Comments after reading Dr. Floyd Toole’s article „The Measurement and Calibration of Sound Reproducing Systems”

Contents:

The author raises a wide range of problems in the field of loudspeaker usage, but is quite skimpy in proposing any real solutions.

He points to a „widely accepted” steadystate amplitude response measurement that is being used just as a „matter of faith”, and which „now has penetrated consumer audio” simply as „an enticing marketing story”. He mentions the use of time windowed measurements as a next stage of development of steadystate amplitude response measurement. At the same time, he points out that „the simple measurements therefore cannot be definitive” and „there are reasons to exercise great caution in the application of equalization based on conventional in-room measurements.”

The author only touches on some solutions very briefly and has not shown the source or basis for such conclusions:

„The on-axis curve by itself is insufficient data. Full 360 data, appropriately processed, is important information.”

„...spatially-averaged measurement to reveal the underlying curve.”

Let’s look at the article, reading more closely.

„For the system to function sensibly, mixing and mastering engineers need to experience sound that resembles what their customers will hear.”

This is an overly simplified statement. Customers will not experience the same sound in most cases because their speakers are not as good as those used in radio, TV, ordinary car audio systems. But it is a well-known fact that the „good” work of a recording engineer (producer) sounds good as a piece of art through any speaker that is used. Of course, it is not the same „sound”, but it is the same work of art. And it seems curious that the recording engineer must use the best possible, ideal monitor system to get the result that will sound good anywhere. It means he must be free of any „fight” with monitor problems to successfully produce his kind of art.

The job of tuning studio monitors should be divided into two tasks:

1) The removing of any colorations (usually narrowband problems)
2) The setting up an overall broadband balance.

If the first task is not done properly, good results are almost out of the question.

If second task isn’t done – it usually isn’t a problem, as it doesn’t “ruin” the artwork, since that balance can be adjusted during the mastering process or even by the listener just tweaking the “old-fashioned bass and treble tone controls”.

„Flat on-axis frequency response is clearly the engineering objective for most of these systems. Those that deviate significantly earn lower ratings in double-blind subjective evaluations.”

This requirement is necessary but still not sufficient. Moreover, finding that axis is a problem. It is problematic even for good loudspeakers (in terms of even directivity). Responses deviate significantly even when they are taken in some +- 5 degrees from the axes of symmetry. Such deviation is enormous for an ordinary Hi Fi consumer 3-way system. But there is a solution in such a heavy case.

It is a serious mistake to make decisions regarding a loudspeaker's performance based on just on-point measurements.

„The goal of this paper is to identify the key variables in sound reproducing systems that can lead to a calibration process for monitoring conditions during the creative process, as well as for reproduction

systems for audiences of all sizes."

While setting such a goal is a nice undertaking, the subsequent content of Dr.Toole's article fails to propose any solution or even talk much about the variables in question.

It is also quite a serious mistake to treat professional monitoring systems and home consumer systems equally.

The creation of a recording that is „a sound image" is very similar to the creation of a photographic image. It is not sufficient to simply have good fixation equipment (camera or audio recorder) and a good technician operating it. It is an art to capture an image in the right way, thus creating a work of art.

Proper and precise monitoring is essential. It saves time and prevents mistakes and overdone efforts.

It seems therefore wrong that „recording engineers" are hired just for the process of producing a recording in that the engineer's position — is just part of the technical staff. One should note that there is also the job of „recording director" in some professional cultures of audio-production that emphasizes the art over the technique of the sound recording process. Therefore, a recording director usually will have received more of a humanitarian education at a music university than a sound engineer.

The level of monitoring quality is not simply a question of some preference found through a double-blind test. It is about clearly pointing out the loudspeaker problems that were corrected. Most highly-qualified recording engineers can point to the problems corrected without use of a „bypass" baton. Recording engineers at the next-lower qualification levels may not be able to point these out so clearly but they do report double the efficiency of their work after such corrections are introduced. And they are able to deliver their work to the producer without any remarks or requests to improve on something.

There was an interesting case where a producer took part in a blind test:

Brant, a mixing engineer, was providing services to producer George, who was located abroad. Brant decided to apply equalization on his monitors J and kept developing the work for George. Later George started asking Brant – 'what is going on, why are your mixes sounding so good all of a sudden?'

2.1 directivity

„The DI can be interpreted as the difference in dB between the on-axis sound and the total radiated sound power"

– strange, up to now it was the difference in dB between the sound intensity level on-axes and the sound intensity level where the loudspeaker does not have directivity (is emitting in all directions equally) and emits the same sound power level. Or in the sound power domain - it is the difference in dB between the sound power level created by the loudspeaker and the sound power level where this loudspeaker is emitting the same sound intensity in all directions as on-axes.

