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A System for Multi-Channel and Binaural Room Response Measurements

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ABSTRACT

The development of a system for room acoustical measurements and analysis is described. The goal of the project was a versatile system for multi-channel and binaural investigations of room and concert hall acoustics. It consists of a portable PC-based workstation with multi-channel AD/DA data acquisition, an omnidirectional sound source, a 3-D microphone grid for directional response registration, a dummy head, standard omnidirectional and cardioid microphones, and MLS-based response computation. In addition to traditional room-acoustical attribute analysis, special algorithms have been developed to investigate the time-frequency behavior of responses in different directions, for example in concert halls. Analysis cases of interesting hall measurements are discussed.

0 INTRODUCTION

The measurement of room-acoustical responses and analysis of related attributes is an important task in audio and acoustics. Several commercial systems are available for the task. If special properties are required, not many systems meet requirements such as good portability, multiple input and output channel analog data acquisition, full set of room-acoustical attributes analyses, binaural dummy head and "real head" measurements and analysis, easy postprocessing of data, as well as open extensibility and programmability for research purposes. In this paper we describe the development of a system that meets these criteria.

The engine of the IRMA (Impulse Response Measurement Application) system [1] was decided to be a portable rack-mounted PC with multi-channel AD/DA converters. In a normal configuration there are 10 input and 2 output channels, but the number of channels can be increased if needed. The impulse response measurement software is based on the maximum length sequence (MLS) method [2], although other test signals can also be used. The entire software system can be controlled and exteded using Matlab [3].

An essential requirement was the ability to perform multichannel measurements. This allows for using different transducers, such as multiple microphones, microphone grids, a dummy head, even simultaneously. It is also convenient to use at least two sound sources such as an omnidirectional loudspeaker and a subwoofer (not necessarily at the same time), so that switching cables is not needed.

The real interest in multichannel features was to develop microphone grid techniques to capture directional responses in rooms and concert halls. For this purpose a 3-D grid with 12 microphones was constructed. It consists of three pairs of small electret capsules, each pair with 1 cm spacing, in x-, y-, and z-coordinate directions, and another similar set of pairs with 10 cm spacing, all pairs concentrically positioned. This microphone grid probe is used for directional room response measurements and analysis.

Standard room acoustical attributes are measured using an omnidirectional capsule, and a stereo microphone with a pair of cardioids for the determination of lateral energy. For binaural measurements, the setup contains a Cortex dummy head.

A standard part of the analysis software includes routines for traditional room acoustic attributes. More interesting from a research point of view are the extensions for directional room response analysis. For example in concert hall measurements it is interesting to study not only the overall response at a specific seat point but also to know the direction of wave component flow in different directions. This makes it possible to see if sound is propagating from the stage to the audience or vice versa, or to isolate discrete echoes and their direction of arrival, as well as to obtain the lateral sound field.

1 THE MEASUREMENT SYSTEM

The measurement system consists of the IRMA system together with a number of transducers and related hardware. A general layout of the system is shown in Figure 1.

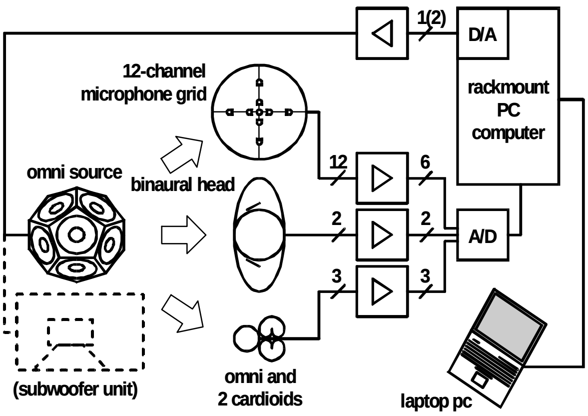

Fig. 1. The measurement system hardware layout.

1.1 The IRMA measurement system

The IRMA measurement system was put together as a part of this work. The system is built upon a Matlab-based application for response measurement and analysis, running on a portable measurement computer with multichannel audio hardware.

