Comparison of Different Impulse Response Measurement Techniques*

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The impulse response of an acoustical space or transducer is one of its most important characterizations. In order to perform the measurement of their impulse responses, four of the most suitable methods are compared: MLS (maximum-length sequence), IRS (inverse repeated sequence), time-stretched pulses, and SineSweep. These methods have already been described in the literature. Nevertheless, the choice of one of them depending on the measurement conditions is critical. Therefore an extensive comparison has been realized. This comparison was done through the implementation and realization of a complete, fast, reliable, and cheap measurement system. Finally, a conclusion for the use of each method according to the principal measurement conditions is presented. It is shown that in the presence of nonwhite noise, the MLS and IRS techniques seem to be more accurate. On the contrary, in quiet environments the logarithmic SineSweep method seems to be the most appropriate.

0 INTRODUCTION

Under the assumption of source and receiver immobility, the acoustical space in which they are placed can be considered a linear time-invariant system characterized by an impulse response $h(t)$. In room acoustics the accurate measurement of the impulse response is very important, since many acoustical parameters can be derived from it. Moreover, in present-day audio applications (that is, virtual reality, auralization, spatialization of sounds) the importance of measuring binaural room impulse responses with a very high signal-to-noise ratio becomes more and more evident. Once the impulse response has been measured precisely, it can be integrated in a complete auralization process [1], [2]. In order to achieve the best quality for this auralization process, the measured impulse response must reach a very good signal-to-noise ratio (more than 80 dB if possible).

A common method for measuring the impulse response of such an acoustical system is to apply a known input signal and to measure the system output. The choice concerning the excitation signal and the deconvolution technique that will permit obtaining the impulse response from the measured output is of essential importance:

- The emitted signal must be perfectly reproducible.
- The excitation signal and the deconvolution technique have to maximize the signal-to-noise ratio of the deconvolved impulse response.

- The excitation signal and the deconvolution technique must enable the elimination of nonlinear artifacts in the deconvolved impulse response.

In general the signal-to-noise ratio is improved by taking multiple averages of the measured output signal before the impulse response deconvolution process is started.

The most commonly used excitation signals are deterministic, wide-band signals known as:

- MLS (maximum-length sequence) and IRS (inverse repeated sequence), which use pseudorandom white noise
- Time-stretched pulses and SineSweep, which use time-varying frequency signals.

1 BRIEF DESCRIPTION OF MEASUREMENT TECHNIQUES

The acoustical impulse response measurements using the MLS technique were first proposed by Schroeder in 1979 [3] and have been used for more than 20 years. Many papers discussed the theoretical and practical advantages and disadvantages of their technique [4]–[15]. Shortly after the publication of Schroeder, the IRS technique was proposed as an alternative, allowing a theoretical reduction of the distortion artifacts introduced by the MLS technique [4], [16], [17].

Two years after the proposition of Schroeder, Aoshima introduced a new idea for the measurements of impulse responses which led to the time-stretched pulses technique [18]. His idea was then pushed further by Suzuki et al.

[19], who proposed what they called an "optimum computer-generated pulse signal."

Recently Farina and Ugolotti introduced the logarithmic SineSweep technique [20], [21] intended to overcome most of the limitations encountered in the other techniques. The idea of using a sweep in order to deconvolve the impulse response is not new [22], but the deconvolution method used is different in Farina.

These techniques have already been described and discussed in many papers. However, it is intended here to focus on some important properties which are necessary to understand the comparison of the different methods.

1.1 MLS Technique

The MLS technique is based on the excitation of the acoustical space by a periodic pseudorandom signal having almost the same stochastic properties as a pure white noise. The number of samples of one period of an mth-order MLS signal is $L = 2^m - 1$.

More theoretical considerations about the MLS sequences can be found in [7], [10], [12], [13], [23] and in the excellent book on shift-register theory [24].

With the MLS technique, the impulse response is obtained by circular crosscorrelation (as shown in [25]) between the measured output and the determined input (MLS sequence). Because of the use of circular operations to deconvolve the impulse response, the MLS technique delivers the periodic impulse response $h'[n]$, which is related to the linear impulse response as

$$h'[n] = \sum_{t=-\infty}^{+\infty} h[n + lL] . \tag{1}$$

Eq. (1) reflects the well-known problem of the MLS technique: the time-aliasing error. This error is significant if the length L of one period is shorter than the length of the impulse response to be measured. Therefore the order m of the MLS sequence must be high enough to overcome the time-aliasing error. Our measurement system allows the generation of MLS sequences up to order 19 (which corresponds to a period of 12 s if the sampling frequency is 44.1 kHz).

1.1.1 MLS Immunity to Signals Not Correlated with the Excitation Signal

Each MLS sequence is characterized by a phase spectrum which is strongly erratic, with a uniform density of probability in the $[-\pi, +\pi]$ interval, as can be seen in Fig. 1.

According to this property, the MLS technique is able to randomize the phase spectrum of any component of the output signal which is not correlated with the MLS input sequence [5], [9]. As a consequence, any disturbing signal (that is, white or impulsive noise) will actually be phase randomized, and this will lead to a uniform repartition of the disturbing effects along the deconvolved impulse response (Figs. 2 and 3) instead of localized noise contributions along the time axis. A postaveraging method can then be used to reduce this uniformly distributed noise appearing in the deconvolved impulse response.

1.1.2 Disadvantages of the MLS Technique

The major problem of the MLS method resides in the appearance of distortion artifacts known as distortion peaks [6]. These artifacts are more or less uniformly distributed along the deconvolved impulse response. The origin of the distortion peaks lies in the nonlinearities inherent in the measurement system and especially the loudspeaker.

These distortion artifacts introduce characteristic crackling noise when the measured impulse response is convolved with some anechoic signal in order to realize the auralization process. These distortion peaks can be attenuated by:

- The use of dedicated measurement methods (such as the inverse repeated sequence technique [4], [16].
- The optimization of some measurement parameters. For example, the amplitude of the excitation signal is, in practice, a compromise between increasing distortions at high levels and decreasing the signal-to-noise ratio at low levels. This optimization is very time consuming because of the practical difficulty of finding the optimum amplification level. Moreover, this compromise level must be chosen carefully for each new measurement situation.
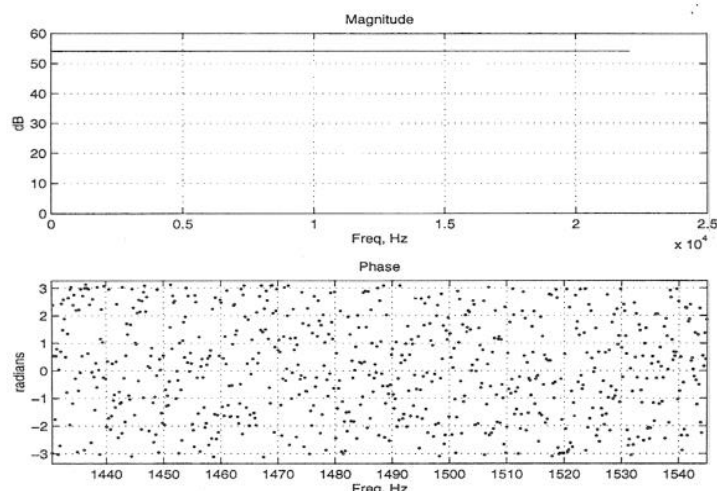

Fig. 1. Magnitude and phase spectra of MLS sequence. Phase spectrum was enlarged to clearly show its uniform random distribution.

Fig. 4 illustrates the quality of the results that can be obtained when particular care is taken in the optimization of the parameters (mainly the output level) conditioning the MLS (or IRS) input signal. It can be seen that the distortion peaks are reduced significantly but not completely removed.

1.2 IRS Technique

Each IRS sequence with a $2L$ sample period $x[n]$ is defined from the corresponding MLS sequence of period L (MLS[n]) by the following relation:

$$x[n] = \begin{cases} \text{MLS}[n], & n \text{ even}, & 0 \leqslant n < 2L \\ -\text{MLS}[n], & n \text{ odd}, & 0 < n < 2L \end{cases} \quad . \quad (2)$$

The deconvolution process is exactly the same as for the MLS technique (circular correlation).

