Original Acoustic Image - a new reconstructive restoration procedure using room impulse responses

(Original Acoustic Image - ein neues rekonstruktives Restaurationsverfahren mit Raumimpulsantworten)

Beitrag zur 24. Tonmeistertagung Leipzig 2006

Klaus - Hendrik LORENZ-KIERAKIEWITZ
PEUTZ GmbH, khl@peutz.de

Abstract

Listening to historical recordings, in spite of their sometimes undoubtful repertoire value, the listening pleasure often is decreased by poor sound quality due to monaural microphone setup and limited frequency characteristics.

Using the procedure "Original Acoustic Image" (OAI), it is possible to restore a realistic original-like timbre and natural sounding stereophonic reverberation. The method consists of suitable filtering applied to the distorted original and the appropriate use of room impulse responses measured in the halls where the recordings took place. The process can lead to an increased enjoyment of listening to these historical recordings. Its advantages, shortcomings, results and artistic aspects are presented here in detail.

1. Introduction

In the early history of sound recording, before the common use of the HiFi-stereophonic technique, many recordings of classical music were made preserving studio sessions or life performances of outstanding interpretative or repertoire value for posteriority.[1] However, listening to such recordings, three main shortcomings seriously reduce the listening pleasure: noise, a totally degraded timbre and the lack of stereophonic spaciousness. Meanwhile, sophisticated algorithms to equalize and denoise historical recordings are available. To overcome the lack of spaciousness in monaural recordings, in the past many pseudo-stereophonic restoration experiments ("Breitklang" etc.) tried to make these recordings more convenient to Hi-Fi ears, but never succeeded so far until now.

In this research work it was examined, how the technique of convolution with 2-channel impulse responses measured in the original concert halls could serve to reconstruct an acoustic image as realistic and original as possible, especially the true stereophonic reverberation of the recording hall.

2. Impulse response measurement technique and locations

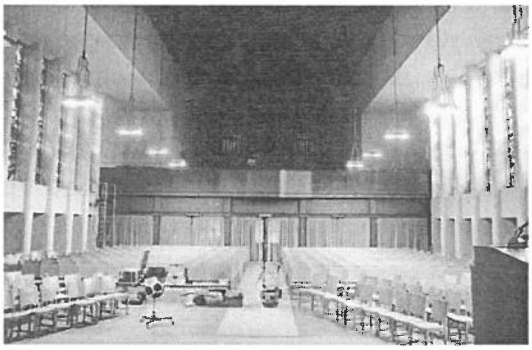
To be able to use impulse responses of historic recording halls for convolution, 2-channel impulse responses were measured in their current state using a PC-based Maximal Length Sequence measuring system (MLS of degree 17B, Fs=44.1 kHz, resolution 16 Bit).[2]

The microphones were placed near the critical distance at possible stereo main microphone positions at a height of 3,85 m above the parquets and 1 m in front of the stage in the hall. The two microphones were set up in AB- time of arrival stereophony. The stereo base **b** was varied in dependence of the maximum angle of sound incidence (27 to 38 cm). **b** results for a reproduction on precise loudspeakers ($\Delta t = 0{,}8$ ms) to:[3] $\Delta t = \frac{\Delta l}{c} = b \cdot \frac{\sin\alpha}{c} \Leftrightarrow b = \frac{\Delta t \cdot c}{\sin\alpha}$.

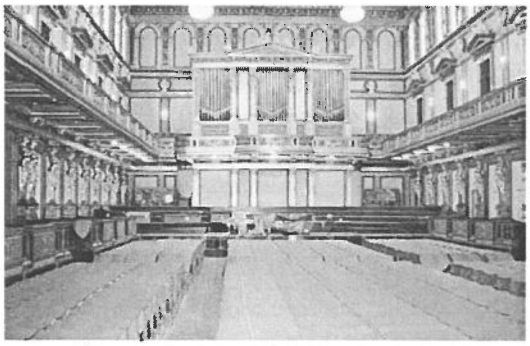
The omni-directional dodecahedron type sound source was placed in the centre of the stage.[2,4] To ensure high-quality convolution, the measured 2-channel impulse responses have to be corrected for the non-linear diffuse-field frequency response of the source.[4]

