Validation of lateral fraction results in room acoustic measurements

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ABSTRACT

The early lateral energy fraction (LF) is one of the most important acoustic descriptors of concert hall sound. This paper describes a procedure for validating LF measurements of halls. The measurement of LF can be problematic due to uncertainties with calibrating, aligning and aiming two microphones of different sensitivities and directivity patterns. The validity and reproducibility of measured LF values needs to be established. A series of simulated sound fields consisting of the direct sound and a single reflection in an anechoic chamber were used to validate LF measurement systems. The reflection was varied in angle and level, and the measured LF values were compared with the known or calculated values. Two commercially available measurement systems were validated and the measured results were generally within one JND of the calculated values.

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1. INTRODUCTION

The terms “spatial impression” and “early lateral reflections” are ensconced in the terminology used to describe the acoustics of concert halls. In the late 1960s, through listening experience and research, Harold Marshall discovered that early reflections arriving from lateral directions created a desirable sense of spaciousness (1). This phenomenon, which he originally called “spatial responsiveness” (later, spatial impression), was then extensively investigated by Michael Barron in his PhD thesis. Later, Barron and Marshall derived the “early lateral energy fraction” (LF) as a linear measure of spatial impression (2). Recent research by Pätynen et al. (3) has also established that the perceived dynamic range is enhanced when the room geometry provides strong lateral reflections. LF has become one of the most important acoustic descriptors that correlates highly with subjective listener preference for concert hall sound. Today the term spatial impression refers to two subjective effects: apparent source width, and listener envelopment. The first corresponds to Marshall and Barron’s work, while listener envelopment is related to the level of the late lateral sound energy (4).

LF is defined as the linear ratio of the lateral early energy to the total early energy. Barron and Marshall found, through subjective listening tests with a simulation system, that the degree of spatial impression was maximised when the sound arrived side on to the listener and zero when the sound arrived from the direction of the source. The test results showed a correlation with $\cos(\theta)$, where θ was the angle between the lateral reflection and the axis through the ears. LF is generally measured from impulse responses obtained using a cosine or “figure-of-8” microphone (to measure the lateral energy) in conjunction with an omnidirectional microphone (to measure the total energy) (5).

$$\mathrm{LF} = \frac{\int_{0.005}^{0.080} p_{\mathrm{L}}^2(t)dt}{\int_{0}^{0.080} p^2(t)dt} \qquad (1)$$

where $p_{\mathrm{L}}(t)$ is the impulse response signal measured with a figure-of-8 microphone, $p(t)$ is the signal measured with an omnidirectional microphone and the null of the figure-of-8 microphone is pointed towards the source.

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Recent measurements in a 2300 seat hall using two different acoustic measurement systems showed quite different values of LF for the same source receiver locations. Some years ago, laboratory validation of a commercially available reverberation time measurement system revealed a 10% error due to a software oversight. These experiences lead the authors to devise a method for laboratory validation of LF measurement systems. The authors proposed that a known physical relationship should be set up and measured to confirm that any system is measuring the correct LF values.

2. METHODOLOGY

2.1 Concept

The technique to validate measurements of lateral fraction involved simulating a series of simple sound fields consisting of the direct sound and a single reflection in an anechoic chamber. The reflection was varied in angle and level, and the known or calculated LF values were compared to values measured with different room acoustics measurement systems. Two commercially available measurement systems were tested.

2.2 Sound Field Simulation Setup

The simulations were conducted in the anechoic chamber at the Acoustics Centre of The University of Auckland. This room is fully isolated with internal dimensions of 5 m x 5 m x 5 m (wedge-tip to wedge-tip), and an acoustically transparent wire mesh floor is suspended above the wedges at the base of the chamber.

Two identical loudspeakers, Tapco S5 studio monitors, were placed inside the chamber to simulate the direct sound and a reflection, as shown in Figure 1. These were placed at a height of 0.7 m above the mesh floor and at a distance of 2.4 m from the receiver position. The first loudspeaker (labelled “Direct Sound”) was fixed at the 0 degrees position and the second (“Reflection”) was placed at different angles as shown in the figure. The idea was to generate the direct sound at 0 degrees and a single reflection at angles of 0, 15, 30, 45, 60, 75 and 90 degrees. At each angle the reflection was set to arrive 40 ms after the direct sound, and at levels of -3 dB and -6 dB relative to the direct sound level. The delay time and the reflection level were chosen as they are typical of early reflections in a real concert hall.