Fig. 5 shows the attenuation of the distortion peaks

when the IRS method is used. These impulse responses have been obtained by performing the measurements in an anechoic room, leaving all measurement parameters unchanged from one measurement to the next.

1.3 Time-Stretched Pulses Technique

This method is based on a time expansion and compression technique of an impulsive signal [18]. The aim of using an expansion process for the excitation signal is to increase the amount of sound power emitted for a fixed magnitude of the signal and therefore to increase the signal-to-noise ratio without increasing the nonlinearities introduced by the measurement system. Once the response to this "stretched" signal has been measured, a compression filter is used to compensate for the stretching effects induced and to obtain the deconvolved impulse response.

Fig. 6 shows the impulse response obtained with this
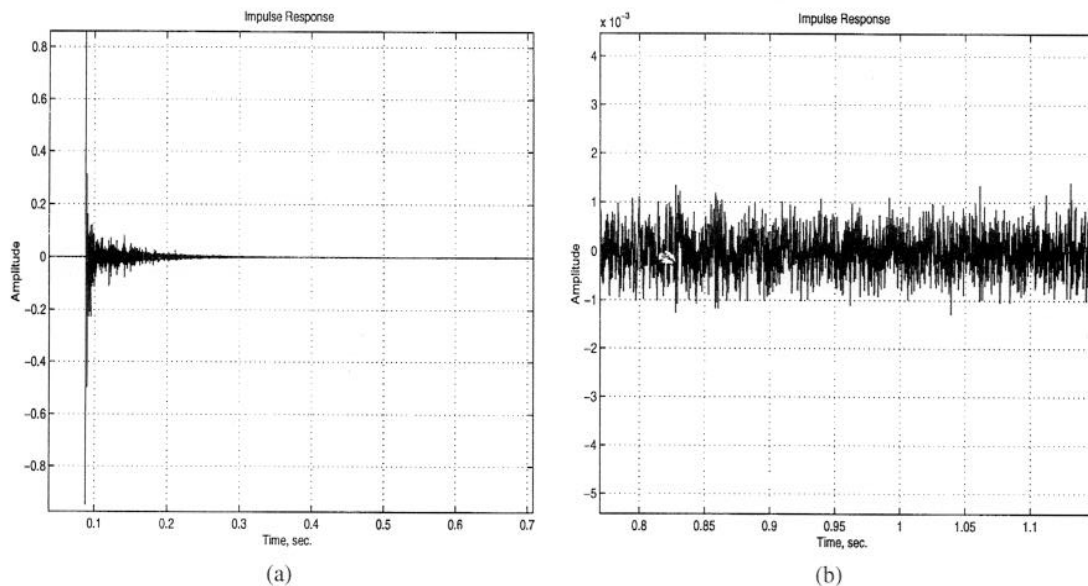

(a)   (b)

Fig. 2. (a) Impulse response obtained with single MLS sequence of order 18 in classroom, when a white-noise generator is simultaneously present. Noise level measured at microphone is 60 dB. (b) Zoom on end of impulse response.

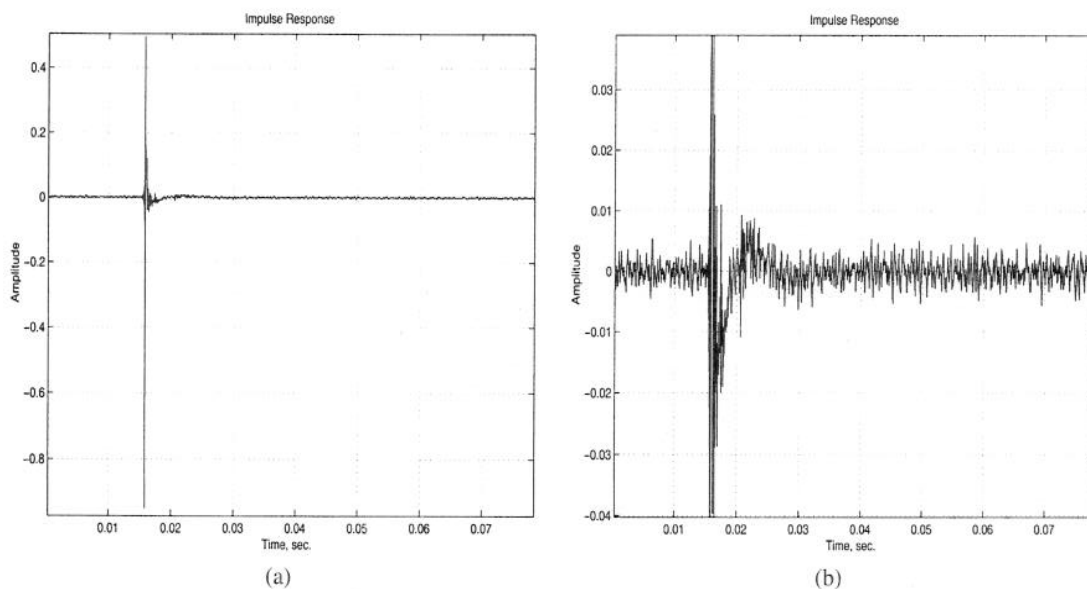(a)   (b)

Fig. 3. (a) Impulse response obtained with single MLS sequence of order 16 in anechoic room when impulsive noise is simultaneously present. (b) Zoom on magnitude scale.

technique. The magnitude scale has been enlarged to clearly illustrate the absence of distortion peaks. However, this does not mean that distortion artifacts are completely removed. They still appear in the impulse response (as a residue of the deconvolution filter), as can be seen in Fig. 7.

1.4 Logarithmic SineSweep Technique

The MLS, IRS, and time-stretched pulses methods rely on the assumption of linear time-invariant (LTI) systems and cause distortion artifacts to appear in the deconvolved impulse response when this condition is not fulfilled.

The SineSweep technique developed by Farina [21] overcomes such limitations. It is based on the following idea: by using an exponential time-growing frequency sweep it is possible simultaneously to deconvolve the lin-

ear impulse response of the system and to selectively separate each impulse response corresponding to the harmonic distortion orders considered. The harmonic distortions appear prior to the linear impulse response. Therefore the measured linear impulse response is assured exempt from any nonlinearity, and at the same time, the measurement of the harmonic distortion at various orders can be performed.

Fig. 8 illustrates the black box modeling of the measurement process common to all four techniques discussed. In this model it is assumed that the measurement system is intrinsically nonlinear, but on the other hand, perfect linearity is considered regarding the acoustical space from which the impulse response is to be derived.

As pointed out in Farina [21], the signal emitted by the loudspeaker is composed of harmonic distortions (consid-
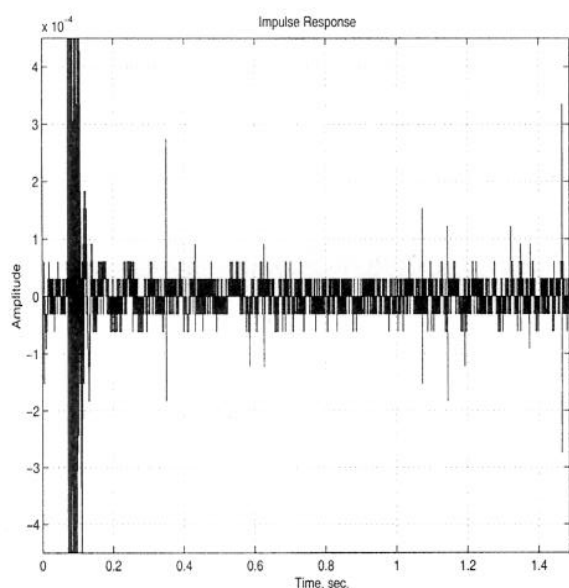