*Table 1: Measured historical concert halls used for recordings, *Seating state: S = audience simulation (more than ca. 90%), PS = ca. 65-85 %, U = unseated*

City	Hall	Date	Audience*	Microphone	Distance r/m	V /m³
Berlin	Jesus-Christus-Kirche	26-07-00	U	M296	9,0	10000
London	Royal Albert Hall	06-08-00	U	M296	11,0	86650
Amsterdam	Concertgebouw	03-08-01	PS,U	4007	4,6	18780
Wien	Musikvereinssaal	20-08-01	S,U	M93	5,2	15000
Hamburg	Musikhalle	27-03-01	U	M93	5,3	11700
Basel	Stadtcasino	23-08-01	PS;U	M93	3,1	10500
Zürich	Tonhalle	06-08-02	PS,U	M93	4,5	11400



Jesus-Christ-Church, Berlin



Großer Saal des Musikvereins, Wien

Concertgebouw, Amsterdam

Royal Albert Hall, London

Figure 1: Recent inner views of the Jesus-Christ-Church (top, left), the Musikverein Vienna (top, right), the Concertgebouw (bottom, left) and Royal Albert Hall (bottom, right)

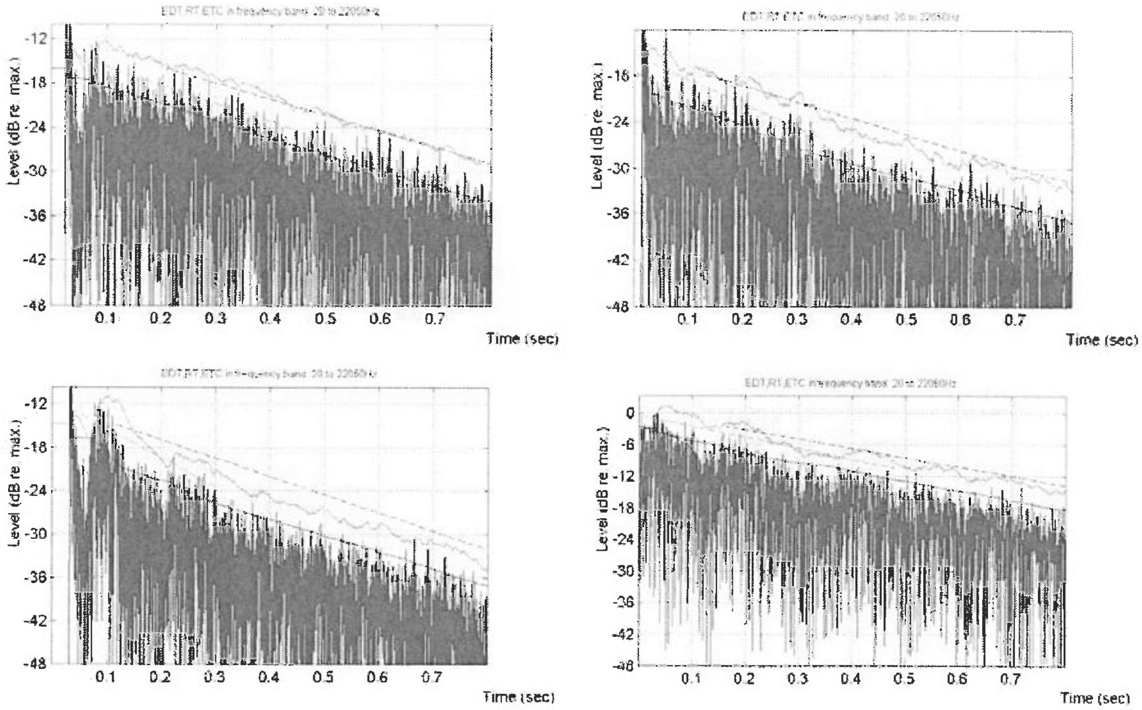


Figure 2: Broad-band ETCs of the left channel on the main microphone position in the Jesus-Christ-Church Berlin 2000 (top, left), the Musikverein Vienna 2001 (top, right), the Royal Albert Hall 2000 (bottom, left), stop-chord in the Musikverein 1944 (bottom, right)

