A simple hearing method for audio quality evaluation.

By: A.J. van den Hul

1: How to define good sound quality.

We all know from experience and especially from other people that quality evaluation of audio reproduction is mainly a matter of taste and sometimes even of experience.

Taste is an ever changing factor and therefore not very reliable. The experience comes with the years and even then, the truth is not always there. Experience and habits are strongly related to this matter.
So what to do to improve both experience and habits?

The answer is to go back to the original sound during the recording session and to study what happens afterwards.

Assuming that the original sound during the recording (performance) is the best possible in all aspects, then afterwards during rework and play back we experience all kinds of quality degradations. The final result is sometimes close but often far away from the original impression we had in our acoustically trained minds.

We all agree that the original performance had the most spacious and accurate sound with the highest resolution. This prime quality sets our reference in all aspects.

2: Good sound is qualified by the following parameters.

The following eight subjective parameters are very important:

1. Detail.
2. Dynamics.
3. Lack of distortion.
4. Balance in timbre.
5. Depth.
6. Spatial impression: Width and Height.
7. High definition.
8. High resolution.

The combination of these eight properties I name: STRUCTURE.
Structured play back is reproduction which the listener highly appreciates; the sound quality is close to that of the original life performance.

3: Why good sound quality can change so dramatically.

When one of the above mentioned properties is missing, our appreciation of the reproduced sound diminishes. With more properties missing, the reproduction quality already gets doubtful; the sound still contains its musical message like the melody and words, but more or less lacks the musical emotion and the many fine details that originally were perceivable: the natural acoustical environment with the orchestra and its dimensions within, the microphone setup, microphone types etc.

For economical reasons in making a recording we (audio engineers) often have to make a choice, meaning that the final listener never exactly gets what the original performance sounded like.

Being in the profession we however have to work seriously on quality and have to define methods to improve the quality level, despite economical or other reasons.

It is not just the equipment (everyone with some money can buy) that establishes our quality level; more importantly our experience and understanding of the profession give the pie its final taste.

From all eight parameters I have given you, I regard the spatial impression to be the most important factor for correct evaluation of reproduction quality.
My following series of thoughts serves to illustrate latter statement. These thoughts form the basic message of this paper.

Obviously, the original live performance is the most spacious. This is before the recording (with its chain of technical operations like editing, copying, mastering and cutting, employing equalizers, special effect equipment and other solutions) attempts to capture and render the “soundscape”.

Mainly due to all the black box work that belongs to our profession, from all eight subjective parameters given, some very quickly fade in quality. The result is that the pleasant original spatiality gets replaced by a more or less equivalently sounding signal, spiced with many minor signals of various origin which are not a part of the original (live) impression.

We thus perceive that when e.g. detail fades, there is also a reduction of depth and resolution; or that due to introduced distortion the natural timbral quality and definition are diminished and are replaced by harshness.

We all know that our equipment in use is very effective in affecting one or more of the eight parameters mentioned. Even such simple things as cables can be responsible for the loss of original quality. Quality that is lost somewhere along the audio chain can never be restored again.

4: The way back again.

Then, how to reach back to the highest possible quality level without needing a too technical understanding of all the electronic processing in our studios?

The answer is simple and effective: use your ears and evaluate the spatial reproduction quality; retrieve the original musical structure from your played music.

The higher the spatial quality rendered, the better your equipment. The lower the spatial character, the harder you have to work to improve the overall performance; also the harder (in many cases louder) you have to listen to find some of the original structure back again. Playing louder however is NOT better; in many cases this only indicates that you work with an unsound system and that you are actually not at all aware of this. With this you are trying to enclose or recapture the residues of the lost structure in the play back. Some real professionals call this their “SOUND”.

Mind your ears, dear friends. They are your valuable tools and can never be replaced, so be very careful with these wonderful instruments.

5: How the musical structure got lost.

The basic idea is very simple: due to its signal processing nature, all our equipment, no matter how good and/or expensive, introduces extra signals.
These extra signals are not a part of the original spatial and structured sound; they are not a part of the musical structure. In essence these extra signals can be viewed upon as separate mono signals mixed together with the spacious left and right channel (additive signals).
They don’t belong there, but are there because of all the major and minor defects of the equipment and interconnections along the entire long chain between live performance and play back at home.
These two added artificial signals have several properties:

1. Their level is higher when the equipment is worse.
2. They act as two mono signals possessing some harmonic relation to the original left and right channel.
3. In practice, their relation to the original left and right channel is mainly composed of harmonic distortion and non-linear phase (but in fact all linear and non-linear distortion effects present build the relation).
4. The stronger they are, the more difficult it gets to evaluate the spatial properties of the original sound and to enjoy the acoustics of the original recording environment.
5. The more and higher the distortion components in the play back signal, the more one’s acoustical attention concentrates towards the left and right loudspeakers themselves.
6. These extra equipment signals make an acceptable positioning of your loudspeakers very difficult.

6: The introduction of M_L and M_R.

