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The V-criterion for good listening conditions in control rooms - on the importance of the first 15ms

Prof. Dr. E.J. Voelker, Institute for Acoustics and Building Physics, Oberursel in the Taunus
E-Mail: E.Voelker@em.uni-frankfurt.de

Summary

Many control rooms suffer from deficient acoustical conditions at the mixing desk listening position. The positioning of the monitoring loudspeakers, whether in-wall mounting with a distance of 3 to 4 m between them or nearfield monitoring, has acoustical consequences mainly for the first 15ms after the arrival if the direct sound radiated from one or more speakers. The reverberant environment with many short time reflections can lead to a masking effect with the result that the recorded studio sound is affected. The threshold for Any Change of Perception AChP is shifted upwards by ca. 6-8dB relative to delayed signals. These signals belong, for instance, to the recorded voice with its plosive sound components or to musical instruments. Their impulse response at the microphone is additionally loaded with first incoming reflections from nearby surfaces, all within the first 10-15ms. When listening to these recorded sound sources in many cases neither the speaker nor the control room meet the requirements defined by the V-criterion. Many measurements have been carried out to find more about the impulse responses of musical instruments when they are affected by impulse signals via a structure borne sender. Impulse responses of many high quality speakers were made and compared with head-phones and well known test-speakers. Actual discussions are focused on optimizing the listening conditions under these aspects. It is now time to do more for the exact reproduction with a better impulse response of high quality speakers and for consequent acoustical conditions.

1. Introduction

The judgment of microphone recordings is only possible when an optimal reproduction of sound via loudspeakers is obtained in control rooms /1/. This optimum is required both for speech and music. The microphone has picked up sound signals from the recording area which must carry their acoustical structure via transmission to, and including, the monitoring speakers. At the listening position, direct sound and short time reflections very often interfere with each other. Both form the sound which is more or less expected. It is not the sound impression which occurs in the studio environment or in the concert hall during the recording of an orchestra.

Instead, it is the desired sound impression in control rooms which is produced by loudspeakers and influenced by room acoustical properties. Positive judgments are equally possible for people *listening* at home /24/.
During a sound recording the so called 'fine structure' composed of direct sound reflections and reverberation has to be regarded. It should not be disturbed by acoustical influences due to incorrect sound mixes. The control room should therefore be neutral. However, the reality of control room design is not so. They are dampened, reverberant or over-dampened. They are too small or too large. The reverberation times can also differ distinctly. Sometimes nearfield monitors in 1 m distances are even necessary. Than the best solution seemed simply to be the use of headphones.
On the other hand, it is desirable to have control rooms to make comparisons. When sound engineers are totally acquainted with their control rooms they have no need for an 'ideal control room'. Perhaps they fear that the great variety of different control rooms may be limited by a lack of artistic freedom. It should not be overlooked that many control rooms nowadays are being built with great efforts and expense although they are doomed to failure. Sound engineers and tonmeisters nevertheless know the 'Sound' of their control rooms and are able to transform recordings. This is proven by investigations in which tonmeisters or sound engineers reached their best results in their own and well known control room /2/.
After the completion of more than 20 control rooms for different purposes such as: large orchestra, dance and pop music, news and drama plays, the author came to the conclusion that an important part of the transmission chain from the studio to the listening place in control rooms were not regarded sufficiently. The question was and still is today: what happens within the first 15 milliseconds when listening to the direct sound of the monitoring speakers?

2. To the human listening ability

The human being hears an arriving sound with an organ that has been developed over millions of years. In the beginning, the rapid acknowledgment of sound was a determining factor between life and death. Still today, this ability to hear is used every day in every situation. It is used for both listening to speech and to music. According to Spreng there is a lower listening plateau (Hoerebene) which is responsible for this fast acknowledgement of sound. The fast reaction follows in the same way. Spreng says:

