19th INTERNATIONAL CONGRESS ON ACOUSTICS
MADRID, 2-7 SEPTEMBER 2007

CONCERT HALL ACOUSTICS 2001-2007

PACS: 43.55.Fw

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ABSTRACT

Concert hall acoustics has received major attention since the search for the text of *Concert Halls and Opera Houses* was concluded [Beranek, Springer-Verlag 2004][i]. This paper presents an overview and analysis of those recent contributions that appear most useful to practicing acoustical consultants. The material begins with the 17th ICA in 2001, extends through Forum Acusticum in 2005, and includes publications in principal technical journals up to the present date. The subjects treated are: Reverberation Time (RT_{30}), Early Decay Time (EDT), Apparent Source Width (ASW), Lateral Fraction (LF), Interaural Cross-Correlation Coefficient (IACC) and Binaural Quality Index (BQI), Listener Envelopment (LEV), Strength (G) and various substrengths including early and late relative levels, and early and late lateral relative levels, Perceived Bass, Texture, Clarity, Sound Localization, and Instrumentation for Measuring Acoustical Parameters.

REVIEW OF SOME BASICS

The Sabine and Eyring equations are commonly used for predicting reverberation time in a concert hall. Since the former is simpler, it is preferred in calculations. The Eyring coefficients α_{ey} can be obtained precisely from the Sabine coefficients α_{sab} by noting the following equality between the two when the same reverberation time is involved,

$$0.161\ V / (S_{total}\ \alpha_{sab} + 4mV) = RT = 0.161\ V /[\{S_{total}(-2.30 \log (1 - \alpha_{ey})\} + 4mV] \qquad (1)$$

The α relationship is: $$\alpha_{sab} = -2.30 \log (1 - \alpha_{ey}) \qquad (2)$$

As usual, $$(S_1\ \alpha_{sab1} + S_2\ \alpha_{sab2} + \ldots)/ S_{total} = \alpha_{sab}, \text{ and} \qquad (3)$$

$$(S_1\ \alpha_{ey1} + S_2\ \alpha_{ey2} + \ldots)/ S_{total} = \alpha_{ey}\ . \qquad (4)$$

where RT is the reverberation time at a given frequency; V = room volume; $S_{total} = S_1 + S_2 + \ldots$; S_1, S_2, etc. = sub-surfaces in a room, 4mv is the absorption in the air and is usually only important at frequencies above 1000 Hz (if the hall is not too large).

Equation (2) is perfectly valid if the Sabine coefficient is allowed to take on all values from 0 to infinity[ii]. With this understanding, the Sabine equation is not an approximation to the Eyring equation, and it is perfectly valid to quote Sabine quotients in excess of 1.0. Wallace Sabine recognized this fact by publishing values in excess of 1.0 for materials he had measured [iii]. Eyring absorption coefficients for the subsurfaces are obtained from Sabine coefficients for those surfaces, by dividing Eq. (4) by Eq. (3) and noting the equalities,

$$\alpha_{ey1} = (\alpha_{ey} / \alpha_{sab})\ \alpha_{sab1} \qquad (5)$$

$$\alpha_{ey2} = (\alpha_{ey} / \alpha_{sab})\ \alpha_{sab2}\ ;\ \text{etc.} \qquad (6)$$

Studies by Joyce[iv] lead to the conclusion that with the same absorption coefficient over all surfaces, even different degrees of diffusion require a different reverberation equation. By extrapolation, if one uses the same reverberation equation for different venues, the absorption coefficients employed must have been determined in closely allied venues. One confirmation of this is presented in Ref. 2 where the sound absorption coefficients obtained using the Sabine equation for 20 concert halls fall into two categories, one is for conventional shoebox types of halls where there are open walls above the highest balcony, and the other is for all other types of concert halls. All halls had volumes within the range of 15,000 to 30,000 m^3. The average audience absorption coefficient for the non-rectangular type hall calculates as larger so that it requires a cubic volume about six percent greater than would be necessary for a conventional shoebox type. In a paper at this ICA conference, T. Hidaka relates this difference in these two groups of halls to their respective mean-free-paths.

Kitamura et al[v] show that different locations of a large area of a sound absorbing material in a room result in different sound absorption coefficients for that material. For example, a given material covering the upper third of the rear wall of a test auditorium yielded an Eyring coefficient of 0.5, but when moved to the middle third, yielded 0.35. They also find that the absorption coefficient for an area of acoustical material changes when an absorbing material is added to another surface of a room. Conclusion, even when using the same reverberation equation, the absorption coefficient used should be determined from a measurement in a similar shape of room in which there is a similar location of the material.

Bradley[vi] tests the use of the international standard ISO-3382[vii] in evaluation of the acoustics of concert halls. Measurements of the well known acoustical parameters G, RT, EDT and C_{80} were made at many positions in several halls. He concludes that hall-averaged values of these parameters should be supplemented by measurements in sub areas of a hall and at various distances from the sound source. In particular, focusing effects cause important changes in these parameters. Because ideal values of the parameters may not be available, it is often useful to compare measurements with values from well-regarded halls, particularly from halls of the same general shape and size. Of course, in any such reference hall, all of the parameters may not be at their ideal values.