I have called these two non-original signals that are added by the equipment M_L and M_R, where M_L (Mono-left) stands for the signal added to the left channel and M_R (Mono-right) stands for the signal added to the right channel.

The audible effect of M_L and M_R combined with the reproduced recorded signal is that the total acoustic/spatial impression is limited and that there is a focus on distorted sound just between the two loudspeakers.

We might think of latter as a kind of common-mode signal, which however is without too many signals in common:
Each of the two channels has its own function and component shortcomings and of course also the electrical input signals are different; the result is that M_L and M_R are never identical. Only the mechanisms causing M_L and M_R are common

because the two channels in your system are about identical.

In the mean time, recording engineers having collected their experience using equipment blessed with higher levels of M_L and M_R are used to this unnatural effect. Moreover, they sometimes even get nervous and complain about the sound quality when this effect is NOT there. The same happens with regular audio-consumers.

All female vocals must come from the exact centre between the two play back loudspeakers (as a common habit). This often also with a strong accent on higher harmonics, giving the impression that she worked very close to the microphone. When despite this she did so, the effect is even more dramatic and gets very unnatural.
Digital recording and play back with a lot of extra signal processing has this unnatural effect automatically incorporated.

Leaving out all technical processing results in a much cleaner and more pure signal; listening becomes a pleasure without stress again. The emotional value of the original recording is still there and brings back the value of the original sound. Another important acoustical result is that there is space again around the soloists and the instruments. Acoustical aspects like distance, direction, dimensions and other characteristics of the original room are simply there (and for free).

7.1: How to detect this M_L and M_R effect.

There is a very easy method:
Without any M_L and M_R incorporated in the play back signal, there is a great spacious experience in the listening.
With a lot of M_L and M_R there is no fun at all. Maybe that is what a lot of engineers are used to, but it is still an artificial result with much reduced acoustic qualities.
If we could subtract the M_L and M_R signals partly or even completely from the signals feeding our loudspeakers or from our loudspeakers acoustic output, it would enable us to detect these signals' influence on musical reproduction.

7.2: And here is the simple method.

Ask your assistant to stand in front of e.g. the left loudspeaker. He now absorbs a part of the M_L signal.
By doing this, the spatial reproduction (a relation in time and amplitude, and therefore also in phase, between the left speaker and the right speaker) is also influenced, but much less. This because the original left and right signals support each other in the reconstruction of the acoustic space the music was performed and recorded in.
This supportive mechanism on the other hand does NOT work for the M_L and M_R signals; these two signals do not originate with any mutual spatial correlation as their basic parameter; in other words, they are not space (phase) related. The phase relation between M_L and M_R is random or at least artificial.

When (with your assistant standing in front of the left loudspeaker) the acoustic result is that there is a strong shift of the acoustic space to the right speaker, you can be very sure that there is a lot of M_L absorbed by your assistant. What happens is that M_R now attracts your attention to such a degree that this is the main reason of the shift of your acoustic interest to the right speaker.
When your assistant now walks to the right speaker and stands in front of it, there will be the same acoustic image shift to the left speaker where the M_L now is present at a dominant level.
With lower levels of M_L and M_R (in case of better equipment or even with less equipment) the acoustic shift will be smaller.
Without any M_L and M_R present (if possible), the result is that the spatial reproduction is hardly influenced by your assistant standing in front of one of the speakers.
This experiment requires a serious audio training and very good recording source material. Here, as a start an analog tape works best.
When analog sources are not available, start with a DAT master tape recorded with 2 omni microphones of highest quality.
After 10 changes of the position of your assistant, you are very well aware of what this experiment is concerned with.

Changes in equipment (or even changing cables with differences in e.g. the connector) to find the best alternative are the target of this listening exercise.
CD players can be changed between, using the same CD. By performing this test the better unit of the two can be found. The same works for amplifiers, mixing desks and even for your own loudspeakers.

The main part of this test is to find the cleanest sound available in your studio or listening room. The cleaner the sound (the less distortion) the more natural the original musical atmosphere of a recording is rendered.
With some ear training you will become able to detect the extra M_L and M_R without any assistant and in any situation. This is the biggest achievement of this whole training.
Signals free of M_L and M_R are your target.

With a lot of M_L and M_R in your audio signal, the original space, timing and thrill are taken over by a nervous making,

attention asking and screaming “over modulation”.
Also, low level signals related to or dealing with detail and resolution are lost or hidden and are not audible anymore. Furthermore, the positioning of your loudspeaker pair becomes very complicated; this because finding a good positioning involves finding a balance between the rendition of space and the audibility of the M_L and M_R signals. The latter two usually require an exact positioning to get the female vocals in centre. Often this calls for extra acoustic (absorbent) measures.

