Title

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CENTER FOR THE BUILT ENVIRONMENT

ABSTRACT

Small conference rooms are often used for either face-to-face communication or for virtual meetings involving an electroacoustical link between a talker and a listener. The intelligibility of speech in such environments depends on a number of factors, one of which is the nature of the reverberant sound within the space. Treating such a room with sound-absorbing materials helps reduce the so-called "cognitive load" for people who are spaced some distance away from a talker or who are listening to monaural speech reproduced by a loudspeaker. This study describes an acoustical retrofit of a small conference room to attain the reverberation time criterion found in LEED version 4.1 ID+C. Several mathematical models were used to predict the reverberation time before and after adding sound-absorbing treatment. In addition, measurements were conducted to quantify the before and after room reverberation characteristics. We found that speech was always intelligible both before and after the retrofit; however, one's cognitive load is noticeably reduced when listening to speech after installation of the treatment.

BACKGROUND

Good quality room acoustics are crucial to successful speech communication in conference rooms and teaching spaces. In conference rooms, an excessively reverberant acoustical environment affects speech intelligibility for people listening to a talker standing at a lectern some distance away.

Reverberant rooms are especially problematic when using tele-communication links. One party who listens to a monaural speech signal reproduced from the transmitting end of the link is unable to benefit from a special listening skill innate to organisms having two ears (called binaural hearing). For humans using both ears simultaneously, binaural hearing enables one to better discern speech from a live talker in the presence of noise and reverberation. This skill is particularly useful in the midst of a party when one is listening binaurally to a nearby individual who is speaking directly to the listener while others talk in the background. In short, binaural hearing represents a kind of noise suppression capability that is partially disabled when one listens to the monaural reproduction of speech in a reverberant room.

When encountering poor listening conditions in classrooms, students have difficulty understanding teachers, concentrating on tasks, and paying attention. This is especially true for students afflicted by a loss of hearing acuity. It is estimated that eight-to-ten percent of students experience learning problems aggravated by inadequate room acoustics. Excessive reverberation or echoes also impact teachers who must exert greater vocal effort to convey information orally across a separation distance of several meters.

Despite the proven need for satisfactory speech communication in conference rooms and teaching spaces, many facility owners and architects neglect this aspect, either due to lack of knowledge or out of a concern for cost — nevertheless, it remains the architect's responsibility to satisfy industry standards for room acoustics.

Speech Intelligibility

Several trade associations and non-governmental code bodies have attempted to establish criteria for speech intelligibility within spaces where oral communication is important (e.g., lecture rooms and conference rooms).

Quantifying speech intelligibility can involve some arcane analyses (see Appendix 3). As a matter of simplicity, speech intelligibility can be reasonably inferred by using an *indirect* figure-of-merit to characterize the acoustical quality of a small conference room.

The indirect figure-of-merit for speech intelligibility is called *reverberation time* (abbreviated RT60). Appendix 1 includes a technical description of RT60.[1]

Specifying RT60 Criteria

In terms of building specifications, the acoustical criteria are subdivided into two categories:

- **Performance**-based (measured reverberation time)
- **Prescriptive** (sound-absorbing treatment area used in an enclosed room, A)

Prescriptive-based criteria are usually expressed as the quantity of sound-absorptive treatment installed on the walls and ceiling of an enclosed room.

For a given room, one can convert between these two categories using the mathematical equivalency:

$$\mathrm{RT60} \propto 1/A$$

Thus, the reverberation time in a room is proportional to the inverse of the quantity of absorptive material (more treatment results in shorter reverberation times). We refer to this equivalency as the Sabine equation named after the first scientist to discover the relationship 125 years ago.

The historical assumption has been that these two categories (performance versus prescriptive) have equivalent impact on the speech intelligibility within an enclosed space.

For small rooms having uniformly-distributed absorptive treatments, the Sabine equation is a satisfactory model; however, with non-uniform (or non-existent) treatments, the results calculated from the Sabine equation ‘*ain’t necessarily so*.’

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[1] Reverberation time (RT60) is specified in LEED documents that address room acoustics. With respect to a typical room having a hard-finish floor, this RT60 inference presumes that sound-absorptive treatments are properly distributed among the five available surfaces.