'The first arriving signal creates an over-emphasized behavior which leads to a dynamic changing with the consequence of higher sensibility of the auditory system. This property is the way to a very rapid detection of equally fast running signals as usual for speech' /3/.
In this instance, the auditory system is switched over to a static behavior. This refers to a listening situation in which a great quantity of information is indicated using the brain's memory. Only with this memory can the human compare long and short reverberation times, tonality or coloration of sound. Cremer describes this very fast reaction of the human auditory system with the ability to produce so-called 'spikes', which are comparable to an 'all or nothing' decision in that an action potential is shot out /4/. They do not have the same form as the arriving sound signal. So far it is almost a digital conversion. Theile calls this special ability of the auditory system an association. He separates it into a very fast running part of acknowledgment and localization and a later association for evaluation of signals /5/.
When these differences between the dynamic and the static behavior of the human auditory system really exist and when the dividing line is approximately 10-15 ms, then a consequence for the design of control rooms must be considered.

2. Early sound within 15 ms - Impulse responses

Speech, musical instruments or sounds are sources for very early sound particles received at the eardrum as sudden changes of pressure. "Very early" means that the sound is reflected directly upon stimulation. Molin found and recorded the first fields of vibrating areas between the F-holes of a violin as early as the first 0.07 ms after a ball struck the body of the instrument /6/. In the following 0.3 ms the entire cover and backside of the instrument vibrated. This fast or early sound reflection was confirmed by own measurements which were made with several musical instruments, even with a trumpet, when the mouthpiece was stimulated by structure borne sound /1/. Based on the TDS-technique according to Heyser, the amplitude above time or frequence can be evaluated and displayed by way of Fast-Fourier-Transformation FFT, if a sweep tone and filter is being used /7/. Figs. 1 and 2 below are examples based on a violin. The instrument's impulse response is displayed on the time curve showing how fast sound is reduced within ca. 10 ms.

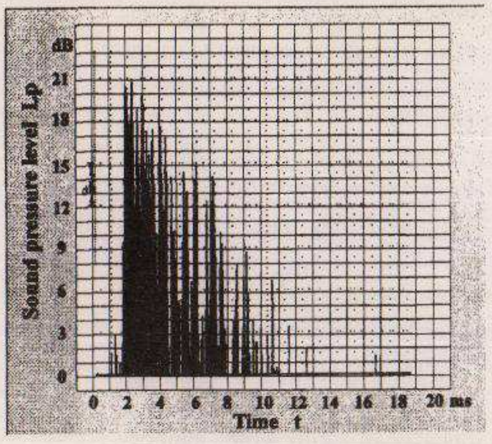
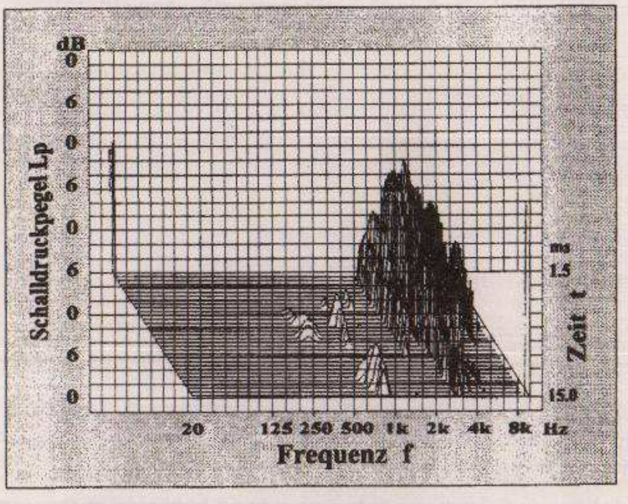


Fig.1 Impulse response of a violin when stimulated by a structure-borne vibration source

Energy-Time-Curve of sound irridiated from the musical instrument, here within 20ms. The violin was tuned, no dampening of strings.

Measurement of sound pressure level at a distance of 0,5m in front of the instrument. Stimulation with a 1,5m steel bar, which was damped by a tube surrounding the bar /1/



Figg. 2 Impulse response of a violin when stimulated by a vibration source and measurement of the sound pressure level at 0,5m distance
Three dimensional display of level, time and frequency with TDS measurement technique

The three-dimensional representation provides an overview of amplitude, time and frequence. Here as well, fast stimulation and reduction within 10 ms can be established. Measurements, however, should only show the time structure of the sound reflection. Evaluation of frequence-dependent stimulations actually caused by bowing or blowing into the instrument was not planned since it is not required here. Instead, impulse stimulation can occur when the string of a double-bass produces irregularities using a revolving collophoned endless band /8/.