Of course, the acoustical conditions of the halls used at the time of the historic recordings may have changed considerably. The question is, whether they changed the sound character the halls are famous for. Therefore, with the new measurements and stop-chords of the old recordings a reverberation time analysis can be performed to reveal the reverberation time differences of the halls between the recording date and the current situation:[2]

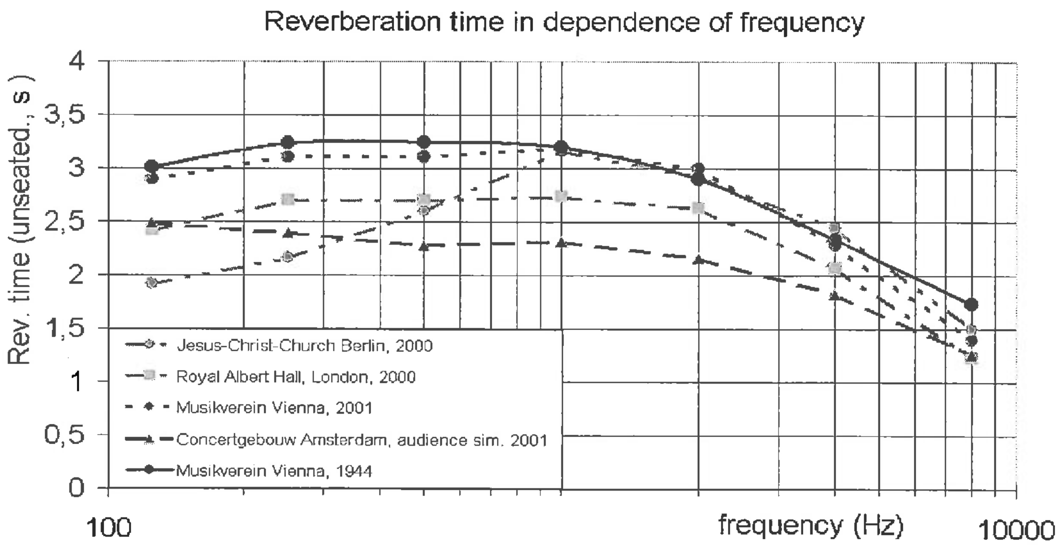


Figure 3: Reverberation time T_{30} in dependence of frequency in the four recording locations

3. Restoration technique and procedure

In the following section, the procedure of how to treat old mono recordings to get an "original acoustic image" is presented. The different stages of the restoration process should be executed in the highest digital resolution available and are described step by step.

3.1 Historical recordings: raw material

Of course, high-quality restoration results require master sources with the highest sound quality available, i.e. as little noise, distortion and artefacts as possible. Hence, precondition of the process is the selection of appropriate material sources, preferrably original master-tapes, vinyl discs of the first series pressed etc..

3.2 Dubbing and AD-conversion

The selected raw material has to be played back on the original and/or highest quality machines available, aligned for the medium to play back and provided with the most appropriate equalization curve (NAB, RIAA etc.) to ensure the purest electrical signal. The signal should be transferred to digital domain by means of high resolution AD-converters and stored as a raw file.

3.3 De-noising

If this raw material is contaminated by noise effects, an appropriate denoising process has to take place. The de-noising should include sophisticated processing of clipping, hiss, rumble, distortion, crackle etc.

3.4 Reference recording

Mostly, old recording chains resulted in an overall linear distortion of the frequency transmission function with sometimes severe bass emphasis, and an extreme drop of the transmission factor above 5 kHz. Even if many people have attuned themselves to this distorted timbre ("patina"), a realistic original-like sound can be achieved with an appropriate timbre of the recording.
Therefore, a modern reference recording can be selected and used, preferrably of the same orchestra playing the same piece in the same hall. This modern recording has to be free of noise and distortions and to exhibit an appropriate timbre. Once such a reference recording is selected, a multi-channel-editor can be used to identify corresponding sound segments.