Industry Standards

LEED version 4.1 ID+C calls for reverberation time to be less than 0.6 seconds in a conference/teleconference room. This value of RT60 pertains to the three most important octave frequency bands comprising the spectrum of speech sounds, *viz*. 500, 1000, and 2000 hertz (abbreviated Hz).

Although not made explicit, the recommended RT60 value is an optimal design target — much shorter RT60 values do not necessarily lead to an improvement in speech communication (see Footnote 2 on Page 7).

OBJECTIVE

The primary purpose of this intervention study is to document the effectiveness of an acoustical retrofit intended to improve the speech intelligibility in a small conference/teleconference space called Room 390 in Bauer Wurster Hall at the University of California, Berkeley. The study includes both pre- and post-retrofit acoustical analyses to quantify the change in the room's performance.

As part of this study, we also wanted to better understand the effect of room acoustics on the subjective intelligibility of speech. In this regard, our efforts went beyond simply meeting the numerical guidelines for rooms treated with sound absorbing materials — we also explored the subjective acoustical benefits perceived by listeners.

The results from our research could be directly applied to small-sized listening environments such as classrooms and lecture rooms.

RESEARCH APPROACH

Predictive Acoustical Models

As part of this study, we created mathematical models of the surfaces in Room 390. The surfaces could then be mathematically altered among the various untreated and treated conditions.

Contemporary modeling software for room acoustics involves certain assumptions about the behavior of sound decaying within an enclosed space. These assumptions can lead to deviations between the predicted and measured reverberation decay, especially when the room acoustical treatments are non-uniform.

Various room acoustical models range from the simple Sabine relationship to very complex computer-based software (see Appendix 3). Our goal was to find a model that would most closely correlate the measured RT60 with the predicted RT60 in a small room.

The three predictive models are:

a) Sabine equation

b) ODEON Room Acoustics Software

c) Enhanced Acoustic Simulator for Engineers (EASE) computer model

Objective Acoustical Testing

Figure 1 illustrates Room 390 as initially configured in 2018. We conducted acoustical measurements in a) the untreated room, b) the same room after the ceiling treatment was installed and, c) the room with ceiling plus wall treatments installed.

Figure 1: Room 390 as found in 2018. The walls are all hard surfaces and the ceiling is exposed concrete. The room has an RT60 of 0.82 seconds in the speech frequency range, thereby exceeding the target value of 0.6 seconds. The reverberant characteristics of Room 390 make it challenging to understand speech, especially during teleconferences.

Sketches of the room treatments are shown in Figure 2, below.