Several key points have been concentrated on in the system integration: an open architecture in both the hardware and the software implementations, practical suitability for complex multichannel field measurements such as concert halls, and the possibility of expanding and adapting the system to a wide range of measurement applications. The system was built using ordinary computer audio and sound studio equipment, which effectively reduced the hardware costs as compared to the price of specialized and proprietary systems.

An emphasis has been made on the adaptability and expandability of both the system hardware and software. Many commercial systems suited for long impulse response measurements under field conditions are currently either limited to one or two channels, or exhibit a closed architecture.

The IRMA system hardware

The IRMA system is built around an industrial PC computer and a flat panel display, housed in a 19" rack case and equipped with a suitable multichannel PCI sound card. In addition to the computer unit, the main measurement system includes an external 8 channel A/D converter, a power amplifier and a connector panel in the same rack..

The sound card provides two channels of analog input and output, digital S/PDIF input and output, as well as digital 8 channel ADAT input and output connections for connecting external A/D and D/A converters. The sound card is available to the system as a number of standard Windows two channel sound devices, all of which can be accessed simultaneously. Interchannel synchronization required for multichannel measurements is provided in the sound hardware. Clock sync connections enable the use of a single master clock signal for all connected digital audio equipment, thus avoiding jitter problems.

For convenient field use, a separate laptop computer can be used for operating the measurement system with remote software over a standard 100 Mbit/s network connection. All data acquisition, processing and storage are still handled by the rack computer, with the laptop only used as a remote mirror of the rack computer's display and input devices. This solution enables operating the measurement system at a distance from the physical rack unit, with only a single twisted pair network cable of optional length attaching the units.

The IRMA system software

Custom software was written for the IRMA measurement system, including Matlab functions for creating stimuli, performing measurements, post processing, filtering and analyzing responses, as well as a graphical user interface for operating the main features in Matlab. A full duplex multichannel sound card handler was written in C as a separate executable which is called by the Matlab system.

A functional layout of the IRMA measurement software is shown in Figure 2. The measurement functions include stimulus generation and response acquisition, as well as special functions for compensating and processing acquired responses prior to acoustical analysis. Standard octave and third-octave filtering is included, as well as a set of functions for calculating standard room acoustic attributes as defined in ISO 3382 [4].

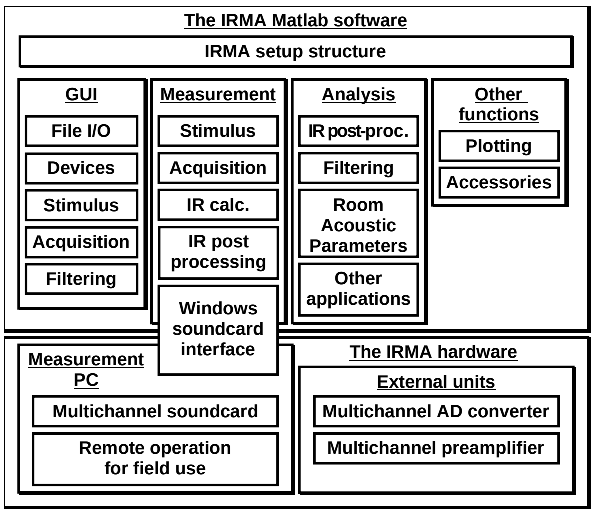

Fig. 2. A functional layout of the IRMA measurement system.

Response measurement methods

A variety of methods may be used for measuring impulse responses. The MLS method gained focus, as it is widely used in room acoustic measurements due to its high accuracy, speed and applicability to the measurement of room acoustic attributes. However, the IRMA system supports impulse response measurements using any arbitrary stimuli and FFT deconvolution instead. This is of practical importance, when electroacoustical systems with nonlinear or time variant properties are involved. Even in these non-LTI cases, variants of the MLS method may also be used [5, 6].

1.2 Transducers and related hardware

The multichannel capabilities of the IRMA measurement system enable a large number of transducers to be simultaneously used. These include a 3-D microphone probe, a custom preamplifier, an omnidirectional loudspeaker, a binaural head as well as standard omni and cardioid microphones.