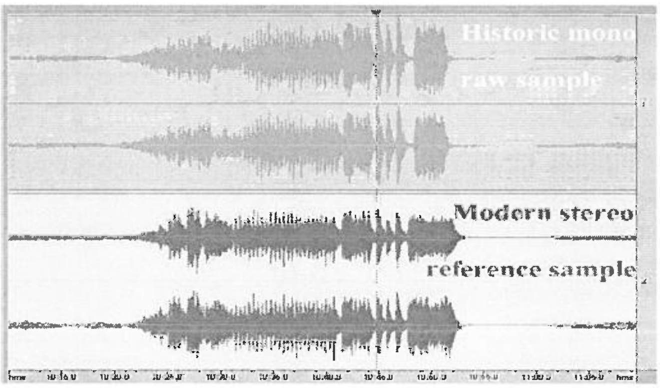


Figure 4: Identification of corresponding passages in historic and reference recording

3.5 Spectral analysis of timbre-differences by long-term FFT spectra

The sound samples should not be shorter than about 30 sec. to 1 min. (best more than 2^{19} samples for Fs=44,1 kHz). These content-analogous time segments are stored as reference and raw sample and can be analysed in a scientific analysis program.

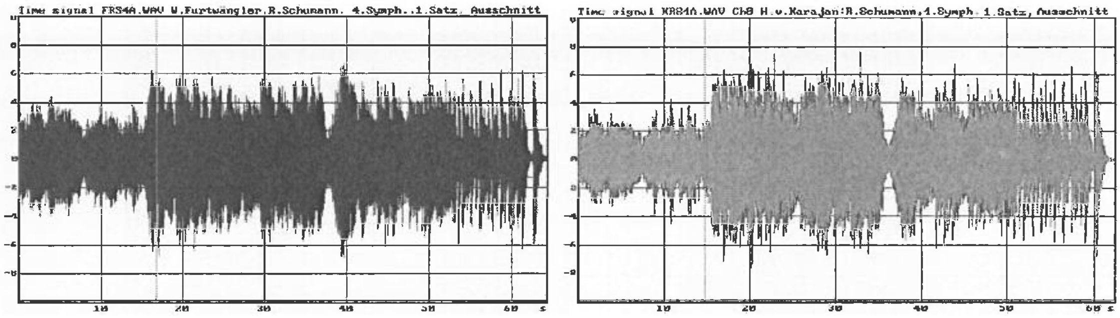


Figure 5: Time domain representation of corresponding segments in historic (left) and reference recording (right) with FFT-window

To identify the quasi-stationary timbre of sound, long-term FFTs can be used containing all important frequency dependent phenomena of the timbre (as distorted frequency transmission factor).[5-7] The scientific analysis program should be able to perform FFTs with 2^{19} to 2^{21} FFT sampling points.

It can be found that (due to their high content of harmonics) f and ff passages, when all instruments are playing, are most suitable.

It is important to analyse musically exactly analogous passages as time windows in reference and raw file, and to choose an identical FFT-length of a power to two samples in both cases (this enables the FFT algorithm to work without resampling). Both amplitude spectra then have to be smoothed. Suitable setting for this purpose is e.g. continuous gliding logarithmic smoothing with 1/1 octave averaging between 15 Hz and Fs/2. This yields the average spectra of modern reference recording and raw material:

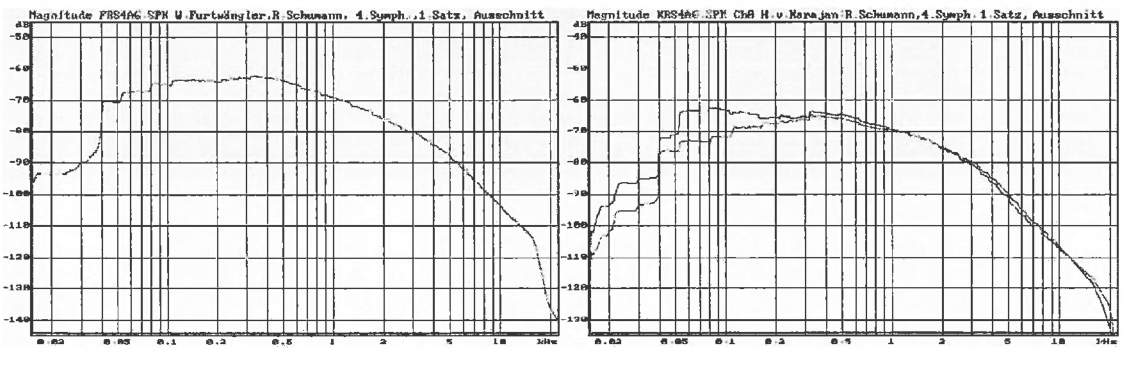


Figure 6: Frequency domain amplitude spectrum representation of corresponding passages in historic (left) and reference recording (right) after long-term FFT

