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Generation of desired sound impulses

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Abstract

There have been two major difficulties that hinder the use of a loudspeaker–pipe system as a short-duration acoustic point source generator. They are the non-flat response of the loudspeaker and the reflection at the pipe exit. Here, a pre-distorted digitally synthesized input signal is used not only to yield a desired impulse waveform but also cancel its first reflection and hence all subsequent reflections from the pipe exit. Our experimental results demonstrate the feasibility of generating repeatable sharp impulses of various desired waveforms and durations as short as 0.5 ms.

1. Introduction

An impulse point sound source capable of generating spherical sound waves of a desired form in space and short duration in time finds many applications, such as the measurement of acoustic impedance [1], barrier diffraction [2] and room reverberation [3]. The response to an impulse point source provides the basic information about an experiment as the Green's function in corresponding analysis. With an impulse point source, it is a simple matter to differentiate the reflected waves from an object and those from other distant objects or confinement. The spatial simplicity of the incident sound wave generated by such a sound source also allows a direct comparison between experimental measurement and theoretical prediction.

There have been several difficulties associated with the design of a good-performance impulse point sound source. Devices such as electric sparks, guns and toy crickets have been used to generate impulse sound waves, but it is difficult for them to produce controllable, repeatable pulse waveforms desired for various acoustic measurements [1,2]. A loudspeaker is a common sound-generation device that meets the requirement of good repeatability, but often too large to be treated as a point source unless the distance between the source and the test sample is sufficiently large. One way to produce a point source is to connect the loudspeaker with a wave-guiding pipe whose exit is small compared with the wavelength. However, it is not straightforward for such a loudspeaker–pipe system to generate a sharp sound pulse due to its non-flat frequency response and the reflections between the pipe exit and its interior. Although it is possible to edit out the pipe end reflection by using a very long pipe [4], the practicality of such a sound source is limited.

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In this paper, a novel sound source technology is developed which digitally control the loudspeaker–pipe system to generate sharp, spherical pulse waves. The basic principle is to use a digitally synthesized pre-distorted input signal to achieve firstly a desired output waveform and secondly the annihilation of the first reflection and hence all subsequent reflections between the pipe exit and its interior. The method of synthesized pre-distorted input has been developed and refined by several authors [4–8], where the problem of the pipe exit reflection was not tackled. Various degrees of spectral and temporal discrepancies were evident in these attempts, and the impulses reported ranged from several to tens of mini-seconds with the exception of the triangular pulse of 200 μs by a 123 cm horn [6]. The present experiment eliminates the pipe exit reflection effectively and demonstrates the generation of sound impulses of various desired waveforms and durations as short as 0.5 ms. In principle, this method is not subjected to the length of the pipe connected to the loudspeaker. In practice, equally good or better waveforms were obtained using a 0.5 m long pipe than a 1.4 m pipe. The digital control method and the experimental results are presented in following sections.

2. Sound source configuration and test equipment

As shown in Fig. 1, the loudspeaker is a 6″ Seas H455 mid-frequency woofer whose recommended frequency range extends from 40 to 4000 Hz. The woofer is selected for its good performance at low- to mid-frequencies. A steel enclosure of 3.0ℓ volume is used to provide sound insulation and stiffness for the loudspeaker to operate. The sound wave from the loudspeaker is guided through a $d = 300$ mm conical reduction to a straight 20 mm pipe of length l. The test equipment includes a power amplifier, an NI DAQ6062E data acquisition board, a BNC 2090 adaptor, a 1/2″ ACO 7012 condenser microphone for receiving the sound signal, and a laptop computer for data processing. The AI and AO channels of the DAQ board are programmed to be synchronized and have the sampling rate of 100 kHz.

