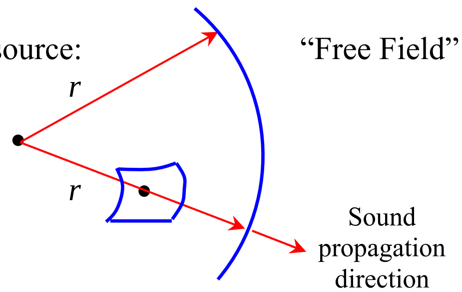
Auditorium & Room Acoustics

Sounds out in the open, distance $r \gg \lambda$ away from sound source:

Sound Intensity $\boxed{I(r) = Power/4\pi r^2}$ sound source:

Intensity, I in *Watts*/m^2 (since *SI* units of Power are *Watts*)



Sound intensity $I(r)$ decreases as $1/r^2$, spreads out radially in all directions from sound source
* Sound Intensity Level, $L_I(r) = 10\log_{10}(I(r)/I_o)$ decreases by 6 *dB* for every doubling of r.
* Ground (grass, weeds, bushes, *etc.*) absorbs sound…
  – sound level $L_I(r)$ falls off faster than $1/r^2$ ($L_I(r)$ falls off more steeply than 6 *dB*)
* Put a reflecting surface behind musicians for focusing sound to audience…

* Confined sound in an enclosure (*e.g.* a room):
  – Get sound reflections off of all walls (just like light bouncing off of mirrors)
  – Angle of incidence = Angle of reflection
  – Law of reflection (light and/or sound) arises from energy/momentum conservation at wall/mirror!)

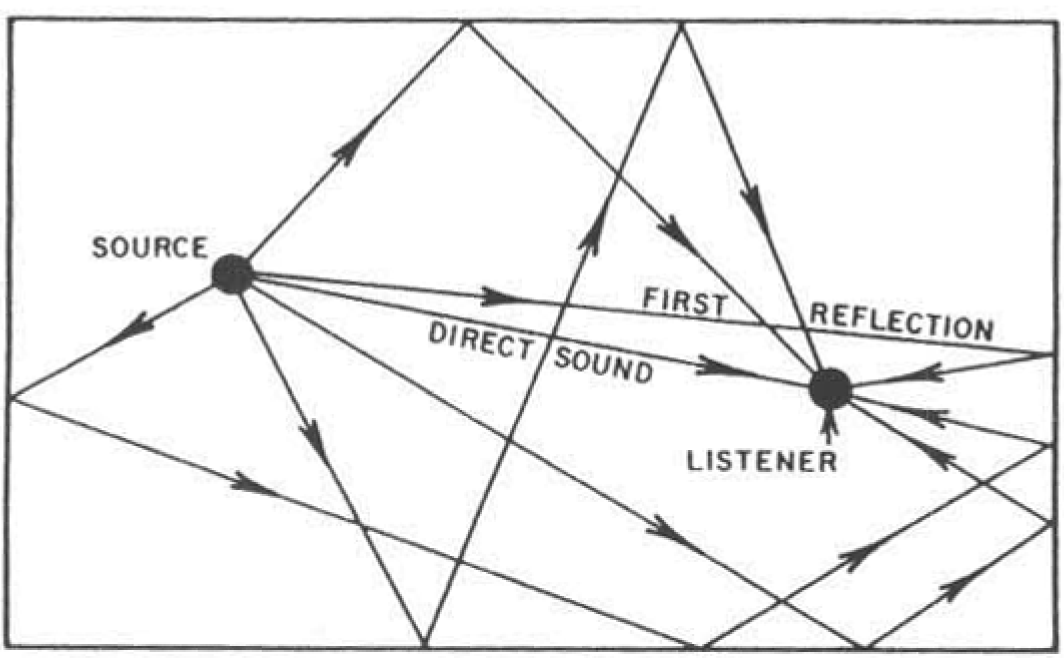


FIG. 1. Multiple reflections from the walls of a room of a single impulse produced by a sound source.