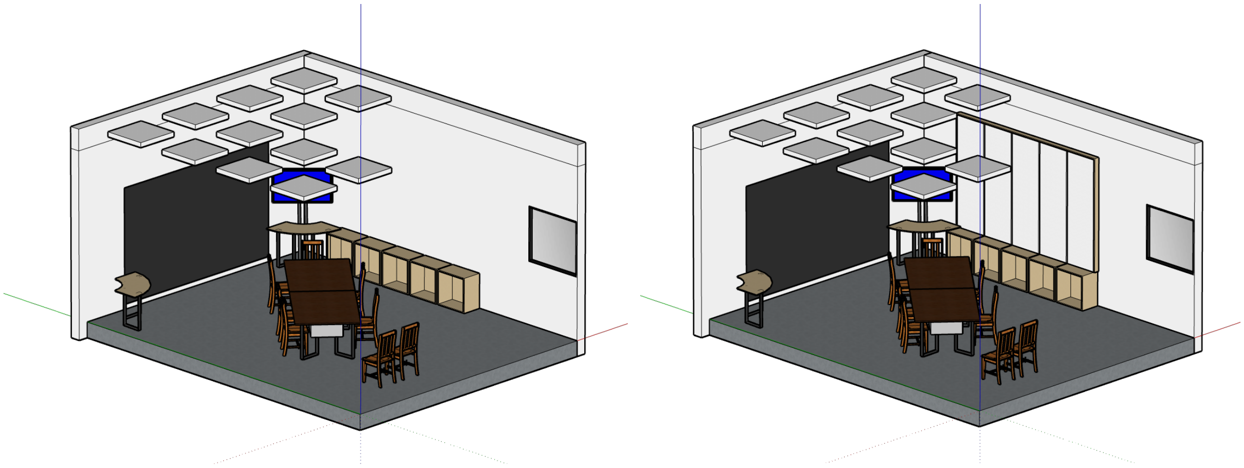

Figure 2: Axonometric sketches of Room 390 looking towards the northeast corner. In the left image, the gray squares represent the new acoustical treatment to be installed within the ceiling coffers. In the right image, further acoustical treatment in the shape of a large rectangle is shown along the east wall. In addition, a narrow, full-height acoustical panel was installed on the west wall near the northwest corner (not shown). The ceiling treatment comprises suspended Armstrong World Industries Capz™ panels. The wall panels are Armstrong TECTUM® backed by a core of 50 millimeter-thick glass fiber. These new treatments are intended to satisfy the RT60 specification in LEED version 4.1 ID+C.

Two of the three new treatment areas are shown in Figure 3, below.

Figure 3: Views of completed ceiling treatment (left) and installation of the treatment along the east wall (right).

Both the objective *reverberation time* and subjective speech intelligibility were assessed for all three treatment conditions. Details of the objective testing protocol are described in Appendix 1.

Subjective Speech Assessment

Speech intelligibility is affected by many factors, several of which pertain to room acoustics. We wanted to assess all these acoustical factors so, in addition to objectively measuring reverberation time, we also evaluated the subjective effects of various room acoustical treatments.

As part of this effort, we placed a manikin head at the listener’s position (see Figure 4 on the following page). The head was equipped with binaural microphones, each of which was fitted to one of the manikin’s “ears”.

This binaural experiment involved capturing a “dry” (anechoic) speech signal as reproduced by a sound source in the room. The intent was to better assess the potential degradation of speech intelligibility in the room relative to the reference anechoic speech recording.

Listening to binaural recordings of either live or reproduced speech gives one a convincing impression of being virtually present in the room.

Using stereophonic headphones to reproduce a binaural recording helps identify the intelligence in speech sounds even when their consonants are somewhat “blurred” by acoustical reflections.[2] In this regard, an individual can use his or her innate binaural hearing skills to help understand speech in unfavorable acoustical conditions.[3]

The manikin head was located about two meters from the sound source; thus, the binaural microphones picked up the speech signal *plus* the reverberant sound from room reflections.

Photographs of the instrumentation used during our onsite evaluation of Room 390 are shown in Figure 4, below.