3. Basic control principle of output

3.1. Determination of transfer function

The first step of the control method is to measure the transfer function of the loudspeaker–pipe system. In principle, the transfer function can be computed from the response of the system to an arbitrary input signal, such as a triangular or rectangular signal [6,7]. However, the spectra of such signals are not flat and may even

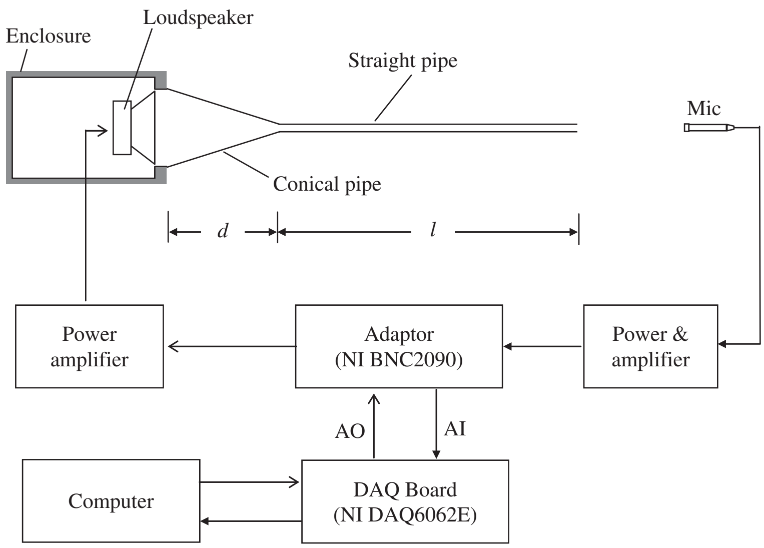

Fig. 1. The sound source configuration and the test equipment.

contain many zero values as the frequency extends from low to high. For example, a 0.5 ms rectangular pulse has the first zero value at 4000 Hz. Since the transfer function is given by the ratio between the output and the input signals, the occurrence of small or zero values in the denominator degrades the numerical accuracy of the computation. To overcome this problem, we use the impulse response of a digital 10th-order Butterworth filter of 10 kHz cutoff frequency and about 0.5 ms duration as the input signal $x_B(t)$, Fig. 2(a). Fig. 2(b) shows the flat, up to 10 kHz, frequency response of the filter, $X_B(\omega)$. Thus, the transfer function of the loudspeaker–pipe system is given by

$$H(\omega) = \frac{Y(\omega)}{X_B(\omega)}, \tag{1}$$

where $Y(\omega)$ is the output from the microphone. A time-domain averaging method has been used to increase the signal-to-noise ratio, and it is demonstrated that the noise level can be effectively reduced after 50 times of averaging.

3.2. Reflection at pipe exit

We consider first the case when a 1.4 m long pipe is connected to the loudspeaker. The measured time response to the input signal $x_B(t)$ is given in Fig. 3. We can see that, due to the reflection of the pipe exit, subsequent weaker pulses appear every 8.2 ms after the initial pulse. The 8.2 ms delay coincides with the distance traveled between the initial and subsequent reflected pulses, or twice the length of the straight pipe.

3.3. Control on the initial pulse

Applying the Fourier transform to the time response truncated just before the reflected pulse, we obtained the transfer function of the speaker–pipe system, $H_e(\omega)$, as shown in Fig. 4(a). For this speaker–pipe system to

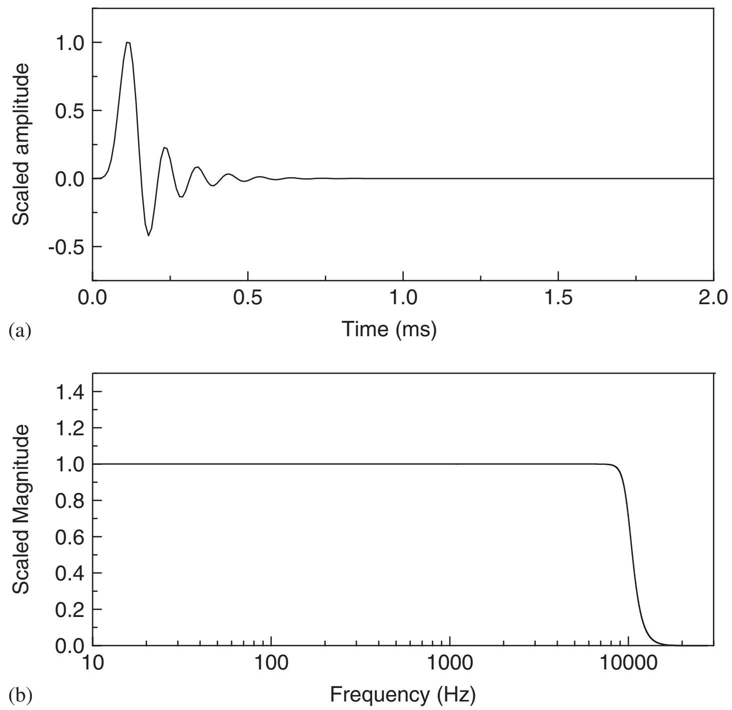

Fig. 2. The input signal for obtaining the transfer function: (a) the impulse response and (b) the frequency response of a 10th-order Butterworth filter of 10 kHz cutoff frequency.