A sound field simulator was responsible for driving the two loudspeakers in real time. This consisted of a USB audio interface (RME Fireface UFX) connected to a computer running Adobe Audition software. The simulator accepted the mono output from the measurement system under test and routed this directly to the direct sound loudspeaker. A copy of this signal was delayed and attenuated before being sent to the reflection loudspeaker. Note that for the 0 degree reflection, the delayed signal was played back through the direct sound loudspeaker.

A total of 14 measurements were conducted for each system under test; 7 reflection angles and 2 reflection levels.

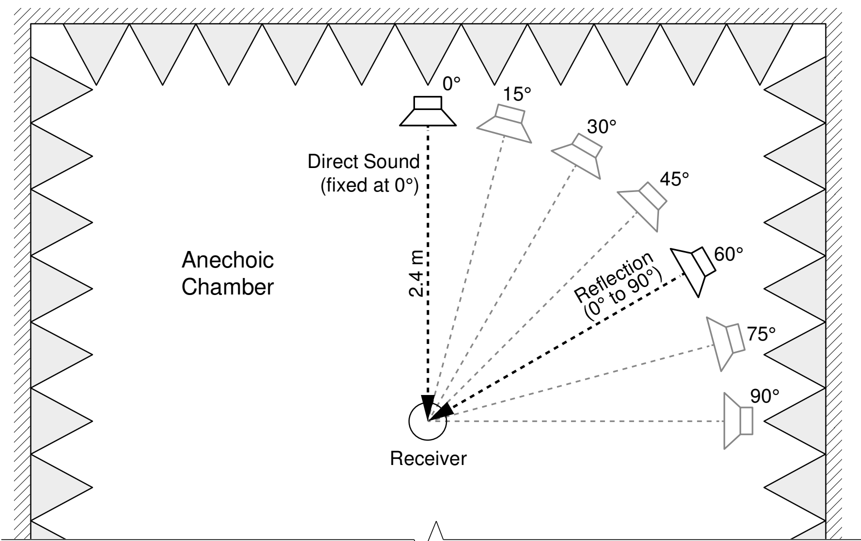

Figure 1 – LF sound field simulation setup in the anechoic chamber

2.3 Calculated LF

The actual LF value of each sound field simulation was calculated using a combination of theoretical and experimental data. This process measures the actual levels from each loudspeaker at the measurement position and then calculates the LF based on the known level and reflection angle as described below:

1. For each different sound field configuration a reference pressure microphone (Brüel & Kjær Type 4007) was placed at the receiver position and the impulse response was recorded. The excitation signal was a logarithmically swept sine signal of 10 seconds from 20 Hz to 20 kHz, followed by a short silence. The recording was conducted using a MOTU 4pre and Adobe Audition software, and converted to an impulse response using custom software routines.
2. The resulting broadband pressure impulse response signal was divided into two segments of 40 ms in length, one encapsulating the direct sound, and the other encapsulating the reflection.
3. The RMS levels of the direct sound and reflection were calculated in four octave bands: 125, 250, 500 and 1000 Hz.
4. The LF was calculated in each band by:

$$\mathrm{LF}_{\mathrm{calculated}_b} = \frac{\left[p_{\mathrm{reflection}_b} \sin(\phi)\right]^2}{{p_{\mathrm{direct}_b}}^2 + {p_{\mathrm{reflection}_b}}^2} \qquad (2)$$

where p_{direct} and $p_{\mathrm{reflection}}$ are the RMS levels in band b for the direct sound and reflection, respectively. The angle ϕ is the reflection angle relative to the direct sound in degrees. The single number value for the calculated LF is the arithmetic average of the four bands above, as defined by ISO 3382-1:2009 (5).

This procedure was repeated for every change in configuration of the sound field and measurement system under test.

2.4 IRIS System

The first system to be tested was IRIS, an integrated hardware and software room acoustics measurement system developed by Marshall Day Acoustics (6).

IRIS is distinctive in that it can measure impulse responses in 3-D. IRIS utilises a compact tetrahedral microphone array, a Core Sound TetraMic (Figure 2), which is able to resolve incoming sound in terms of level, time and direction. The microphone array interfaces to a PC running the IRIS software via a MOTU 4pre. The 4pre is also used for playing back the stimulus.

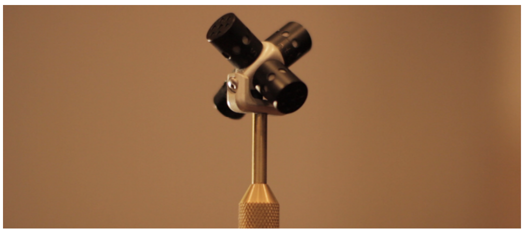

Figure 2 – The IRIS microphone array, a Core Sound TetraMic.