Reverberation/Reverberant Sound:

Totality of sound = direct sound, multiple echoes and "clutter"

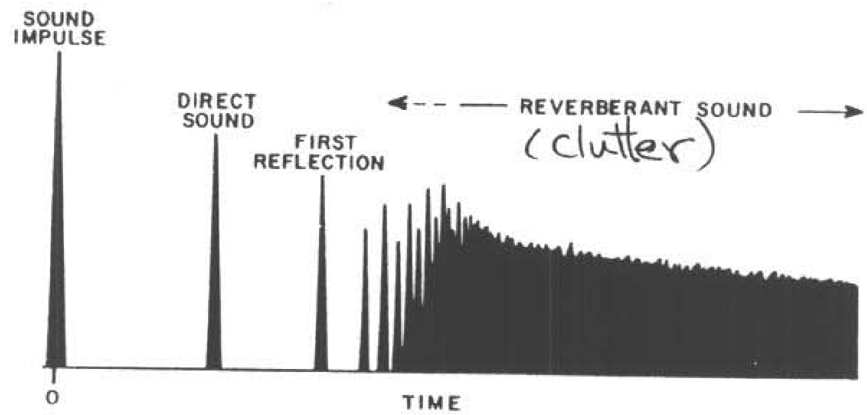


FIG. 2. Multiple reflections of a sound impulse as heard by a listener.

Reverberation Time, T = time for sound to decay to 10^{-6} (one millionth) of its original intensity, I.
Corresponding change in Loudness Level/SPL: $\Delta L = 10\log_{10}\left(I_2/I_1\right) = 10\log_{10}\left(10^{-6}\right) = -60\ dB$.
Hence, reverberation time T is also known as T_{60}.

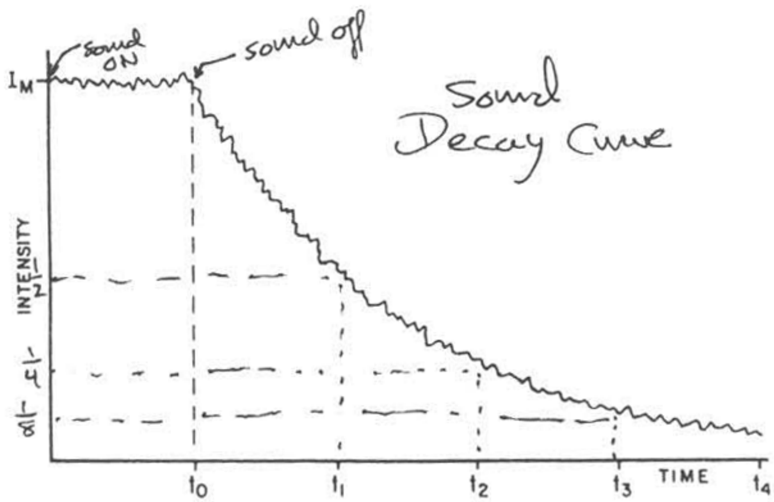


FIG. 3. Decay of reverberant sound in a room.

Reverberation Time, $T \propto$ (= proportional to) room volume, V - *i.e.* $T \propto V$

Reverberation Time, $T \propto 1/Area$ of "hole(s)" in room, A $\quad T \propto 1/A$

Sabine Equation: $\boxed{T = K\dfrac{V}{A}}$ where K = constant of proportionality $= T\dfrac{A}{V}$

If we know V and A and then measure T, we find K=0.049 *sec/ft* (= universal number!!!)

Sabine Equation

$$\boxed{T = K\frac{V}{A}}\ (seconds)$$

V = Room volume in ft^3 (m^3)

A = "hole" area in ft^2 (m^2)

K = 0.049 in $secs/ft$ (= 0.161 $secs/m$)

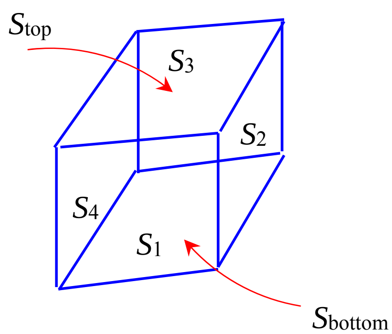
Reverberation Time, $T = T_{60} = 0.049\left(\frac{V\left(ft^3\right)}{A\left(ft^2\right)}\right) = 0.161\left(\frac{V\left(m^3\right)}{A\left(m^2\right)}\right)$ seconds

= time for sound to decay to 10^{-6} of its original intensity.