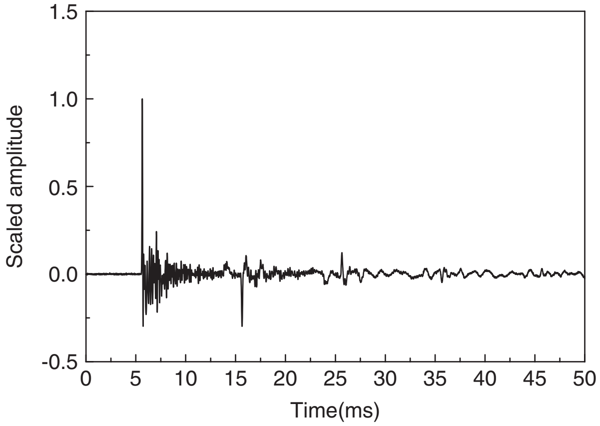

Fig. 3. The time response of the speaker–pipe system to the input $x_B(t)$, the length of the sound-conducting pipe is 1.4 m.

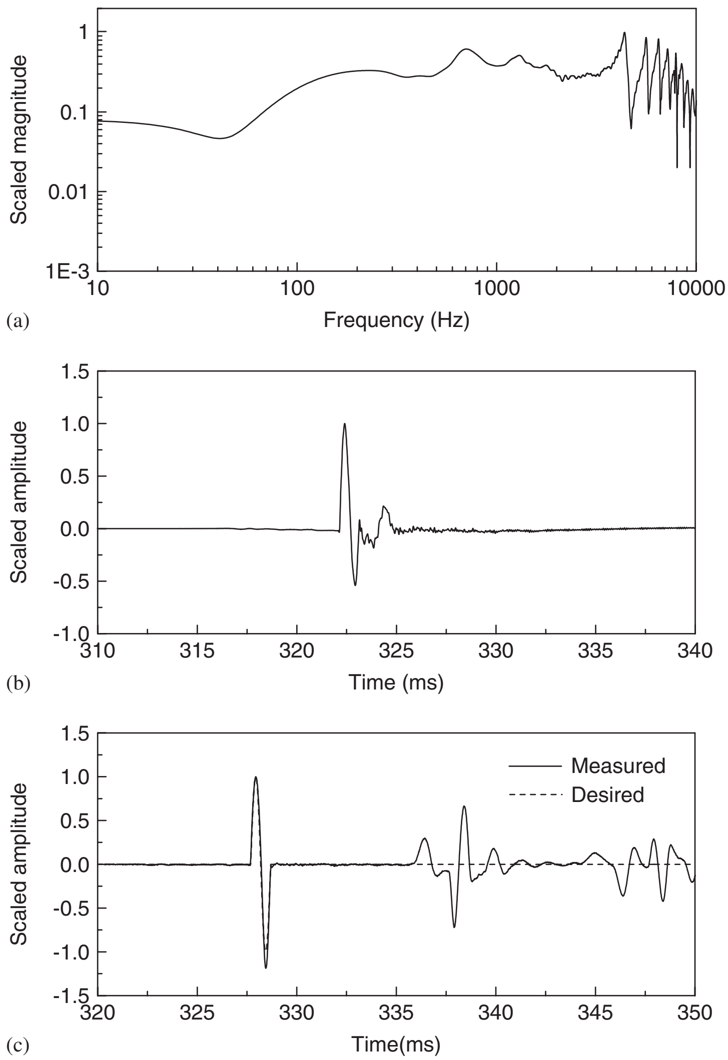

Fig. 4. Initial control on the generation of a desired one-cycle Sine wave of 1 ms duration on a 1.4 m long pipe: (a) transfer function not including the effect of pipe end, (b) synthesized input signal, and (c) measured output signal.