It is widely accepted that the use of good-sounding studio acoustics helps capture the sound of acoustic instruments. The acoustics being received by the microphone, consisting of the number of reflections of almost all the sound emitted by an instrument is a proven method of achieving recordings rich with timbre. By the same token, finding the right placement for a close-microphone on an acoustic instrument usually presents challenging problems.

We can enjoy the full richness of an acoustic instrument's timbre only from its studio „ambience" – from its reflections in the far field.

2.3

„The on-axis curve by itself is insufficient data. Full 360 data, appropriately processed, is important information."

Here the importance of SPFR is stressed. By why not make SPFR measurements for a particular place?

„In an acoustically dead room, the room curve will be identical to the on-axis response of the loudspeaker."

This is a serious mistake – the room can sound quite „dead" but it at least has a floor on which the loudspeakers are placed. And, at least, two loudspeakers are working together with the respective effect of „mutual" work – each of the speakers receives sound pressure from its neighbor and this increases the sound-emitting efficiency of that loudspeaker group (known as „mutual work") to some frequency that depends on the distance between the speakers.

And you can't find any „on-axes" spot between those two speakers (though sometimes some point on the floor is proposed ...).

But why go through so much effort just to see if we can measure SPFR in one place?

The author points out the importance of SPFR, but does not refer to work that is already using on-field SPFR measurement.

2.4 A very nice description of the problem and situation. But the lack of a solution with some half-word „spatially-averaged measurement to reveal the underlying curve"

Yes, it provides the SPFR of a loudspeaker.... Let's try to equalize it?

„In conclusion, there are reasons to exercise great caution in the application of equalization based on conventional in-room measurements."

Yes, but what to do about it?

4.2 Car audio.

The article pays little attention to this important and complex field. It seems as if no solutions have been proposed. Just talk of targets. But you must get the reference „flat" before talking about (or applying) any target.

The complexity of the car audio field created competition among solutions with a very „serious" approach – with local, regional, national and continental levels of competition. This would require some objective and unbiased judging. And this is achieved at some acceptable level – according to the level of funding spent by competitors on their car audio systems and participation costs in such contests.

So this is a good place to "see" how one or other solution is objectively deemed to be better than another.

And you won't see "multi-national luxury" brands here. But you can find some real working solutions that have been tested in the "race" and are now working in the Sound Power Density domain.

7 „ ... while maintaining a stable perceptual impression of the sound sources.

Changing the venue does not fundamentally alter the timbral character of voices and musical instruments. They are merely those voices and instruments in different venues. Sound production and reproduction are not treated differently."

„For the entire history of sound reproduction a flat and smooth on-axis frequency response has been the performance target for loudspeaker designers."

„Later it was found that maintaining similarly smooth performance off axis resulted in even higher subjective ratings."

These claims are mostly self-evident. But where are the solutions offered?

7.1

„In discussions to this point it has been indicated that measurements in a room are, by themselves, insufficient data upon which to base a prediction of sound quality”

This is very unclear – what constitutes „measurements in a room”? Any field measurements or just „room measurement” as it is stated at the very start of the article?

I can’t agree with the statement that „field measurements in a room” are insufficient.

For stadystate room measurements – I would confer.

The author is neglecting to mention work that already offers usable methods for taking measurements and using them for correction decisions.

„The art is not completely preserved (Fig. 1).

Adaptable humans will still find information and entertainment, no doubt, but there is no escaping the fact that the sound quality is different. Movie soundtracks need to be compatible with the world outside cinemas, and cinemas may wish to play musical programs created outside the movie context. There is a problem.”

The first sentence is over expressed. The overall balance LF-HF is very easily adjustable and it is easy to get accustomed to slight imbalances. The problem is not so serious as is claimed.

Yes, the film industry is trying to tune up their systems and has accepted some curve as a standard but it has gone away from „widely accepted” one. What is that?

Let’s take some loudspeakers from a store shelf and bring them home. The manufacturer, in the process of production, set the amplitude frequency response to flat in an anechoic chamber. Now we put this speaker on the floor of our room - +3dB in LF. On the floor near the wall - +6 dB in LF. In the corner - +9 dB.

In this real case, we are getting something in between those numbers.

But this is true for recording studios as well!

Thus we get a widely accepted standard that describes why there is no problem with program compatibility between the studio and home.