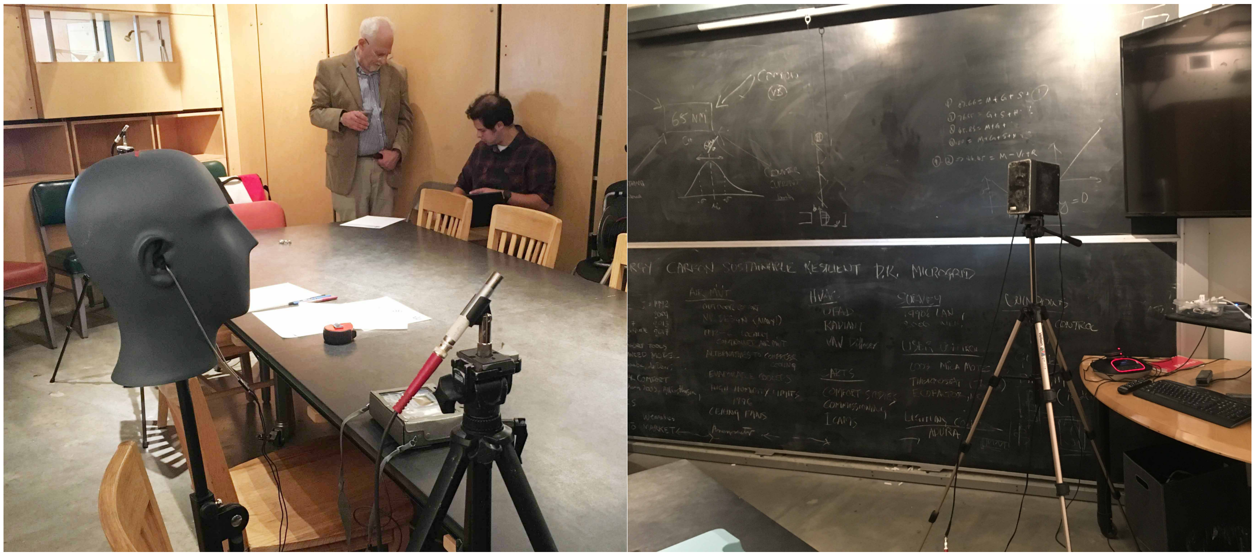

Figure 4: In the photograph on the left, the dark blue manikin head is seen fitted with a pair of microphones (one per ear) to obtain binaural recordings of speech signals reproduced by a sound source. The small silver cylindrical device seen near the manikin is one of three microphones used for measuring the decay of random noise in Room 390. The right photo depicts the small (black-colored) sound source located at the northeast corner of the room (i.e., to the right side of the manikin). This is one of two sound source locations intended to simulate a teacher talking to seated students. The size and directional characteristics of the sound source closely resemble those of a human talker.

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[2] Some reverberation is still desirable for good hearing conditions in a small room fitted with an optimal amount of properly-distributed, sound-absorptive material. Conversely, in a highly-absorptive (“anechoic”) room, the directional nature of a human talker would render speech very difficult to understand if the front of a talker’s head were aimed away from a straight path between the talker and the listener. Blind individuals, for example, need some local sound reflections to help guide their way through a room and would be flummoxed if they encountered an anechoic acoustical environment. With respect to sound-absorbing treatments in a small room, more is not necessarily better.

[3] The binaural recording/listening technique is superior to a typical teleconference room acoustical environment where only a single microphone is used to record speech sounds in combination with some reverberant “blurring” of consonants. At the other end of the communication link, the single (monophonic or monaural) microphone signal is reproduced by a single sound source, thereby partially degrading one’s innate binaural hearing skills.

By listening with stereo headphones, the reproduced anechoic speech signal could be subjectively compared under several conditions:

a) the original anechoic speech signal itself (i.e., without influence from the room)

b) the reproduced speech signal in the untreated room as recorded by binaural microphones fitted to the manikin's "ears"

c) same as b) except with only the ceiling acoustical treatment added

d) same as c) with ceiling *plus* wall acoustical treatments added

This test protocol enabled us to subjectively assess the blurring of speech consonants caused by the various reverberant characteristics of the room itself.[4]

This subjective speech experiment was quite revealing with respect to one's impression about the relative differences in speech clarity among the three categories of room treatments.

For all three treatment categories, the reproduced binaural speech signal was *completely intelligible* to the five acoustical consultants onsite — i.e., all the spoken words could be understood, especially with some assistance from contextual clues in the English language lecture.[5]

For the untreated room condition, however, the listeners had to concentrate more intently to understand information contained in the binaurally-recorded speech signal.[6] This increased effort was due to some blurring of speech consonants; thus, the listeners needed to depend more heavily on contextual clues in the spoken sentences. This additional interpretive task is more burdensome for a non-native speaker of English and/or one who suffers from a loss of hearing acuity.

Even after treating both the ceiling and walls, there was still a noticeable blurring of speech consonants when comparing the anechoic source signal to the same signal binaurally recorded in the treated room. When comparing c) the *treated ceiling only* condition to d) the *ceiling + wall* treatment condition, there was a just-noticeable improvement in clarity between the two binaural recordings.

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[4] It is this "blurring" or "smearing" of consonants that increases the so-called "cognitive load" for listeners who are spaced some distance away from a talker (see Footnote 6, below). One can have too much treatment — the amount of acoustical treatment is optimal when the cognitive load approaches a minimum.

[5] The five consultants included two whose native language is Spanish.

[6] In the field of haptics, this sensory effort is called cognitive load[ing]. The cognitive load factor is further increased for a non-native speaker of the language and/or one with a loss of hearing acuity. Refer to the book, *Cognitive Load Theory* by Sweller, J. et al., ISBN 978-1-4419-8125-7, DOI 10.1007/978-1-4419-8126-4, Springer, 2011. For information regarding cognitive abilities, refer to the paper, "Noise pollution and human cognition: An updated meta-analysis of recent evidence", by Thompson, R., et al. published in *Environment International*, 158 (2022) 106905, January 2022.

Figure 5: Photograph showing the general layout of measuring instrumentation in Room 390. A sound level meter is installed on a tripod close to the left side of the blackboard. The device on the silver tripod to the right is the transmitter portion of a specialized RASTI system (see Appendix 2). The companion receiver portion of the RASTI system is on the table in the foreground. The dark blue manikin head can be seen next to the right side of the table. Additional instrumentation can also be seen in Figure 4 on page 7.