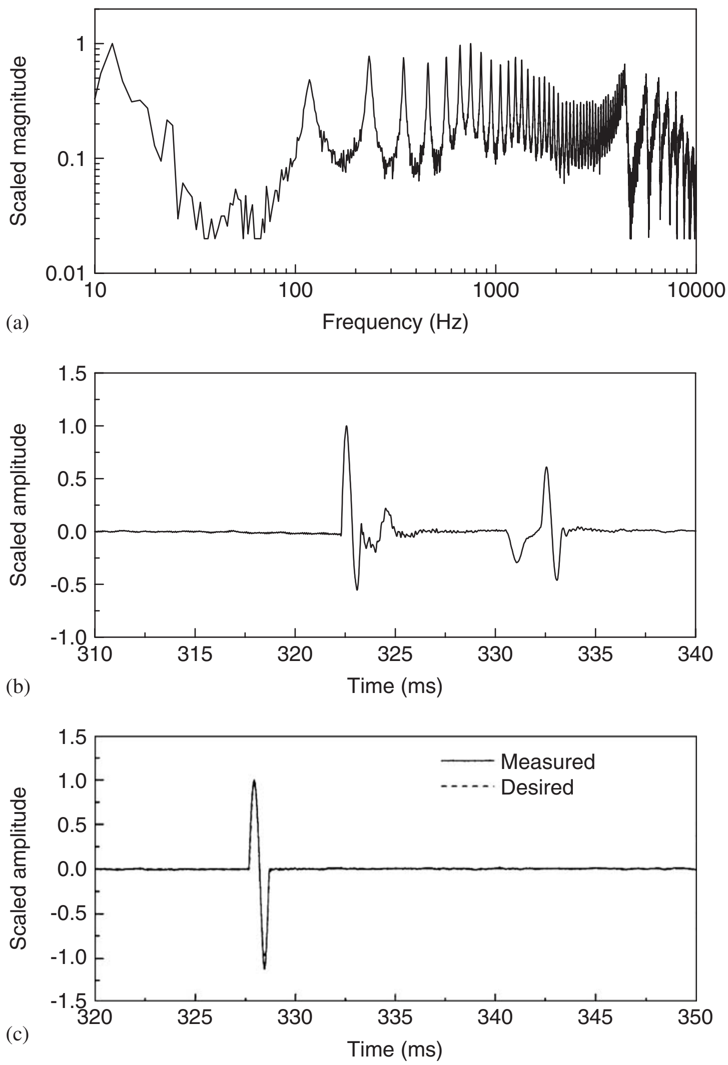

Fig. 5. Reflection-corrected control on the generation of a desired one-cycle Sine wave of 1 ms duration on a 1.4 m long pipe: (a) transfer function including the effect of pipe end, (b) synthesized input, and (c) measured output.

Table 1
The parameters for the pulse waveform of Eq. (3)

Pulse duration	$\lambda_n^{\pm}$	$A_n^{\pm}$
1 ms	$\pm 2598.7+4394.7i$	$+1.2002 \pm 2.6113i$
	$\pm 7462.2+4223.7i$	$-1.2277 \mp 1.2012i$
	$\pm 13456+2635.4i$	$+0.02748 \mp 0.10279i$
0.5 ms	$\pm 5357.6+10867.8i$	$+4.9361 \pm 8.6336i$
	$\pm 15726.6+10118.5i$	$-4.7103 \mp 0.97651i$
	$\pm 27545.1+7653.3i$	$-0.22584 \mp 0.53226i$

emit a pulse of desired waveform $z(t)$, instead the synthesized pre-distorted signal $s(t)$ is fed into the speaker:

$$s(t) = F^{-1}[Z(\omega)/H_e(\omega)] = F^{-1}[Z(\omega)X_B(\omega)/Y_e(\omega)], \tag{2}$$

where F^{-1} denotes the inverse Fourier transform of $Z(\omega)$, the spectrum of $z(t)$, conditioned by the Butterworth filter $X_B(\omega)$ and the truncated time response $Y_e(\omega)$.