One more thing is advocating such a natural LF boost. There are „advanced” maximizing techniques that tend to create a program with a „pink” spectrum – equal energy in each octave.

Such a maximized program sounds good on our „widely accepted” LF boost.

We should tune our systems to flat to get a well-balanced cinema soundtrack for listening in our rooms.

We need an easy and reliable measurement system that „reveals” the underlying loudspeaker curve.

And there is such a solution that is ready to use.

I agree that the quick solution for this what the author proposes:

"Consumers who are sensitive to timbral imbalances may find old-fashioned bass and treble

tone controls to be beneficial."

7.2 Questions

1. "room reflections" should be divided into two parts:

a) room reflections that are caused by the placement of a loudspeaker on the floor, near a wall or in a corner and the effects of the mutual work of those 2, 4 or 8 speakers.

b) the reverberation field of sound – sound energy that is accumulated and traveling in the room.

And the answer - the first must be measured to „see” how those 2, 4, 8 speakers are mutually working together.

The „widely accepted” target should be used to keep compatibility with the majority of distributed sounding programs.

2. „To quantify room reflectivity, is it better to use the early decay of conventional reverberation time data, or to use a measure of early-reflection energy accumulation time?”

C coefficients introduced in the 1970’s-80’s can provide the quantified evaluation of how room/hall acoustics are perceived. They are very well described by Ahnert and Reichardt [Ahnert, W. and Reichardt, W., Grundlagen der Beschallungstechnik (Foundations of sound reinforcement engineering) (Berlin: Verlag Technik, 1981)]

Now it is possible to obtain the frequency responses of C parameters at a very high frequency resolution.

3. „A flat, timbrally neutral, direct sound appears to be the logical objective. How best can it be achieved in practice? Anechoic loudspeaker data (corrected if necessary for screen loss) and in-situ time-windowed measurements are possibilities.”

The word ‘appears’ still seems to suggest little more than ‘appearances’.

And so the serious claim that all possibilities have been accounted for is in fact preventing further development.

And of course, the first sentence of question is true insofar as the direct sound is created by the mutual work of a number (2, 4, 8) of loudspeakers.

The two cases should be treated differently:

a) a near-field loudspeaker setup when the loudspeaker is placed on a stand where the 3:1 law is in force,

b) placement on the floor, near a wall or in a corner.

In the case “b”, much of the „non-axes” sound reaches the listener by placement reflections, and the SPFR measurement describes the mutual performance of real or imaginary speakers quite well. And that is the “direct sound” of a group of speakers.

In case “a”, it appears that the „on-axes” curve, which is measured in an anechoic chamber in the classic way, ought to work. But no, it doesn’t. It is impossible to find „axes” even for good loudspeakers in terms of smooth directivity (studio monitor). AFR measurements that are taken at +- 5 degrees from a plane of symmetry (axes) show a very wide distribution, even for this angle of distance. 3-way HiFi systems are an especially difficult case.

However, from the information is collected in that +- 5 degrees segment, it is possible to create usable corrections that can approximate the performance of studio monitors from HiFi speakers. And it wouldn’t be false to label this measurement an „on-axes”.

4. „What is the optimum frequency resolution for acoustical measurements?”

Measurements should be as accurate (high resolution) as possible. What are we trying to hide by decreasing resolution and finding an optimum? To make curves as „nice” as we would like to see and to lose important details?

It depends what we are willing to do – just talk about the curves or actually use them to make decisions – correction decisions – to create an equalizer.

The right way of making „beautiful” curves is to increase the time resolution respective to a decreased frequency resolution. It is emulating human sound perception that is insensitive to most soundwave interference because of a high time resolution.

Because of the high precision of current measurement systems it is nonsensical to decrease accuracy artificially. Let's do measurements the right way and exploit all the accuracy available in order to get incredible-sounding results.

The interference „noise" in an „ordinary" room (steadystate) measurement is so high that it masks serious problems even as serious as being out of phase on crossover frequency.

5. „At low frequencies adjacent boundaries (baffle wall, floor, nearby rear or side walls, etc.) contribute to what is measured and heard. These effects can be quantified by some number of in-room measurements and the appropriate equalization applied. A methodology is needed to address these phenomena ..."

As mentioned above, there is no problem measuring SPFR of such a „stack" of speakers and then equalizing it. This method has been known for at least 10 years...

6. „Is it necessary to compensate for high frequency air attenuation over the propagation distances in cinemas or do humans expect this?"

a) If we are working with a reinforced or recorded program (cinema) – yes, it will add a nice „air" and make the result more commercial.