So a spectral representation can be calculated by separately frequency-analyzing the sound samples by means of long-term FFT analysis of every channel and every movement (or even every passage).

To get a frequency correction curve, the resulting smoothed spectra are divided (reference : raw) resulting in a difference spectrum.

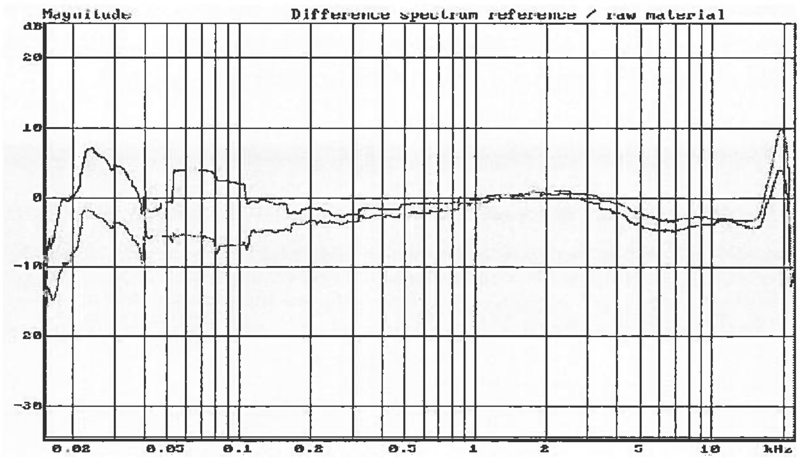


Figure 7: Frequency domain amplitude difference spectrum of reference : historic sample

This spectral difference serves as timbre correction pattern to restore the raw recording.

3.6 Difference spectrum based equalizing

Once the difference spectrum per passage, channel and movement is known, the timbre correction can be performed using a multi-band parametric equalizer, contouring the difference spectrum as correction curve:

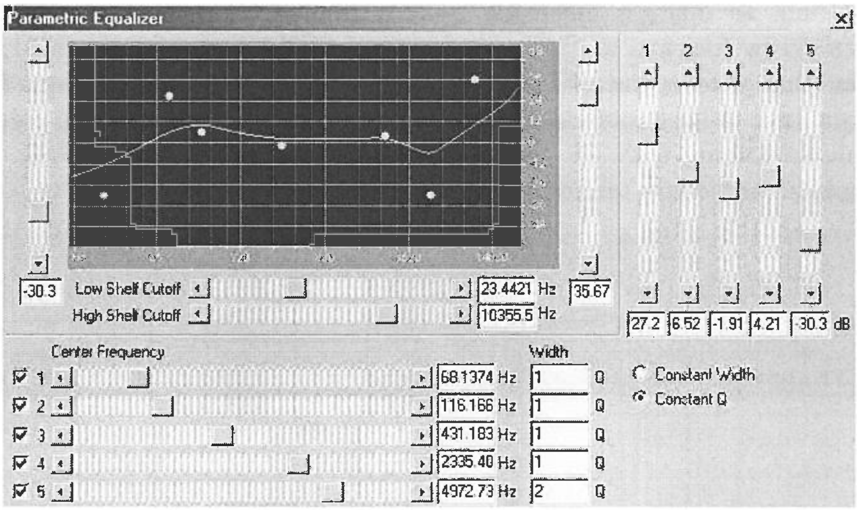


Figure 8: Contouring the correction spectrum with multi-band parametric equalizer

Because of a possibly exaggerated amplification of high frequencies, a second stage of de-noising may be necessary to eliminate the additional hiss.