Technique Used to Measure Reverberation Time

Several techniques can be used to measure reverberation time in an enclosed space. For this research project, we decided to use the generally accepted interrupted noise method described in Section 5.2 of an international standard entitled, ISO 3382, *Acoustics - Measurement of the reverberation time of rooms with reference to other acoustical parameters*.

We used timed bursts of pre-recorded random noise emitted from a sound source (a loudspeaker) to repeatedly excite the room while a data recorder captured the resulting decays of acoustical signals from three measurement microphones, each of which was spaced apart from the others by two meters.

Assuming the rms sound pressure within the room is expressed on a logarithmic scale, its exponential decay characteristic will trace a straight, sloping line as first discovered by Sabine.[7]

The exponential decay of sound in an enclosed space can be quantified as a constant *nn* decibels per second. The time for rms sound pressure to decay to 0.1 percent of its initial value (a reduction of 60 decibels) is arbitrarily defined as the reverberation time (RT60). Reverberation time is expressed in seconds.

If the rms sound pressure decays to 0.1 percent of its initial value in one second, the RT60 would be one second (i.e., a decay rate of 60 decibels per second). Similarly, a decay rate of 120 decibels per second would be equivalent to an RT60 of 0.5 seconds, etc.

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[7] See Appendix 1 for a glossary of terms.

Mathematical Models

We used several mathematical models to help predict the reverberation time for the room treatment conditions in Room 390.

The three treatment conditions are:

1. As-found
2. Treated ceiling only
3. Treated ceiling *plus* walls

For all three treatment conditions, we calculated the reverberation time at 1000 hertz using three mathematical models:

1. Sabine equation
2. ODEON Room Acoustics Software
3. Enhanced Acoustic Simulator for Engineers ("EASE") — a computer model

In the 1000-hertz octave band, for example, the nine calculated results among the three treatment conditions and three mathematical models are shown in Table 1 below. For comparison, the average *measured* RT60 data obtained from all sound source and microphone locations are also shown.

Table 1. Reverberation Time (RT60) in 1000-hertz Octave Band, seconds

	Sabine Model	ODEON Model	EASE Model	Measured (1000 Hz)	Criterion
As found (no treatment)	2.5	1.9	1.7	0.82	**0.6**
Treated ceiling only	0.6	0.7	0.8	0.53	
Treated ceiling and walls	0.4	0.5	0.6	0.41	

For the special case involving the untreated condition, it is obvious that all three prediction models seriously overestimate the reverberation time. The likely cause of this discrepancy is extra energy losses as the traveling sound waves diffract around sharp corners. The prediction models do not sufficiently account for these extra losses, particularly for small rooms.

When treatment is added to a few room surfaces, the prediction models come much closer to the measured values of RT60. It is possible that the decay of sound in the room depends on a mix of absorption and diffraction mechanisms; hence, as sound absorption begins to dominate the decay process relative to [the constant] diffraction process, the models continue to overpredict the RT60 since the diffraction mechanism is still not taken into account. Once the ceiling and walls are treated, the prediction error from the computer models is less significant as the contribution from diffraction becomes relatively small.

For the two treated room conditions, the Sabine model is the most reliable predictor of reverberation time. The ODEON model is second-best. The EASE model is last.

RESULTS

The chart in Figure 6 depicts a reverberation decay as measured in Room 390. A sound source in the room generated random noise at 77 decibels; the signal was then muted 0.2 seconds after an arbitrary "zero time" point at the chart origin.