If the room has NO holes in it, the area A physically represents the effective area of the room that *behaves* as if it were a hole, *due to sound absorption*.

$\boxed{1\ ft^2 = 1 \text{ absorption unit}}$

Suppose a room with volume V has a surface area S made up of same material on all 6 sides:



Total surface area of room S:

$$S = \overbrace{S_1 + S_2 + S_3 + S_4}^{\text{area of sides}} + \overbrace{S_{top} + S_{bottom}}^{\text{area of top and bottom}}$$

$$\boxed{\begin{array}{c} A = aS \\ \\ a \equiv \dfrac{A}{S} \end{array}}$$

$a \equiv$ absorption coefficient, $0 \le a \le 1$

$a = 0 \Rightarrow$ **no** sound absorption (no "hole", *i.e.* $A = 0$)
$a = 1 \Rightarrow$ **total** sound absorption ("hole" = room area, *i.e.* $A = S$!!!)

For a more complicated/realistic room:

$$\boxed{\begin{aligned} A &= A_1 + A_2 + A_3 + A_4 + \ldots A_N = \sum_{n=1}^{N} A_n \\ &= a_1 S_1 + a_2 S_2 + a_3 S_3 + a_4 S_4 + \ldots a_N S_N = \sum_{n=1}^{N} a_n S_n \end{aligned}}$$ for N objects (surfaces) in room.

The "Optimum" Reverberation Time:
* If reverberation time is too short, room sounds "dead"
* If reverberation time is too long, room sounds muddled/obscured

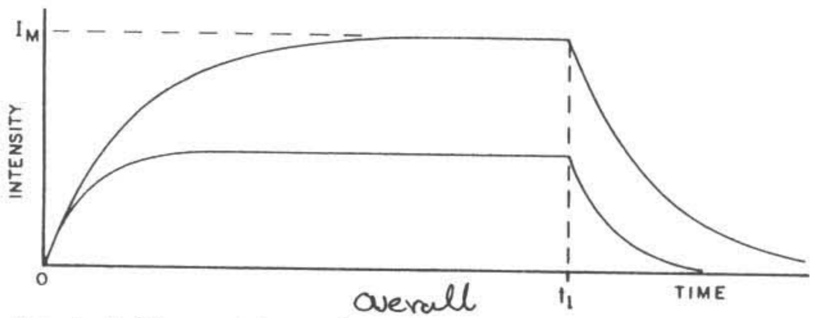


FIG. 4. Buildup and decay of overall sound intensity in an auditorium when a source of steady sound is present.

Max Intensity $I_{\max} = P/A$, where P (*Watts*) = acoustic power of sound source and A = total absorption (measured in square meters).

Suppose *e.g.* we input P = 1 *Watt* of acoustic power into a room, allow time for the sound to build up to a steady level, and then use an *SPL* meter, *i.e.* a device to measure the max *SPL* (in *dB*) in the room. Suppose we find (*i.e.* we measure) max *dB* = 99.54.

We then invert the *dB* formula: max $dB = 10\log_{10}(I_{\max}/I_o)$ to obtain: $I_{\max} = 10^{9.954}\, I_o = 0.009\ W/m^2$.

<u>Thus</u>:

$$\boxed{\begin{aligned} A &= \frac{P}{I_{\max}} = \frac{1\ Watt}{0.009\ W/m^2} \\ &= \quad 110 \text{ square meters} \\ &\simeq 1200 \text{ square ft } (= 1200 \text{ absorption units}) \end{aligned}}$$

The "Optimum" Reverberation Time:

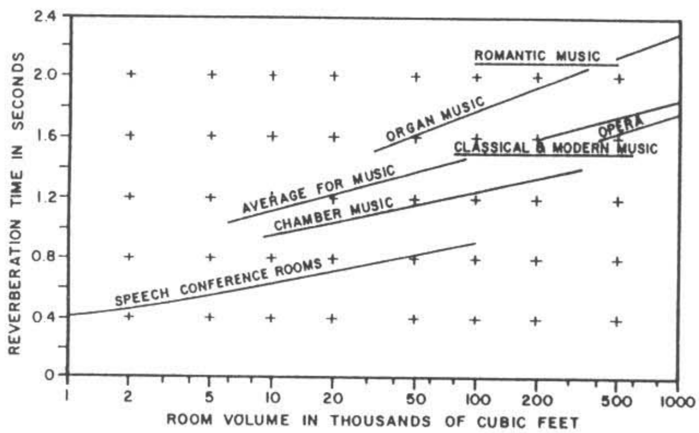


FIG. 5. Optimum reverberation time for auditoriums of various sizes and functions at a frequency of 500 hertz.