The directional ability of IRIS means it is relatively straightforward to calculate LF. The output of the microphone array, after post-processing, is a coincident set of virtual microphone patterns – a pressure microphone (omnidirectional pattern) and three velocity microphones (figure-of-8 pattern) arranged in the X, Y and Z directions. In Ambisonics this is known as first order B-format. From these signals it is possible to derive the outputs of any simple microphone pattern in any direction. In the case of LF two signals are required, the signal from an omnidirectional microphone and the signal from a figure-of-8 microphone in the lateral direction.

IRIS has two advantages over the traditional dual microphone LF measurement techniques. First, a user does not need to be concerned with matching the sensitivities of multiple microphones. The TetraMic is calibrated by the manufacturer and this data is taken into account in the IRIS software. Second, there is no need for accurately aiming the TetraMic as long as it is in the upright position. The software indentifies the direction of the direct sound using a sound intensity technique, and then synthesises a horizontal figure-of-8 microphone with the null in the direction of the direct sound.

The IRIS system was configured to output a logarithmically swept swine stimulus of 10 seconds, from 20 Hz to 20 kHz, followed by a short period of silence. The microphone array was placed in an upright position with its X-axis pointing approximately towards the direct sound loudspeaker. A photograph of the microphone array in the simulation system is shown in Figure 3 below.

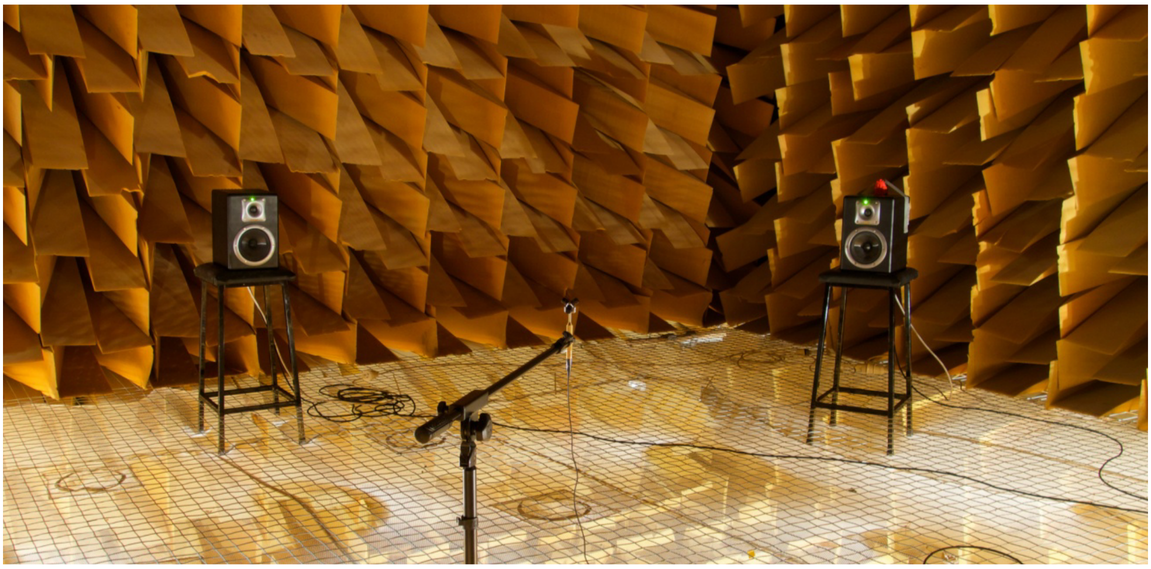

Figure 3 – The IRIS microphone array set up in the simulation system

2.5 System 2

A more traditional acoustic measurement system using dual microphones for measuring LF was also tested. This consisted of commercially available acoustic measurement software with two commonly used microphone arrangements. For the purposes of this paper, the software will be referred to as “System 2”.

The first microphone arrangement consisted of an omnidirectional microphone (Brüel & Kjær Type 4007) coupled with a figure-of-8 microphone (AKG C414 B-ULS). A photograph of this combination is shown below in Figure 4. The second arrangement used a single switchable pattern microphone (AKG C414 B-ULS) to record the omnidirectional and figure-of-8 signals in a dual-pass measurement. System 2 was set up to play back a similar logarithmically swept sine excitation signal to the IRIS system, in terms of frequency range and duration.

Figure 4 – The first microphone arrangement for “System 2”: a Brüel & Kjær Type 4007 pressure microphone (top) with an AKG C414 B-ULS microphone set to a figure-of-8 pattern (bottom). This is setup in a reverberation chamber at The University of Auckland for a calibration measurement.

