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Assessing the Acoustic Characteristics of Rooms: A Tutorial with Examples

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Abstract

In both practicing 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.

Keywords

room acoustics; speech acoustics; speech perception; voice use

Assessing Rooms

Speech communication is achieved through a dynamic interaction among three components: (1) the *talker* who creates a spoken message; (2) the *transmission channel* over which that message travels; and (3) the *listener* who interprets the message as received. Physical changes to any 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 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 in reverberation time (Black, 1950). Changing a talker's or a 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, & Braid, 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 sound 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. And 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). And both can affect the accuracy of sound localization by both normally hearing and hearing impaired listeners (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 non-native 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 S12.60–2009/2010) make two recommendations. First, the noise level in an unoccupied classroom should not exceed 35 dBA (see Section III below for a discussion of A-weighted sound level measurement). And, second, the reverberation time should not exceed 0.6 seconds in smaller classrooms and 0.7 seconds 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 sec) 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 Section II) 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). And finally, Pelegrín-García, Smits, Brunskog, & 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 Section I, we describe those rooms and the equipment used when making measurements within them.

In Section II, 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*. 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 Section III, 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 Section IV 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.

I. Rooms and Equipment

Faculty and students in the Department of Communicative Sciences and Disorders (CSD) 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 CSD 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, mid-sized, and large clinical rooms. A description of each of them is given in Table I.

Acoustically Special Research Rooms

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

Equipment and Software

The equipment used when making audio recordings in the example rooms is listed, and briefly described, in Table III. Also described there are software programs/plugins used for acoustical analysis in this study.¹

II. 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: (1) reverberation time, (2) early decay time, and (3) clarity factor.²

Reverberation Time

RT60.—The reverberation time of a room is the time it takes for the sound pressure level (SPL) of an impulsive sound to decay by 60 decibels (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 third-octave-center frequencies or at octave-center frequencies. However, they are sometimes calculated at just 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 seconds; chamber music auditorium, 0.9 – 1.4 seconds; orchestral hall or other large performance space, 1.5 – 2.4 seconds.

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 the present 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.

¹All 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.

²These locations were identified by applying the same decision rules that were described above for the multi-location recording of impulse responses.

In order 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 dB to -25 dB). A signal-to-noise ratio 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 from 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 early decay time (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 dB 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 since 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 milliseconds 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 milliseconds of the direct sound have been shown to integrate with it and to have positive effects on 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-millisecond time window.

Reverberation Measurement Examples

Measurements of room reverberation were obtained from impulse response recordings made in all of the rooms described in Section I. In each room, three arbitrary points were selected, all of them at least one meter away from the room's surrounding surfaces. A total of 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, balloon pops). In the examples reported here, the source was always a balloon pop.⁴ Twelve-inch balloons were pumped until nearly maximum size and then pricked with a pin. The pop impulse was then recorded using the omnidirectional microphone and digital recorder described in Table III (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.

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 show 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 IV.

Some Observations

Using Table IV 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 two second 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 seconds for the small room, 0.22 seconds for the mid-size room, and 0.48 seconds for the large room (Table IV). We sometimes use the large clinical room for classroom instruction. Average T20 and EDT values for this room (0.48 and 0.31 seconds, 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 Hz, 1000 Hz) T20 approached the upper limit of what is recommended for classrooms (0.6 – 0.7 seconds).

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

³Recording 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.

⁴Impulsive sounds (such as a "balloon pop") have a large amplitude and a very short duration. They also have a broad spectrum that is quite regular (Durrant & Lovrinic, 1984).

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.

III. 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 equal energy at all frequencies, or pink noise, which has 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, like the noise from an overhead fan, coming from inside the room itself, others, like 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 (dB), and it should always be measured with a calibrated sound level meter (SLM). SLMs typically have user selectable options for frequency weighting and for 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 one meter 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 (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 one meter 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 milliseconds) or over a longer window (1000 milliseconds). 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.

Signal-to-Noise Ratio

Signal-to-noise ratios (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 1/3-octave frequency band from 125 Hz to 16,000 Hz. The measurements were made from 30-second background noise samples taken with the SLM described in Table III. (The SLM was mounted on a tripod and placed at least one meter from any reflective surfaces in each room). The upper panel of Figure 4 shows the *Leq* dBZ results, the lower panel shows the *Leq* dBA results.

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 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 (like these background noises) that have a relatively low intensity overall.

Table V 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 Hz to 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.

IV. Acoustical Insulation

All three of the clinical rooms described in Section I 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 mid-sized 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 1/3-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 1/3-octave frequency band from 63 Hz 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 – 4,000 Hz). These highlighted frequencies are especially important for speech communication.

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 to 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. While it is unlikely that any of these conversations would be intelligible to a

⁵These locations were identified by applying the same decision rules that were described above for the multi-location recording of impulse responses.

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 mid-size 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 – 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, particularly when the participants have a speech, language, or hearing limitation. This tutorial reviewed key acoustical parameters regarding room reverberation, background noise level, and acoustical insulation, and it outlined best-practice procedures to be followed when measuring those parameters. Measurement data from example rooms used for both clinical and research purposes were presented.

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Terms

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 occurring 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.

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

Figure 2.

Reverberation measurements (T20, EDT, C50) for the three clinical rooms, plotted at octave-band center frequencies. *Small room*: dotted line; *Mid-size room*: dashed line; *Large room*: solid line.

Figure 3. Reverberation measurements (T20, EDT, C50) for the two research rooms, plotted at octave-band center frequencies. *Anechoic chamber*: dashed line; *Reverberant room*: solid line.

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 1/3-octave-band center frequencies. *Small room*: dotted line; *Mid-size room*: dashed line; *Large room*: solid line.

Figure 5.

Insulation measurements (level difference, D) for the clinical rooms, plotted at 1/3-octave-band center frequencies. *Corridor > Small room*: dotted line; *Corridor > Mid-size room*: Short-dashed line; *Corridor > Large room*: Long-dashed line; *Small room > Mid-size room*: Solid line.

Table I.

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.
Mid-size 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.

Table II.

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 like speech intelligibility or musical quality that may be strongly affected by room reverberation.

Table III.

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, www.nti-audio.com).
Omnidirectional microphone	Behringer ECM8000 omnidirectional condenser microphone with flat frequency response. (www.behringer.com).
Digital Recorder	TASCAM DR-40 multi-channel digital recorder. (TASCAM, Japan, www.tascam.com).

Impulse source 12 inch Balloons, balloon pump, and pins.

Loudspeaker Rokit5 G3 studio monitor speaker. (KRK Systems, www.krksys.com).

Audio Editing Software Audacity multi-track audio editor and recorder. (www.audacityteam.org).

Acoustical Analysis Plug-ins AURORA signal processing modules (plugins) for Audacity (and for Adobe Audition). (Aurora Plugins, www.aurora-plugins.forumfree.it).

Table IV.

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

Room	T20 (sec)	EDT (sec)	C50 (dB)
Small clinical room	0.18	0.15	19.50
Mid-size 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

Table V.

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

Room	Leq dBZ	Leq dBA
Small clinical room	31.1	26.8
Mid-size 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

Table VI.

Acoustical insulation level difference, D; reported values are averages over all frequency bands measured.

Noise Source Room (L1)	Receiving Room (L2)	L1-L2 = D (dB)
Corridor	Small clinical room	25.2
Corridor	Mid-size clinical room	25.8
Corridor	Large clinical room	22.4
Small clinical room	Mid-size clinical room	33.9