The Impact of Sound Control Room Acoustics on the Perceived Acoustics of a Diffuse Field Recording

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Abstract: - Live recordings of music and speech in concert halls have acoustical properties, such as reverberation, definition, clarity and spaciousness. Sound engineers play back these recordings through loudspeakers in sound control rooms for audio CD or film. The acoustical properties of these rooms influence the perceived acoustics of the live recording. To find the practical impact of 'room in room' acoustics in general, combinations of random room acoustic impulse responses using convolution techniques have been investigated. To find the practical impact of a sound control room on the acoustical parameter values of a concert hall when played back in that control room, combinations of concert hall impulse responses and sound control room impulse responses have been investigated. It is found that to accurately reproduce a steady sound energy decay rate (related to the reverberation time), the playback room should have at least twice this decay rate, under diffuse sound field conditions. For energy modulations (related to speech intelligibility) this decay rate should be more than four times higher. Finally, initial energy ratios (related to definition and clarity) require auditive judgement in the direct sound field. ITU-recommendations used for sound control room design are sufficient for reverberation and speech intelligibility judgement of concert hall recordings. Clarity judgement needs a very high decay rate, while judgement of spaciousness can only be done by headphone.

Key-Words: Sound control, Sound studio, Control room, Room acoustics, Concert hall, Recording, Playback, Convolution, Head and torso simulator, HATS

1 Introduction

From experience it is clear that a recorded reverberation time can only be heard in a room having a reverberation time shorter than the one in which the recording was made. The smallest details and the finest nuances with regard to colouring, definition and stereo image can only be judged and criticized when there is little acoustical influence from the playback acoustics on the recorded acoustics [1]. However, usually the playback room in combination with the used sound system affects the recorded acoustics. This happens in class rooms [2], congress halls [3], cinemas and even in sound control rooms.

Using formerly measured impulse responses, a first step is made in investigating the impact of the reproduction room acoustics on recorded acoustics [4]. This first step is to find the practical impact of 'room in room' acoustics for 66 combinations of one room acoustics with another. In this case the impact on reverberation, speech intelligibility and clarity has been investigated, using convolution techniques. From the results presented in chapter 5, criteria for the listening room's reverberation time have been derived for the 'proper playback' of each of these parameters, starting from a more or less diffuse sound field and the JND (Just Noticeable Difference) as allowable error. Measurement conditions are given in chapter 4.

The next step in this research is to find the practical impact of direct field room acoustics on the perceived acoustics of a reproduced sound. In this case the impact of the control room acoustics on live recorded acoustics has been investigated, using the same techniques. To this end the convolution has been applied to binaural impulse responses of six control rooms, a symphonic concert hall, a chamber music hall and a professional headphone. For 28 combinations of concert hall acoustics and sound control room acoustics the impact on reverberation, speech intelligibility, clarity and inter-aural cross-correlation has been investigated using convolution techniques. From the results, a first step is made to judge the quality of a sound control room using this new approach, starting from the JND (Just Noticeable Difference) as allowable error. Measurement conditions are given in chapter 4 and measurement results in chapter 6.

2 Convolution

The convolution y of signal s and system impulse response h is written as and defined as:

$$y(t) = s(t) * h(t) \tag{1}$$

or

$$y(t) = (s * h)(t) = \int_{-\infty}^{\infty} s(t) \cdot h(t - \tau) d\tau \tag{2}$$

In words: the convolution is defined as the integral of the product of two functions s and h after one is reversed and shifted. From a room acoustical point of view $s(t)$ is a sound that is recorded in an anechoic room (dry recording) and played back in a standard room, $h(t)$ the impulse response of the standard, more or less reverberant room and $y(t)$ the convolved sound as it is heard in that standard room. Therefore, an impulse, for instance a hand clap, recorded in an anechoic room, played back in a reverberant room, is heard as an impulse response of that reverberant room. A recorded impulse in the reverberant room that is played back in the anechoic room is again heard as the impulse response of the reverberant room. In both cases the derived room acoustic parameter values will be the same.

When both the recording room and the playback (listening) room are reverberant, smoothing of the sound occurs. Therefore, in some cases it is impossible to judge the original recordings in detail. The room acoustics in the sound recording that we want to demonstrate or judge will be affected by the acoustics of the listening room. With a double convolution by which an impulse response from one room is convolved with a dry recording and afterwards the result is convolved with the impulse response of another room, it is possible to hear how a recording, made in a reverberant room, sounds when played in another reverberant room. The result is usually an unwanted smoothed sound signal. By using a pure impulse (Dirac delta function) instead of a normal sound signal to be convolved with both room impulse responses (eq 3 and 4) we can examine what one room does with the other concerning the values for the room acoustic parameters (eq 5). So it is possible to derive a 'room in room' acoustic parameter value from the smoothed impulse response (Figure 1).