Hidaka[viii] investigated the effects on measurements owing to source locations and number and positions of receivers. The study included RT, EDT, G, C_{80}, and IACC as measured in 15 symphony halls, 7 chamber halls, and 4 opera houses. Where possible the measurements were made in conformance with ISO-3382. The significance of the averaged values of the above parameters was taken from psychological difference limens, measurement errors, or simply the statistical significance of the numerical numbers. The criteria for judgment were RT_M (middle frequencies) 4%; RT_L low frequencies 6%; EDT_M 5%; C_{80} 0.5 dB; G_M and G_L 0.5 dB, and $IACC_{E3}$ 0.09. He used a dodecahedral loudspeaker source, placed at 12 positions on the stage, an omnidirectional microphone and binaural head at 20 positions in front, center and rear of the hall. In conclusion, he deemed to be adequate one position on stage, 3 m from front of stage on the centerline and all audience positions at least one meter off the center line. For concert halls, he presents a table in which the number of measuring positions Nr can be determined when the number of seats N is known. For RT, the table shows that N/Nr for measuring RT is 300-400, for EDT it is 150-250, for G it is <100 and for $IACC_{E3}$ it is 150-250. For chamber halls, the number of measuring positions can be one-third as many. For example, for a 2000 seat hall, the number of measuring positions for G should be 20 or more. He concludes that the 6 to 10 measurement positions of ISO-3382 may lead to unsatisfactory results, except for RT.

REVERBERATION TIME RT AND EARLY DECAY TIME EDT

Measurements of RT and EDT are generally made in conformance with ISO-3382.The standard requires an omnidirectional source which usually is a dodecahedral loudspeaker.

Koga and Okubo[ix] describe three round-robin tests involving 10 Japanese organizations. They found that neither different dodecahedral loudspeakers, nor their rotation from one test to the next, made any difference in RT (T_{30}) in any octave frequency band from 125 to 4k Hz. However, EDT was affected by different loudspeakers and different rotations of them. The deviations in the values of EDT owing to the 10 different dodecahedral's, was about $\pm$ 0.08 s around 1.5 sec in all frequency bands. The results owing to rotation of the loudspeakers showed no differences in EDT in the lower four octave bands and about $\pm$ 0.05 s in the 2k and 4k Hz bands.

Miyazaki et. al.[x] made acoustical measurements in 352 concert halls, multi-purpose halls and theaters in Japan. It is interesting to observe the change in practice in the reverberation times of auditoriums built in different time periods. For concert halls and multi-purpose auditoriums (with stage enclosures) that were built in the 1970's and earlier, they found that the mean volumes were 2000 m^3 and mean RT's were 1.5 s. In the 1980's the mean volumes were 6000 m^3 and the RT's were 1.9 s. In the 1990's and after, the mean volumes were 15,000 m^3 and the RT's were 2.1 s. As a side observation, the mean noise levels (averages of all the halls) were NC-30 in the 1960's, NC-25 in the 1980's and NC-18 in this century.

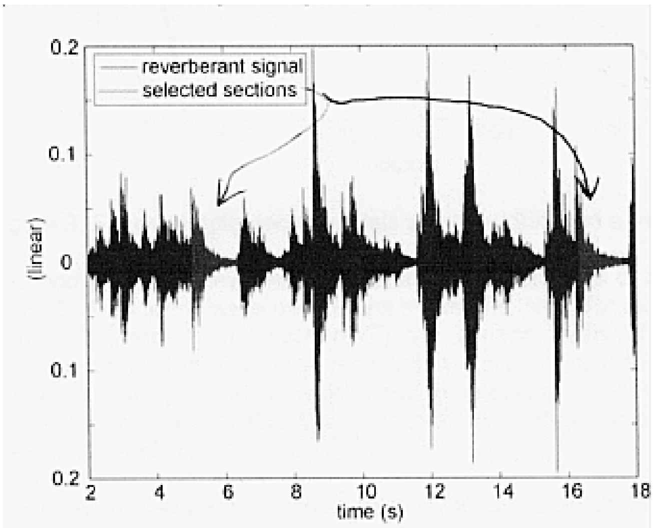
Choi and Fricke[xi] made subjective preference comparisons of the sounds at equivalent seats in two halls in the same city which had nearly the same reverberation times at mid-frequencies, RT = 2.1 s. The larger hall had a lower acoustical reputation among concertgoers. The larger hall seated 2679 and the smaller 1238. A carefully calibrated dummy head and loudspeakers were used to obtain room impulse responses RIR at different seats. Anechoically recorded music was convolved with the RIR's and presented to subjects with earphones. Their first conclusion was that the EDT values averaged in the 500 - 1k Hz bands gave higher correlation with the overall preference than EDT unfiltered. The subjects expressed positive opinions for EDT's in the smaller hall and expressed negative opinions for EDT's in the larger hall.