Optimum reverberation time is ***subjective***!

- tempo dependent
- sound level dependent
- complexity dependent
- frequency dependent

Calculation of Reverberation Time

TABLE I

Absorption coefficients of some building materials

	FREQUENCY—HERTZ					
	125	250	500	1000	2000	4000
Marble or glazed tile	.01	.01	.01	.01	.02	.02
Concrete, unpainted	.01	.01	.01	.02	.02	.03
Asphalt tile on concrete	.02	.03	.03	.03	.03	.02
Heavy carpets on concrete	.02	.06	.14	.37	.60	.65
Heavy carpets on felt	.08	.27	.39	.34	.48	.63
Plate glass	.18	.06	.04	.03	.02	.02
Plaster on lath on studs	.30	.15	.10	.05	.04	.05
Acoustical plaster, 1″	.25	.45	.78	.92	.89	.87
Plywood on studs, ¼″	.60	.30	.10	.09	.09	.09
Perforated cane fiber tile, cemented to concrete, ½″ thick	.14	.20	.76	.79	.58	.37
Perforated cane fiber tile, cemented to concrete, 1″ thick	.22	.47	.70	.77	.70	.48
Perforated cane fiber tile, 1″ thick, in metal frame supports	.48	.67	.61	.68	.75	.50

Note frequency the dependence of absorption coefficients for various building materials!

Example Calculation of Reverberation Time, $T=T_{60}$:

From example calculation of T in John Backus' book – "The Acoustical Foundations of Music"

With this information, let us calculate the reverberation time of a hypothetical auditorium, which we will arbitrarily assume to be 100 ft long, 60 ft wide, and 40 ft high. The calculation will be for a frequency of 500 hertz. The walls and ceiling will be assumed to be plaster, with an absorption coefficient of 0.10, and the floor covered with carpet on felt, with an absorption coefficient 0.40, as given in Table I. The total absorption is then calculated from Eq. (7) as follows:

	Area, sq ft		Abs. Coeff.	Abs. Units
Floor	100 × 60	= 6000	0.40	2400
Ceiling	100 × 60	= 6000	0.10	600
Two side walls	2 × 40 × 100	= 8000	0.10	800
Two end walls	2 × 40 × 60	= 4800	0.10	480
		Total absorption		4280 ≈ 4300 units.

The volume of the room is 40 × 60 × 100 = 240,000 cubic feet. The reverberation time will then be

$$T = 0.049 \times \frac{240 \times 10^3}{4300} = 2.7 \text{ sec} \qquad \text{Too long!}$$

If $T \le 1.5$ sec, then:

$$A \ge 0.049 \frac{V}{T} = 0.049 \frac{240 \times 10^3}{1.5} = 7800\ ft^2 (= \text{Abs. units})$$

If 4300 absorption units already present,

Then need to <u>add</u> 7800 – 4300 = 3500 absorption units

1" thick perforated tiles have absorption coefficient, $a = 0.70$ (@ 500 *Hz*)

$A = aS = 3500\ \text{ft}^2$

$$\therefore \quad \boxed{S = \frac{A}{a} = \frac{3500\ ft^2}{0.70} = 5000\ ft^2}$$

$\Rightarrow$ Need 5000 ft^2 of 1" thick perforated tiles.

* Need to be careful here: seats, people, *etc*. are sound absorbing too!!!
* Acoustical properties of an empty auditorium are not the same as when full!!!