It is clear that $X_B(\omega)$ in Eq. (2) serves as a low-pass digital filter for the synthesized input signal. According to Nicolas et al. [5], it is important to apply filters to eliminate the low- and high-frequency components that affect the rendering of a desired waveform. The present work shows, however, that with a loudspeaker of good low-frequency performance the filtering of the low-frequency components is unnecessary.

Since the desired output signal is a pulse, it is expected that the synthesized input signal is also pulse-like. To obtain a crisp synthesized pulse-like input signal and avoid the division by the zeros or very small values of $Y_e(\omega)$ in Eq. (2), $Y_e(\omega)$ is clipped or set to the critical value ε when $|Y_e(\omega)| < \varepsilon$. The use of a woofer of good low-frequency performance is also helpful in obtaining a clear pulse-like synthesized input. It was shown previously that when high-frequency tweeters were used instead of a woofer, the inability to generate low-frequency sound wave below 100 Hz had resulted in the non-impulse-like synthesized input and thus long tails in the controlled output signal [4,5].

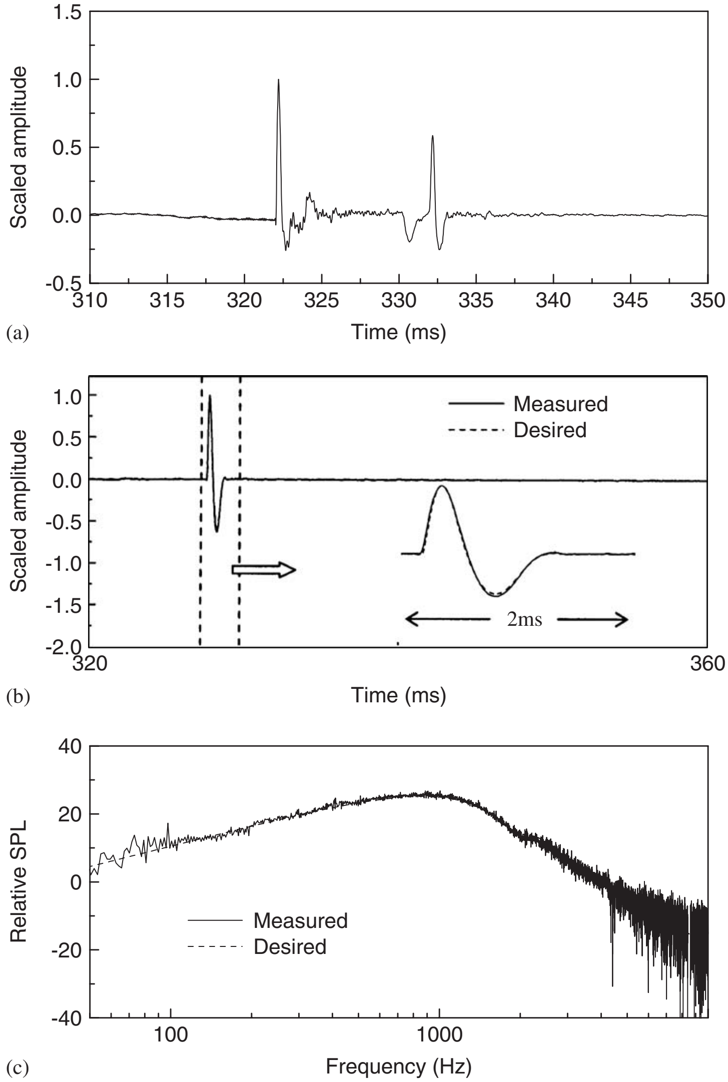

Fig. 6. Digitally controlled generation of the 1 ms impulse of Eq. (3) on a 1.4 m long pipe; (a) synthesized input, (b) measured pulse, and (c) measured spectrum.

Fig. 4 shows the synthesis of a one-cycle Sine wave of 1 ms duration. The synthesized input impulse in Fig. 4b is not very Sine-like, but the initial output impulse is satisfactorily controlled to have the desired one-cycled-Sine waveform, only followed by its first reflected impulse 8.2 ms after the initial pulse, as shown in Fig. 4c. Obviously, the reflection is undesired in an impulse measurement method. It is possible to edit out or separate the initial pulse from the pipe end reflection [4], using inconveniently a very long pipe however.