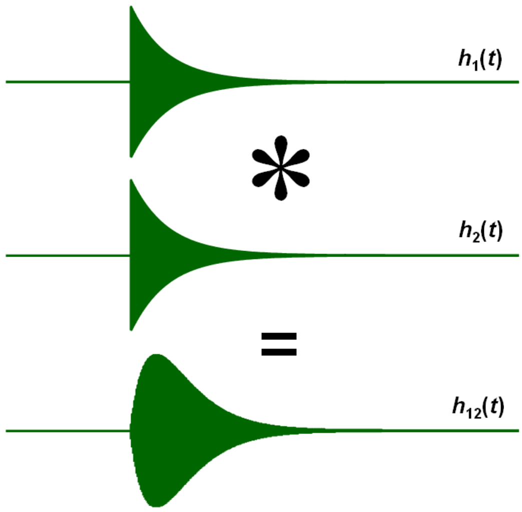


Fig 1. *Impulse response smoothing by convolution.*

Mathematically:

$$h(t) * \delta(t) = \delta(t) * h(t) = h(t) \tag{3}$$

Where:
$h(t)$ = room impulse response
$\delta(t)$ = Dirac delta function (ideal impulse)

$$h_{12}(t) = \delta(t) * h_1(t) * h_2(t) = h_1(t) * h_2(t) \tag{4}$$

Where:
$h_{12}(t)$ = 'total' impulse response room 1 $*$ room 2
$h_1(t)$ = impulse response room 1
$h_2(t)$ = impulse response room 2

Substituting equation (4) into equation (1) results in:

$$y_{12}(t) = s(t) * h_{12}(t) \tag{5}$$

Where:
$y_{12}(t)$ = convolution of a random sound signal with the 'total' impulse response
$s(t)$ = random sound signal

3 Room acoustic parameters

Many objective room acoustic parameters are derived from the room's impulse responses according to ISO 3382-1 [5] and IEC 60268-16 [6]. Examples of such parameters are the reverberation time, which is related to the energy decay rate, the clarity, the definition and the centre time, which are related to early to late energy ratios, the speech intelligibility, which is related to the energy modulation transfer characteristics of the impulse response and the latereral energy fraction, the late lateral sound energy and the inter-aural cross-correlation, which are related to the lateral impulse response measurements. Five of them have been investigated, being the reverberation time T_{20} and T_{30}, the clarity C_{80}, the modulation transfer index MTI and the inter-aural cross-correlation.

3.1 Reverberation time T

The reverberation time T is calculated from the squared impulse response by backwards integration [7] through the following relation:

$$L(t) = 10\lg \frac{\int_t^{\infty} p^2(t)dt}{\int_0^{\infty} p^2(t)dt} \quad [dB] \tag{6}$$

where L(t) is the equivalent of the logarithmic decay of the squared pressure. For this investigation the T_{20} with its evaluation decay range from -5 dB to -35 dB and the T_{30} with its evaluation decay range from -5 dB to -45 dB are both used to determine T.

3.2 Clarity C_{80}

The parameter C_{80} [8] is an early to late arriving sound energy ratio intended to relate to music intelligibility and is calculated from the impulse response using the following relation:

$$C_{80} = 10\lg \frac{\int_0^{80ms} p^2(t)dt}{\int_{80ms}^{\infty} p^2(t)dt} \quad [dB] \tag{7}$$

3.3 Modulation Tranmission Index

The Modulation Transfer Function m(F) [9] describes to what extent the modulation m is transferred from source to receiver, as a function of the modulation frequency F, which ranges from 0.63 to 12.5 Hz. The m(F) is calculated from the squared impulse response using the following relation:

$$m(F) = \frac{\int_{-\infty}^{\infty} p^2(t)\cdot e^{-j2\pi Ft}dt}{\int_{-\infty}^{\infty} p^2(t)dt} \quad [-] \tag{8}$$

The m(F) values for 14 modulation frequencies are averaged, resulting in the so called Modulation Transmission Index MTI [10], given by:

$$MTI(F) = \frac{\sum_{n=1}^{14} m(F_n)}{14} \quad [-] \tag{9}$$

3.4 Inter-aural cross-correlation coefficient IACC

Although the IACC is still subject to discussion and research, the parameter IACC [11] is used to measure the "spatial impression" and is calculated from the impulse response using the following relation (inter-aural cross-correlation function):

$$IACF_{t_1,t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_l(t)\cdot p_r(t+\tau)dt}{\sqrt{\int_{t_1}^{t_2} p_l^2(t)dt \int_{t}^{t_2} p_r^2(t)dt}} \quad [-] \tag{10}$$

where $p_l(t)$ is the impulse response measured at the left ear and $p_r(t)$ is the impulse response measured at the right ear of the HATS. The inter-aural cross-correlation coefficient IACC is given by:

$$IACC_{t_1,t_2} = |\, IACF_{t_1,t_2}(\tau)\,|_{\max} \quad \text{for } -1\text{ms} < \tau < +1\text{ms} \tag{11}$$

For this investigation only the interval between $t_1 = 0$ and $t_2 = 80$ ms (early reflections) is used.

3.5 Just noticeable differences

The just noticeable differences (JND) for all used objective room acoustic parameters are shown in table 1.

Table 1. *JND (Just Noticeable Differences).*

T_{20}, T_{30}	C_{80}	MTI	IACC
10 %	1 dB	0.1	0.075

4 Impulse responses and measurements

4.1 Diffuse field in diffuse field acoustics

4.1.1 Measurement conditions

The subset selection from the original set of impulse responses is based on the measurement quality, the measurement equipment, the rooms in which the measurements are performed and the positions of the sound source and the measurement microphone. Finally 11 impulse responses have been selected from which the 500, 1000 and 2000 Hz octave bands have been used.