TABLE II

Sound absorption by theater seats and audience, in absorption units

	FREQUENCY—HERTZ					
	125	250	500	1000	2000	4000
Wood or metal seats, unoccupied	0.15	0.19	0.22	0.39	0.38	0.30
Cloth-covered upholstered seats, unoccupied	1.4	2.8	4.2	5.0	4.6	4.4
Audience in upholstered seats, per person	2.9	4.3	6.0	7.0	6.9	6.0

n.b. Air absorbs sound somewhat too – *i.e.* air temperature & relative humidity also matter! Taking into account air absorption, the Sabine Equation is modified as:

$$\boxed{T_{60} = 0.049\frac{V\left(ft^3\right)}{A\left(ft^2\right)+mV\left(ft^3\right)} = 0.161\frac{V\left(m^3\right)}{A\left(m^2\right)+m\cdot V\left(m^3\right)}}$$

where m is a temperature, humidity and frequency-dependent parameter, varying from $m \sim 0.01/m$ at 2 KHz to $m \sim 0.1/m$ at 8 KHz for ~ NTP conditions with RH ~ 30-50%.

Nowadays, all of this is done using acoustical computer simulation programs (*e.g.* EASE, LARA), all "tuned" from real measurements. Input all of the gory details of shape, size, and volume of rooms, exact shapes, sizes, and locations of all sound absorbing elements, *etc.* (also frequency dependence – see figure below)!

