Braida, L. D. (1986). Speaking clearly for the hard of hearing II. *Journal of Speech and Hearing Research, 29*(4), 434–446. https://doi.org/10.1044/jshr.2904.434

Rakerd, B., Aaronson, N. L., & Hartmann, W. M. (2006). Release from speech-on-speech masking by adding a delayed masker at a different location. *The Journal of the Acoustical Society of America, 119*(3), 1597–1605. https://doi.org/10.1121/1.2161438

Roberts, R. A., Koehnke, J., & Besing, J. (2003). Effects of noise and reverberation on the precedence effect in listeners with normal hearing and impaired hearing. *American Journal of Audiology, 12*(2), 96–105. https://doi.org/10.1044/1059-0889(2003/017)

Schroeder, M. (1965). New method of measuring reverberation time. *The Journal of the Acoustical Society of America, 37,* 409–412.

Yacullo, W., & Hawkins, D. (1987). Speech recognition in noise and reverberation by school-age children. *International Journal of Audiology, 26*(4), 235–246. https://doi.org/10.3109/00206098709081552

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Appendix A. Terminology and Abbreviations

IR, impulse response	The response of a room to an impulsive sound such as a balloon pop. The impulse response captures acoustic characteristics of the room, including its reverberation.
RT60, reverberation time	The time it takes for the sound pressure level of an impulse to decay by 60 dB.
T20, estimated reverberation time	The time it takes for the sound pressure level of an impulse to decay by 60 dB, as extrapolated from a measurement of the time needed for a decay of 20 dB (from −5 to −25 dB).
EDT, early decay time	The time it takes for the sound pressure level of an impulse to decay by 60 dB, as extrapolated from a regression line measurement of the time needed for a decay of 10 dB (from 0 to −10 dB).
C50, clarity	The intensity ratio between early and later occurring portions of the impulse response. High values correspond to better quality of speech received.
D, level difference	The drop in sound power (measured in decibels) that takes place when a barrier is introduced in between a sound source and a receiver.