Fig. 4. Zoom on impulse response obtained after optimization of measurement parameters (output sound level), MLS technique.
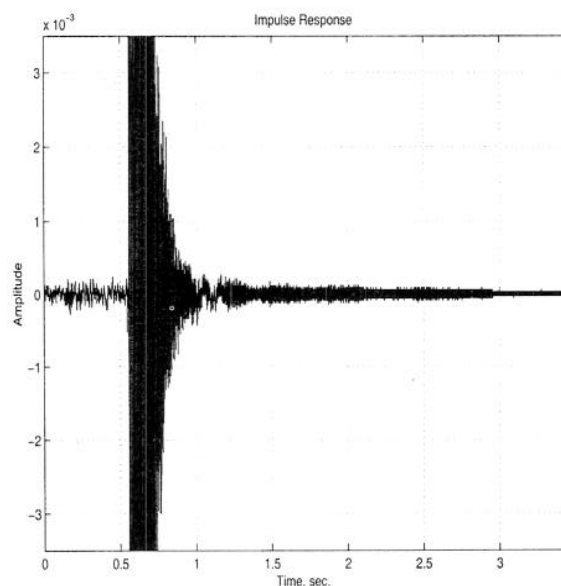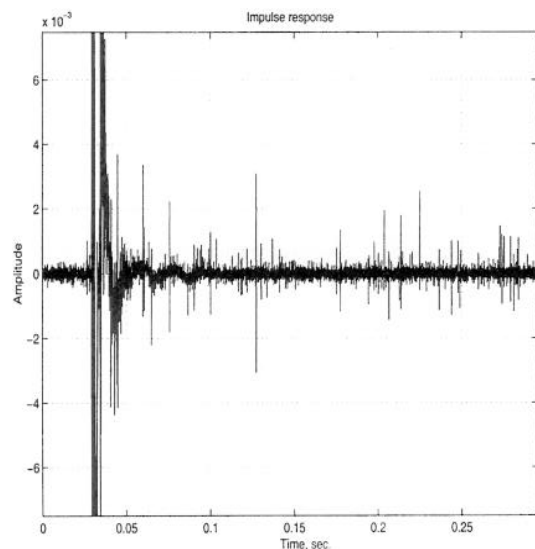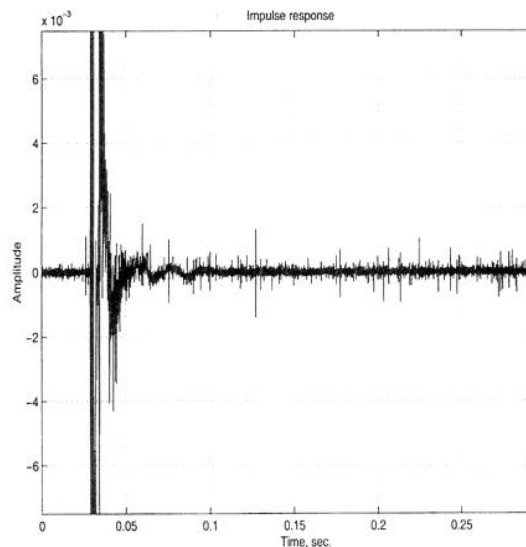

Fig. 6. Zoom on impulse response in classroom when using time-stretched pulse of about 6 s.

(a)

(b)

Fig. 5. Zoom on impulse responses. (a) MLS. (b) IRS.

ered here without memory) and may thus be represented by the following equation (see Fig. 8):

$$w(t) = x(t) \otimes k_1(t) + x^2(t) \otimes k_2(t)$$
$$+ x^3(t) \otimes k_3(t) + \cdots + x^N(t) \otimes k_N(t) \qquad (3)$$

where $k_i(t)$ represents the ith component of the Volterra kernel [21], which takes into account the nonlinearities of the measurement system.

In practice it is relatively difficult to separate the linear part (the reverberation part in the impulse response) from the nonlinear part (distortions). In the following we will consider the response of the global system (the output signal from the system represented in Fig. 8) as being composed of an additive Gaussian white noise $n(t)$ and a set of impulse responses $h_i(t)$, each being convolved by a different power of the input signal:

$$y(t) = n(t) + x(t) \otimes h_1(t) + x^2(t) \otimes h_2(t)$$
$$+ x^3(t) \otimes h_3(t) + \cdots + x^N(t) \otimes h_N(t) \qquad (4)$$

where $h_i(t) = k_i(t) \otimes h(t)$. Eq. (4) underlines the existence of the nonlinearities at the system output.

In the case of the logarithmic SineSweep technique, the excitation signal is generated on the basis of the following equation (see [21] for more theoretical information):

$$x(t) = \sin\left\{ \frac{T\omega_1}{\ln(\omega_2/\omega_1)} \left[e^{(t/T)\ln(\omega_2/\omega_1)} - 1 \right] \right\} \qquad (5)$$

where ω_1 is the initial radian frequency and ω_2 is the final radian frequency of the sweep of duration T.

Fig. 9 shows the time and spectral representations of a logarithmic sweep with initial and final frequencies at 10 Hz and 1000 Hz, respectively.

The impulse response deconvolution process is realized by linear convolution of the measured output with the analytical inverse filter preprocessed from the excitation signal. Using linear convolution allows time-aliasing problems to be avoided. In fact, even if the time analysis window has the same length as the emitted SineSweep signal (and is shorter than the impulse response to be measured), the tail of the system response may be lost, but this will not introduce time-aliasing. This is a first advantage over MLS and IRS methods.

In practice a silence of sufficient duration is added at the end of the SineSweep signal in order to recover the tail of the impulse response.

The deconvolution of the impulse response requires the creation of an inverse filter $f(t)$ able to "transform" the ini-
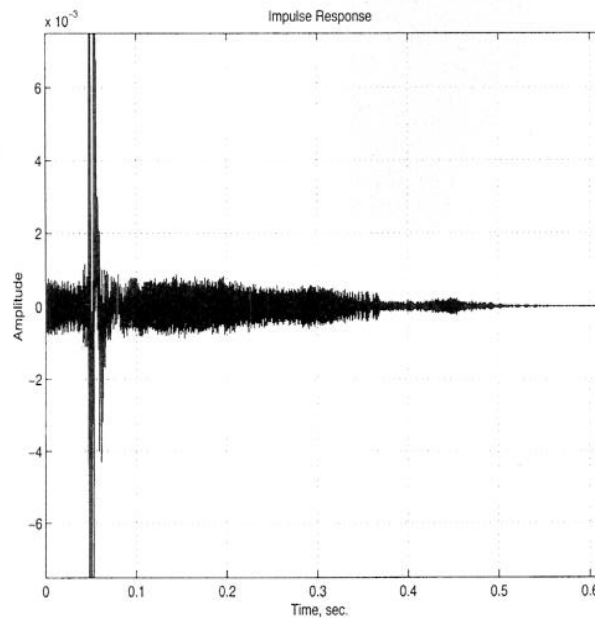

Fig. 7. Zoom on impulse response in anechoic room when using time-stretched pulse of about 1 s. In this case a poor-quality loudspeaker was used to emphasize the nonlinearity of the measurement system.
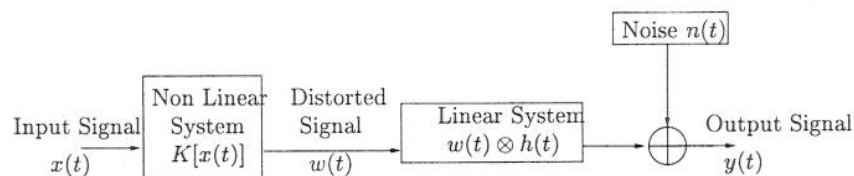

Fig. 8. Modeling of global system, including loudspeaker (considered a nonlinear element) and acoustical space (considered a perfectly linear system).

tial sweep into a delayed Dirac delta function:

$$x(t) \otimes f(t) = \delta(t - K) . \qquad (6)$$

The deconvolution of the impulse response is then realized by linearly convolving the output of the measured system $y(t)$ with this inverse filter $f(t)$:

$$h(t) = y(t) \otimes f(t) . \qquad (7)$$

The inverse filter $f(t)$ is generated in the following manner:

1) The logarithmic sweep (which is a causal and stable signal) is temporally reversed and then delayed in order to obtain a causal signal (the reversed signal is pulled back in the positive region of the time axis). This time reversal causes a sign inversion in the phase spectrum. As such the convolution of this reversed version of the excitation signal with the initial SineSweep will lead to a signal charac-

terized by a perfectly linear phase (corresponding to a pure delay) but will introduce a squaring of the magnitude spectrum.