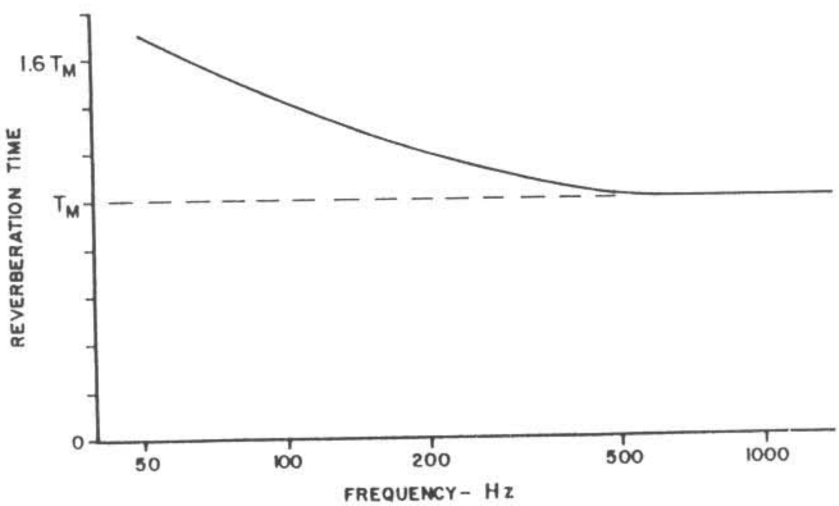


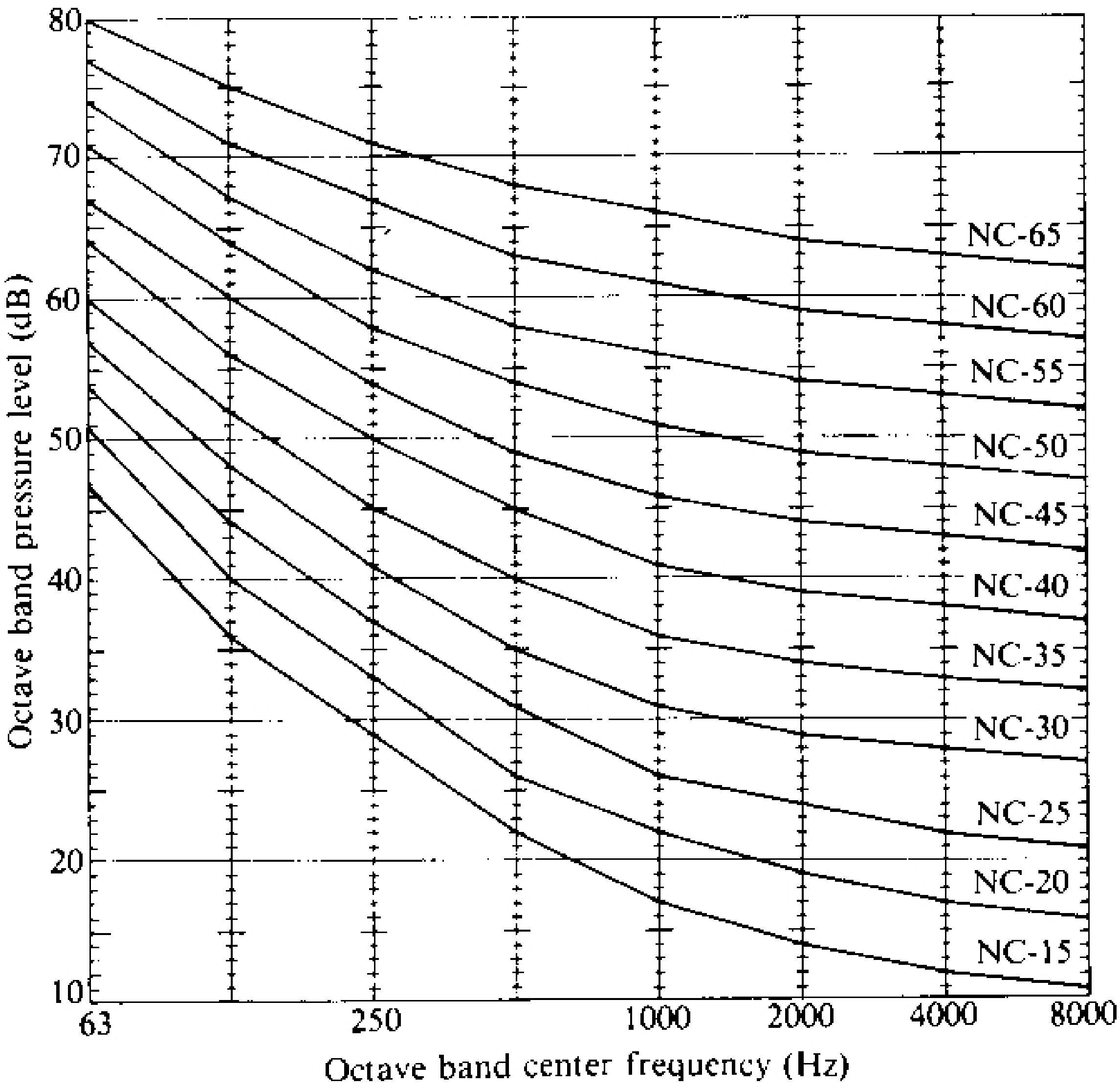
FIG. 6. Recommended variation of reverberation time with frequency.

Again, "Recommended" => subjective decisions were made about this….

Auditorium Background Noise:

Ambient/background noise levels in auditoriums/concert halls – *e.g.* from noise sources such as ventilation systems, nearby traffic, *etc.* needs to be controlled such that it does not detract (or distract) from the audience's enjoyment of the music performance. Poorly designed ventilation systems often generate ~ $1/f$ type noise (or even resonances!) as well as broad-band noise from air flow in ducts and through grilles. Inadequate isolation from corridor noise, noisy and/or squeaky doors, etc. can also contribute unwanted sounds.

Over the years, Noise Criteria (NC) curves have been developed that specify maximum permissible noise levels in octave frequency bands for a particular noise rating. A specification for NC-20 (or below) requires octave-band sound pressure levels *SPL* (in *dB*) at or below the NC-20 curve shown in the figure below (*n.b.* which closely follow the Fletcher-Munson apparent loudness level curves):



A good concert hall should at least meet the NC-20 curve, and preferably the NC-15 curve. For lecture halls and classrooms, the noise levels should be at (or below) the NC-30 curve, or better yet, the NC-25 curve. T_{60} reverberation times should also be < 0.5 seconds in lecture halls and/or classrooms, in order to avoid speech interference problems, particularly in the speech intelligibility range of 500 – 4000 *Hz*. A teacher using a normal voice will produce a sound pressure level of ~ 46 *dB* at the ears of a student 30 *ft* away; the NC-30 curve corresponds to an average background noise level of ~ 36 *dB* in this frequency range, thus a 10 *dB* difference in speech *vs*. noise level, as required for speech intelligibility.

Today, audio/acoustic engineers can harness the power of the computer and use sophisticated computer programs to create accurate 3-D simulations of the acoustical environment/acoustical properties of an arbitrarily-shaped room – be it a concert hall, a theatrical stage, a church, *etc*. Ray-tracing techniques are used to simulate 3-D sound propagation from accurately-modeled sound sources, or even actual recorded sounds, reflecting off of the 3-D surfaces of the room, including frequency-dependent absorption and diffusivity coefficients of these surfaces.