The impulse response shows two important aspects:

* a very fast irradiation of impulse sound of the instruments, the same applies to plosive sounds of speech as indicated by Spreng /3/

* Fast decrease of sound level to -20 dB and below within 10 ms

In Fig. 3 it is shown that for 7 musical instruments researched all results are comparable at 1.8 dB/ms as "response attenuation" (RA) of the instrument. The first pressure change is transmitted nearly unchanged through the ear canal to the eardrum via the hammer, incus and stirrup to the oval window, as Hudde showed /9/. Theile mentions early local acknowledgment in his association model followed by later perceptions comparable to Gestaltassoziation /10/ which takes place on the frequency or tonal domain. The fast perception of sound will be clarified as follows.

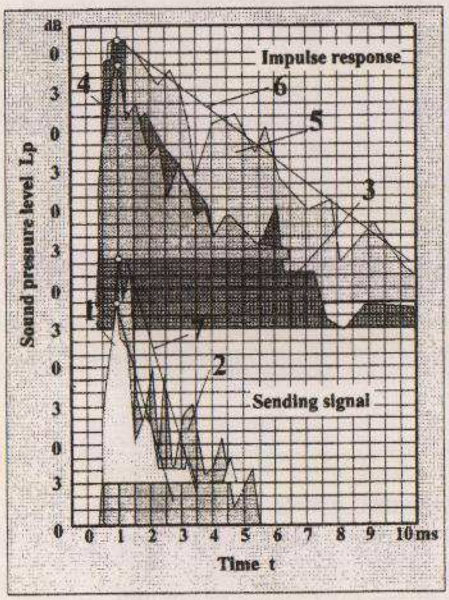


Fig. 3 Sound irradiation from musical instruments, comparison between sending signal and response of musical instrument

Attack by a structure borne loudspeaker via a metal bar on the instrument or the mouthpiece of a trumpet. Measurement of the irradiated sound in appr. 1 m distance. Measurement of the source alone by cutting the connection to the instrument. The sending source is totally sound proofed in order to avoid interferences.

1: Only the metal bar measuring the airborn sound/ 2: as 1, but with the structure borne pick up at the end of the bar / 3: Irradiation from a guitar/ 4: Irradiation from a cello/ 5: Irradiation from a violin/ 6: Average value of all responses from 7 musical instruments: 1,8 dB/ms / 7: average falling off of sound irradiation of the sending source alone: 8,0 dB/ms

The just perceptual shifting of the virtual sound source between two stereo speakers is 1.2 degrees (11) received by a time difference of only 0.021 ms. This kind of fast perception is being activated with each impulse response of any musical instrument, e.g. in case of stimulation changes by way of bowing or blowing. Consequently, no distortions should be allowed at this first pressure front; for example, by way of delays, phase influences or additional pressure fronts. Manger calls these distortions dangerous for optimal sound perception and fatigue during long term listening /12/13/.

4. Direct sound package (DSP)

The normal direct sound of a source is that which is audible in a non-reverberant or dead room. This sound is well known as somehow unpleasant and disturbing. In a life environment on a stage or in a studio additional short term reflections reach the musician's ear or the microphone, as well as the ears of the conducter by delays of ca. 15 ms. Reflections from walls or ceiling will be received later, reflections from the hall, e.g. from the back wall even later. Important sound elements are the direct sound und the first reflections, which form the DSP in Fig. 4. The time gap of up to ca. 15 ms is therefore important /1/.