The sensitivities of the microphones, for both arrangements, were matched using a diffuse field calibration method. One of the reverberation chambers at the Acoustics Centre of The University of Auckland was used for this purpose (see Figure 4). This room has a volume of 202 m^3 and a reverberation time ($T_{30,\mathrm{mid}}$) of 6.9 seconds. A total of six measurements were conducted for each microphone configuration, using two source positions and three receiver positions. The stimulus was a full range (20 Hz to 20 kHz) logarithmically swept sine signal of sufficiently long duration to capture all the decaying components in the space, played back through a Tapco S5 studio monitor loudspeaker. System 2 provided the functionality to process these diffuse field measurements and apply the resulting calibration data to subsequent measurements.

3. RESULTS

3.1 IRIS System Results

Table 1 gives a summary of the results of the IRIS test. The LF values are given as single numbers (average of the four octave bands from 125 Hz to 1 kHz), for each angle and level of the reflection. The error columns give the difference between the measured and calculated values. Note that the error values are calculated from the raw data, not the rounded values as displayed in other columns of the table.

The measured LF values are slightly lower than the calculated values, with a maximum absolute error of 0.02 from the calculated values. This is well within the just-noticeable-difference (JND) for LF of 0.05. The absolute error appears to increase with increasing angle and level of the reflection.

Figure 5 plots the measured LF values against the calculated, for all angles and levels. The dashed line corresponds to the calculated values and the solid line gives the measured values. The measured values track the calculated values closely, but deviate slightly as the LF increases (i.e. with increasing angle and level of reflection).

Table 1 – IRIS LF results compared to calculated values

	Reflection at -3 dB			Reflection at -6 dB		
Angle	Calculated LF	Measured LF	Error	Calculated LF	Measured LF	Error
0	0.00	0.00	0.00	0.00	0.00	0.00
15	0.02	0.01	-0.01	0.01	0.01	0.00
30	0.08	0.06	-0.01	0.04	0.04	-0.01
45	0.15	0.14	-0.01	0.09	0.08	-0.01
60	0.22	0.21	-0.02	0.13	0.12	-0.01
75	0.28	0.27	-0.02	0.17	0.16	-0.01
90	0.31	0.28	-0.02	0.18	0.17	-0.01

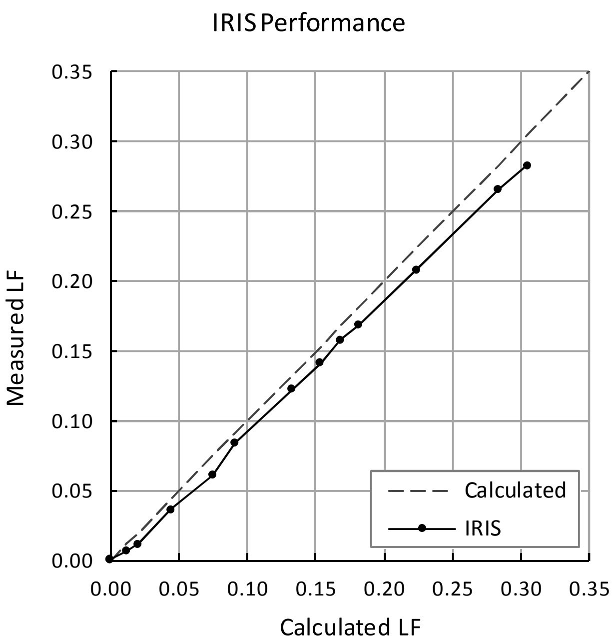

Figure 5 – LF values measured with IRIS plotted against calculated values

3.2 System 2 Results

The results for the test using the Brüel & Kjær Type 4007 (omnidirectional) and AKG C414 B-ULS (figure-of-8) microphones with System 2 are given in Table 2. The maximum absolute error is 0.06, just outside the JND of 0.05. The measured LF values are plotted against the calculated LF values in Figure 6 below (trace labelled "B&K + AKG").

Table 2 – System 2 LF results using Brüel & Kjær Type 4007 with AKG C414 B-ULS

	Reflection at -3 dB			Reflection at -6 dB		
Angle	Calculated LF	Measured LF	Error	Calculated LF	Measured LF	Error
0	0.00	0.00	0.00	0.00	0.00	0.00
15	0.02	0.02	0.00	0.01	0.01	0.00
30	0.08	0.07	-0.01	0.05	0.04	0.00
45	0.16	0.13	-0.03	0.09	0.08	-0.02
60	0.23	0.19	-0.04	0.14	0.12	-0.02
75	0.28	0.23	-0.06	0.17	0.14	-0.03
90	0.30	0.25	-0.06	0.18	0.15	-0.03

The results for the test with the single AKG C414 B-ULS microphone and switching directivity patterns in a two-pass measurement are listed in Table 3. The maximum absolute error is 0.04, just within the JND. These results are also plotted in Figure 6 (trace labelled "AKG").