The most common measure of the reverberation time for these programs is T_{30}, (the time for the sound intensity to decay to 1/1000th of its steady-state value – noting that $T_{60} = 2\ T_{30}$) which is calculated from the slope of the curve fit through the simulated ray-tracing generated reverberation time data for the sound intensity level *vs*. time as the sound intensity level decays from −5 *dB* to −35 *dB*. The T_{30} reverberation time obtained in this manner is a more sensitive indicator of the true reverberation properties of a room than that obtained from the Sabine equation, because it takes into account both the absorption <u>and</u> the <u>*diffusivity*</u> of the room surfaces, as well as the detailed specifics of the geometry of the room (limited only by the accuracy input to the 3-D model of the room). In such acoustics ray-tracing programs, one can also investigate T_{30} *e.g*. in different octave bands (*i.e*. as a function of frequency) much more easily than T_{60}, which requires significantly more computation time.

Ray-tracing acoustical simulation software programs can also obtain accurate estimates of the sound pressure levels everywhere in the simulated room (and as a function of frequency) to enable the {as uniform as possible}sound pressure levels independent of the location of a person in the room, thus eliminating and/or minimizing dead spots or hot spots in the room.

Please see/read the P406 Lecture Note hand-out on “EASE Examples” for more details.

Speech intelligibility is another important acoustical attribute of a listening room, particularly for lecture halls, theater- and church-goers – *i.e*. any room or acoustical environment where public speeches are important. One statistic for speech intelligibility is Definition D_{50} , defined as the ratio of the integral of the square of the overpressure within the first 50 *msec* of the initiation of sound associated with *e.g*. a very short sound impulse (< 50 *msec* duration) to that integrated over all time for that same sound, expressed as a percentage:

$$\boxed{D_{50} \equiv 100 \times \left[\frac{\int_{t=0}^{t=50ms} p^2(t)dt}{\int_{t=0}^{t=\infty} p^2(t)dt} \right] \ (\%)}$$

Thus, D_{50} is a measure of the per-cent total sound energy arriving within 50 *msec* after an initial pulse of sound. If most of the energy of the sound impulse is within this 50 *msec* window, then it will be {much} easier for people in this room to understand speech than if *e.g.* there are many echoes over a longer time for people to try to comprehend. This sound parameter can only be determined (reasonably easily) with ray-tracing acoustical simulation software, for a realistic room. A "good" listening room from a speech-intelligibility perspective has $D_{50} > 50\%$.

A related statistic is ***speech*** Clarity, C_{50} defined as:

$$\boxed{C_{50} \equiv 10\log_{10}\left(\frac{\int_{t=0}^{t=50ms} p^2(t)dt}{\int_{t=0}^{t=\infty} p^2(t)dt - \int_{t=0}^{t=50ms} p^2(t)dt}\right) \quad (dB)}$$

For > 80% syllable intelligibility, a clarity of $C_{50} > -2dB$ is required, and is considered the minimum admissible limit for good speech intelligibility.

Music Clarity, C_{80} defined as:

$$\boxed{C_{80} \equiv 10\log_{10}\left(\frac{\int_{t=0}^{t=80ms} p^2(t)dt}{\int_{t=0}^{t=\infty} p^2(t)dt - \int_{t=0}^{t=80ms} p^2(t)dt}\right) \quad (dB)}$$

Another statistic is the Center Time $\langle t_s \rangle$, the mean/average time associated with a sound impulse, defined as:

$$\langle t_s \rangle \equiv \left[\frac{\int_{t=0}^{t=\infty} t \cdot p^2(t)dt}{\int_{t=0}^{t=\infty} p^2(t)dt}\right]$$

The {subjective} mean/average syllable intelligibility $\langle V_s \rangle$ is related to the center time $\langle t_s \rangle$ by:

$$\boxed{\langle V_s \rangle \equiv 96 \cdot \left(1 - 10^{-5} \langle t_s \rangle^2\right) \quad (\%)} \quad \textit{n.b.}\ \langle t_s \rangle \text{ in } \textit{msec} \text{ time units, here.}$$

For mean/average syllable intelligibility $\langle V_s \rangle > 80\%$, a center time of $\langle t_s \rangle \le 130$ *msec* is required. If the center time is measured *vs*. octave bands center frequencies, then for speech one wants $\langle t_s(f_{ctr}) \rangle \le 60 - 80$ *msec* for the 4 octave band centers at 500 *Hz*, 1000 *Hz*, 2000 *Hz* & 4000 *Hz*.