2) The magnitude spectrum of the resulting signal is then divided by the square of the magnitude spectrum of the initial SineSweep signal.

The time and spectral representations of the inverse filter corresponding to the SineSweep (Fig. 9) are shown in Fig. 10. To minimize the influence of the transients introduced by the measurement system and appearing at the beginning and end of the emission of the excitation signal, the ends of the SineSweep signal are exponentially attenuated (exponential growth at the beginning and exponential decrease at the end).

In order to perform acoustical measurements over the entire audible range, the excitation signal must extend from 20 Hz to 20 000 Hz. As the transients outside this range have to be included, the choice of $f_1 = 10$ Hz (initial sweep frequency) and $f_2 = 22\,000$ Hz (final sweep fre-

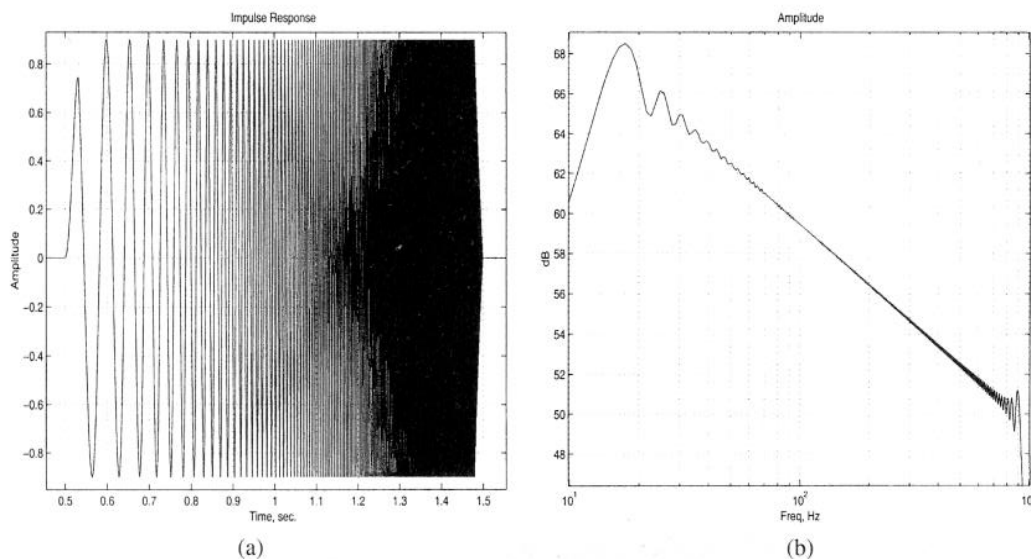(a)   (b)

Fig. 9. (a) Time representation of SineSweep excitation signal ($\omega_1 = 2\pi 10$ rad/s, $\omega_2 = 2\pi 1000$ rad/s). (b) Corresponding magnitude spectrum.

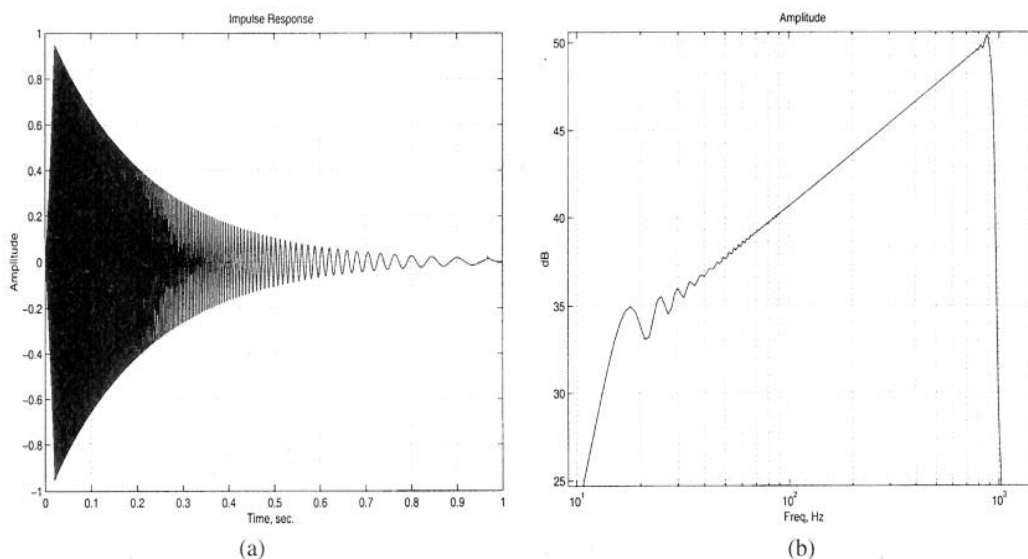(a)   (b)

Fig. 10. (a) Time representation of inverse filter corresponding to SineSweep signal in Fig. 9. (b) Corresponding magnitude spectrum.

quency) is realized.

In practice the SineSweep deconvolution leads to the apparition of a sequence of impulse responses clearly separated along the time axis (Fig. 11). Fig. 11 shows that the different harmonic distortion orders appear separately prior to the linear impulse response in increasing order, from right to left.

2 IMPLEMENTATION AND EXPERIMENTAL SETUP

2.1 Measurement System

A complete measurement system has been designed and realized to enable fast, reliable, and simple comparisons between the different methods. While several dedicated measurement systems already exist, they are generally expensive, immutable (because of the hardware implementation of the algorithms), and bulky. Therefore a graphical, highly configurable, portable program written

under Matlab 5.3 has been developed for automatically obtaining the impulse response with common elements such as a microphone, a loudspeaker, and a computer. This has led to a global, cheap, and adaptable measurement system (Fig. 12), which allows fast and accurate measurements of the impulse response.

The Matlab program controls the generation of the different excitation signals, their emission through the loudspeaker connected via a power amplifier to a full-duplex soundcard, and the simultaneous recording of the signal at the microphone. The deconvolution technique is then performed automatically. The time required for one measurement is very short: only a few seconds are necessary to obtain the impulse response of an acoustic space.