Omnidirectional and cardioid microphones

A standard omnidirectional microphone and a stereo microphone consisting of a pair of closely spaced cardioids were included in the hardware setup for reference. The cardioid pair can be used for lateral energy measurements, as the sum and difference of the signals result in omnidirectional and figure-of-eight directivity patterns.

Binaural microphones
For binaural measurements, the setup contains a Cortex dummy head. Real-head recordings can also be done with a pair of ear-canal microphones.

3-D microphone probe
In order to acquire sound directivity information in the halls to be investigated a 3-D sound intensity probe was designed. Figure 3 shows a picture of the construction.

Six pairs of miniature electret microphone capsules were mounted and soldered on two wire grids. The grids were tightened on a pair of circular frames, which were attached together. The inner microphone pairs are set with a spacing of 10 mm between capsules for high frequency use (700 Hz – 7 kHz). The outer pairs are spaced with 100 mm between capsules for low frequencies down to about 100 Hz.

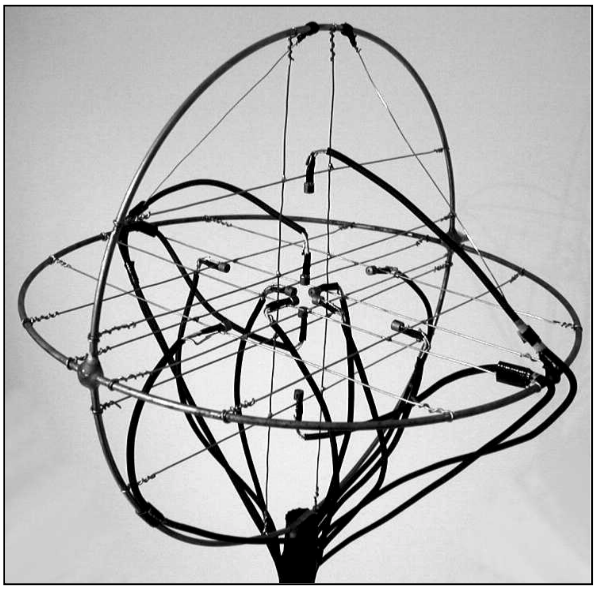

Fig. 3. A picture of the 3-D microphone probe.

Custom preamplifier
A 12 channel microphone preamplifier was specially designed for the probe capsules. It consists of 12 inputs (one for each of the electrets of the probe) and 12 balanced XLR outputs. Gain settings ranging from 20 to 50 dB in 10 dB steps are possible for the whole 12 channels simultaneously. Furthermore, it is possible to individually attenuate each input channel by 20 dB.

Omnidirectional loudspeaker
The hardware setup includes an omnidirectional loudspeaker, constructed in a plywood dodecahedron of 26 cm diameter, containing 12 driver elements. It covers the frequency range of about 100 Hz to 10 kHz. A subwoofer can be added or used separately for the lowest frequencies.

2 ANALYSIS

The IRMA system has been employed in analyzing response data, with a focus on novel methods for studying the directional information of the sound field in concert halls. As the measurement system is based on the Matlab environment, analysis may be performed both in-situ on the measurement computer, as well as later using only the IRMA software together with arbitrary additional Matlab code.

In this section, both standard and directional response analysis methods are considered, preceded by several issues on compensating unidealities inherent to the acquired responses.

2.1 Transducer compensation

Both the omnidirectional sound source and the 3-D probe exhibit unidealities, which should be compensated for prior to response analysis. The properties of these custom transducers were first verified by laboratory measurements.

3-D microphone probe
The probe's properties were verified by performing free-field measurements on the sensitivities and impulse responses of each of the 12 electret microphone capsules in the probe. The custom microphone preamplifier and the IRMA system's multichannel A/D converter unit were included in the calibration measurements.

The calibration measurements were repeated after the field measurement tournee to ensure that the probe exhibited stable acoustical properties. The calibration data has been used for compensating acquired responses prior to multichannel analysis.