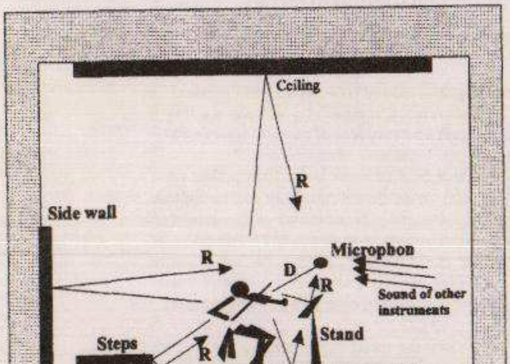


Fig. 4 Direct sound and sound reflections as DSP

Reflections are produced near the sound source. They arrive within appr. 15 ms after the direct sound. Later reflections come from the ceiling or from walls, but with greater delay of appr. 20 to 30ms:

Direct sound	0 ms
From a music stand	2 ms
From the floor	6 ms
From steps on the podium	12 ms
From adjacent musicians	5-8 ms
From near walls	18 ms
From other walls	48 ms
From the ceiling	35 ms

This DSP was confirmed by measurements executed inside orchestras, as shown in Fig.5. After these 15 ms a gap follows which Berank called Initial Time Delay Gap ITDG. This gap should not be filled with reflections in order to avoid a small room-impression. Other reflections following a time gap of only 25 ms would lead to the impression that the room is too small to be a concert hall /14/. In other words: DSP forms the direct sound followed by the ITDG as a distinct reflection free zone.

When listening in a recording studio or living room the DSP shows that it should reach the listener's ear undistorted within 10 ms. The limit of 15 ms includes response of musical instruments, which has to be considered from 1.8 dB/ms. By the time of 15 ms the sound reflection is reduced to ca. -27 dB compared to the first sound at t=0 ms.

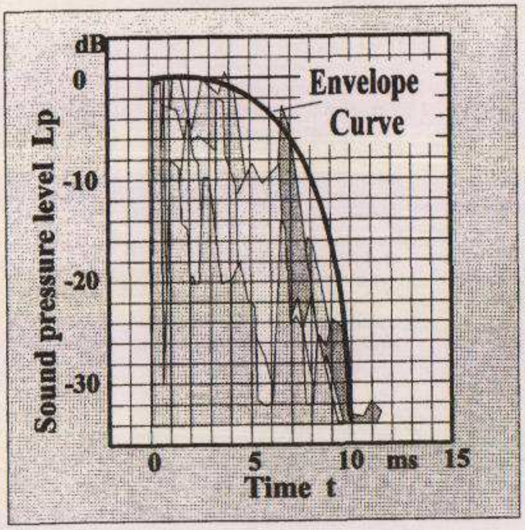


Fig. 5 Impulse response of early reflections near the musical instrument or the voice

Stimulation by an omni directional speaker. Measurement with an omni directional microphone at different distances

Exposition of the "direct-sound-package" DSP consisting of direct sound and early reflections near the sound source. Both parts are significant for the sound recording in the studio. Later reflections are not to be seen here. They belong to the overall room acoustics of the hall or studio.

-20 dB in a time of 9,5 ms
-30 dB in a time of 9,8 ms

Medium value of 6 measurements, an envolope can be found

6. Consequences for the control room

No other contributions to this clear, first pressure-front-signal are permitted to reach the listeners ears, except for later reflections after more than around 15 ms as shown in Fig. 6. A reflection-free zone is required from 0 to 15 ms regarding the DSP from the studio recording. The V-shaped criterion was derived from listening tests when a single sound reflection with a delay of up to 15 ms arrived at the sound engineers position in a control room /1/. The limit of around 15 ms is reached when a reflection becomes inaudible with a level of approx. -22 dB compared with the direct sound. Later reflections beyond 15 ms are permitted and to some extent, desired. They can follow the curves 4 or 5. Many early reflections indicate a reverberant listening aria obtained for instance by diffusors while curve 5 can be found in fairly dampened control rooms as proposed by the author.