If in performing the test there is hardly any shift in spatial reproduction, you can be sure that there is hardly any M_L and M_R present in the play back signal. In this case the loudspeaker positioning is less critical and in any listening position there will always be enough musical space to enjoy the play back. There is rarely any need for “repair” of the room with absorbent materials or other acoustic tricks.
Experiments showed that audio contaminated with M_L and M_R required the distance between the two loudspeakers not to exceed 3 meter (9 ft.) to keep critical listeners happy. Without M_L and M_R on the other hand, the distance could be enlarged to 8 meters (24 ft.) without any of the fifty listeners present experiencing any critical loudspeaker positioning at their place in the room.

7.3: Conclusion.

The more flawed the electrical play back signal is, the more complicated the positioning of your speakers gets.

8: How to get away from M_L and M_R.

The answer is not simple; there are too many factors playing an important role here.
Let’s make a start:

1. Start with a clean mains power input. Clean this as good as possible and/or generate your own power.
2. Use very heavy power lines with HF damper rings to reduce switching and other noise entering your audio equipment.
3. Keep line level equipment separate from power level equipment by using different power lines.
4. Keep all your electrical contacts in good shape by cleaning them once a month with a good contact cleaning liquid like our “The SOLUTION”. Don’t forget your mains plugs.
   It also helps to clean the fuses in your power lines and inside your equipment. This despite the fact that AC will not create heavy layers of oxide. DC power lines clean up a lot after this treatment.
5. Have a star ground system to ground your whole audio setup.
6. Be very critical about all your equipment: the less equipment working, the less electronics, the better the result. Don’t impress your friends with the quantity of your equipment but with the quality.
7. Use a single master clock for digital work. The reduction of interface induced jitter distortion dramatically improves spatial reproduction and harmonic purity.
8. Avoid an unlimited number of copies of copies of copies of the master tape. Since bit errors can accumulate, the lower the generation number of the final result the better the quality.
9. Don’t save money on cables and connectors. With hundreds of meters of cable actually being used in a recording studio, the total length of metal involved here is much higher than that of all the copper of all the PC boards the signal passes.
   The higher the screening factor of all cables the better. Choosing a cheap cable type with a single and not entirely closed screen would have you end up with a lot of cross-talk and noise from other sources. Important and critical signal lines must minimally have three screens.
10. Never let power lines run in parallel with signal lines; the result is that extra noise will be radiated into your signals, causing higher levels of M_L and M_R.
11. A common situation which should be avoided is that signals are amplified at one stage and are attenuated at a later stage, this sometimes even over and over again. Every amplification stage adds M_L and M_R, so the amount of “dirt” in the signal is increasing.
12. Be very critical about the quality of the components inside your equipment. Especially the signal lines are an often overlooked part. Avoid electrolytic capacitors where possible and prefer high quality film capacitors; the higher their voltage rating the better.
13. Reduce the amount of microphones used in recordings. In many situations two is enough. Consoles with 96 channels are visibly impressive but are too complicated in e.g. power and ground management to work properly.
14. Properties of solder will change under exposure to high temperatures and mechanical stress. (Latter e.g. also caused by thermal cycling of printed circuit boards). Especially tube equipment can benefit from internal rework.
    Never think that what you bought will always function at the same quality level.
15. Handle your cables with care; any repeated stress or twisting will eventually reduce the transmission quality.
16. A lot of digital equipment acts as HF transmitter. But they also have sensitive inputs like your analog equipment

does. As a result often an unpredictable blanket of “noise” is spread over all your channels (again M_L and M_R). Though most digital equipment is well shielded nowadays, the following measures cannot hurt:
Give digital equipment a separate mains feed. Isolate all incoming and outgoing signal and power lines with HF damper rings. Keep analog and digital equipment separated by at least 2 to 3 meter. Keep digital equipment switched off when not in use.

17. Due to the nature of digital signals and their processing, standing waves (signal reflections due to impedance mismatch) are very common in digital interconnections. So called interface induced jitter due to inter symbol interference is the result. The application of typical 75 Ohm coaxial (SPDIF) or 110 Ohm balanced (AES/EBU) connections is not enough. Better is to work with products which exhibit a natural damping (e.g. our carbon products: The FIRST ® Ultimate and The FIRST ® Metal Screen for 75 Ohm and The SECOND ® for 110 Ohm). Their natural and smooth damping creates a much more quiet electrical behavior of all digital lines. Signal reflections due to interface impedance mismatch are damped out, effectively putting and end to interface induced jitter. This causing some signal attenuation is irrelevant since digital signal levels most often are high enough.
18. The application of pure carbon or at least hybrid products in your whole setup will dramatically reduce M_L and M_R. This has directly to do with the high structure level of these materials themselves. Metals or other conducting materials with less, and a less stable, structure are always introducing extra M_L and M_R.
    Politicians without a good structure always create higher or extra taxes to cover up their mistakes.
19. Use more analog equipment. By its nature analog equipment exhibits less M_L and M_R distortion since less complicated signal processing is involved.
20. Train your ears to HEAR the M_L and M_R signals. This helps you in finding more sources of M_L and M_R and to get away from these signal sources.

Signed: A.J. van den Hul