Sum and Pan[xii] propose a new technique for determining the reverberation time, and in particular early decay time. In contrast to the usual method of obtaining the best straight-line fit to the measured decay curve smoothed by the Schroeder integrated impulse method, they propose a technique that accounts for the 50-ms average response time of the human ear and the 3-dB just-detectable loudness change of hearing. They begin by determining the total acoustic energies in successive 50 ms intervals of the sound decay in a hall. (There are 30 such intervals in 1.5 s). The direct sound is included in the first 50 ms interval. The time at which the energy reaches the -10 dB level (the EDT) is obtained by interpolation between the two adjacent 50 ms intervals whose energies are just above and below the -10 dB level. In a case where the energy in the interval (or intervals) following the first interval is less than -3 dB, the EDT is calculated starting from the interval where the energy is first below the –3 dB level. The total energy in this 50 ms, -3 dB, interval is taken as 0 dB and it is used to represent the onset of the perceived sound decay. The same interpolation approach in successive 50 ms intervals is used to determine the reverberation time for the energy decay from -10 dB to -25 dB (LDT, late RT).

In measurements made in different parts of one hall, Sum and Pan found that the EDT's determined by this method differed greatly from those determined by the Schroeder smoothing method. For example, if an EDT equal to 0.5 s was obtained by this method, then Schroeder's EDT was 0.75 s. Also, another EDT was 0.35 compared to Schroeder's 0.85 s and in all seats they found the EDT's by the new method lower by at least 0.1 s. They then made subjective judgments of the "impression of reverberance" at the different seats and found that the subjective responses correlated well with the new type of measurements. They also concluded that new EDT correlated higher with *subjective reverberance* at different seats than LDT. If the decay is near exponential from the onset, the EDT and the LDT obtained by this method are close to those measured by the Schroeder method. They identify the EDT differences in a room, "During the early decay, the sound field in a hall is dominated by the first and following reflections that can be sparsely and irregularly distributed in time, and may have greatly different magnitudes. The number of these reflections within EDT can also be small. Thus, the sound field can decay in unsteady rates, so that various portions of the EDT can have very different slopes, in other words, the entire early decay is non-exponential."

Fricke and Nannariello[xiii] attempted to find better design guidelines for concert halls. The information in Ref. 1 on the grouping of 58 concert halls according to their subjective acoustical quality and on the halls' physical dimensions provided the data for the study. A statistical analysis of the relation between the "geometrical" and "acoustical" data for the halls in the three subjective groupings, (*Best, Medium* and *Worst)* was undertaken using a Student t-test of significance. All of the usual acoustical parameters were considered. The results: The most important acoustical discriminator was sound strength at mid-frequencies G_{mid}. In rectangular halls the width W was the most important geometrical factor and in the best halls was less than 24m. The best combined parameter for all halls was $N*G_{mid}$, where N is the number of seats and for the best halls was greater than 11,000.

Cox et al[xiv] seek to determine RT, EDT and C_{80} from music performed in a live concert situation. Their method is promising but has not yet been tested for real performances. The proposal has three steps. (1) A polyfit pre-processing unit separates the running music sample into 0.5 s segments and performs a polynomial fit to each section to determine rough estimates of the decay rate for each segment; (2) for the maximum likelihood estimation procedure, the ten longest continuous segments of decay are selected. A typical result for this step is shown in the Fig. below. From these, they found that the model which best fits the selected segments embodies two decay terms. This model yields estimations of the decay parameters for each of the sections. The decay parameters then provide the envelope of the decay and with them estimates of RT and EDT are made using the usual backward integration procedure. (3) Selection of the best estimate is made by choosing the shortest time for RT and EDT and the longest value of C_{80}. Trials were made of these parameters for reverberated music at 1000 Hz. The results for RT_{30} were excellent for reverberation times of 0.6 to 4 s , while EDT differed by about 10% at values around 3 s. For the range of C_{80} between -7 dB and -3 dB the fit was excellent, but it assigned less high values by 2 dB to 3 dB for C_{80} values between -2 dB and + 3 dB. They say the results are promising, but the quality of the estimation depends greatly on the music being used and as a result the technique needs to be improved to allow a greater variety of pieces.



APPARENT SOURCE WIDTH ASW AND LATERAL FRACTION LF.

Witew, et. al.[xv] in their attempt to find difference limens (DL's) for LF found the results not conclusive, partly owing to the uncertainties in the response of the figure-of-eight microphones. They state "As part of the equipment discussion it is noteworthy that the figure-of-eight microphones seem to represent a weak link in the measurement chain."

Bork[xvi] reports that in five-team, round-robin comparisons of three different specimens of figure-of-eight microphones (Neumann KM 86) level differences in free field were detected of more than 3 dB between sound irradiation from the front and from the back. Aging is attributed to causing the differences in sensitivity of the two spatially-separated capacitor-microphone capsules involved. Before undertaking measurements, accurate calibration of figure-of-eight and omnidirectional microphones in an anechoic chamber are essential.