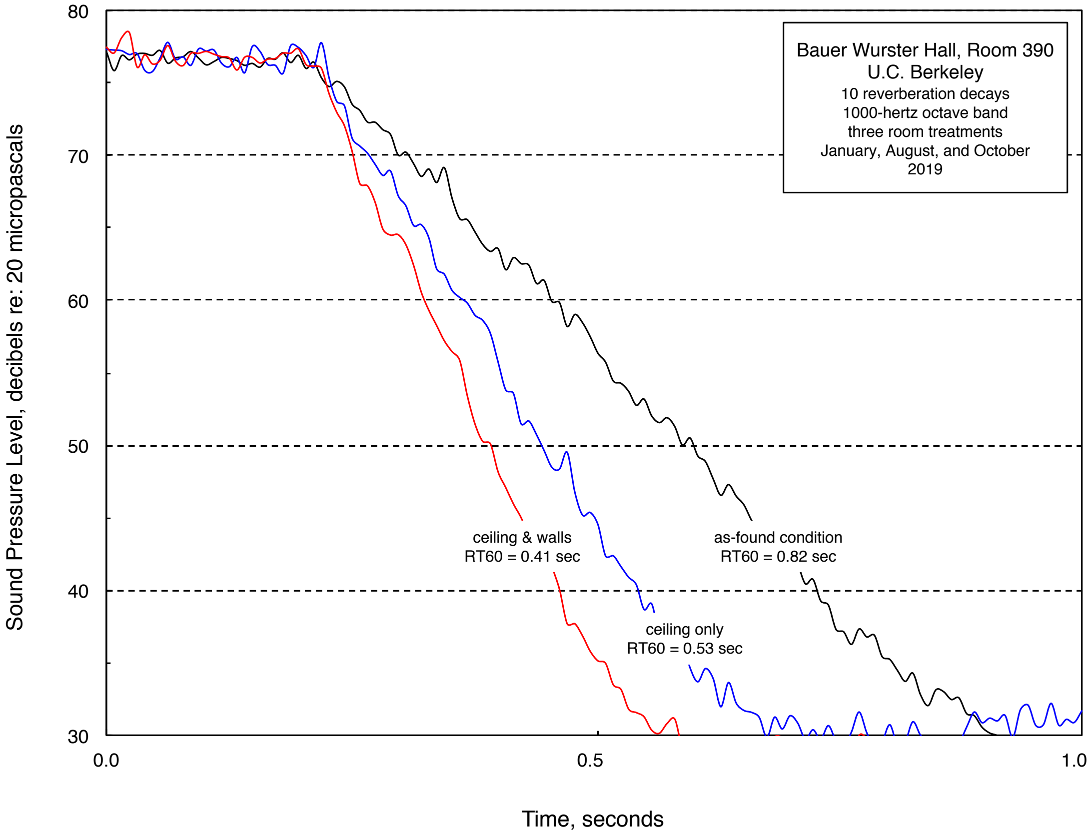

Figure 6: Decay of rms sound pressure for three different room treatment conditions. The vertical (ordinate) axis is the measured sound pressure level expressed in decibels. The horizontal (abscissa) time axis is shown in seconds.

The decay of sound pressure approximates a straight line, implying that the room conforms to the exponential Sabine model within this 1000-hertz octave band. Data are shown for a single sound source and single microphone location — the ensemble means of all RT60 data differ slightly from those shown here.

When treatments were applied to both the ceiling and walls, the resulting RT60 satisfied the recommended criterion in the three discrete octave bands pertaining to speech communication (500, 1000, and 2000 hertz).

Summary of Findings

1. The untreated condition in Room 390 exhibited a higher-than-recommended *reverberation tim*e (RT60) for a conference room. The RT60 was 0.82 seconds, exceeding the maximum LEED criterion of 0.6 seconds.
2. Speech was always intelligible to the five acoustical consultants who listened to all three test conditions (untreated, ceiling only, and ceiling *plus* walls).
3. For the untreated condition, a typical listener needed to concentrate more intently when perceiving speech information affected by "blurring" of speech consonants, thereby imposing a greater cognitive load on the listener.
4. For the untreated room condition, the listening quality was judged "poor" — after treatment, the quality was judged "satisfactory".

CONCLUSIONS

- Speech intelligibility in small teleconference rooms can be adequately quantified by using the reverberation time (RT60)
- Attaining an RT60 of 0.4 to 0.6 seconds in small rooms helps reduce the "blurring" or "smearing" of consonants
- It is important that the sound-absorptive treatments be properly distributed among the five surfaces of the room to help prevent deleterious acoustical "fluttering" between parallel reflective surfaces
- Architects responsible for meeting an RT60 criterion in critical rooms need expertise to perform reliable calculations

ACKNOWLEDGMENTS

The authors wish to thank several contributors who made this work possible. William Frantz, Senior Principal Scientist at Armstrong World Industries, provided the installation details and procured the materials that were donated by Armstrong. Semar Prom, Director of Fabrication Services for UC Berkeley's College of Environmental Design, supervised the installation. Gail Brager and David Lehrer of CBE assisted with identifying the test space and advising the research team.