Table 2. *Properties of used room impulse responses.*

Room	INR [dB] [1]	T_{20} [s] [2]	C_{80} [dB] [2]	RASTI [-] [3]
Ice chapel	62	0.20	25.63	0.89
Lecture room	66	0.63	7.23	0.70
Auditorium 1	59	0.85	4.38	0.61
Conference hall	56	1.09	2.45	0.61
Measurement room 1	80	1.11	2.05	0.54
Chamber music hall	63	1.21	4.93	0.58
Concert hall 1	60	1.28	1.74	0.56
Concert hall 2	54	2.00	2.08	0.50
Concert hall 3	53	2.92	-2.34	0.37
Auditorium 2	68	4.90	-6.23	0.27
Measurement room 2	73	5.30	-5.84	0.30

[1] Average over 500, 1000 and 2000 Hz
[2] Average over 500 and 1000 Hz
[3] Calculated from MTI 500 and MTI 2000 Hz [lit 6, 9, 10]

All impulse responses are obtained from diffuse sound field measurements using deconvolution techniques [12] with MLS and e-sweeps [13], resulting in INR values > 50 dB [14]. Some properties of the selected impulse responses are shown in Table 2.

4.1.2 Measurement equipment
The measurement equipment consisted of the following components:

- *Microphone:* omnidirectional, sound level meter (RION -NL 21);
- *power amplifier:* (Acoustics Engineering - Amphion);
- *sound source:* omnidirectional (B&K - Type 4292);
- *sound device:* USB audio device (Acoustics Engineering - Triton);
- *measurement software:* DIRAC (B&K - Type 7841);
- *signal:* synchronous or asynchronous [15][16].

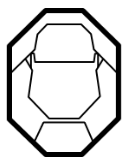
4.2 Diffuse field in direct field acoustics
The impact of the control room acoustics on live recorded acoustics has been investigated. To this end the convolution has been applied to binaural impulse responses of six control rooms, a symphonic concert hall, a chamber music hall and a professional headphone.

4.2.1 Measurement conditions
All measurements, both single channel and dual channel, were performed using a HATS or an artificial head [17]. The decay range (INR) [14] for all measured impulse responses is larger than 52 dB for all octave bands used.

4.2.1.1 Large and small concert hall
Impulse response measurements were performed in the large and small concert hall of "The Frits Philips Muziekcentrum Eindhoven" [18] with a volume of approx. 14400 m^3, an unoccupied stage floor and $T_{empty} \approx$ 2 s for the large (symphonic) concert hall and a volume of approx. 4000 m^3, an unoccupied stage floor and T_{empty} $\approx$ 1.5 s for the small (chamber music) hall. Figures 2 and 3 give an impression of the halls and the schematic floorplans with the source position S as indicated, placed on the major axis of the hall, and the microphone positions R1 and R2, where R1 is placed at approx. 5 m from the source S, equal to the critical distance, and R2 is placed at approx. 18 m from S (diffuse field). More specifications of both concert halls are presented in table 3, using the total average over both microphones (ears) of the HATS, the 500 and 1000 Hz octave bands and the receiver positions R1 and R2. The INR for all measured symphonic and chamber music hall impulse responses had an average of 60 dB for all used octave bands, with a minimum exceeding 54 dB.

Fig 2. *Symphonic (left) and chamber music hall (right).*

Table 3. *Concert halls specifications.*

Hall type	Symphonic music	Chamber music
Floor plan 10 m		
Volume	14400	4000
Number of Seats	1250	385
Stage area [m^2]	200	70
T_{avg} [s]	2.0	1.5
$C80_{avg}$ [dB]	1.1	2.7
MTI [-]	0.51	0.57
IACC [-]	0.58	0.44