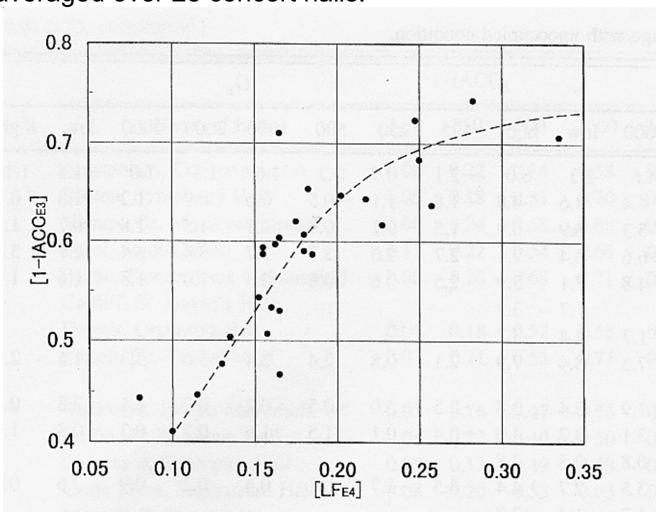
Okubo, et. al.[xvii], have developed new equipment for measuring LF and associated parameters which permits better understanding of sound arriving from lateral, overhead and fore-aft directions. The probe microphone and associated electronic apparatus provide a directional pattern that is half (namely, a single lobe) of a figure-eight microphone and with it the values of $p^2(t)$ arriving at a point from different directions can be determined. Their apparatus achieves this directional pattern by multiplying cardiod and figure-of-eight responses. This single-lobe pattern enables the measurement of $p_0^2(t)$, $p_L^2(t)$, $p_R^2(t)$, $p_F^2(t)$, $p_B^2(t)$, (omni, left, right, forward and back directional components of sound power), respectively, at a point.. The probe is comprised of four pin microphones on the four corners of a square with the fifth in the center. Diagonally opposite microphones are separated by 2 cm. The five microphones connect to a preprocessor which sends two figure-of-eight signals (X and Y directions) and one omni signal to the main processor which in turn determines the $p^2(t)$'s. A signal to noise ratio of 40 dB is maintained over the frequency range of 200 to 8.5k Hz. For measurements at lower frequencies a greater separation of the microphones is necessary to overcome inadequate S/N ratios.

Marshall[xviii] and Barron and Marshall [xix] determined their subjective measurements of apparent source width ASW using the averages of lateral fraction LF over the four octave bands from 125 Hz to 1000 Hz. They express the opinion that signals in the upper bands can be disregarded. However, Blauert and Lindemann[xx] and Morimoto and Maekawa[xxi] have shown that the higher frequency bands also contribute to ASW. In fact, anyone listening to a musical performance in a well-rated shoebox-shaped concert hall, hears only the sounds in the frequency bands from 500 Hz and above as contributing to ASW.

Beranek[1] (Appendix 2, 585-613) gives LF data for 30 halls, 17 measured by Bradley and 7 by Barron, and in every case the averages of the LF's in the 500 to 2000 Hz octave bands are nearly identical to the averages in the 125 to 1000 Hz octave bands. Hence, it appears that either average can be used in making comparisons with subjective determinations of ASW. There are two arguments for formally selecting the average of the higher-bands when determining LF. First, for sounds reaching a listener laterally, the signals in the 125 and 250 Hz bands are nearly the same at the two ears as those arriving from overhead because the wavelengths are long compared to the width of the head (i.e., IACC ≈ 1.0). This fact weighs in the direction of averaging LF in the higher bands where sounds arriving laterally at the two ears are different from those arriving from overhead, i.e., where the lateral interaural cross-correlation coefficient, IACC, drops to as low as 0.3 in actual halls. Second, the hearing mechanism is more sensitive to high frequency sounds. Hence, it is more logical to tie LF to higher-frequency octave frequency octave bands e.g., 500 to 2k Hz. It must be emphasized that sounds in the low-frequency bands are very important in judging the acoustical quality of halls, but for the reason that their influence is strongly tied to the strength (sound energy) at lower frequencies, e.g., G in the lower bands (especially G_{125} and lower).

INTERAURAL CROSS-CORRELATION COEFFICIENT IACC AND BINAURAL QUALITY INDEX BQI

The acoustical parameter Binarual Quality Index BQI, introduced in 2004[1], is equal to [1 – $IACC_{E3}$] where "E" means integration over the first 80 ms after arrival of the direct sound and "3" means average of data in the octave bands 500 to 2k Hz. The quantity is zero for the case of no lateral reflections. Like the lateral fraction LF, it is a measure of the strength of lateral reflections. Measurements of BQI and LF (80 ms and 125-1000 Hz bands) are compared in the figure below[xxii] averaged over 28 concert halls.



Bork[10] has compared the measurements of $IACC_E$ from four laboratories. All were required to use dodecahedron loudspeakers and all measured similar spaces. The values of BQI = [1 -$IACC_{E3}$] ranged from 0.1 to 0.7 with the average at about 0.6. The participants used three different dummy heads, always aligned with the source, and the maximum differences between systems were of the order of magnitude of BQI = 0.1. Rotation of the dodecahedron loudspeaker between 0 and 120^0 caused variations in BQI of about 7% in the 2k and 4k Hz bands, and little variation in the lower bands.

In Choi and Fricke's experiment[11] relations were explored between binaural quality index and subjective acoustic quality in two halls. The subjects expressed negative opinions about values of BQI between about 0.4 and 0.6 (found in the larger hall, 2679 seats) and positive opinions for BQI between about 0.6 and 0.75 (in the smaller hall, 1238 seats). These results agree closely with a comparison of BQI vs subjective quality ratings for 25 halls in Figure 4.5, p. 509 of Ref. 1. The halls in that reference had seating capacities ranging from 1546 to 2839 (median 2218).