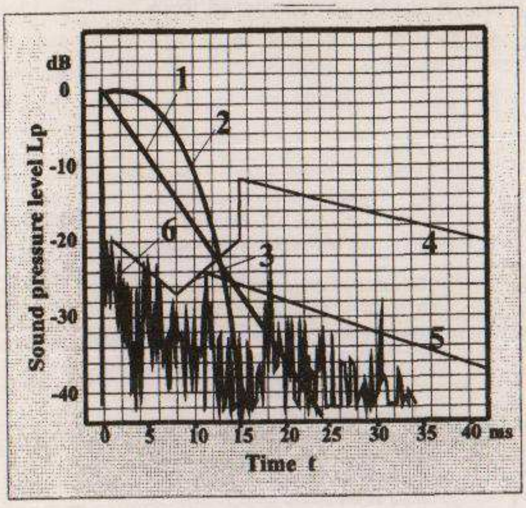


Fig. 6 Permitted, expected and measured sound levels at the listening position in a control room

1 Impulse response of musical instruments stimulated by structure borne sound
2 Direct sound package DSP at the recording microphone and the listening position.
3 V-criterion for permitted sound reflections arriving within 0-15 ms
4 Expected maximum sound reflections in reverberant control rooms
5 Expected sound reflections in usual control rooms with low reverberance
6 Measured room impulse response at listening point with the left loudspeaker

5. Problems with monitoring speakers

The first impulse signal as part of DSP should be irradiated from the monitor with a first pressure-front not being affected by additional sounds which arrive later due to phase shift or time delays. If the signal is distorted by the loudspeaker, differences can be made out clearly as hearing tests have shown. Therefore, it is of no surprise that on the consumer level loudspeakers are chosen very carefully.

Fig. 7 shows impulse responses of different loudspeakers measured with TDS /7/. Type 5 reflects a response as defined by the production company /12/. Type 6, however, shows another maximum level after 1 ms, which means additional pressure fronts to be processed by the human auditory system. These lead to unwanted information contributing to fatigue during listening /13/. Schmid and Jung /15/ as well as Braun /16/ confirmed the perceptibility of these irregularities. Phase changes were caused on purpose. Small broadband loudspeakers can show convenient impulse reactions and, therefore, may be used for certain purposes, e.g. nearfield listening around mixing-desks or computers. Specially developed headphones or small loudspeakers are sometimes used for optimal listening tests. The impulse responses indicate the distinct strong first pressure-front which is certainly necessary for these listening tests. Using the impulse response of curve 4 and adding artificially the time delayed signals of curve 6 a comparison is possible which is to be seen in Fig. 8. Based on an A-B-comparison, 5 test persons listened to different performances inside a recording studio with low sound reflection.

Differences could be experienced very clearly so that a recognition ability of 100% may be assumed.

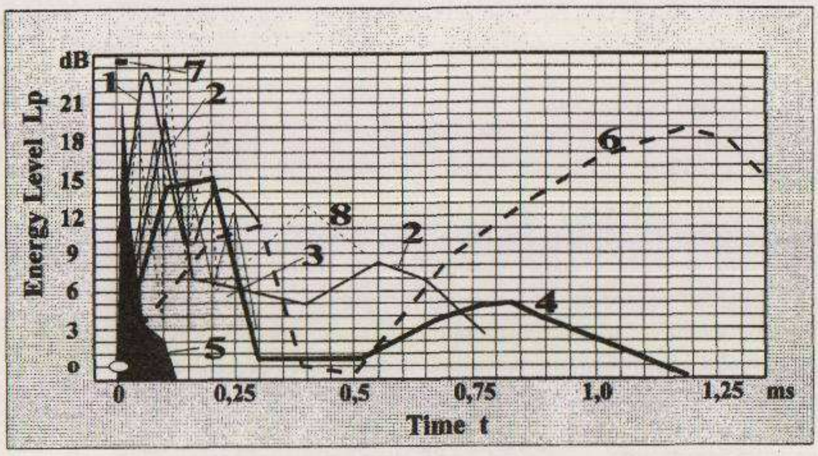


Fig.7 Early irradiation of sound - impulse responses of loudspeakers and headphones

1 Headphone, Typ AKG 240 DF after Hudde 1991 (9)
2 Headphone, Typ HD 424 Measurement 5cm in front of the receiver, after Voelker 1996
3 Measurement speaker for listening tests, Typ Vifa M10 MD-39, 155mm steel ball after Moller und Sorenzen 1995 (15)
4 Test speaker, in a stell ball 150mm, 1 Koaxsystem after Voelker 1995
5 MSS, Manger-Studiobox after Manger 1986 (12)
6 B+W S.2 Studiobox
7 Time gap in which the human ear can just detect a change in localization between the two stereo speakers (0,02 ms)