2.2 Room Impulse Response Measurement

To measure room impulse responses accurately, the measurement system characteristics must be taken into consideration. Calibration of the entire measurement
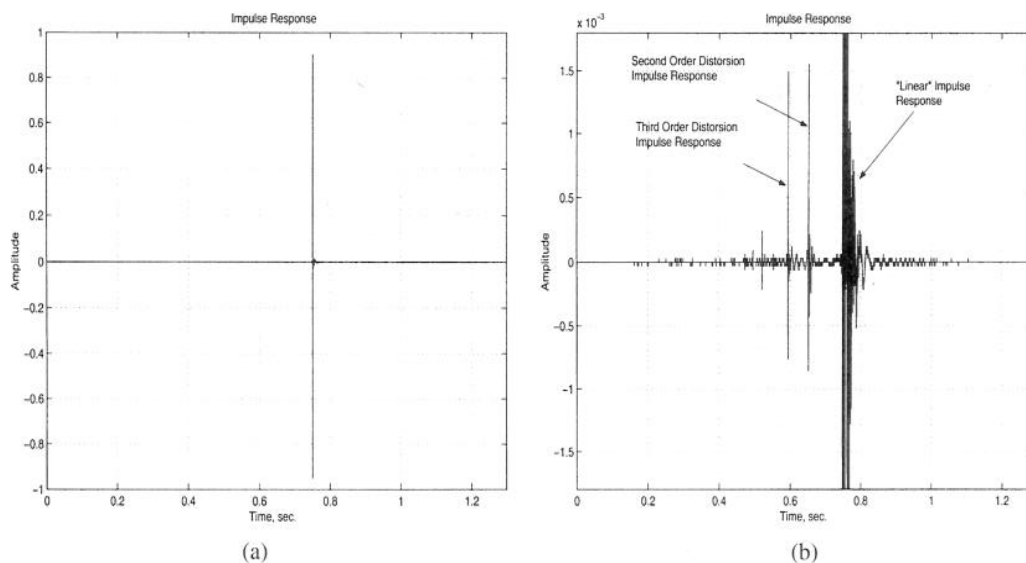
Fig. 11. (a) Impulse response obtained in anechoic room with a logarithmic SineSweep of 1 s characterized by $\omega_1 = 2\pi 10$ rad/s and $\omega_2 = 2\pi 22\,000$ rad/s). (b) Zoom on this response, showing extraordinary precision of achievable results.
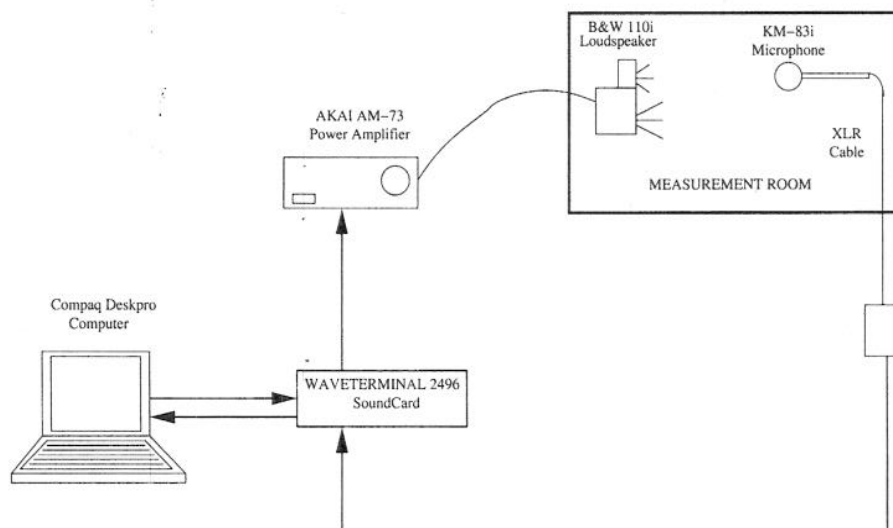
Fig. 12. Schematic representation of measurement system.

chain requires an inverse filtering correction which, in this case, is realized by the application of the time reversal mirror filter technique [26]. This technique generates a preequalized excitation signal in three steps:

1) Determination of the measurement system impulse response in anechoic room (for example, by using one of the techniques described earlier).

2) Time reversal of the system impulse response after appropriate truncation and addition of a time delay in order to obtain a causal result.[1] This "reversed" impulse response is linearly convolved with the excitation signal (MLS, sweep, . . .) that has to be preequalized.

3) Division of the spectrum magnitude of the signal obtained in step 2) by the square of the measurement system magnitude response [fast Fourier transform of the impulse response obtained in step 1].

The results of the preequalization technique are presented in Fig. 13. It can be seen that the phase spectrum is perfectly linear and the amplitude spectrum is almost constant (the residual oscillations around the mean value do not exceed ±0.4 dB) in the range between 40 Hz and 18 kHz.

3 COMPARISON OF METHODS

A comparison of the four impulse response measurement methods was first carried out in the anechoic room in order to ensure individual control of the set of parameters conditioning the measurement. The characteristic parameters of each method were chosen in order to allow an objective comparison. For example, the duration of the excitation signals and the number of averages used have been maintained constant during all measurements.

The program written under Matlab allows the following parameters to be modified:

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[1]The aim of this step is to inverse the phase polarity of the signal.

1) Parameters common to all methods

- Sampling frequency
- Number of averages (number of times the emitted signal will be sent to the loudspeaker)
- Recording mode: mono or stereo (for example, for head-related impulse response measurements).

2) Parameters specific to each method

- *MLS*: Order of the MLS sequence (maximum order = 19; number of samples contained in one period of an mth-order MLS sequence = $2^m - 1$).
- *IRS*: Order of the IRS sequence [maximum order = 19; number of samples contained in one period of an mth-order IRS sequence = $2 * (2^m - 1)$].
- *Time-stretched pulses*: Total duration of pulse and stretching percentage (ratio between amount of time during which pulse has a nonnegligible amplitude and total duration of pulse).
- *SineSweep*: Initial frequency, final frequency, sweep duration, and duration of silences inserted after each sweep.

Fig. 14 illustrates the arrangement of the transducers in the anechoic room for comparing the different measurement techniques.

3.1 Optimal Parameters

Fig. 15 gives a general survey of the impulse responses measured with the four methods presented when the measurement parameters (that is, the amplification sound level) were optimized individually for each measurement technique. The magnitude scale was enlarged to focus on the residual noise present.

The reverberation time and the ambient noise level in the anechoic chamber being very low, the choice of an MLS sequence of order 16 seemed to be a reasonable compromise between measurement time and good signal-to-noise ratio. The durations of the excitation signals used

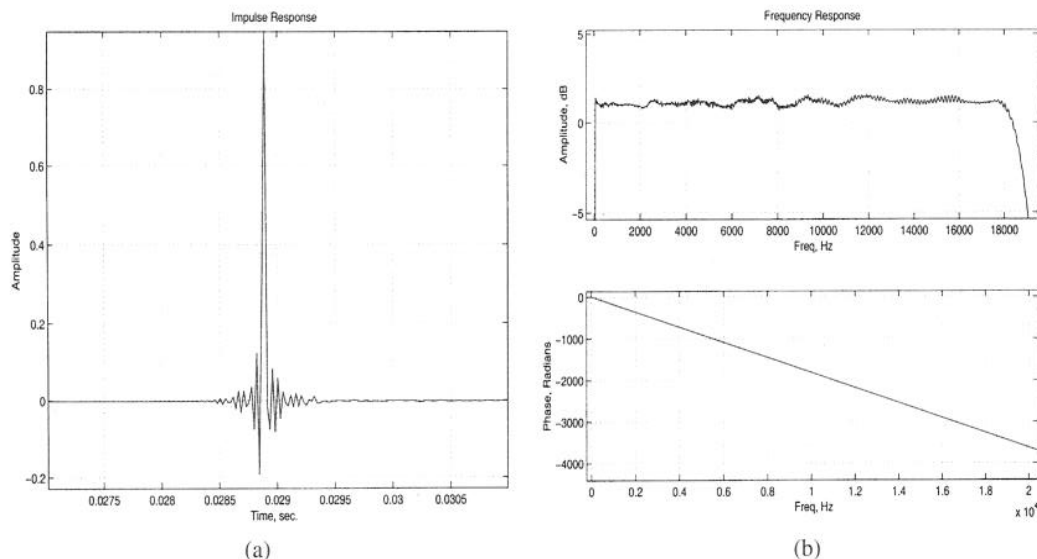(a)   (b)

Fig. 13. (a) Impulse response obtained in anechoic room when excitation signal has been preequalized (logarithmic SineSweep technique was used). (b) Corresponding magnitude and phase spectra.

in the other measurement methods were then adjusted according to the duration of the MLS excitation signal (that is, 1.5 s for a sampling frequency of 44 100 Hz).