But the compensation curve must very accurately follow the curve of real losses. This must be measured, not taken from some tables and parametric EQ used to compensate.

It is almost impossible to follow a simple falling curve by using classic parametric EQ.

For large and very reverberant halls, the hall sound absorption coefficient curve (derived from the reverberation time curve) should be used to introduce a correction that respects the sound energy losses in the hall, including air losses.

b) If we are working to reinforce a live performance – no. Because live sounds will be „processed" by air losses, and playback from the speakers should sound the same.

7. „It is evident that ear-level microphone locations experience elaborate seat dip and acoustical interference effects. ... Are there better ways to characterize the direct sound?"

Of course, this would be Sound Power Frequency Response.

8. „If bass management is selected as an alternative to the existing LFE configuration, how many subsystems, e.g., screen, side, rear, elevation channels, are necessary to avoid distracting localizations? What are the optimum crossover frequencies and attenuation slopes?"

Car sound enthusiasts have an answer to this question because they put subwoofers in the back of a car and use low and very steep crossover at 50 ... 60 ... 70 Hz at the cost of introducing a serious amount of delay caused by such low and steep crossover.

But, if we're talking about LF quality in reverberant halls, the directivity of the LF system must be increased by using multiple drivers 8 ... 16 of 15` ... 18` at each side of the stage instead of just using one subwoofer box, as is the usually case in cinema systems.

Simple, but at considerable cost ...

8. „DISCUSSION AND CONCLUSIONS"

It is unclear why we should predict room curves or only „fundamental aspects of room curves", if we can measure not only room curves (cited as insufficient for making decisions) but also take complete control of the performance of a loudspeaker in terms of amplitude/power and time/phase responses based on reliable field measurements?

I must disagree with the author that, if we put a loudspeaker in a real room/hall, we can't evaluate just that loudspeaker. The „room curves" as they are defined in the paper, cannot of course be used, but at the same time the author claims „... spatially-averaged measurement to reveal the underlying curve" without any reference to a source for such conclusions. But this is the basis of SPFR.

"A binaural human listener is vastly more complex and capable than an omnidirectional microphone and analyzer, and we are not close to having a computer equivalent."

The statement is much too general and does not reflect the current state of development of measurement systems that offers measurements of incredibly high time resolution for use in field applications. That would be high enough for such a system to emulate human perception in terms of high time resolution that causes insensitivity to interference.

„This is a logical parallel to common experience in live unamplified concerts, where humans are able to substantially separate the sounds of instruments and voices from the sounds added by the different venues in which they perform, even though the venue is truly part of the performance. If the goal is good sound it is hard to escape the notion that the starting point for a calibration scheme is free-field data on the loudspeakers."

I fully agree with this statement urging a focus on the loudspeakers. Especially as regards the use of SPFR.

But „free-field" data can be captured in a real room/hall. Especially if the loudspeaker is placed on a stand and a 3:1 law is in force.

The author proposes the use of "total radiated sound power" but with the disclaimer that it should be evaluated in an anechoic chamber. The fact is forgotten that it was exactly the opposite – a reverberation chamber, that was used for sound power evaluation. And, to use such a chamber with full accuracy, you should know and use the curves of the respective chamber.

Therefore any room or hall can be a reverberation chamber. Its curves will just be even more important.

But, as already mentioned, there is no need for an anechoic chamber. There are ready-to-use tools to measure SPFR in field applications in a very short time, to characterize the direct sound of a loudspeaker.

"Therefore, neither a steady-state room curve nor a direct sound curve can be a definitive descriptor of timbre for all programs in all venues at all frequencies."

I agree with this statement, especially because not even one of the mentioned curves complies with the repeatability requirement. And why should we express so much unusable information? And that is true not only for „all" but even for each particular one.

"Adjacent-boundary issues exist in all rooms. These will be revealed by spatially-averaged in-room measurements and they are responsive to equalization."

This is a very promising statement, however with no reference either to a source or to any practical use.

"periodic routine checks employing simple measurements can ensure the continued functioning of all the elements. Standardizing an anechoic data set for loudspeakers would be a significant simplification to the entire process."

As the repeatability of measurements proposed here is very low, the claim of „simplification" is simply not true. Any measured curve will be different even where no damage has happened to a loudspeaker system.

It is unclear why "standardizing an anechoic data set" would give any benefit.

"The starting point would be the delivery of an accurate, neutral, direct sound."