Finally, many thanks to the Salter technical staff who donated their professional expertise to make this project a success.

APPENDIX 1: TECHNICAL FUNDAMENTALS

Reverberation Time (RT60)

The decay of sound in a small enclosed space is dependent on the amount of acoustical energy lost by dozens of reflections as the traveling sound waves encounter room boundaries. Assuming the interior surfaces of the room are treated with a substantial amount of sound-absorptive material, then the decay of sound will be rapid and vice versa. Statistically speaking, the decay of sound in a room occurs as an exponential function of time.

Sound Pressure Level (SPL)

In acoustics, it is common practice to characterize the root-mean-square (rms) sound pressure as a logarithmic quantity expressed in decibels (a unitless logarithmic ratio of two sound pressures). For people working in acoustics, the use of decibels is very convenient because simple integers can be used to represent rms sound pressures ranging over many orders of magnitude.

The explicit term, *sound pressure level* (SPL) is defined as the ratio of sound pressure referenced to the approximate human threshold of hearing in the speech frequency range. With respect to the term, sound pressure level, this ratio is always expressed in decibels.

In accordance with international standards, the reference threshold sound pressure is defined to be 20 micropascals. Thus, a sound pressure of 20 micropascals is equivalent to an SPL of zero decibels, a sound pressure of 200 micropascals is equivalent to an SPL of 20 decibels, a sound pressure of 2000 micropascals is equivalent to an SPL of 40 decibels, and one million micropascals (one pascal) is equivalent to an SPL of 94 decibels.[8]

Using decibel notation is convenient because the enormous numerical range of sound pressures one encounters in the environment can be compressed into a series of integers extending over a three-digit range (e.g., zero to 120 decibels SPL).

Loudness

The term used to express the human sensation of sound intensity is *loudness*. Loudness is generally proportional to sound pressure but the relationship between the human sensation and the physical sound pressure is strongly dependent on the acoustical frequency range.

For sound in the middle portion of the speech frequency range, an increase (or decrease) of 10 decibels in sound pressure is perceived as a doubling (or halving) of loudness. For example, the median SPL of male normal human speech effort at one meter is about 57 decibels. If one speaks in a raised tone of voice, the median SPL would increase to 65 decibels at the same distance — nearly a doubling of loudness. For a “loud” speech effort, the median SPL would be 76 decibels— a quadrupling of loudness.

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[8] Typically, sound pressures represent a very small perturbation around static atmospheric pressure. By international agreement, static atmospheric pressure is defined to be 101,325 pascals.

APPENDIX 2: INSTRUMENTATION

The instrumentation used for recording and analyzing the data are listed below:

- Rion NL-52 class 1 integrating-averaging sound level meter
- Rion Type DA-20 digital data recorders
- Brüel & Kjær Type 4176 class 1 one-half inch diameter, pre-polarized microphones
- Brüel & Kjær Type 2671 microphone pre-amplifiers
- Brüel & Kjær Type 4101 binaural microphone system
- Sennheiser MZK 2002 manikin head
- ADS 2002 self-powered loudspeaker system equipped with 75-mm diameter “woofer” and 25-mm “tweeter”
- Brüel & Kjær Types 4419 receiver and 4225 speech transmitter that together form a complete RASTI system
- Brüel & Kjær Type 2133 one-third-octave-band real-time analyzer

APPENDIX 3: ALTERNATIVE DESCRIPTORS FOR SPEECH INTELLIGIBILITY

Over the past 70 years, the acoustical community has considered a number of descriptors for directly assessing speech intelligibility including:

- the legacy *articulation index* (AI)
- *percentage articulation loss of consonants* ($\%AL_{cons}$)
- *speech transmission index* (STI)
- the legacy *rapid assessment of speech transmission index* (RASTI)
- *speech intelligibility index* (SII)
- *speech transmission index for public address* (STIPA)
- *common intelligibility scale* (CIS)

These alternative descriptors do not seem to have any better correlation with perceived intelligibility than does reverberation time, particularly for a small classroom treated with properly-distributed sound absorption material.

In summary, reverberation time appears to be a satisfactory figure-of-merit for scaling speech intelligibility in a classroom having properly-distributed treatments.