Thus in order to obtain comparable measurements, the following parameters were used:

• Sampling frequency: 44 100 Hz

• MLS and IRS sequence orders: 16 (corresponding to signals of 1.5- and 3.-s duration, respectively, according to the chosen sampling frequency)
• Output amplification level: optimized in accordance with the method used by trial-and-error adjustment of the amplifier knob (the different levels used are listed in Table 1)
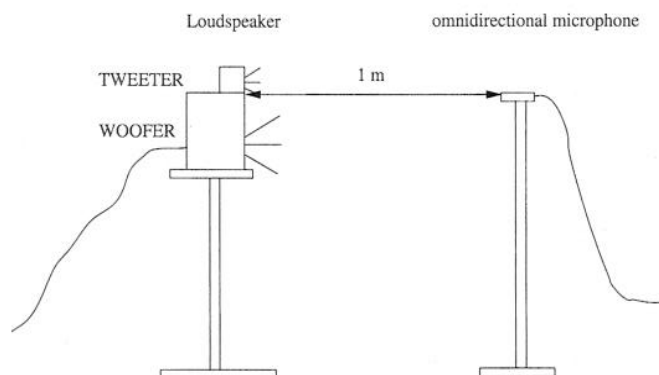

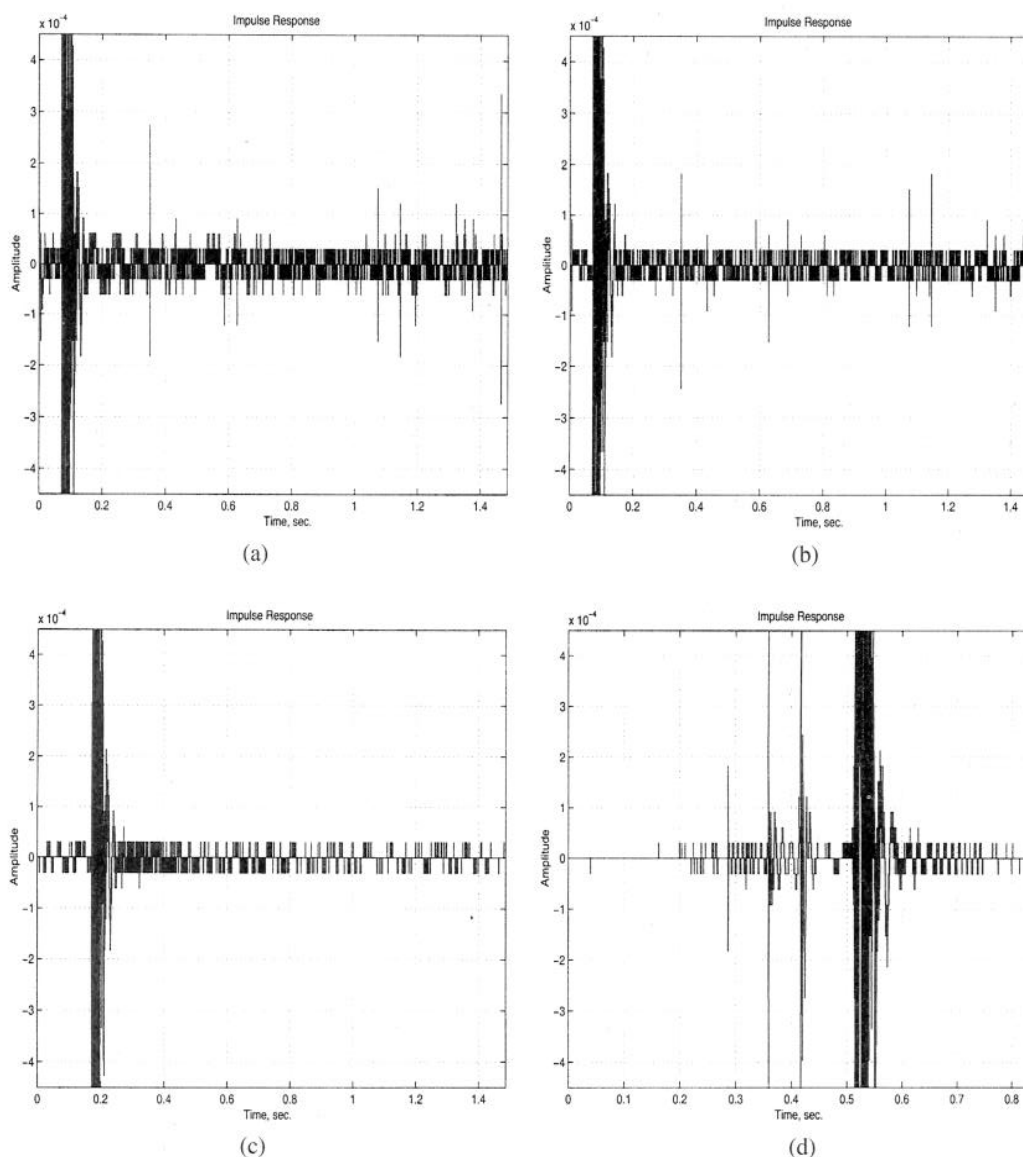Fig. 14. Arrangement of measurement elements in anechoic room.

Fig. 15. Zoom on impulse responses obtained in anechoic room. (a) MLS. (b) IRS. (c) Time-stretched pulses. (d) SineSweep.

- Time-stretched pulse duration: 1.5 s
- Time-stretched pulse stretch percentage: 80%
- Initial and final SineSweep frequency: $f_1 = 10$ Hz, $f_2 = 22\,000$ Hz
- SineSweep duration: 1.5 s
- No averaging
- Noise level in the anechoic room when the computer is present: 30 dB.

When the measurement parameters are optimized, the differences between the MLS and IRS methods tend to vanish [see Fig. 15(a) and (b)]. Furthermore the use of a relatively low output level and the timbre (a white noise is less disturbing than a sweep) of these methods is an advantage if the measurements are to be made in occupied rooms. The advantage of the IRS technique may still be considered. For example, at the extreme right of Fig. 15(b) the peak existing in Fig. 15(a) has disappeared. The disappearance of the distortion peaks when the time-stretched pulses method is used is clearly shown in Fig. 15(c). Finally, the perfect separation of the harmonic distortions from the linear impulse response with the SineSweep method is evident in Fig. 15(d).

The advantage of this last method lies in the fact that a tedious optimization process of the measurement parameters is not needed to obtain optimal results since the limitation on the output amplification level to avoid significant distortions no longer exists. (The nonsuperposition of the tail of the impulse response corresponding to the second-order distortion with the linear impulse response is the only precaution that has to be taken by choosing a sufficient duration of the SineSweep signal [21].)

In Fig. 15 the time axes do not have the same origins in order to focus on details particular to each method.

3.2 Signal-to-Noise Ratio

In order to perform an objective comparison of the impulse response qualities, the optimum signal-to-noise ratios achievable for each technique have been compared. In the following, the signal-to-noise ratio definition used is the ratio expressed in dB between the average power of the signal recorded by the microphone and the average power of the noise and distortions present in the tail of the deconvolved (linear) impulse response. Obviously, we might expect a better signal-to-noise ratio for the SineSweep method since there are no distortion artifacts present in the tail of the deconvolved (linear) impulse

Table 1. Optimum sound levels at the position of the microphone for each method.

MLS	IRS	Time-Stretched Pulses	SineSweep
75.5 dB	75.5 dB	83.9 dB	92.5 dB

Table 2. Optimum signal-to-noise ratios for each method.

MLS	IRS	Time-Stretched Pulses	SineSweep
60.5 dB	63.2 dB	77.0 dB	80.1 dB

response. This affirmation is confirmed by the results presented in Table 2.

The maximum signal-to-noise ratio when a 16-bit quantization is used corresponds to 98 dB [27]. This upper limit will of course never be reached in practice because of undesired contributions such as acoustical noise, electrical noise in the measurement system, quantization errors, and nonlinear distortions, principally due to the loudspeaker.

Table 2 shows the optimum signal-to-noise ratios (that is, the signal-to-noise ratios obtained when the measurement parameters have been optimized) for each of the four methods presented.

Bleakley and Scaife [11] have shown that the signal-to-noise ratio for the MLS sequence increases by 3 dB when the period length of the MLS sequence is doubled. It is thus logical to obtain a 3-dB gain for the IRS technique in comparison with the MLS technique since the length of an IRS sequence is twice the length of the corresponding (same-order) MLS sequence.

The noticeable gain of 14 dB of the time-stretched pulses method over the IRS method can be explained by the use of an optimum output sound level far above the one used in the MLS or IRS cases, as well as by the disappearance of the spurious distortion peaks.

Finally the excellent signal-to-noise ratio (80 dB) obtained with the SineSweep technique is due to the total rejection of the nonlinear distortion artifacts prior to the linear impulse response. The output signal level is no longer limited by the need to reduce the nonlinear influence since all nonlinear distortions are measured separately. This leads to the optimum signal-to-noise ratio (20 dB better than for the MLS method). This signal-to-noise ratio is only 3 dB above the one given by the time-stretched pulses method, but the nonsuperposition of the distortion artifacts is guaranteed in this case.