Yes, it would be. But why just only 'would be'?

„It means that a steady-state room curve should rise by some amount at low frequencies. It is a worthy topic for research."

Indeed, two cases should be discussed:

1) the manufacturer is making loudspeakers with flat SPFR and we will enjoy a natural rise in LF when the loudspeakers are placed in our room on the floor, near a wall or into a corner,

2) let's measure SPFR of the speaker in place (1 real and 7 imaginary for placement in a corner) and introduce correction and then add a target curve depending on our taste.

In his final conclusions the author has neglected to consider recording studios in his citing of places where loudspeakers are used. There is an explanation for this. The ideas proffered by the author are geared to the mass consumer rather than those working in studios. Every trial conducted was unsuccessful.

The recording studio is a critical and exacting place to test any ideas of loudspeaker measurements and correction. We need to use the double-blind method to find a mass consumer reaction to a new solution and to get it without bias. There is no need for that with studio engineers. They are capable of „absolute" judgment. How unfortunate that the author's paper is focused on the mass consumer's very incompetent valuation just with preference, while the professionals, who can describe the problems and changes in sound in a very detailed manner, are not invited to the test.

The most important skill for the sound recording/mixing engineer is to be free of any bias.

It is impossible to do the sound recording/mixing engineer's job well if one cannot be unbiased.

Therefore the opinion of an experienced sound recording/mixing engineer is as close as it comes to being objective in the absence of a blind test.

And their comment usually is not "this sounds better" but rather that "the problem here and other problem here have been removed."

The sound perception of the ordinary listener is adapting very quickly to any coloring present.

The trained sound recording/mixing engineer can keep his perception fresh for a much longer period of time.

But, still he must refresh it from time to time by listening to something neutral – even just the sound of rainfall on a windowsill.

Too much effort spent trying to determine the needs of ordinary customers could, I imagine, finally lead to some piano manufacturer abandoning all efforts to make good pianos that stay in tune, because a certain number of customers, may not be capable of detecting a slightly out of tune instrument when asked to do so in a blind test.

Thankfully this isn't happening and piano manufacturers understand that they are working for the discriminating piano player rather than for the average listener.

The same holds true with loudspeakers and the professionals who use them.

I find it unfortunate that the author, even while trying to refer to the current state of development, did not seem ready to suggest the available solutions in field applications for addressing some of the problems mentioned in his paper.

But experience has shown that the use of precise and unbiased measurement for the purpose of sound correction is the best method for understanding and judging the audio product.

Because if predistortions (correction, equalization), created from measurement, is not compensated by distortions in systems element's distortions (what we are trying to correct) we will get additional distortions - new problems.

„ ... and others are simply wrong."

It looks perhaps more like not so simply. No solutions have been proposed.

Raimonds Skuruls   2015.11.01

Acoustic Power Lab

2. Dr. Floyd Toole:

A response to Raimonds Skuruls' comments on my recent paper.

I thank Mr. Skuruls for his response to my paper. Dialog is needed on this topic, and that, in part, was a motivation to write it. I am accused of "being skimpy in proposing any real solutions". I think a cautious approach is necessary in a peer-reviewed paper that is proposing changes to long-established internationally-standardized measurements. In fact, several dimensions to the needed solutions are, I think, unambiguously identified in the paper, and others that need further research are noted. Even in 30 pages (!) something has to be left out. Most of the detailed explanations are in the references, especially in my 550-page book, and the numerous references therein. There is a lot of relevant data relating technical measurements to perceptions. We are close to being able to assert in unambiguous terms what needs to be done, but my "research scientist" caution inclined me to postpone that until some ongoing investigations are complete.

Mr. Skuruls is in the business of selling his devices and services to customers in the audio industry. As such he can choose to ignore the widely-used recommendations from the ITU, ISO and SMPTE that employ calibration methods based on steady-state in-room measurements and minimum-phase equalization. For decades past, and right now, all of these recommendations are being employed in calibrating movie sound dubbing stages and cinemas worldwide, and in setting up listening venues and facilities for broadcasting as well as university research. The widespread assumption is that they are all that is needed to ensure both good and consistent reproduced sound quality.