Nakagawa et. al.[xxiii] believe that, because of the cost of a dummy head and inconveniences in transportation, IACC measurements are not often made in practice. They compared IACC measurements made using a dummy head with those made using: (1) two points in space, i.e., two omnidirectional microphones, and (2) a ball of 15 to 18 cm diameter on which two microphones were located. They conclude that both the 15 cm ball and a 21 cm separation between two microphones "indicate a possibility to obtain $IACC_{E3}$ approximately without a dummy head."

LISTENER ENVELOPMENT, LEV

Listener envelopment LEV is a subjective acoustical parameter in a concert hall that relates to the listener's perception of being surrounded or immersed in the sound field. LEV has received more research attention in recent years than any of the usual acoustical parameters. Maximum LEV is believed to occur when the sound field is completely diffuse, that is, when the sound arrives at the listener's position equally from all directions. But, there are other parameters that affect LEV.

Morimoto et al[xxiv] made subjective judgments of music (Mozart's *Divertimento*) in the sound field of a simulated concert hall to show that the components of reflections beyond the upper limit of "the law of the first wave front" contribute to listener envelopment LEV. The law of the first wave front has been extensively explored in the literature, as follows: Those lateral reflections that start soon after arrival of the direct sound (e.g., less than 1.5 ms), merge with the direct sound. After a few milliseconds, the reflections no longer merge but they do not change the listener's perception of the location of a source, instead they only spread it, i.e., they contribute to apparent source width ASW. At a particular further separation in time, the image splits into early sound and reverberant sound, i.e., the subjects report being enveloped by the sound, i.e., LEV. The time that the image split occurs is called the upper limit of the law of the first wave front. The results reported here "support the hypothesis that the components of reflections beyond the upper limit of the law of the first wave front contribute to LEV."

Soulodre et. al.[xxv] performed a series of experiments to determine which acoustical parameters are important to subjective LEV. Their results were obtained with a standard five-loudspeaker system (0^0, $\pm$ 30^0, and $\pm$ $110^{0)}$ surrounding a subject in a deadened room. The 0^0 loudspeaker emits the direct sound. The equivalent of early reflections, numbering from five to nine, delayed 10 to 15 ms from the direct sound and from each other, are emitted by the 0^0 and $\pm$ 30^0 loudspeakers. Reverberant signals with varying starting times from 40 to 120 ms were emitted by the 0^0, $\pm$ 30^0, and $\pm$ 110^0 loudspeakers. The subjects were asked to rate only their perception of being enveloped or surrounded by the sound. Their answers were labeled LEV on a scale from 0 to 100. The sound stimulus was a 20 s segment of anechoic music (Handel's *Water Music*). The parameters measured in octave frequency bands from 63 to 8000 Hz were the sound pressure level, the reverberation time, the late lateral fraction energy (LF_L), and the late total energy (G_L). First off, their data show that there is very little change in perceived LEV for reverberation times between 1.7 and 2.0 sec, which are common RT's for concert halls.

Soulodre et. al. noted that because of forward masking, the temporal delay time used for determining the late lateral fraction energy and the late total energy should vary with frequency, that is, it should not be fixed at 80 ms. They searched their data and came up with the following times (after arrival of the direct sound) at which integration of the signal should start taking place, namely, 160 ms for OBA's from 125 to 500 Hz, 75 ms at 1k Hz, 55 ms at 2k Hz and 45 ms from 4k to 8k Hz. If these values are plotted on a linear/log graph, a smoothed curve yields the numbers in Table 1. No basis is given by the authors for the derivation or for the probable accuracy.

Table 1. Integration limits

Octave bands Hz	Integration Limit ms	Octave bands Hz	Integration Limit ms
125	160	2000	60
250	155	4000	48
500	130	8000	45
1000	90		

They then determined the correlations between LEV and various octave-band averages of the late total energy G_L. They found, "the correlations are fairly independent of how the various octave bands are grouped." They decided to use the averages of G_L in the octave bands from 125 to 1000 Hz, but it is obvious that the averages of G_L's from all of their experiments correlated as well with LEV using the 500 and 1000 Hz bands (0.939 for 500/1k and 0.937 for 125/1k). Their reason was, "We wished to include as many of the octave bands as possible and there was no statistical reason to exclude the 125 and 250 Hz octave bands from the measure." [Obviously, there is no statistical reason to use other than the 500 and 1000 Hz octave bands.]

From their experiments they developed a formula for determining a physical quantity GS_{perc}, a quantity that correlated the best with their subjective LEV determinations,

$$GS_{perc} = 0.5\ G_{perc} + S_{perc} \quad \text{dB} \qquad (5)$$

where G_{perc} is the total late energy level (not lateral) and S_{perc} is the 10 log of the late lateral energy fraction, where "perc" indicates using different integration limits for each frequency band, and averages of results in the 125 to 1000 Hz octave bands. The 0.5 coefficient for the first term was introduced to give a higher correlation with the subjective LEV's.