All these signal-to-noise ratios have been obtained through direct measurements (no averaging). Of course, better signal-to-noise ratios would have been obtained if averaging had been used.

3.3 Impulsive Noise Immunity

Fig. 16 shows the impulse responses obtained when measurements are performed in an environment where impulsive noise is simultaneously present. The IRS method gives results approximately identical to those of the MLS technique, and thus its corresponding impulse response is not shown. As announced previously, only the pseudorandom noise techniques (MLS and IRS) possess the ability of randomizing the phase of any component in the recorded signal that is not correlated to the input signal emitted in the acoustical space. Thus any additional noise (white or even impulsive) will be distributed uniformly along the deconvolved impulse response. Therefore additive impulsive noise (appearing as additive white noise in the deconvolved impulse response) is subject to posterior attenuation by averaging techniques.

On the contrary, the presence of impulsive noise (Fig. 16) when the time-stretched pulses or the SineSweep techniques are used, heavily compromises the impulse

response deconvolution process through the presence of excitation signal residues in the deconvolved impulse response. These residues being strongly correlated with the excitation signal will not be eliminated properly if posterior averaging techniques are used.

As a first conclusion we may say that in a (nonrandom) noisy environment, the MLS (IRS) method is subject to giving better results than the other two methods.

3.4 Impulse Response Measurements in Classical Rooms

The properties described so far were mostly illustrated by measurements performed in anechoic rooms. In this section we will show the results obtained when the measurements are performed in more classical rooms such as auditoriums or lecture rooms.

As can be seen in Figs. 17 and 18, most of the properties that were illustrated in the case of anechoic measurements are still visible. In particular we can focus on the separation of the different harmonic distortions from the

linear impulse response in the case of the SineSweep technique. In this case the impact of reverberation on each impulse response can be clearly seen and illustrates the need for nonsuperposition of the second-order distortion with the linear impulse response by using a sufficiently long SineSweep.

4 CONCLUSIONS

A complete, inexpensive, and parameterizable measurement system has been realized in order to compare different impulse response measurement methods. This system is based on a computer program written under Matlab 5.3, allowing an automatic and easy measurement to be realized. On the basis of this measurement system, four different methods have been compared.

The comparison of the four methods leads to the following conclusions.

1) The MLS (IRS) method seems the most interesting method when the measurements have to be performed in
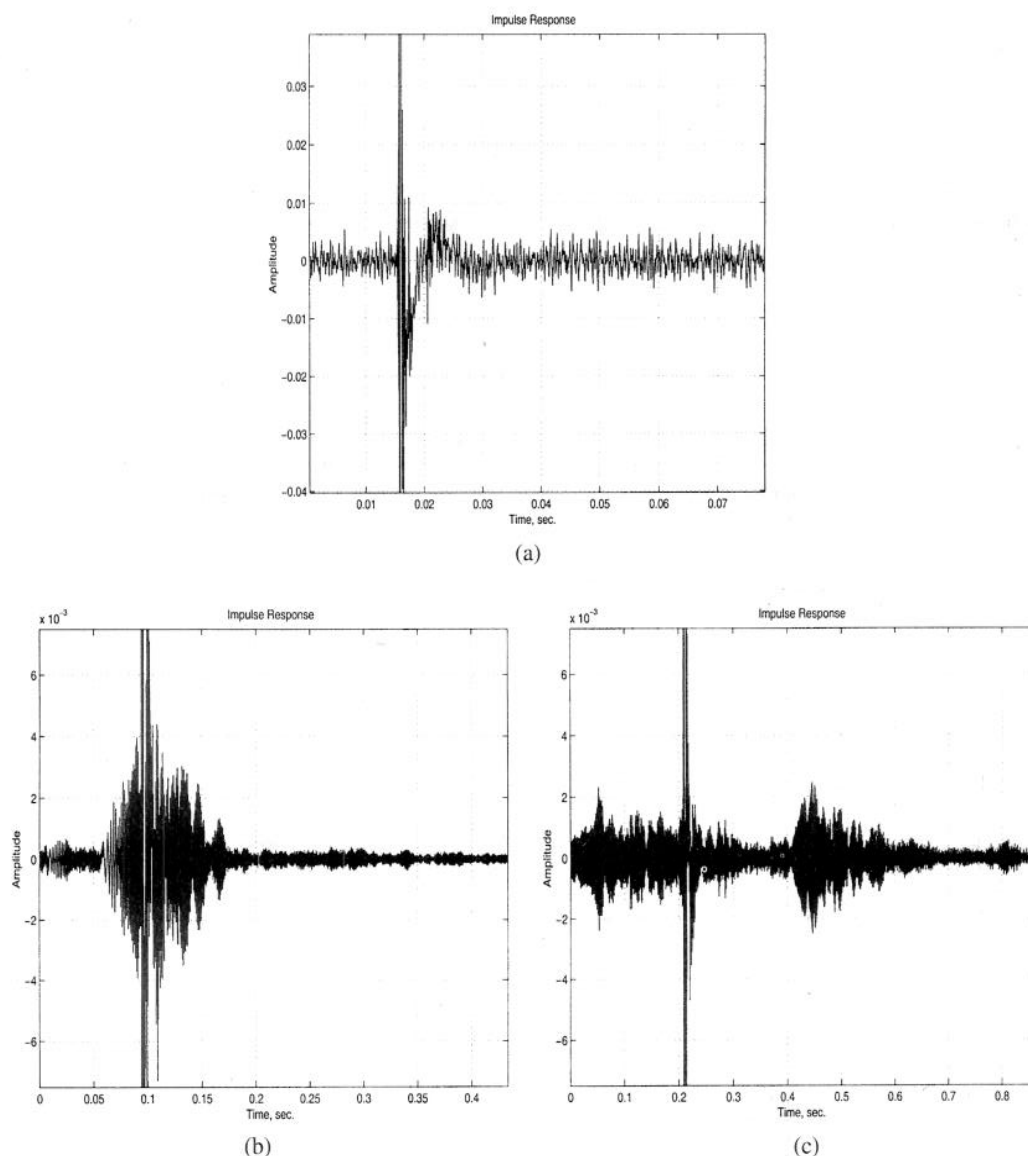

(a)

(b)

(c)

Fig. 16. Impulse responses obtained in anechoic room when intense impulsive noise is simultaneously present. (a) MLS. (b) Time-stretched pulses. (c) SineSweep. Note that amplitude scales are not identical.

an occupied room or in the exterior because of its strong immunity to all kinds of noise (white, impulsive, or others), its weak optimum output sound level, and its timbre (white noise is more bearable and more easily masked out than are sweep signals). However, its major drawback lies in the tedious calibration that has to be carried out to obtain optimum results and in the appearance of spurious peaks (distortion peaks) due to the inherent nonlinearities of the measurement system.

2) The time-stretched pulses method avoids the appearance of the distortion peaks. However, the remaining nonlinear artifacts can possibly be superimposed with the deconvolved "linear" impulse response. The presence of a residue of the excitation signal in the deconvolved impulse response is a result of such superposition problems. This residue can be almost completely eliminated if a precise calibration of the measurement parameters (mainly the output level) is realized. However, its timbre and the high optimum output signal level needed to mask out the ambi-

ent noise make it unusable in occupied rooms.

3) The perfect and complete rejection of the harmonic distortions prior to the "linear" impulse response, their individual measurement, and the excellent signal-to-noise ratio of the SineSweep method make it the best impulse response measurement technique in an unoccupied and quiet room. Moreover, unlike the preceding methods, it does not necessitate a tedious calibration in order to obtain very good results (no compromise between the signal-to-noise ratio and the superposition of nonlinear artifacts in the room impulse response). However, like the time-stretched pulses method, the SineSweep technique is not recommended for measurements in occupied rooms.