I am currently engaged in committees looking to update some of these recommended practices, employing improved measurement methods and applying objectives guided by psychoacoustic relationships learned from disciplined, double-blind, listening tests. All of these documents exist because of a belief that the quality of reproduced sound in recording and broadcast studios, dubbing stages, cinemas, and homes should be fundamentally good and similar. Mr. Skuruls appears to disagree, saying: "But it is a well-known fact that the "good" work of a recording engineer (producer) sounds good as a piece of art through any speaker that is used. Of course it is not the same "sound", but it is the same work of art." So, he asserts that the "circle of confusion" in Figure 1 of my paper is irrelevant – that it is sufficient to recognize the "melody", the "rhythm" and "lyrics" of a song, and that the bandwidth, spectrum, linear- and non-linear distortions and sound levels do not alter the "art". I beg to differ.

At several points in his comments it is asserted that my focus is on consumer products, that "professional" loudspeakers are inherently superior. I admire his faith, but numerous measurements in Section 2.4 and Chapter 18 of my book show that professional loudspeakers are as susceptible to design inadequacies, as are consumer loudspeakers. The best consumer loudspeakers and the best professional loudspeakers, as they measure and sound, are almost indistinguishable. The flawed ones exhibit an infinite variety of misbehavior, not all of which are capable of "correction" after the fact. Identifying and correcting the flaws is made greatly more difficult if the only data come from measurements in reflective spaces.

Mr. Skuruls claims that recording/mixing engineers are unbiased – "as close as it comes to being objective in the absence of a blind test". The listening ability of recording/mixing engineers was elaborately tested early in my career and published as "Subjective Measurements of Loudspeaker Sound Quality and Listener Performance", J. Audio Eng. Soc., vol. 33, 1985 (30 years ago!). In those tests, professional recording engineers and producers were mixed in with audiophiles as subjects in double-blind subjective evaluations of loudspeakers intended for use as broadcast/recording monitors. As a result of noting that several of the professionals were unable to repeat their subjective ratings in subsequent randomized presentations, a problem was revealed that is now widely acknowledged, but rarely discussed. Hearing loss is an occupational hazard in the audio business, and, as a result, the opinions offered by people so afflicted are less reliable, and may exhibit more bias than those from people with more normal hearing. The topic is also discussed in sections 17.4 and 19.1.2 in my book. Professional and "amateur" listeners with relatively normal hearing exhibited closely similar preferences in sound quality. Some of the highest scoring loudspeakers in those tests were consumer products. A couple of the recording engineers commented at the end of the tests that they had never heard such good sound before. They had rejected some of their previously favored monitors, even after some repeat tests using their own master tapes. Needless to say, the highly rated loudspeakers exhibited the least-flawed anechoic measurements, which was discussed in the paper, and others that followed.

Section 17.5 in my book shows how both professional and amateur listeners can be biased by visual information about the products they are listening to – we are all human; blind testing is important. Subjective opinions are the basis for evaluating any sound reproducing system, and controlling the variables, including who is doing the listening, is essential. See Figure 17.6 in my book, and Olive, S. (**2003**). "Difference in Performance and Preference of Trained versus Untrained Listeners in Loudspeaker Tests: A Case Study", J. Audio Eng. Soc., **51**, pp. 806-825. It shows that experience is a significant factor in terms of the consistency and range of sound quality ratings, but that, in the end, the relative ratings of the products are essentially identical for all listeners – that is unless one's "microphones" (ears) are damaged.

Mr. Skuruls is focused on performing useful in-situ measurements – not a bad thing – but this inclines him to downplay the usefulness of comprehensive anechoic data on loudspeakers. That is fine, but the reality is that anechoic data are required by the designers of the loudspeakers (if not, why not?) and as such should be available. These data can be elaborated to the level of the "spinorama" shown in Figure 20 in the paper. The frequency resolution is not limited by the measurement venue (time windowing), and is sufficient to reveal audible resonances over the entire audible frequency range: see Section 19.2.1 in my book and Toole, F.E. and Olive, S.E. (**1988**). "The modification of timbre by resonances: perception and measurement", J. Audio Eng. Soc., **36**, pp. 122-142. Such high resolution is "difficult" in non-minimum-phase rooms, so such problems may go unnoticed. The finding that the spectral "bump" is a more reliable indicator of audibility than time-domain ringing is of importance.

The calculation of a sound power estimate is a natural byproduct of doing full anechoic orbits of frequency-response measurements, and is more accurate than a reverberation chamber estimate for devices that are strongly directional, like loudspeakers (at the NRCC I had access to both kinds of chambers). The frequency resolution of the sound power calculation is that of the basic measurements – in this case 1/20-octave – which helps in identifying the presence of resonances – see Figure 21 in the paper.