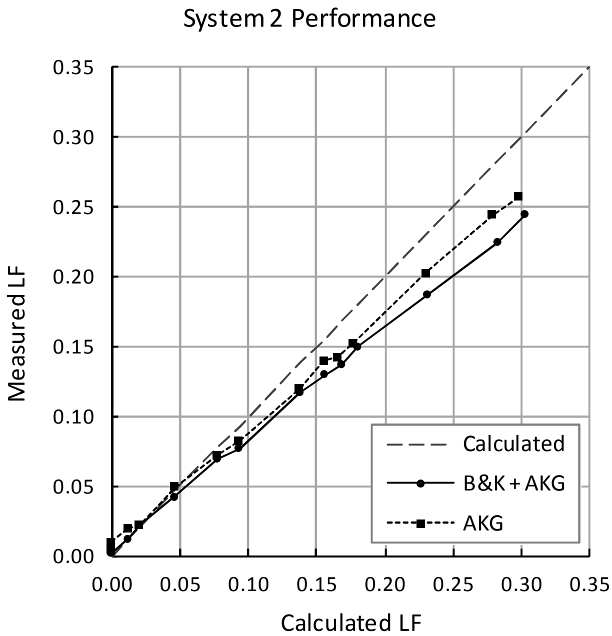

Figure 6 – LF values measured with System 2 plotted against calculated values

Table 3 – System 2 LF results using AKG C414 B-ULS in a two-pass measurement

	Reflection at -3 dB			Reflection at -6 dB		
Angle	Calculated LF	Measured LF	Error	Calculated LF	Measured LF	Error
0	0.00	0.01	0.01	0.00	0.01	0.01
15	0.02	0.02	0.00	0.01	0.02	0.01
30	0.08	0.07	-0.01	0.05	0.05	0.00
45	0.16	0.14	-0.02	0.09	0.08	-0.01
60	0.23	0.20	-0.03	0.14	0.12	-0.02
75	0.28	0.25	-0.03	0.17	0.14	-0.02
90	0.30	0.26	-0.04	0.18	0.15	-0.02

4. DISCUSSION

For each test, the measured LF values appear to fall on a relatively straight line when plotted against the calculated values, as seen in Figures 5 and 6. The line slopes vary with each test.

In each experiment the measured LF values equal the calculated values for a calculated LF of 0. The attenuation at the figure-of-8 null is significant, i.e. for a reflection at angle $\theta = 0°$, $\cos(90° - \theta) = 0$. This would have more of an effect on the results than mismatched microphone sensitivities. At 90° the figure-of-8 microphone is at its maximum sensitivity and any errors in the matching of microphone sensitivities would be more noticeable here. In this way, the slope of the line would be partly determined by how well matched the two microphones are.

The excellent results from the IRIS system imply the TetraMic has been well calibrated by the manufacturer.

The different microphone combinations with System 2 yield slightly poorer results than the IRIS system, but still very good. The difference in results between the two microphone arrangements for System 2 is probably related to the physical spacing of each arrangement. Using the single AKG microphone and switching patterns in a two-pass measurement would result in a more coincident set of signals compared to two different microphones placed close together. As expected, the results from the single AKG microphone are slightly closer to the calculated values. The downside is that a two-pass measurement takes twice as long to complete.

The fact that each curve in Figures 5 and 6 is mostly straight implies the geometrical setup of the simulation system and the directional responses of the microphones were mostly correct.

In practice, using a reverberation chamber to match the microphone sensitivities (System 2) is not always feasible and a more common approach is to perform an in-situ free field calibration. It would seem that this is more susceptible to errors than a diffuse field calibration, but this was not tested.

The validation procedure discussed in this paper requires the use of an anechoic room, which may not be accessible to some practitioners and researchers. Further work is necessary to consider how this might be developed into a more practical procedure for use in the field.

5. CONCLUSIONS

This paper has described a technique to validate measurements of the early lateral energy fraction using simulated sound fields. The calculated values of each sound field were compared to the measured values to assess the LF performance of the measurement system under test. Two commercially available measurement systems were tested. The IRIS system, which uses a 3-D microphone array, was able to measure LF accurate to within half a JND of the calculated values. Another commercially available system, which uses two microphones to measure LF, was also tested. This system also gave good results which were mostly within one JND of the calculated values.

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