Omnidirectional loudspeaker
Source frequency response, directivity, and distortion figures were measured for the omnidirectional loudspeaker. For frequencies above 250 Hz, the unit is capable of producing sound pressure levels exceeding 95 dB at 1 m with 1 % THD figures. The useful frequency range of the sound source is about 100 Hz – 10 kHz.

The results for frequency response and directivity fulfill the requirements set for omnidirectional sound sources in the ISO 3382 standard [4]. When required, the sound source frequency response has been compensated for in the analysis of acquired impulse responses.

2.2 Standard omnidirectional analysis

Figure 4 shows an overview of the phases required for the correct analysis of standard room acoustic attributes from real-life impulse responses. These have been incorporated into the IRMA software.

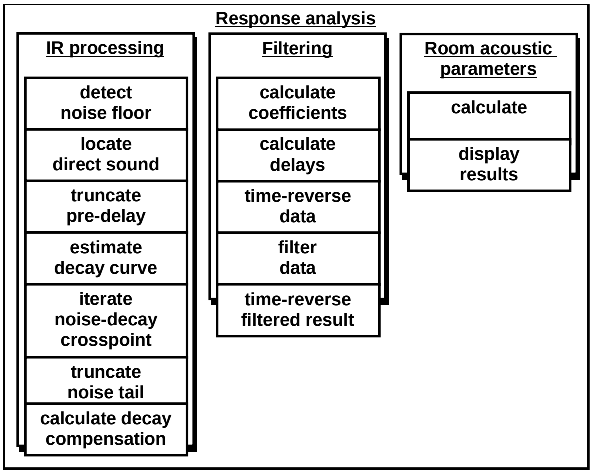

Fig. 4. Response processing and filtering prior to standard room acoustical analysis with the IRMA measurement system.

Filtering
For standard omnidirectional room acoustical analysis, responses are commonly filtered prior to the determination of room acoustic attributes. Bandpass filtering of responses into octave or third octave bands causes a time-spreading of signal energy due to filter delay and decay times. Time-reversed filtering may be used to minimize this effect for the accurate determination of short reverberation times. However, EDT and energy-time ratios should be analyzed using ordinary filtering and compensation for filter delay, in order to avoid initial time-smearing of the response.

Determination of direct sound and background noise level

Impulse responses must be processed to remove unidealities prior to calculating standard room acoustic attributes. These attributes are based on an ideal exponential decay model, so the background noise and initial delay present in real responses must be removed prior to analysis. Various methods have been studied for this purpose. The iterative algorithm described by Lundeby *et al.* [7] has been implemented in the IRMA system.

2.3 Auditorily motivated analysis

To study room impulse responses in more detail in time and frequency, we have applied the auditorily motivated analysis method [8]. The applied analysis method resembles traditional one-third octave band spectrogram analysis, but it better respects the frequency and time resolution of human hearing. In the analysis method, the impulse response is filtered to several subbands, followed by a temporal smoothing to the energy envelope of each band.

First, the input signal is fed through a gammatone filterbank that divides the signal into 40 ERB bands [9, 10], which resemble the frequency bands in auditory perception. After the division to ERB bands, the signals are squared in a similar fashion to the half-wave rectification performed by the hair cells in the inner ear. Then at each subband a sliding time window roughly simulating the ear's time resolution is applied. Finally, by taking the logarithm of the rectified and temporally processed signal for each frequency band, we can depict the decibel values for a time-frequency plot. Unfortunately, the plots cannot be printed here in color.

The method applied in directional analysis is not an accurate auditory model but rather an audio engineer's approach to the modelling of perception. We used directional microphones for capturing the directional components of the sound field, and applied the described auditorily motivated analysis method. In this way, the physics related to the arriving sound wavefronts is easily interpretable. For example, a pair of cardioid microphones can be used to capture the sound field component arriving on the lateral axis. If this first order directional accuracy is not sufficient, microphones with higher directivity can be applied as well. A case study of this kind of analysis is presented in Section 4.2.

2.4 Three-dimensional analysis

Each of the electret microphones in the 3-D probe exhibits omnidirectional directivity. Directional data can, however, be extracted by exploiting the information on the spatial distribution of the microphones.