I could go on, but all that I would say is either already in the literature, or awaits the results of the psychoacoustic investigations needed to fill in the few blanks left in the Journal paper that provoked these remarks. It is hoped that a new generation of peer-reviewed sound quality recommendation documents will emerge at some point. Then Mr. Skuruls may see the "real solutions" he desires – I was not being deliberately skimpy. In the meantime he has created his own solution, described on his website.

Floyd Toole, Nov. 15, 2015

3. Raimonds Skuruls:

A response to Dr. Floyd Toole`s answer to Raimonds Skuruls comments.

„Cautious approach" – it is a usual thing for a serious sound engineer to make same change and check result. It is true for mixing job and system tuning as well.
The mixing job is about training the decision making skills and ability to keep fresh hearing and thinking.

„My "research scientist" caution" – sound engineers in studio and concert are trained to make decisions (usually artistic) with certainty. The „caution" should be lived in rehearsal time. The certainty must be in use when public performance takes place and any caution should be removed by appropriate work done beforehand.

„He can choose to ignore the widely-used recommendations from the ITU, ISO and SMPTE that employ calibration methods based on steady-state in-room measurements and minimum-phase equalization."-
Sorry, incorrect. It is not chosen by me. They are my customers who decide to use the tuning and calibration offered. And they results (usually 2 times less work time to finish the work) are encouraging me to keep doing what I am doing.
„Minimum-phase equalization" – sorry, I did not know any tool offering that (except APL). Please correct me if I am wrong!

„long-established internationally-standardized measurements" – Might be they are standardized but do not have force as a law and usually are not in use because the complexity and lack of reliable instruments, repeatable results and laziness of technicians as well. The example, with a cinema operating company FK that had 13 cinemas with 200 screens, is very enlightening. Yes, they used „steady-state measurement" and nevertheless were not able to see out of phase between sub and mids in the interference „noise". That was nothing close to meeting a standard and having any control on system's performance.
At the same time, Hollywood based recording studio ("The Lord of the Rings", "The Golden Compass") step by step equipped all three of their mixing rooms with products that meet producers requirements (the standard also). Without saying, their monitor systems were not the cheapest ones.

„The widespread assumption is that they are all that is needed to ensure both good and consistent reproduced sound quality."
Yes – still an assumption ...

The offer to use an accurate and effective, in time consumption, measurement of Sound Power Frequency Response are not in contradiction to „long-established internationally-standardized measurements". Especially, when the understanding of need to use multiple measurement points to get some „the Average" measurement with recommendation to do that in power domain with a substantiation that it just would be nice and with research how many points is needed to get some accuracy level (because each point is expensive ...) is taking place.
Let's add new level of accuracy by using a solution and tools already offered for 10 years...

Strange, but no one was offering me to take a look on results acquired by use of „internationally-standardized measurements" and much more – to compare that results with results gained from use of SPFR measurement. It's never too late, let's do that!
„... learned from disciplined, double-blind, listening tests."
If it is possible to avoid any biasing for some researchers only by use of such tests (with respective budget). Ok, let them find such budgets and perform that.
As it was expressed in previous comment, the use of high skills and competence is most important for that than use of masses.
It is not so big a problem to hide the test object from the expert.

„ ... that it is sufficient to recognize the “melody”, the “rhythm” and “lyrics” of a song, and that the bandwidth, spectrum, linear- and non-linear distortions and sound levels do not alter the “art”. I beg to differ.” - understand

The “melody”, the “rhythm” and “lyrics” of a song are „properties” of a live performance. When such a performance is recorded to a carrying media it becomes an „image” of that live performance and will be a new object of art as it is in analogy with optical photography.
If such an image is artistic and rich, those properties will not be lost even by playback of such image (actually recording) by very low quality equipment.
If recording sounds well only on system with wide bandwidth and low distortions, it was badly recorded and the recording engineer was having fun from wide bandwidth and absence of distortions and does not understand the fact that his work has a low artistic value.
Engineers who know that, are placing some crap loudspeakers (narrow bandwidth, not even FR, non linear distortions) next to main monitors and switching to those from time to time to avoid becoming blind of „nice” sound of main monitors. It is needed less and less when enough experience and artistic skills are collected.
But, one very important aspect is that it’s possible only when main monitors not only have wide bandwidth and are free from non linear distortions but their FR is set to some very well equalized target – no sound coloring by monitors. Usually it is fall of 7 dB from LF to HF.
Otherwise, the engineer is fighting with monitoring problems and is loosing focus on his artistic work.
That is creating a curious fact for some engineers –the work created by use of ideally tuned monitors will sound good in any place with crap loudspeakers. The idea that engineer should use crap loudspeakers for his main work is completely wrong.
Same is with the idea that FR accuracy of +- 3 dB (400% error in power domain) is enough for monitoring. The accuracy of sound engineer’s decisions is in 1...2 dB (not 6) therefore the accuracy of +- 0.5 dB should be used. That lets the engineer to do his job (making decisions) even 2 times faster than usually ...