3.4. Control on both the initial pulse and the pipe exit reflection

The need to get rid of the pipe end reflection is clear if a sharp pulse is to be generated without the inconvenience of a very long pipe. The basic principle is rather simple. Fig. 5(a) shows the transfer function of the whole speaker–pipe system, $H(\omega)$, which has many resonance peaks due to the reflection from both ends of the 1.4 m long straight pipe. If this $H(\omega)$ is used instead of $H_e(\omega)$ in Eq. (2), the corresponding synthesized input signal, shown in Fig. 5(b), has the striking feature of two pulses. The first one is identical to that of Fig. 4(b), which is for shaping the initial pulse. The second one is actually for annihilating the first reflection from the pipe exit so that no subsequent reflections will occur. As expected, the time delay between the two

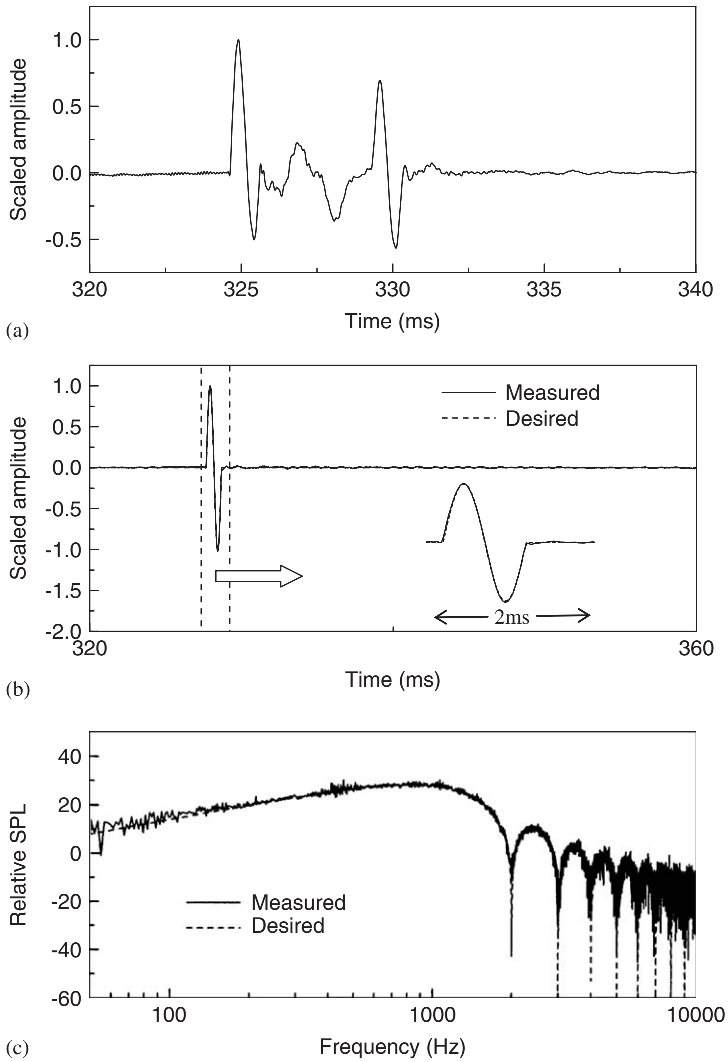

Fig. 7. Digitally controlled generation of a 1 ms one-cycle Sine pulse on a 0.5 m long pipe, (a) synthesized input; (b) measured pulse; (c) measured spectrum.

pulses is about 8.2 ms for a 1.4 m long pipe. This simple control strategy is automatically realized by the digital synthesis approach of Eq. (2). Surprisingly good result is shown in Fig 5(c), where no discernible reflection is found and the sharp 1 ms impulse so synthesized and generated agrees well with the desired signal.