In the following, methods for cardioidal beamforming as well as traditional cross-spectral sound intensity calculation are briefly reviewed. These well-known methods act as a good starting point for future research on novel directional analysis methods.

Cardioidal directivity patterns

A sound pressure component with a cardioidal directivity pattern can be calculated from the signals of a microphone pair as follows: Let $p_1(t,\omega)$ and $p_2(t,\omega)$ be the sound signals of two omnidirectional microphones. For a plane wave and closely spaced idealized microphones, the only difference between the signals is a time difference depending on the direction of propagation of the acoustical wave:

$$p_2(t,\omega) = p_1(t,\omega)\, e^{-j\omega\frac{d\cos\theta}{c}}$$

where d is the distance between the microphones, c is the velocity of sound, ω is the angular frequency and θ represents the angle between the direction of sound propagation and the line connecting the two microphones.

Delaying $p_2(t,\omega)$ by the maximum time difference between the microphone signals and subtracting it from $p_1(t,\omega)$ results in the signal

$$p'(t,\omega,\theta) = p_1(t,\omega) - p_2(t,\omega)e^{-j\omega\frac{d}{c}} = p_1(t,\omega)\left(1 - e^{-j\frac{\omega d}{c}(1+\cos\theta)}\right).$$

It can be shown that

$$\left|p'(t,\omega,\theta)\right| \approx |\, p_1(t,\omega)\,| \frac{\omega d}{c}(1+\cos\theta), \quad \text{when } \frac{\omega d}{c} << \frac{1}{2}$$

i.e., the directivity pattern is approximately cardioidal and the frequency response is of first-order highpass nature for frequencies

$$f << f_L = \frac{c}{4\pi d} \Leftrightarrow \lambda >> 4\pi d$$

For 10 and 100 mm distances the theoretical limiting frequencies are thus about 2.7 kHz and 270 Hz. In practice, however, $p'(t,\omega,\theta)$ can be used for much higher frequencies.

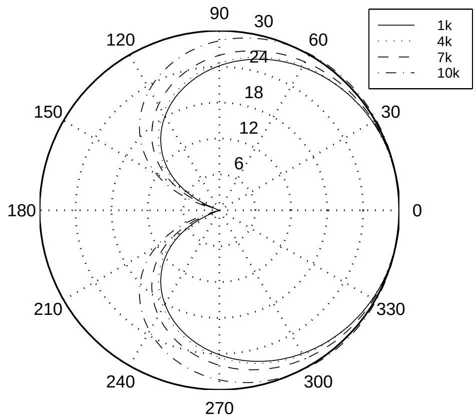

Fig. 5. Polar dB magnitude plots of $p'(t, \omega, 0)$ with d = 10 mm on four different frequencies.

Below f_L, the highpass effect can easily be compensated with a simple lowpass filter. On higher frequencies, the magnitude of $p'(t,\omega,0)$ is no more strictly first order highpass. For better magnitude response of frontal sound, the compensation filters can be designed to be the exact inverse of $|p'(t, \omega, 0)|$ on the frequency range of interest. This will introduce a small relative boost for high frequencies of sound arriving from other directions. However, as can be seen from the polar magnitude plots of p' with d=10 mm in Figure 5, the results are quite close to a cardioid for frequencies up to about 7 kHz. Correspondingly the practical upper frequency limit for d=100 mm is about 700 Hz.

The lower frequency limits depend mainly on phase errors caused by the measurement system. The errors should be kept significantly smaller than the phase difference due to different locations of the microphones forming a pair. For the designed measurement system, practical lower limits are about 600 and 60 Hz for 10 and 100 mm spacings, respectively.

Figure 6 shows the results of an actual measurement. The graph on top represents the measured free-field magnitude response of a loudspeaker. The bottom graph shows the magnitude spectra of 6 calculated cardioids with the direction θ=0 at different coordinate axes. The probe was directed so that X+ axis was pointing towards the loudspeaker. As can be seen, the level of X+ is 6 dB higher than Y and Z -signals and X- is significantly lower than any of the others.