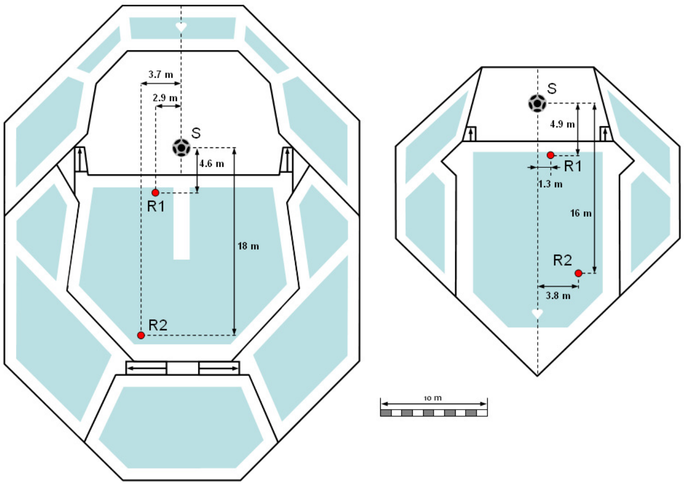


Fig 3. *Sound source S and microphone R positions.*

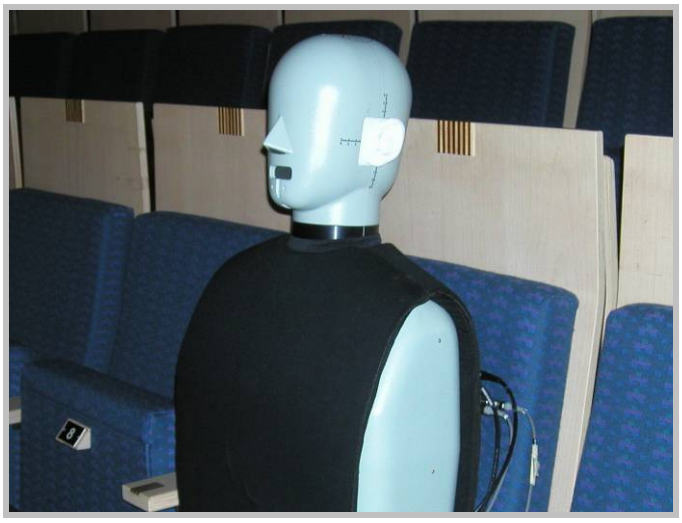


Fig 4. *Concert hall measurement using a Head and Torso Simulator (HATS).*

4.2.1.2 Control rooms

The control rooms under test are all Dutch control rooms and qualified as very good by the sound engineers as well as the designers. For the sake of privacy the measured control rooms are marked from CR1 to CR6. The control rooms were investigated extensively with microphone positions placed on a grid consisting of 15 measurement positions [19]. Based on these measurements several important room acoustical parameters were computed. The results of the reverberation time measurements revealed that in the lower frequencies all control rooms under test met the general criteria of ITU-R BS.1116-1 [20]. In the higher frequencies only control room CR2 met this criterion. Specifications of all control rooms under test are presented in table 4. For all control rooms the monitor configuration is a two channel stereo arrangement, according to the ITU-R BS.775-1 [21]. The loudspeakers are placed with respect to the sound engineer in an arc of 60° as shown in figure 5. The control room impulse response measurements were performed at the sound engineer position, known as the 'sweet spot', the focal point between the main (wall mounted) loudspeakers. Control room CR5 was only suitable for near field monitoring. Manufacturer, type and frequency range of all used loudspeakers (monitors) are shown in table 5. The INR for all measured control room impulse responses had an average of 57 dB for all used octave bands, with a minimum exceeding 52 dB.

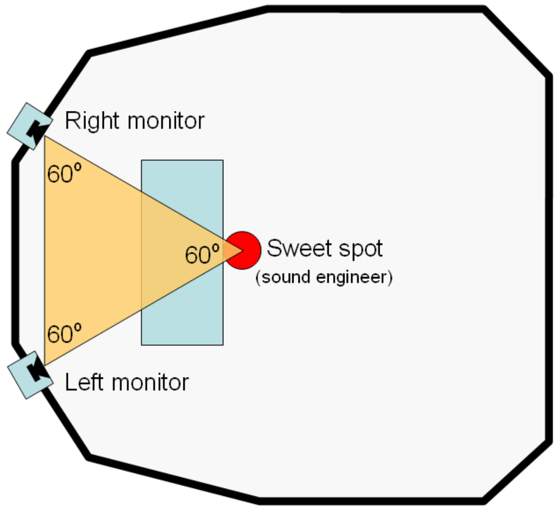


Fig 5. *Monitor placement according to the ITU [21] with respect to the listener in an arc of 60°.*

Fig 6. *Sweet spot measurement using a Head and Torso Simulator (HATS) or an artificial head.*

Table 4. *Control room specifications.*

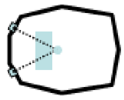
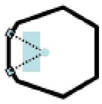
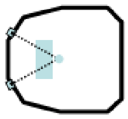
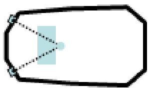
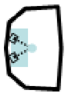
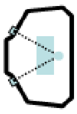
Control room	Floor plan (5m scale)	Floor area m^2	Vol. m^3	T_{avg} s
CR 1		33	103	0.32
CR 2		33	98	0.22
CR 3		35	100	0.20
CR 4		40	120	0.19
CR 5		25	75	0.14
CR 6		25	75	0.14

Table 5. *Monitor specifications.*

Control room	Monitor manufact.	Monitor type	Frequency range Hz
CR 1	Tannoy	SRM 15X	52-20000
CR 2	Genelec	1036A	21-20000
CR 3	Dynaudio	M3	30-20000
CR 4	Genelec	1031A	48-22000
CR 5	Yamaha	NS-10M	60-20000
CR 6	Dynaudio	M2	35-20000

The ITU recommendation gives the tolerance limits for the reverberation times in a critical listening environment. In figure 7 the recommended tolerance limits are presented.