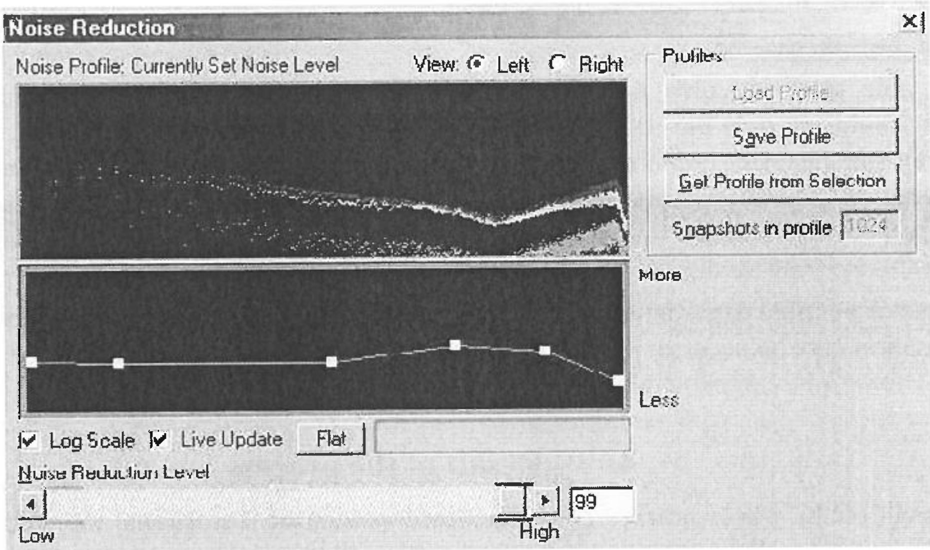


Figure 9: Additional de-noising because of exaggerate amplification of high frequencies

3.7 Convolution with original impulse response

Now a realistic, original-like sound-field can be reconstructed out of the equalized material by convolution with the measured 2-channel impulse response of the original room the recording once took place.

The technique of convolution is known and commonly used nowadays for reverb purposes. The crucial parameter is the ratio of direct sound to reverberation, and the appropriate setting remains – as with all mastering modifications – a question of the recording engineer's taste. Realistic results can be expected with low percentages of direct sound (0…10%).

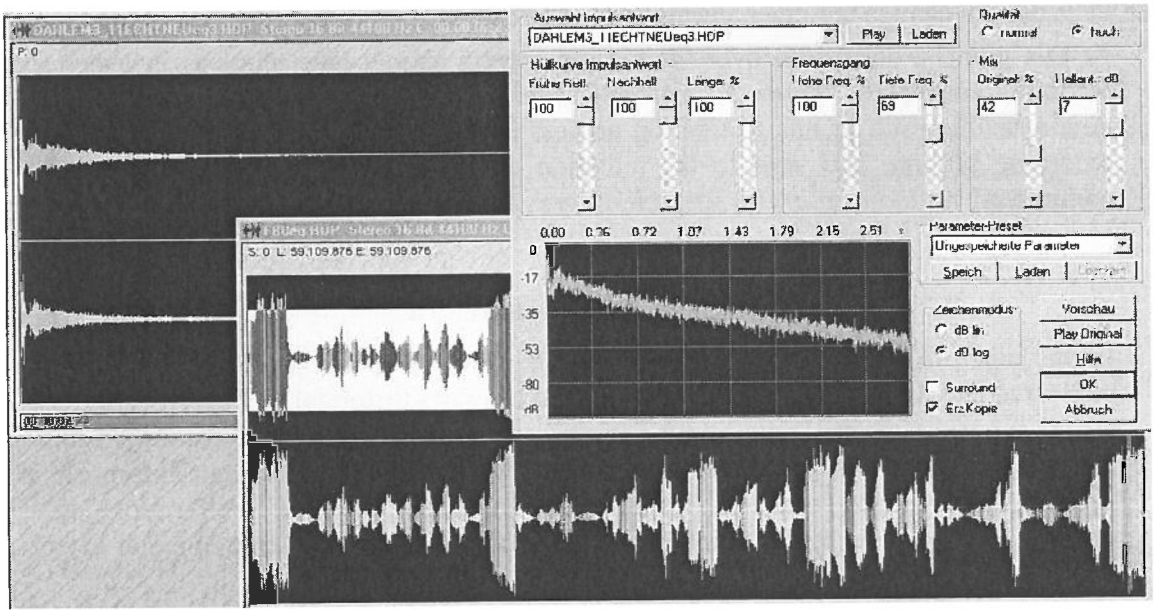


Figure 10: Reconstruction of stereo reverberation by convolution with measured 2-channel impulse response