When one considers the accuracy of the physical measurements that are customarily made in a concert hall and the probable differences of judgment in LEV among casual listeners (as opposed to laboratory subjects), this cumbersome measurement can be simplified without affecting its usefulness (Note: Soulodre et al indicate that the correlations with LEV for the simplifications to follow and the formula above are approximately 0.94 and 0.96, respectively). The simplified equation proposed here is,

$$GS_{mid} = 0.5\ G_{Lmid} + LF_{Lmid} \quad \text{dB} \qquad (6)$$

where, G_L is the late energy determined by using a starting integration time of 80 ms, and LF_L is the late lateral energy fraction determined under the same conditions, and "mid" indicates averaging over the 500 and 1k Hz octave-frequency bands. The formulas for these are,

$$G_L = 10 \log \{(\int p_0^2 (t)\, dt) / (\int p_A^2 (t)\, dt)\} \quad \text{dB} \qquad (7)$$

where, the integration time for p_0^2 is 80 ms to infinity and for p_A^2 is zero to infinity, p_0 (t) is the late sound pressure measured with a dodecahedral sound source and an omnidirectional microphone at a point in a hall, and p_A (t) is the total sound pressure measured at 10 m from the same source operating with the same power level but measured an anechoic chamber, and,

$$LF_L = 10 \log \{(\int p_8^2 (t)\, dt) / (\int p_0^2 (t)\, dt)\} \quad \text{dB} \qquad (8)$$

where, the integration times for both integrals are from 80 ms to infinity, p_8 (t) is the late sound pressure level measured with a figure-eight microphone at the same position and time in a hall as p_0 (t) was measured. Note that a figure-eight microphone only measures the sound energy arriving from both right and left lateral directions. The accuracy of either Eq. (5) or Eq. (6) for determining LEV in concert halls has not been ascertained.

Furuya et al[xxvi] set out to determine whether the contribution of late arriving energy to LEV is dependent on all directions from which the late energy arrives. The results bear on whether measurements of lateral sound energy by a figure-eight microphone, as in Eq. (8), above is adequate. Their motivation was to help designers choose reflecting surfaces in a hall in comparison with today's belief that only lateral sound influences LEV. If they should find that sound from other directions is important, how important are those directions, i.e., what is the relative contribution of energy to LEV from overhead surfaces and from the rear of a hall? In their experiments, the sound was presented to the subjects by six loudspeakers in the horizontal plane, 0^0, $\pm\ 45^0$, $\pm\ 90^0$, and 180^0 and one loudspeaker directly overhead. The structure of the sound fields was seven early reflections and staggered late reverberation, which could be introduced in any or all of the loudspeakers. In their first experiment, each of the directions of the late (reverberant) sound was independently varied. In the second experiment the directional late energy ratios were varied keeping the total late energy constant. In experiment three, the late-to-early sound ratio was additionally varied. They conclude, "Although the degrees of contribution of the late overhead and back energy ratios to LEV are smaller than that of lateral energy ratio, they are definitely effective in the perception of LEV, especially when the late energy is not smaller than the early energy." They conclude that late vertical energy and late energy from behind affect LEV by approximately 40 and 60 percent, respectively, of the late lateral energy.

Morimoto et al[xxvii] investigated in their first experiment the degree to which reverberation time affects LEV and to which the ratio of the early-to-late energy (C_{80}) effects it. The results of their first experiment show that a reverberation time of 2.0 s yields a much greater LEV than 1.0 s. and a similar increase in LEV results when the C_{80} is 0.7 compared to 6.0. In their second experiment the highest LEV was obtained for a flat response of RT vs. frequency. Significantly, LEV was much lower when the RT in the low frequency bands was lowered compared to the middle and high frequency bands and also when the RT in the high frequency bands was lowered compared to the middle and low frequency bands. They conclude, "RT at low frequencies affects LEV about equally well as RT at high frequencies."

Hanyu et al[xxviii] carried out listening tests to examine the effects of early reflections on LEV. In their first experiment, the level of the early reflections was increased with no increase in the later reverberation so that C_{80} changed in steps from 0.5 to 7.1 dB. As expected ASW increased with increasing C_{80}, but at the same times there was a significant *decrease* in LEV. In their second experiment, C_{80} was kept at 2.6 dB and the listening level and the reverberant field were held constant and the early reflections were varied in position from $\pm$ 22.5^0 to $\pm\ 90^0$. The results in that test were that ASW increased between 22.5^0 and 67.5^0 and was constant from there to 90^0. Surprisingly, LEV also increased in the same manner. They conclude that C_{80} should probably not be allowed to increase above 1.0 in concert halls [Note: From Beranek[1], in the best concert halls C_{80} is generally between -1.0 and -4.0).]

Wakuda et al[xxix] have measured in six halls the variation of late energies in the lateral, longitudinal and vertical directions as a function of source-receiver distance. Two halls are shoebox, one is fan shaped, two are multipurpose, and one is for classical music without being described. Their first experiment was to determine the attenuation of these parameters as a function of source-receiver distance (plotted in dB per 10 m). As expected, the dB/10m of the overall strength G and the late overall strength G_L was greatest for the halls with the largest average Sabine sound absorption coefficient. The attenuations for the lateral LG_{late}, vertical VG_{late} and longitudinal LG_{late} components of the strength were highest in the fan shaped hall and were lowest in the shoebox halls. It was found that the dB/10m decrease of the vertical component, VG_{late}, was by far the highest of the three directional components, possibly because of seat absorption. They conclude, "Directional distribution of late sounds depends on the diffusivity of sound fields and a directional measure might be an important factor to evaluate the degree of sound diffusion in the sound fields of auditoria."