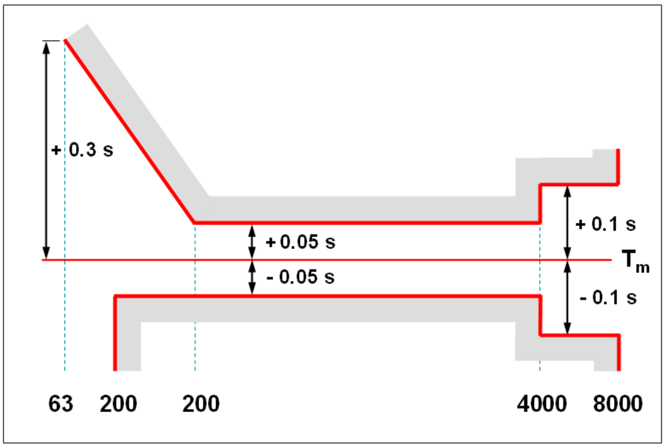


Fig 7. *Tolerance limits for the reverberation time, relative to T_m according to ITU 1116.1* [20] .

The average reverberation time T_m [s] is given by:

$$T_m = 0.25\left(\frac{V}{V_0}\right)^{\frac{1}{3}} [s] \tag{12}$$

where V is the volume of the room in m^3 and V_0 is a reference volume of 100 m^3. Figure 8 shows the reverberation time T_{30} as a function of the frequency relative to the ITU tolerance limits. Because of the different volumes of the control rooms, their absolute tolerance limit values also differ slightly.

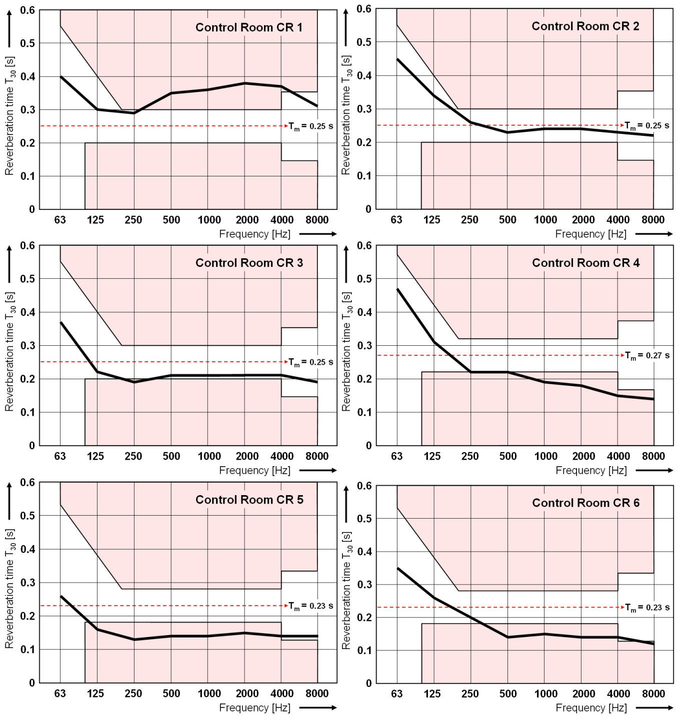


Fig 8. *ITU Tolerance limits for the reverberation time versus the measured control room reverberation time.*

4.2.1.3 Headphone
To complete the set of impulse responses a pure free field measurement was performed using the HATS and a high quality headphone [22] as shown in figure 9. The minimum INR for the measured impulse response reached a value of 88 dB for both the 500 Hz and the 1 kHz octave band.

Fig 9. *Headphone transfer measurement using a HATS.*

4.2.2 Measurement equipment
The measurement equipment consisted of the following components:

- *Head And Torso Simulator:* used in concert halls and control rooms (B&K - Type 4128C) [23];
- *artificial head:* used in control rooms (Sennheiser - MZK 2002);
- *microphones:* used with artificial head (Sennheiser - MKE 2002);
- *power amplifier:* used in concert halls (Acoustics Engineering - Amphion);
- *sound source:* omnidirectional, used in concert halls. (B&K - Type 4292);
- *sound device:* USB audio device (Acoustics Engineering - Triton);
- *headphone:* used as a reference source; (Philips: SBC HP890);
- *measurement software:* DIRAC (B&K - Type 7841).

5 Results and discussion

5.1 Diffuse field in diffuse field

5.1.1 Procedure
From an extensive set of measured impulse responses, a selection was made. From each pair (h_1, h_2) out of this selection the first one is considered as a recorded impulse response and the other one as a listening room impulse response. Using the convolution function in the acoustic measurement program DIRAC, each pair of components are mutually convolved to obtain the impulse response h_{12}, heard when playing back the recorded impulse response in the listening room. h_{12}, thus representing the h_1 affected by h_2, is then compared with h_1, with respect to the reverberation time T [7], the Modulation Transfer Index MTI [9][10] and the Clarity C_{80} [8].

5.1.2 Measurement results (diffuse field conditions)
In figure 10 through 12 the results of the convolutions are depicted as scatter diagrams. Each graph shows the difference between 2 values of a parameter, one calculated from h_{12} and one from h_1. Using for example the reverberation time as the base parameter, on the x-axis the ratio $T_{20}(h_1)/T_{20}(h_2)$ of the reverberation time calculated from h_1 (= $T_{\text{recorded room}}$), and the reverberation time calculated from h_2 (= $T_{\text{listening room}}$) is given. For symmetry reasons ($h_{12} = h_1{*}h_2 = h_2{*}h_1$), only impulse response pairs with $T_{20}(h_1) > T_{20}(h_2)$ have been depicted.