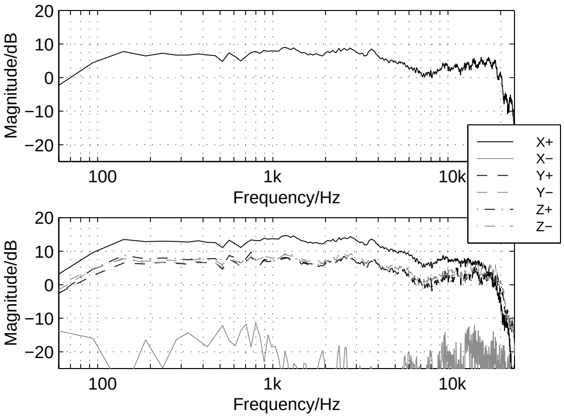

Fig. 6. Top: Magnitude response of the loudspeaker used as a sound source. Bottom: Responses of 6 cardioids pointing to different directions.

The magnitude spectra of the microphone pairs with 10 and 100 mm spacings have been combined to yield a single response on a wider frequency band.

Sound intensity

A well known method for calculating sound intensity from the cross spectrum of two sound pressure microphone signals was introduced by Fahy [11]:

$$I(f) = -\frac{1}{2\omega\rho d}\mathrm{Im}\{S_{12}(f)\}$$

where ρ is the density of air and d is the distance between the microphones. The cross spectrum $S_{12}(f)$ can be calculated from the Fourier transforms of the pressure signals $p_1(t)$ and $p_2(t)$

$$S_{12} = P_1(f)P_2^*(f)$$

where * denotes the complex conjugate. This gives the time-averaged sound intensity as a function of frequency. The time resolution can be controlled by the length of the Fourier transform. Furthermore, different lengths can be used for 10 and 100 mm microphone spacings so that a better time resolution can be achieved for high frequencies, if desired.

As with the directional sound pressure components, the lower frequency limit for a given microphone spacing depends on the phase errors of the measurement system. The higher limit comes from the fact that volume velocity is approximated from the pressure difference between the microphones. At high frequencies this approximation is not valid anymore. Chung [12] has stated that a reasonable upper frequency limit for this method is

$$kd \approx 1 \Leftrightarrow f_L \approx \frac{c}{2\pi d}\,.$$

For spacings of 10 and 100 mm the limits are about 5.4 kHz and 540 Hz respectively. The intensity error resulting from the approximation is about 1 dB at f_L. If more error can be accepted, the limits can be stretched further. Figure 7 shows the results of intensity calculations for the same loudspeaker measurement as in Figure 6.

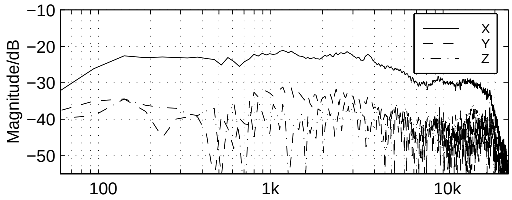

Fig. 7. Sound intensity of the loudspeaker measured in an anechoic chamber.

Angle of arrival estimation

Angle of arrival can easily be estimated from the sound intensity calculations. The intensity components on x- y- and z-axes form a vector pointing to the direction of propagation of sound energy. Thus the azimuth angle of arrival is given by

$$\theta = \arctan\left(\frac{I_y}{I_x}\right) + n\cdot 180^\circ, \quad n = 0, \pm 1, \ldots$$

where n must be determined by the signs of I_x and I_y. Correspondingly the elevation angle is given by

$$\phi = -\arctan\left(\frac{I_z}{\sqrt{I_x{}^2 + I_y{}^2}}\right).$$

Since sound intensity describes propagation of energy, the total intensity of two waves with the same energy and arriving from opposite directions is zero. On the other hand, the sound rejection for θ =90° in a cardioidal directivity pattern is too low for precise angle of arrival estimation. In [13], an attempt has been made to combine cardioidal and figure-of-eight directivity patterns, which both can be calculated from the signals of a microphone pair.