Hanyu et al[xxx] set out to determine the effect of sound level on subjective listener envelopment LEV. The signal was comprised of the direct and, after a time delay tau, a series of reflections 70 ms in length at a level ΔL relative to the direct sound. The subject was surrounded in different experiments by several different sound fields, $\pm\ 45^0$, $\pm\ 90^0$, $\pm\ 135^0$ and combined ($\pm\ 45^0$ and $\pm\ 135^0$). For three values of tau, 10 ms, 80 ms, and 150 ms, they determined three levels of ΔL, (A) that level at which ASW begins, i.e., when the reflections are just heard, meaning that they cease to be merged with the direct sound; (B) that level when the sound begins to contribute to listener envelopment LEV, i.e., this is the time at the end of the law of first wave front—for some increase in ΔL after that time, ASW and LEV are both heard, and then ASW falls off; and (C) the time at which the reflected sound is heard as equally loud as the direct sound—about this time the reflected sound becomes so loud as to be disturbing. At tau equal to 80 ms, the difference between A and B is about 8 dB, and between B and C it is about 20 dB, in other words there is a large range of levels (20 dB) in which LEV is heard. For tau equal to 150 ms, these conditions occur at ΔL's that are about 5 dB lower. The experiments showed that sound coming from different directions sounds different, but the amount of ASW and LEV does not change much from the results above.

Barron[xxxi] investigated in 17 concert halls the two acoustical parameters that are usually associated with listener envelopment LEV, namely, late lateral energy level G_{LL} and late lateral energy fraction LF_{late},

$$G_{LL} = 10 \log [\int p_8^2 (t)\, dt / \int p_A^2 (t)\, dt] \qquad (9)$$

where, the integration times in both integrals are 80 ms to four times RT, p_8 (t) is the sound pressure measured with a figure-eight microphone at a place in a hall using a dodecahedral loudspeaker as a source of sound on the stage, and p_A (t) is the sound pressure measured with an omnidirectional microphone in an anechoic chamber at 10 m distance from the same loudspeaker operating with the same electrical input, and

$$LF_{late} = \int p_8^2 (t)\, dt / \int p_0^2 (t)\, dt \qquad (10)$$

where, the integration times are 80 ms to four times RT and p_0^2 (t) is the sound pressure measured by a omnidirectional microphone located at the same place in a hall and time as the figure-eight microphone of the previous equation.

He concluded that the variation in LF_{late} among the 17 halls was very small and that G_{LL} is related to LEV. However, G_{LL}'s magnitude is determined almost entirely by RT/V, namely the total sound absorption in the hall $S\alpha_{avg}$ [See Eq. 1].

Evjen, et al[xxxii] set out to determine to what extent the direction of arrival of the late sound energy had on perceived LEV, with other aspects of the sound field held constant. They conclude, "In concert halls, G_{LL} (125-1000 Hz) (the level of late-arriving sound from all directions) is an excellent predictor of listener envelopment LEV. However, the late lateral sound strength is a good predictor of LEV."

Hanyu and Kimura[xxxiii] introduce a complication into the search for a physical measurement that correlates well with LEV. They say, "It became clear [from seven subjective experiments] that the contribution of a reflection to LEV depends on the arrival direction of other reflections, but LEV cannot be optimally evaluated with objective measures that weight and integrate individual reflections by the arrival direction like that of lateral energy ratios. It is necessary to consider the mutual effects of these reflections by some means." In the paper they propose a spatially balanced center time (SBT_s), using a center time T_S for each direction, as a measure that correlates highly with LEV to quantify the effects on LEV by spatial distribution of reflections." Space here does not permit detailed presentation of their experiments. They judged LF_{late} and G_{LL} in comparison with their new SBT_s and obtained, for the three respectively, the correlations 0.76, 0.84 and 0.96. However, they did not consider the energy of the sound field in their experiments and said that further study was necessary to consolidate absolute energy into their findings. In addition, they gave no practical way for determining the necessary T_S's from field measurements.

Hanyu et al[xxxiv] studied the effect of different acoustics on the ability of listeners to hear contrasts in music. They compared six different types of sound fields (shoebox shape, fan shape, square shape, large bulgy rectangular shape and randomly generated artificial sound fields) on LEV, using as sources, a) different musical instruments, i.e., cellos and high notes of violins, b) violins that were played fast for short durations, and c) violins that played slowly for long durations. The subjects were asked to say that they heard contrast whenever they felt major differences in LEV. The experiments showed that in highly diffuse sound fields and in the fan shaped hall almost no LEV contrast was heard. They conclude, "In the diffuse sound fields a listener was surrounded by sound so LEV was always very high, and in the fan shape hall LEV was always small and, therefore, in these sound fields no contrast of LEV occurred." By far the greatest LEV contrast was heard in the shoebox shaped hall, which they observe is consistent with the acoustical reputation of this type of hall. For the square and bulgy shaped halls the LEV contrasts were in-between.