The differences were calculated for three acoustical parameters. One is a decay related parameter T_{20}. The second one is a modulation related parameter MTI, a value between 0 and 1, used to calculate the speech intelligibility. The third one is an energy distribution related parameter C_{80}.

The scatter plot of the reverberation time T_{20} in figure 10 shows that for the selected impulse responses (table 2) the variation of the percentual difference between $T(h_{12})$ and $T(h_1)$ lies within a band of +/- 10% around the trend line. Starting from a JND (Just Noticeable Difference) of 10%, a diffuse field recording, and the reverberation time as the only judgement criterion, we can conclude that for an accurate demonstration a listening room should have a reverberation time less than half that of the recording room.

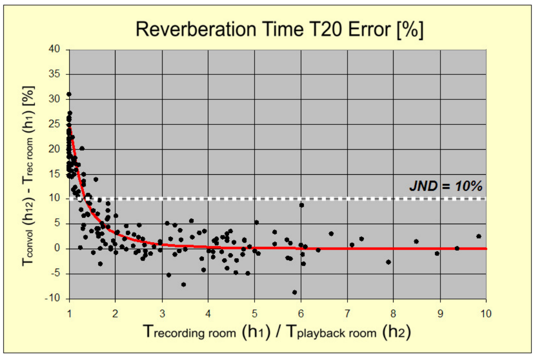


Fig 10. Difference between T_{playback} and $T_{\text{recording}}$.

The scatter plot of the modulation transmission index MTI in figure 11 shows that for the used impulse responses (table 2) the variation of the differences between $MTI(h_{12})$ and $MTI(h_1)$ lies within a band of +/-

0.05 around the trend line. Speech intelligibility experiments and demonstrations require a playback or listening room with a reverberation time at least 4 times shorter than that of the recorded impulse response, using a JND for the MTI of 0.1.

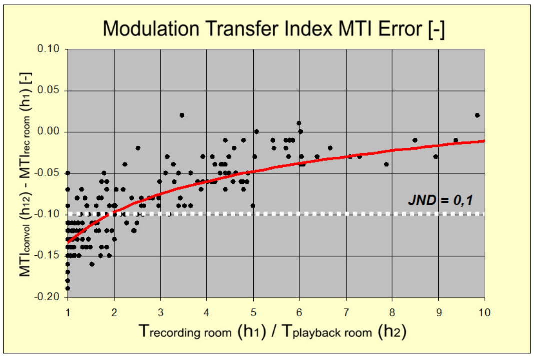


Fig 11. Difference between $MTI_{playback}$ and $MTI_{recording}$.

The scatter plot of the Clarity C_{80} in figure 12 shows that for the used impulse responses (table 2) the variation of the difference between $C_{80}(h_{12})$ and $C_{80}(h_1)$ lies within a band of +/- 3 dB around the trend line. When it is important to demonstrate or judge the details of sound definition or brightness, using a JND of 1 dB for the Clarity, you have to use a playback or listening room with a reverberation time more than a factor of 10 lower than the reverberation time of the 'recorded' impulse response.

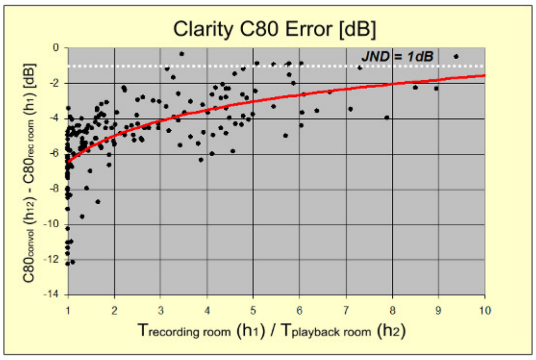


Fig 12. Difference between $C80_{playback}$ and $C80_{recording}$.

5.2 Diffuse field in direct field

5.2.1 Procedure

Starting from a set of 6 binaural control room impulse responses, 1 binaural headphone impulse response and 4 binaural concert hall impulse responses, 28 pairs of impulse responses are defined. From each pair (h_1, h_2) the first is considered as a concert hall impulse response and the other as a control room impulse response. Using DIRAC, each pair (h_1, h_2) is convolved to obtain the impulse response h_{12}, heard when playing back the recorded concert hall impulse response in the sound control room. h_{12}, thus representing h_1 affected by h_2, is then compared to h_1, with respect to the reverberation time T_{30}, the clarity C_{80}, the modulation transfer index MTI and the inter-aural cross-correlation coefficient IACC [2,3].

5.2.2 Measurement results (direct field conditions)

In Figure 13 through 20 the results of the convolutions are depicted as an average over the 500 and 1000 Hz octave band. Each graph shows the difference between 2 values of a parameter, one calculated from h_{12}, the convolution of the concert hall with the control room and one from h_1, the impulse response of the concert hall. On the x-as the control rooms CR1 to CR6 are given in order of the decay rate. The differences are calculated for four acoustical parameters: T_{30}, C_{80}, MTI and IACC.