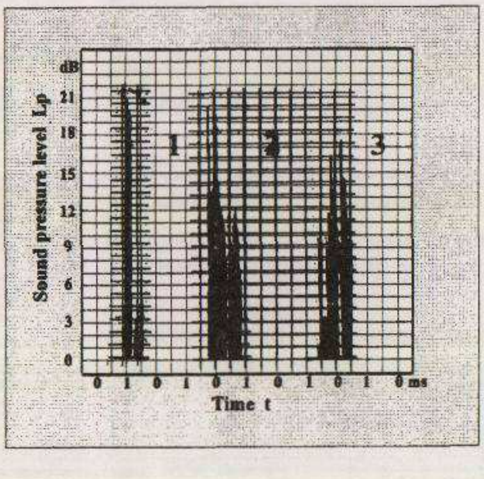


Fig. 8 Loudspeaker-packages

Artificial loudspeaker package by adding short time reflections

Measurements with TEF 12 plus in the Studio of IAB

1 Small test loudspeaker
2 Monitoring Speaker Typ B+W S.2, Comp.. Bauer and Wilkensen
3 as 1, but with artificial short time reflections

In a test all 6 subjects found a distinct difference between 1 and 3 when presented in A-B-comparison with the same results for speech and music.

This clarifies the importance of correct reproduction of the approaching pressure front. It is repeated within the DSP time window. The results are obvious. Interfering sound reflections have to be avoided or their level reduced as much as possible.

7. Multi channel sound reproduction

Recording studios still work with "stereo". However, the market has opened yet some more. Sound techniques in movie theaters are already based on multi-channel sound reproduction to be experienced by everyone visiting them. On the other hand, multi-channel sound reproduction requires multi-channel recording and mixing techniques. Krause, Hensel and Schaller for example suggest orthophonic sound reproduction /16/ based on three eight-way directional recording microphones as well as one omni directional microphone. Sound reproduction needs 11 to 15 loudspeakers positioned around the listening area. Völker, 1989, mentioned one example of recording the sounds of a reverberent office space as if it were a radio play production. This was compared to an artificial sound field produced by a processor. As a result the artificial space sounded much more "real" than the actual sounds of the typical office /17/. Sounds were recorded with two studio monitors. Advantages offered by multi-channel sound reproduction is well known from various simulations made in experimental recording studios with more than 120 loudspeakers positioned at walls, ceilings and floors /18/19/. This kind of sound system is not of major concern in this context. Multi-channel sound reproduction has to be limited to 5 loudspeakers, three of which are located in the front and two on each right and left sides in the back. Spikofski, Pehrs and Theile described 3 / 2- or 3 / 4 loudpseaker systems based on 5 or 7 loudspeakers /20/21/.

Fig. 7 shows loudspeaker positions according to 3/2 and 3/4, as evaluated by Völker in a recording studio /22/. The experiments were first aimed at useful recordings based on a performance of a music group in the IAB studio. Sound mixing was made with Soundcraft 6000 on 5 or 7 channels. Single sound delays and reverberations were mixed additionally to the last channels SL, SR, HL and HR. These channels had to simulate room impressions. During sound mixing and checking all kinds of special effects were discovered. As room impressions were eliminated prefering direct sound only, or mostly the listener no longer seemed to be inside a room but much closer to the orchestra.

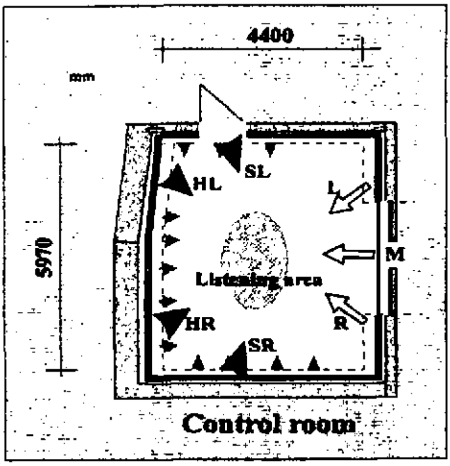


Fig. 9 3/2- and 3/4- sound reproduction in the IAB control room
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Multichannel reproduction with:
L,R: Monitoring speaker, Typ K+H O96, Company Klein aund Hummel
M: JBL 4425
SL, SR: Monitoring speaker Pyp P 7302 of Company Heco
HL, HR: Small broadband boxes HECO 2000

Volume of the control room: V= $80m^3$ net.
Reverberation time, medium T= 0,18s
Attitude 3,03 m net.