Being able to deconvolve the mono main microphone response before convolution, of course would improve the result of the reverberation reconstruction. However, this has not yet been satisfactorily implemented. Therefore, up to now, only an additive convolution of the stereo-reverberation is possible: the reverberation portion of the recording increases and the direct-to-reverberant ratio decreases (as one would move in the hall away from the stage).

Because of possible degradations of the timbre by the convolution process, a second stage of equalisation may be necessary to finish the restoration process.

4. Result of the process

The result of the whole process is an equalized version of the original monaural recording provided with a realistic stereophonic reverberation field.

If all steps of this process have been performed carefully, the results gained by the method "Original Acoustic Image" can lead to very impressive restorations.

Obviously these interpretations could not be heard for decades this realistically.

5. Discussion

The method presented in this paper goes far beyond the usual limits of conventional restoration and mastering. Moreover, it has the shortcomings of only one source while measuring the impulse responses and the current impossibility to deconvolve the monaural impulse response from the raw recording.

Finally, the process is based on some assumptions, which may be regarded questionable: mainly, that frequency characteristics of orchestra playing and the room acoustics of the halls did not change too much during the last decades.

At this point, a discussion over the artistic aspects should take place as in every serious recording concept.[1,8,9] Even if there will never be a definite answer, it is worthful to ask, how near the result of this restoration process comes to the original acoustic image of the recording session, how near to the intention of the recording engineers, conductors and composers?

6. Conclusion

In this paper, the new restoration technique "Original Acoustic Image" (OAI) was presented, explained and discussed in detail. This method provides historical monaural recordings with a timbre-correct stereophonic reverberation using a modern reference recording, measured 2-channel room impulse responses and sophisticated signal processing. The result of the process is a spectrum-corrected version of the original monaural recording provided with a realistic stereophonic reverberation field. This lead to an increased enjoyment of listening to historical recordings.

7. Acknowledgements

I wish to express deep gratitude to Peutz bv for their kind support of this research work.

8. References

[1] Burkowitz, P.: "Die Welt des Klanges. Musik auf dem Wege vom Künstler zum Hörer", *Stereoplay 5/91 ff.*

[2] Lorenz, K.-H., Metkemeijer, R.: "Objektive und subjektive Erforschung der Aufnahmeakustik europäischer Konzertsäle", *Beitrag 23. Tonmeistertagung 2004*

[3] Gernemann, A.: "Meßtechnische Untersuchung der akustischen Vorgänge beim natürlichen Hören im Vergleich zu den Vorgängen bei der Laufzeit- und ‚Intensitäts'-Stereophonie", *Aachen, 1995*

[4] Lorenz, K.-H.: "Development of new Auralization Techniques in Room Acoustics based upon Binaural Impulse Responses", *Diploma Thesis, University Düsseldorf 2000*

[5] Heike, G.: "Das gemeinsame Klangideal von Belcanto und altitalienischen Geigen", *in: Festschrift zum 90jährigen Jubiläum der Sprechwissenschaft in Halle. Hallesche Schriften zur Sprechwissenschaft und Phonetik. Bd. 3. 1997*

[6] Heike, G.: "Listener's judgements and acoustic properties of violins", *Proceedings of the Institute of Acoustics, Vol. 19, Edinburgh 1997*

[7] Heike, G.: "Untersuchungen zu objektiven und subjektiven Klangeigenschaften von Singstimmen und Streichinstrumenten", *DFG-Projektabschlussbericht Az. He 434/28-1, 1998*

[8] Urbanski, J.: "Die Tonaufnahme als Problem der künstlerischen Interpretation", *IMZ Report 1967*

[9] Kier, H.: *"Der fixierte Klang"*, Köln 2006