CLARITY AND SOUND LOCATION

Griesinger[xxxv] presents new ideas on the perception of clarity and of source location in concert halls. First, he combines into the direct sound those reflections that occur within a few milliseconds of each other. By his definition, reverberant sound starts with those early reflections that have not been merged and, he adds, if the note is held long enough, such reflections grow into a steady state. For the listener to determine the direction and clarity of the sound source in a concert hall, a sufficiently long initial time gap between the merged direct sound and the reverberant sound is necessary and this depends on the strength of the direct sound relative to the reflected sound. He says that the reverberant sound is part of the initial gap (or is submerged in the initial gap) up until the time a direct to reverberant energy (d/r) ratio of approximately 1.0 is reached. Let's determine the onset and rise of sound in Boston Symphony Hall, which has a mid-frequency, occupied, reverberation time of 1.9 s. For a source located at 0.1 the hall's length and a listener located at 0.7, the direct sound and the gap dominate the effective reverberant sound for about 20 ms, i.e., before the time that the d/r = 1.0 condition is reached. At the 0.7 listener's location, perception of clarity, the direction of the source, and the feeling that the source is close to the listener are excellent. As the listener moves closer to the source, the d/r increases but the sound impression does not change much. As he moves farther away, the initial gap narrows and the ability to perceive direction decreases and clarity diminishes. [Note: These factors are still sufficiently satisfactory at all distances in the Boston hall although the differences are noticeable if one is seeking them.]

Griesinger continues: Listener envelopment LEV occurs after longer time delays, i.e., 100 to 200 ms after arrival of the merged direct sound. In addition, to be enveloping, the reverberant sound must have achieved a reasonably large state of directional diffusion (sound arriving at listener's position equally from all directions). A longer reverberation time in a large concert hall automatically (basic acoustical theory) means slower growth of the reverberant energy and a longer length of the initial gap. Also, in such halls because of the long reverberation time it is possible to perceive the reverberant sound between successive notes. In a small hall, because of the short gap, it is difficult to separately perceive the direction of the sound and the clarity unless the d/r ratio is higher than that in the large hall. He says, "In a small hall the initial gap is shorter than that in a large hall because the early reflections arrive sooner. If the reverberation times are the same in the two halls, the d/r ratio must be maintained higher in a small hall than in a large hall or else the azimuth detection in the small one becomes difficult, clarity is low, and envelopment is less."

STRENGTH G

Shimokura et al[xxxvi] recommend a new acoustical parameter G_{re} as a means for yielding more information to designers. It is defined as,

$$G_{re} = 10 \log \{ \int p^2 (t)\, dt / \int p_d^2 (t)\, dt \} \text{ dB} \qquad (11)$$

Where, integration time for both integrals is from 0 to infinity, p (t) is the sound pressure measured at a distance d from the source in a hall, and p_d (t) is the sound pressure measured at the distance d in a free sound field from the same source operating with the same electrical input power. This differs from the usual G in that the denominator for usual G is determined at a distance of 10 m in a free field. Their measured data in a rectangular concert hall and an opera house show that G_{re} throughout a hall is influenced most by the reflections from the side walls and ceiling closest to the receiver position, while G is influenced most by the walls closest to the source position.

PERCEPTION OF BASS

Bradley and Soulodre[xxxvii] varied reverberation time and low-frequency sound level to determine their effect on the perception of bass using Handel's *Water Music*. They found that the perceived level of bass sound is not influenced by low frequency reverberation times, but rather it is influenced by the energy levels of both the early and late arriving low frequency sound, G_E and G_L. Surprisingly, they found that the subjective evaluations of bass levels were more sensitive to late arriving low frequency sounds (after 80 ms) than to early arriving ones. Finally, levels of sound in the 250 Hz octave band were less important than those in the 125 Hz octave band. Their simple equation for relation to bass perception is :

$$G_{weighted} = 10 \log \{ [E_{80,125} + 3E_{late,125}] + 0.5 [E_{80,250} + 3E_{late,250}] \} \qquad (12)$$

where the E quantities are strengths analogous to G_{80} and G_L.

In Ref. 1, a comparison of the measured strengths in the 125 Hz band (unoccupied) (G_{125}) to the rank-ordering of acoustical-quality of thirty-one concert halls was presented. There was a significant correlation of 125-Hz bass level with the quality ratings. The formula of Eq. (12) was not tested because of the lack of the data required. Bradley[6] recommends that measurements be extended to include the 63 Hz octave band as the perception of bass could likely extend there.

The oldest attempted measure of bass response was the ratio of the reverberation times (in occupied halls) at low frequencies (average of RT's in 125 and 250 Hz bands) to those at mid frequencies (average of RT's in 500 and 1000 Hz bands). In both Refs. 1 and 37, this measure was found to have little relation to the quality of sound in concert halls.

TEXTURE

Its definition in Ref. 1 is "Texture is the subjective impression that listeners derive from the patterns in which the sequence of early sound reflections arrive at their ears. In an excellent hall those reflections that arrive soon after the direct sound follow in a more-or-less uniform sequence. In less good halls there may be a considerable interval between the first and those reflections that follow. Good texture requires a large number of early reflections, uniformly but not precisely spaced apart and with no single reflection dominating the others." As a means of counting the number of early reflections Hidaka[xxxviii] proposes an envelop-function based on the Hilbert Transform of a band-passed impulse response. Compared to a reflectogram, this method has an advantage of erasing the influence of phase in the impulse response. He applied this method to the subjective rank-ordering of 22 opera houses in a number of countries and a satisfactory correlation was obtained between the judged acoustical quality and the number of significant reflections. He also found a reasonable correlation of the texture parameter with the binaural quality index, BQI = [1 – $IACC_{E3}$].

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