3 MEASUREMENTS

The IRMA system has been used for extensive multichannel measurements in a number of Finland's major concert halls. In addition to standard room acoustical measurements, a three-dimensional probe with 12 microphones and a binaural head have been used to gather a representative amount of response data in each hall. This has provided thorough in-situ testing and evaluation of the measurement system's capabilities and performance.

Measurements in concert halls were performed according to the guidelines published by Gade [14]. Responses were taken from three stage source points to 5–10 receiver points in the audience and three receiver points onstage. The number and locations of the representative audience receiver points were chosen individually for each hall.

A wealth of multichannel data was gathered from each hall. Impulse responses were acquired at each audience receiver point using an omnidirectional microphone, the cardioid pair, the binaural head and the 12-channel 3-D probe. With up to 39 source-receiver point combinations in a single hall, the total amount of acquired impulse response data exceeds 500 megabytes for each hall, yielding a valuable asset for future research and analysis.

4 RESULTS AND DISCUSSION

As a case study, a set of responses acquired in the Tampere-talo concert hall are presented and analyzed. The responses portray a single source-receiver pair, with the receiver located at the back of the floor, just below the balcony overhang.

Only the omnidirectional and cardioid responses are portrayed in this case, as the methods for multidimensional analysis are currently under research. However, as shown in section 2.3, the virtual cardioid pair analysis leads the way to true multidimensional analysis methods.

A broader selection of the standard room acoustical attributes from the measured concert halls is discussed in another paper [15].

4.1 Standard room acoustical analysis

Standard room acoustical attributes calculated from the response are shown in Table 1.

The responses in question were chosen for analysis, as the standard omnidirectional room acoustic attributes yield adequate results, whereas the energy-time curve shows a large amount of late energy with very little direct sound. This is also later seen in the time-frequency analysis.

According to general guidelines to the acceptable ranges of room acoustic attributes in concert halls [16], the reverberation time indices T_{20} and T_{30} as well as the clarity index C_{80} in Table 1 do not indicate any certain flaws in the room acoustics. The early decay time T_{10} and clarity C_{50} values confirm the presence of strong late reflections, but do not give any further information. The lateral energy fraction (LEF) values are also within normal limits.

	125	250	500	1000	2000	4000	8000
SNR, dB	-53	-61	-61	-63	-63	-68	-69
T_{10}, s	1.67	1.50	1.64	1.59	1.61	1.37	0.92
T_{20}, s	1.89	1.78	1.78	1.65	1.68	1.55	1.16
T_{30}, s	1.92	1.89	1.85	1.71	1.7	1.56	1.36
C_{50}, dB	-6.9	-6.4	-5.8	-5.5	-4.7	-2.0	-1.0
C_{80}, dB	-0.5	-0.7	-1.8	-1.8	-0.5	1.6	3.4
D_{50}, dB	-7.6	-7.3	-6.8	-6.5	-5.9	-4.1	-3.5
T_S, ms	136	126	137	131	125	98	77
LEF, %	3	23	20	26	20	15	19

Table 1. Calculated standard room acoustical attributes in octave bands.

Figure 8 portrays the energy-time curves of the omnidirectional response at the 125, 500 and 2000 Hz octaves. The absence of direct sound in comparison to the late reflections within the first 50–100 ms can be clearly seen in the 125 Hz and 2 kHz bands. However, the ETC curves cannot give any directional indications to these phenomena.

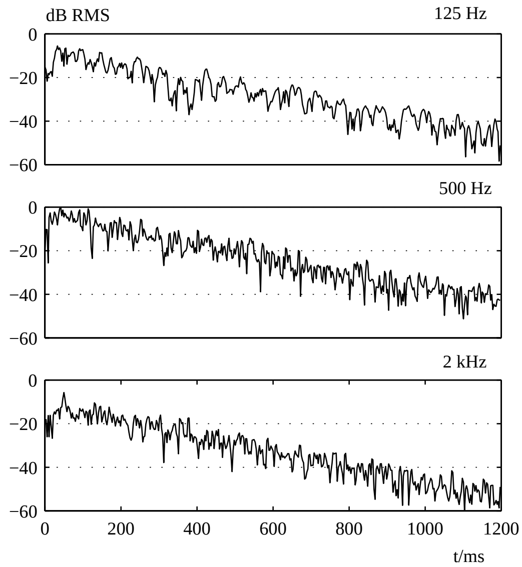

Fig. 8. Energy-time curves for the 125, 500, and 2000 Hz octave bands.