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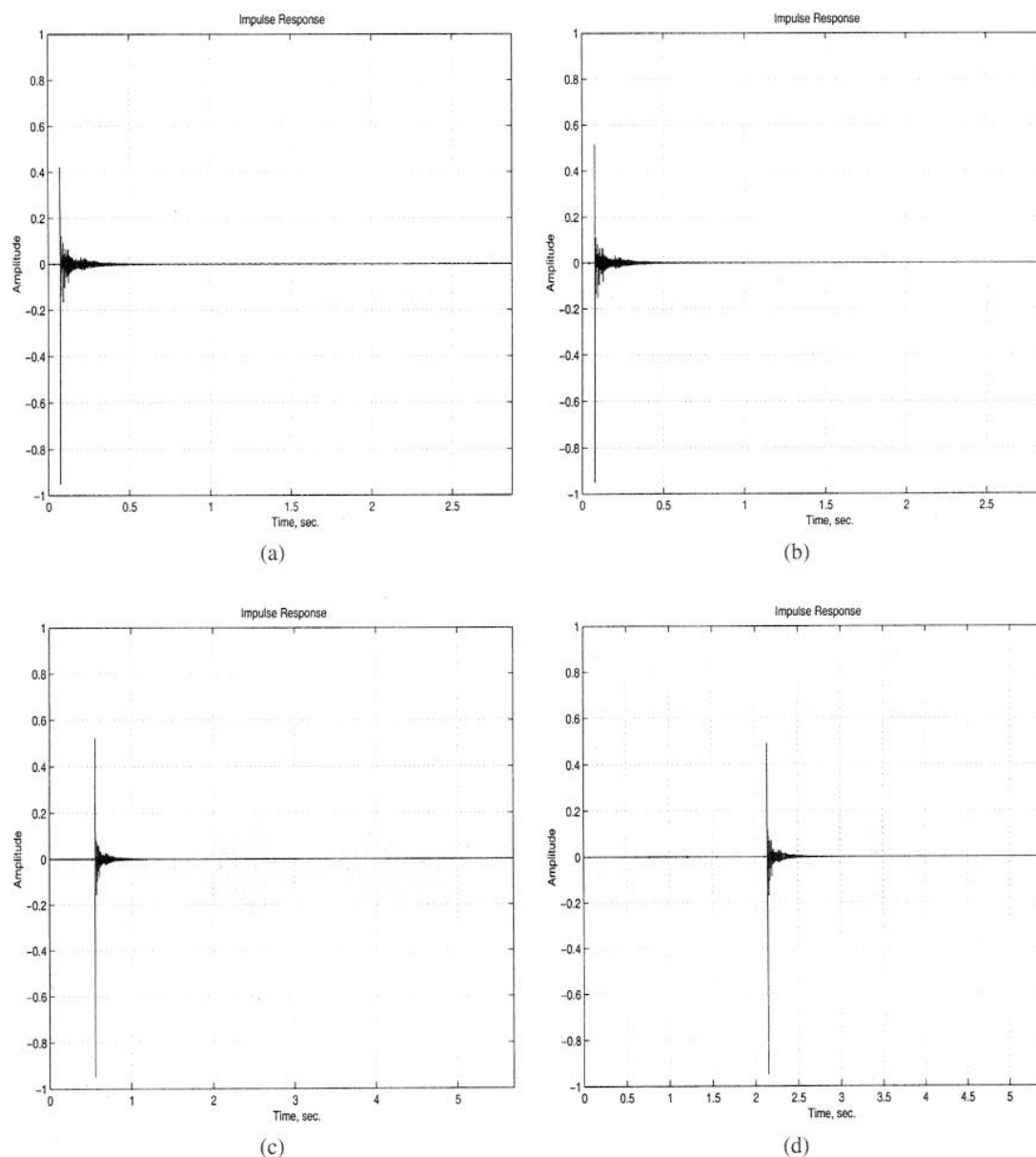

Fig. 17. Impulse responses obtained in auditorium 604, Europe Amphitheater, University of Liège (Belgium). (a) MLS. (b) IRS. (c) Time-stretched pulses. (d) SineSweep.

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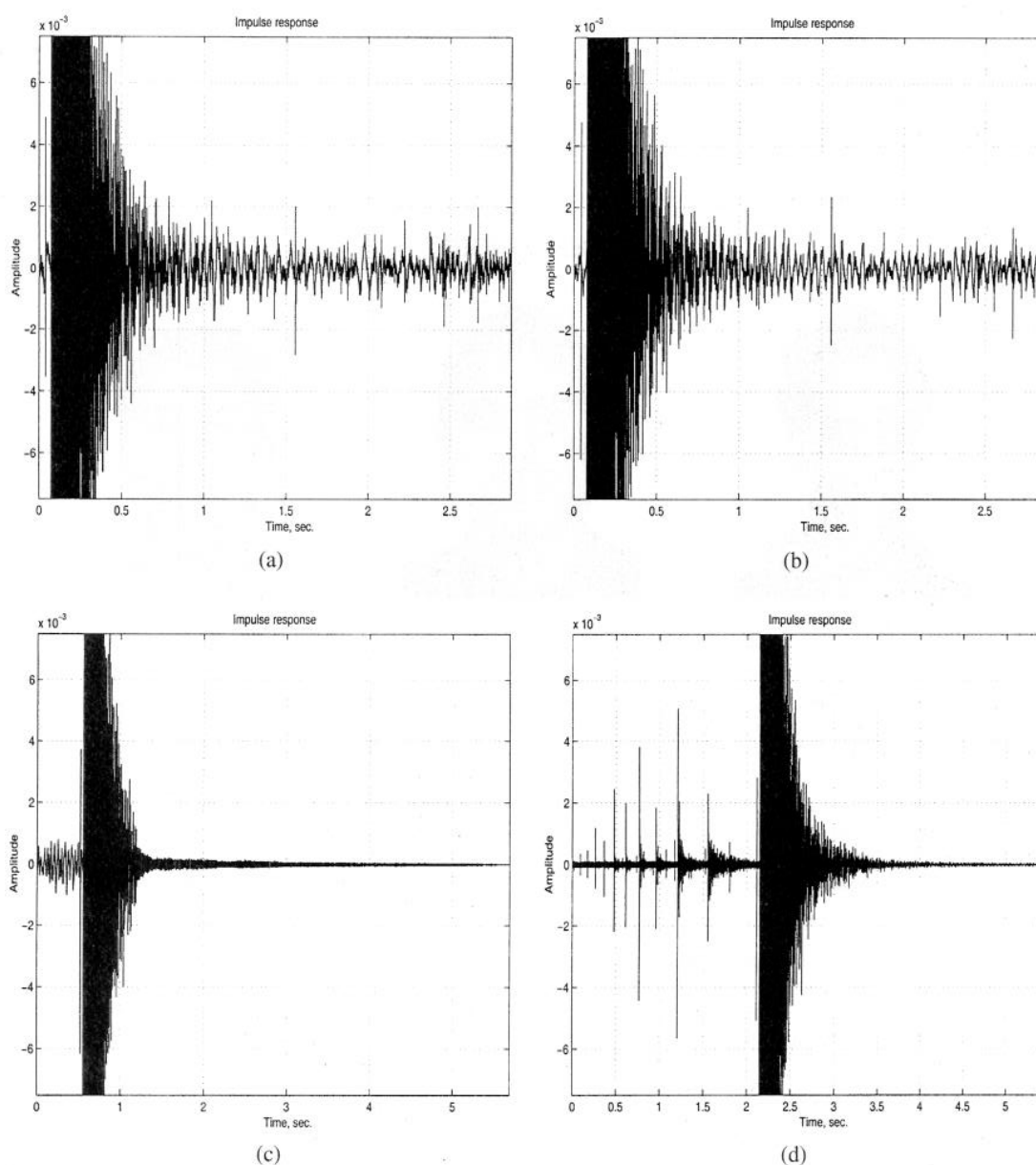

Fig. 18. Zoom on impulse responses obtained in auditorium 604, Europe Amphitheater, University of Liège (Belgium). (a) MLS. (b) IRS. (c) Time-stretched pulses. (d) SineSweep.

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