„At several points in his comments it is asserted that my focus is on consumer products, that “professional” loudspeakers are inherently superior.”
It would be nice if they would be somehow inherently superior. But we must make them superior by our respective attention. They are two completely different things – to listen for an enjoyment and to listen for a creative work.

„The best consumer loudspeakers and the best professional loudspeakers, as they measure and sound, are almost indistinguishable.”
- It is an injury against professionals.

„The flawed ones exhibit an infinite variety of misbehavior, not all of which are capable of “correction” after the fact. Identifying and correcting the flaws is made greatly more difficult if the only data comes from measurements in reflective spaces.”
- it takes a different approach but it is solved already.

„The listening ability of recording/mixing engineers was elaborately tested”
Looks like the 30 years old experiment was developed in a flawed way.
It was a listening test. It should have been a working test.
The ultimate accuracy of monitor speakers is showing its importance only in work process. The quick „listening to” by use of some unfamiliar program is not showing value of accurately tuned monitors instantly. The recording engineers usually report their impressions next day after they get new quality in their mixing room. Usually, as a sign of 2 times more effective working.

„In those tests, professional recording engineers and producers were mixed in with audiophiles as subjects in double-blind subjective evaluations of loudspeakers intended for use as broadcast/recording monitors.
As a result of noting that several of the professionals were unable to repeat their subjective ratings ... ”
- Why focus on „several who were unable”? There were several who were able, right?
Looks like to prove the idea that loudspeakers must serve requirements of 90% of public.
The requirements of professionals „who are able” are not important in such a case ...

„ ... shows how both professional and amateur listeners can be biased by visual information about the products"

I see Dr. Toole expression on blind test as most important to make any decisions.
I made a lot of test that should be named as blind. But most importantly, they were sufficient to me to keep working in this field and keep serving very critical demands of professional sound engineers and see their happiness. :)
It means, my tests were enough objective to not to make mistakes and to make future investments. It is possible to conduct any new tests to secure any investment right now.

Some examples of tests, collected in 10 years, which can not be called as not being blind.

1. Daughter is playing with dolls next room. I asked, let's listen to music. But two versions - A and B. Did you here difference? -Yes I did! What kind of difference you hear? -The loudspeaker disappears in version B! How you can characterize version A? -Everything is too much ...

2. The loudspeaker manufacturer RRR is using Sound Power measurement (instead "on axes" measurement in anechoic chamber) for final tuning of their loudspeaker M1. They decided to participate in a contest conducted by magazine S. Magazine has chosen 7 brands (including RRR with their M1) for that contest. Results are published as broad report on 15 pages with measured curves included. The M1 gets second place but without any negative comment about it's sound. Most important part of the comment on sound is "it is not understandable why it sounds so good". The speaker Y gets first place but with negative comment on its sound, being 2 times more expensive and 2 times heavier (biasing by sponsoring? :)). A lot of other participant's curves are like drawn with a ruler. Curves of M1 are not such, therefore - not understandable for magazine's experts.

3. Peter was making sound for an event. TV channel was making reportage from this event.
John is a competitor of Peter and is seeing and listening to that reportage. Then, when he meets Peter the first time, he says - it was a good idea to give TV a line signal from your mixing console. - No, TV was capturing everything by their camera mic, Peter answers.

4. Brant is a mixing engineer and serving his producer George abroad.
Brant decides to apply equalizing on his monitors J and is keeping working for George. Some time after that George is asking Brant - what is happening Brant, why your mixes get so good sound now?

5. Car audio competitions - judges are usually not biased by equipment installed on cars they are judging.

6. When rehearsal of an event was going on, some performers went down from stage, listened and turned to sound engineer asking - why FOH is not working!?! Only the MUTE of all FOH proved the opposite fact. That was happening many times.

Let's offer to „standard committee" tools that are ready to use by ordinary technician not only for initial fine tuning but also for regular check because of high repeatability of such measurements and insensitivity to kind of operator's behavior!

What about such experiment?
Let's ask experts - how much sound coloring still remains in loudspeaker sound in comparison with live sound, instead of question – which speaker do you prefer?

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Acoustic Power Lab