4.2 Time-frequency and directional analysis

The same impulse response is analyzed with the auditorily motivated method and the result is depicted in Fig. 9. Analysis is performed in the frequency range from 100 Hz to 17.5 kHz (covering 40 ERB bands). The figure clearly shows the strength of the early reflections in comparison to the weak direct sound at around 50 ms especially at low and high frequencies. Also, the profile of decay attenuation versus frequency is clearly seen at the higher frequencies.

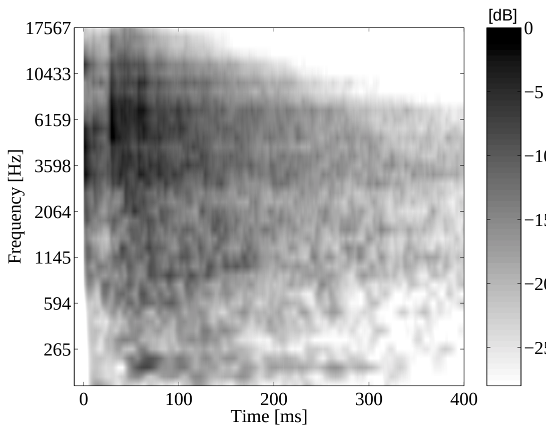

Fig. 9. An example of auditorily motivated analysis on the studied impulse response. Both the source and the microphone used exhibit omnidirectional directivity patterns.

To study the directional characteristics of the impulse responses, the same impulse response measurement has been performed with two cardioid microphones, pointed left and right laterally. The two impulse responses obtained with these microphones yield some information on the directional characteristics of the lateral sound field at the measurement point.

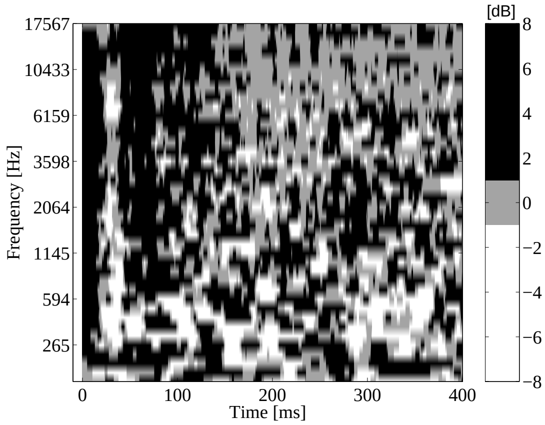

Fig. 10. An example analysis of lateral energy flow. Black areas are obtained when the signal from the left side cardioid microphone is dominating, and white areas for the right side respectively.

If the two responses are analyzed with the applied method and subtracted from each other, an estimation of the direction of sound energy flow at each time moment is acquired. Due to the temporal integration of the analysis method, this subtraction is more reliable than a subtraction of two ETC curves. The result, plotted in Fig. 10, quite interestingly shows that most of the energy is coming from the left side (black areas in Fig. 10). However, this is not the case between 200 and 400 Hz, where the sound pressure captured by the right side microphone is stronger almost all the time.

5 CONCLUSIONS AND FUTURE WORK

In this paper, a room acoustical measurement and analysis system is described, focusing on the multi-channel, binaural, and directional features. Implemented methods for both omnidirectional and directional analysis is described. As a case study, a set of responses acquired in a concert hall are presented and analyzed using a number of the described methods.

An auditorily motivated model has been applied to the directional components of the room response to derive a perceptually motivated directional time-frequency representation. Such a time-frequency-direction visualization of room responses is a potential technique to characterize and visualize important properties of the room or hall.

Future work includes further research on the directional analysis methods for the 3-D probe data, as well as development of binaural auditory modeling methods, using dummy head data, for new perceptual characterizations of room acoustics.

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