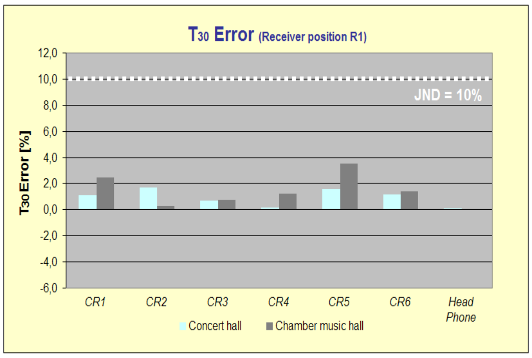


Fig 13. *Percentual difference between $T_{h_{12}}$ and T_{h_1} (T_{30} error) measured at (hall) position R1.*

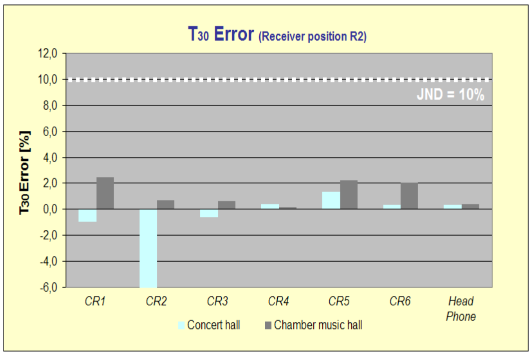


Fig 14. *Percentual difference between $T_{h_{12}}$ and T_{h_1} (T_{30} error) measured at (hall) position R2.*

Figure 13 shows the T_{30} error in % at receiver position R1 (equal to the critical distance) for the symphonic concert hall and chamber music hall when played back in control rooms CR1 to CR6 and using a headphone. Figure 14 shows the results at receiver position R2 (diffuse field). It is shown that all situations result in T_{30} errors much smaller than the JND of 10%, which supports the earlier conclusion that for an accurate reproduction of T_{30} a listening room should have a reverberation time below half that of the recorded hall. However, there is no clear relation between the T_{30} of the sound control room and the T_{30} error. No explanation was found for the fairly large and negative T_{30} error of -6 % in control room CR2 when listening to the concert hall recording at position R2.

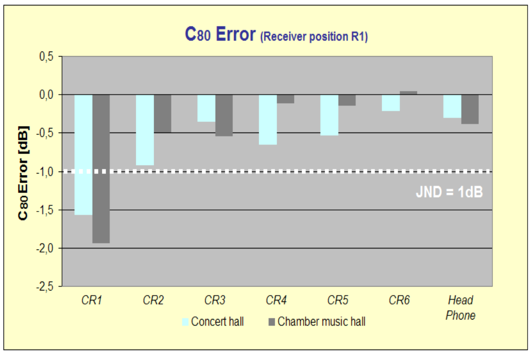


Fig 15. *Difference between $C_{80h_{12}}$ and C_{80h_1} (C_{80} error) measured at (hall) position R1.*

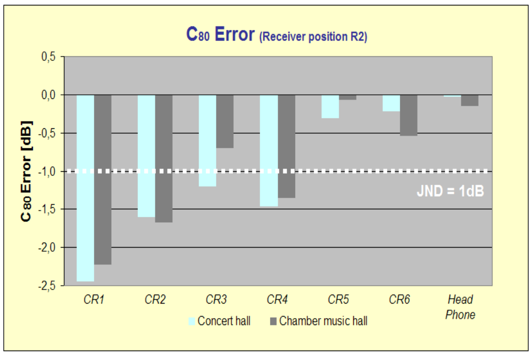


Fig 16. *Difference between $C_{80h_{12}}$ and C_{80h_1} (C_{80} error) measured at (hall) position R2.*

Figure 15 shows the C_{80} error in dB at receiver position R1 (equal to the critical distance) for the symphonic concert hall and chamber music hall when played back in control rooms CR1 to CR6 and using a headphone. Figure 16 shows the results at receiver position R2 (diffuse field). It is shown that some situations result in C_{80} errors larger than the JND of 1 dB, dependent on the reverberation time of the control room and the receiver position being at the critical distance or in the diffuse field in the symphonic concert hall and chamber music hall. In general, the C_{80} error decreases when the reverberation time of the control room decreases. Also, for the control rooms CR1 to CR4 the C_{80} error increases at the listeners position R2 in the diffuse field.

Only in control room CR5 and CR6 with a $T_{30} \leq 0.15$ s or using the headphone the C_{80} is reproduced within the JND. This supports the earlier conclusion that for an accurate reproduction of the clarity C_{80} you have to use a playback or listening room with a reverberation time less than 1/10th of the reverberation time of the recorded hall.

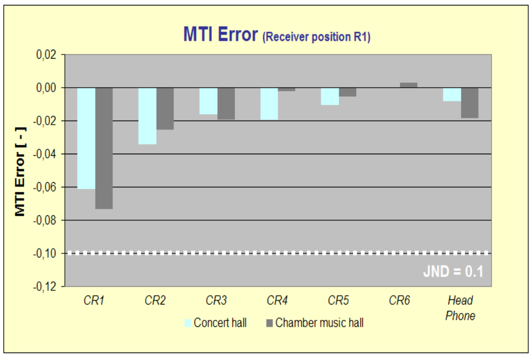


Fig 17. *Difference between $MTI_{h_{12}}$ and MTI_{h_1} (MTI error) measured at (hall) position R1.*