He could even reach the orchestra and, in one case, was moved into the orchestra itself.

This effect was already established in the recording studio with 8 direct sound channels with high quality monitoring speakers and 108 small loudspeakers based on 24-channel reproduction. The listener felt like he was inside the orchestra and, with increase of reverberation sounds, the impression was switched from room acoustics to acoustics inside a stadium or a church. There was a special interest in the effects regarding the musicians' listening experiences /22/.

The V-criterion applies both for the two stereo loudspeakers and to the additional speakers in the middle in both 3/2 or 3/5 combination. Exact and optimal listening requires the reflection-free zone. This is obtainable for the three direct systems: left, right and middle. For the other speakers left and right in the back only or mainly room impressions will be reproduced. Therefore a delay of 15 ms can be introduced and seems to be acceptable. When using all five or seven speakers for direct sound with high power, problems may occur.

The acoustics of this control room or reproduction area must fulfill other requirements. What was reached in classic control rooms with the „Aimed Acoustics" is not simply to copy for multi-channel reproduction /23/. The principles, however, apply in the same way. It is required to avoid first reflections in the 15 ms gap. This was easy for two speakers and even for the third one in the middle. The others lead to distinct consequences. The room must be equally absorbed. An another solution is to use nearfield monitoring. This is to say increasing the DSP and excluding the control room acoustics.

8. Tonmeisters with a great sense of adaptation

Years ago, it was found that Tonmeisters, sound engineers and producers can work with more success in their own, well-known control room or acoustical environment. Although that control room was acoustically insufficient, it was an adapted environment in which outstanding recordings were made. In these control rooms even a certain and, for the broadcasting company, typical sound was created which was successfully transmitted via antennas, records, or CD's over many years. It would be wrong to state that due to some disadvantages in these control rooms, music and speech recordings were made with defects.
On the contrary. There is an adaptation over a long period of time to get acquainted with the acoustical environment. Sometimes disadvantages can be masked. This process is of course evident in today's control rooms where some kind of personal conversion exists to translate the microphone signals into the listening situation at the mixing desk. This is probably the reason that up until now no unique standard control room exists. It is not really necessary to force the introduction of such a control room facility. However, one disadvantage remains: every recording sounds different when it is replayed in other control rooms. In other words, a mix-down from a multi-track recording will lead to another result in every different control room.

8. Summary and open questions

Control rooms differ distinctly in their acoustics. Many questions arise and are not answered yet. Some examples: Who uses consequently the time gap for the first 15 ms according to the requirements of the V-criterion? Can the reverberance of strong first or secondary reflections be accepted in the sense of more or less reverberance or diffusion? Who differs between the last association of the human auditory system with very fast recognition compared with the later association of impressions, the sound of a room, or the evaluation of reverberation times? Are the requirements for the impulse response of loudspeakers high enough? Can the listening system really detect the microphone signals from the studio which contain the impulse behavior of speech and music in addition to very first reflections from the very near environment such as from the floor or a music stand? Are there smearing effects when during the reproduction in the control room, reflections are added with the consequence being that the listener is burdened by irritations? Is it sufficient that reverberation times in control rooms can differ between e.g., 0.2 - 0.5 s? When the frequency response

shows from 300-500 Hz, a level change of 5-6 dB, can coloration of sound be accepted? Is a nearfield monitoring in order to exclude the acoustical conditions a sufficient solution for control rooms?
The questions could be continued. Nevertheless, it becomes obvious that discussions will go on. Nobody wants to have a reflection-free control room as in a test laboratory. On the other hand, reverberance and a certain amount of reflections can be introduced to reach a reasonable compromise, together with the transmission chain including microphone and loudspeaker.

Acknowledgments
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