In the ***reverberant*** portion of the sound field of a large listening room/auditorium (*i.e.* far enough away from a sound source, *e.g.* located at the front of the large room/auditorium – please see/read UIUC Physics 406 Lecture Notes 10 *p*. 1-3 for more details), the center time $\langle t_s \rangle$ associated with short impulsive sounds is related to the reverberation time T_{60} by $\langle t_s \rangle \simeq T_{60}/13.8$.

Two other statistics are the Echo Criterion for ***speech*** $EK_s(t,\tau_s)$ where $\tau_s = 9\,msec$ and the Echo Criterion for ***music*** $EK_m(\tau_m)$ where $\tau_m = 14\,msec$. Defining the time-dependent statistic:

$$T_n(t) \equiv \frac{\int_{t'=0}^{t'=t} t' \cdot |p(t')|^n \, dt'}{\int_{t'=0}^{t'=t} |p(t')|^n \, dt'} \quad \text{where } n = 2/3 \text{ (1) for speech (music)}$$

Then the Echo Criterion for Speech $EK_s(t,\tau_s)$ and the Echo Criterion for Music $EK_m(t,\tau_m)$ are respectively defined as:

$$\boxed{EK_s(t,\tau_s = 9\,msec) \equiv \frac{\Delta T_s(t,\tau_s)}{\tau_s} = \frac{T_s(t+\tau_s) - T_s(t)}{\tau_s} = \frac{T_s(t+9\,msec) - T_s(t)}{9\,msec}}$$

$$\boxed{EK_m(t,\tau_m = 14\,msec) \equiv \frac{\Delta T_m(t,\tau_m)}{\tau_m} = \frac{T_m(t+\tau_m) - T_m(t)}{\tau_m} = \frac{T_m(t+14\,msec) - T_m(t)}{14\,msec}}$$

An echo occurs when $\max\{EK_s(t,\tau_s)\} > 1.0$ and/or $\max\{EK_m(t,\tau_m)\} > 1.8$, respectively. A ***flutter echo*** can exist (*e.g.* due to an impulsive-type sound bouncing rapidly back and forth between two parallel reflecting surfaces – *i.e.* axial modes), when $\max\{EK_s(t,\tau_s)\} > 1.0$ and/or $\max\{EK_m(t,\tau_m)\} > 1.8$ occurs ***periodically***, *e.g.* at intervals of ~ 50 *msec* for speech, and at intervals of ~ 80-100 *msec* for music.

The clarity associated with the ***direct*** sound level in a large listening room/auditorium can be characterized by the C_7 statistic, defined as:

$$\boxed{C_7 \equiv 10\log_{10}\left(\frac{\int_{t=0}^{t=7ms} p^2(t)dt}{\int_{t=0}^{t=\infty} p^2(t)dt - \int_{t=0}^{t=7ms} p^2(t)dt}\right) \quad (dB)}$$

The direct sound level clarity statistic C_7 should be well-correlated with the sound source-listener separation distance, and hence should not fall below a range of $C_7 \sim -10\,dB$ to $-15\,dB$.

Numerous other, often more complicated, frequency-dependent room acoustics measurement statistics have also been developed over the years by sound engineers, such as the Speech Transmission Index (STI), Room Acoustics Speech Transmission Index (RaSTI) , Clarity for music C_{80}, Inter-Aural Cross Correlation (IACC), Strength Measure (G), Early Decay Time (EDT), Reverberance (R) – a measure of the acoustic "liveliness" of a reverberant room, an Echo Criterion for music ($\langle EK_m(\tau)\rangle$), Lateral Efficiency (LE) and Lateral Fraction (LF), Bass Ratio (BR), Warmth (W), Brilliance (B), … The speech intelligibility statistic, %ALCONs – per cent Articulation Loss of Consonants in Physics 406 Lecture Notes 10, *p*. 4-6, will be discussed after first discussing various aspects of the nature of a sound field in a listening room.

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