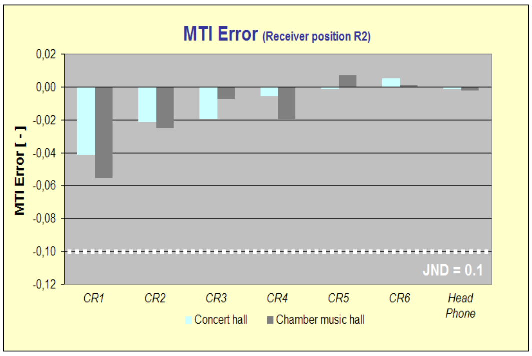


Fig 18. *Difference between $MTI_{h_{12}}$ and MTI_{h_1} (MTI error) measured at (hall) position R2.*

Figure 17 shows the MTI error at receiver position R1 (equal to the critical distance) for the symphonic concert hall and chamber music hall when played back in control rooms CR1 to CR6 and using a headphone. Figure 18 shows the results at receiver position R2 (diffuse field). It is shown that all situations result in MTI errors much smaller than the JND of 0.1, which supports the earlier conclusion that for an accurate reproduction of speech intelligibility MTI a playback or listening room is required with a reverberation time of at least 4 times shorter than that of the recorded hall. Again a clear relation is found between the reverberation time of the control room and the MTI error. No clear difference is found between receiver position R1 and R2.

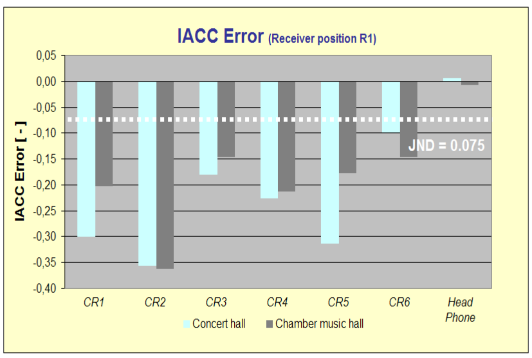


Fig 19. *Difference between $IACC_{h_{12}}$ and $IACC_{h_1}$ (IACC error) measured at (hall) position R1.*

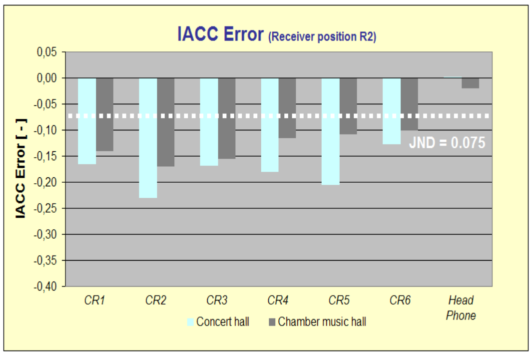


Fig 20. *Difference between $IACC_{h_{12}}$ and $IACC_{h_1}$ (IACC error) measured at (hall) position R2.*

Figure 19 shows the IACC error at receiver position R1 (equal to the critical distance) for the symphonic concert hall and chamber music hall when played back in control rooms CR1 to CR6 and using a headphone. Figure 20 shows the results at receiver position R2 (diffuse field). It is shown that all situations result in IACC errors larger than the JND of 0.075 except when using the headphone. In most cases the IACC errors for the chamber music hall recording are smaller than the errors for the symphonic concert hall recording. This may imply that the IACC error when listening in a typical control room with a good reputation is dependent on the reverberation time of the room where the recording has been made. However, an accurate judgement of spaciousness of a binaural concert hall recording apparently requires the use of a headphone.

6 Conclusions

6.1 Diffuse field investigation

Starting with 11 more or less randomly selected high quality room impulse responses and the Just Noticeable Difference of three calculated room acoustic parameters (T_{20}, C_{80} and MTI) it can be concluded:

- To accurately reproduce a steady sound energy decay rate (related to the reverberation time), the playback room should have at least twice this decay rate, under diffuse soundfield conditions.
- For energy modulations (related to speech intelligibility) the decay rate of the playback room, under diffuse sound field conditions, should be at least 4 times higher than that of the recording room.
- Initial energy ratios (related to definition and clarity) require auditive judgement in a predominantly direct sound field. It seems that the decay rate of the playback room, under diffuse sound field conditions, should be at least 10 times higher than that of the recording room.

6.2 Sound control room investigation

Starting with 6 qualified as good, more or less standardised sound control rooms, 2 concert halls, a headphone and the Just Noticeable Difference of four calculated room acoustic (ISO/IEC) parameters (T_{30}, C_{80}, MTI and IACC), the following can be concluded:

- The ITU-recommendations for sound control room design are adequate for evaluation of reverberation in concert hall recordings.
- When it is important to assess the details of sound definition of a concert hall recording, you need a control room with a very high decay rate. Only the control rooms with a reverberation time below 0.15 s can be used. This confirms the conclusion from the diffuse field investigation that you need a room with a reverberation time less than 1/10th of the reverberation time of the recorded concert hall. This is lower than the recommended value of the ITU.
- The ITU-recommendations for sound control room design are adequate for evaluation of speech intelligibility in concert hall recordings.
- An accurate judgement of spaciousness of a binaural concert hall recording apparently requires the use of a headphone. This requires further investigation.

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