4. Results and discussion

Short-duration pulse of different waveforms can be generated by the present method. Besides the one-cycled Sine impulse, we consider the type of waveform defined by the sum of damped sinusoids,

$$p(t) = \sum_{n=1}^{3} A_n^{\pm} \exp(\mathrm{i}\lambda_n^{\pm} t), \tag{3}$$

where $\mathrm{i} = \sqrt{-1}$, and $A_n^{\pm}$ and $\lambda_n^{\pm}$ are the complex pairs listed in Table 1, giving two sets of parameters for impulses of 1 and 0.5 ms duration, respectively. Fig. 6 shows that a single sharp pulse is generated over the entire time span. If there were reflections from the pipe exit, a secondary pulse would have appeared at about

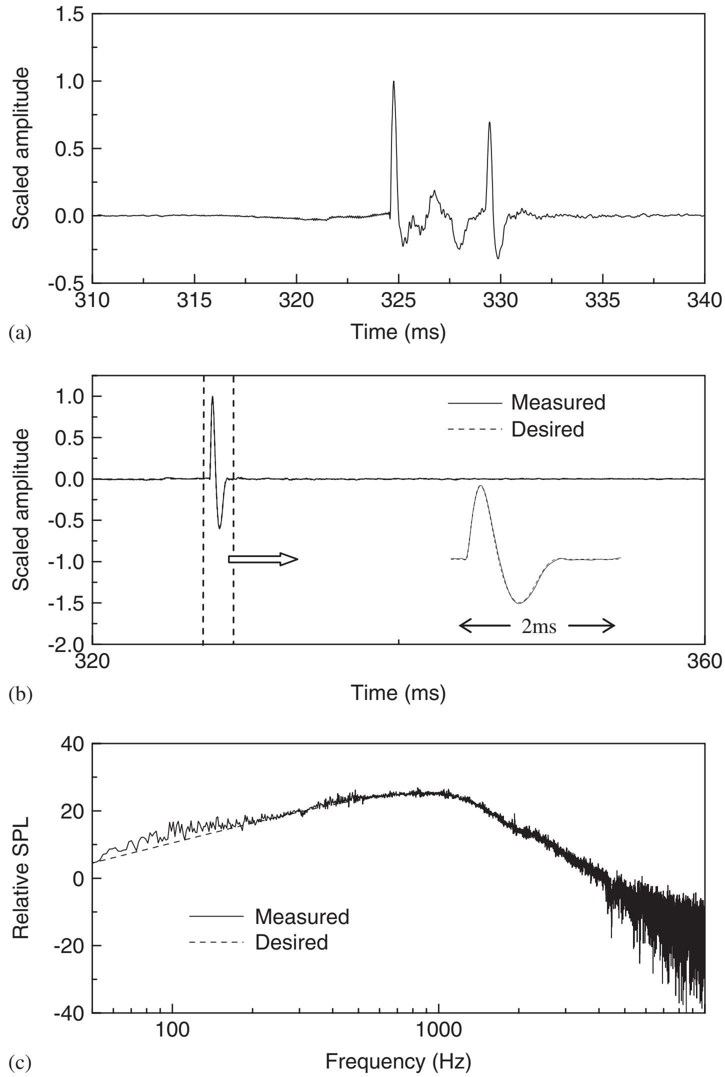

Fig. 8. Digitally controlled generation of a 1 ms pulse of Eq. (3) on a 0.5 m long pipe; (a) synthesized input, (b) measured pulse, and (c) measured spectrum.

335 ms, or 8.2 ms after the initial pulse. Also, the measured output signal agrees fairly well with the desired waveform as shown in Fig. 6(b). A comparison between the spectra of the measured and the desired pulses is presented in Fig. 6(c).

Previous studies have indicated that it is possible to edit out the pipe end reflection if a long straight pipe is connected to the loudspeaker [4]. However, this makes the sound source inconvenient or even impossible to use in circumstances where space is limited. It is demonstrated that the present method is applicable regardless of the pipe length. Figs. 7 and 8 show the same desired pulses on a straight pipe of only 0.5 m long. As shown in Fig. 7(a), similar to the case of a 1.4 m long pipe, the synthesized input signal consists of two pulses, but the delay between them is shortened to about 3 ms in relation with the pipe length. Figs. 7(b) and (c) show that the controlled output signal has the desired temporal and spectral fidelity of a one-cycled Sine. Similar results are shown in Fig. 8 for the desired waveform of Eq. (3).

We also tried to use the method to generate pulses whose duration is shorter than 1 ms. Again the desired waveforms of one-cycled Sine and Eq. (3) are considered but for the shorter duration of 0.5 ms. We can see from Figs. 9 and 10 that very sharp pulses are generated over the entire time duration, and the waveforms of

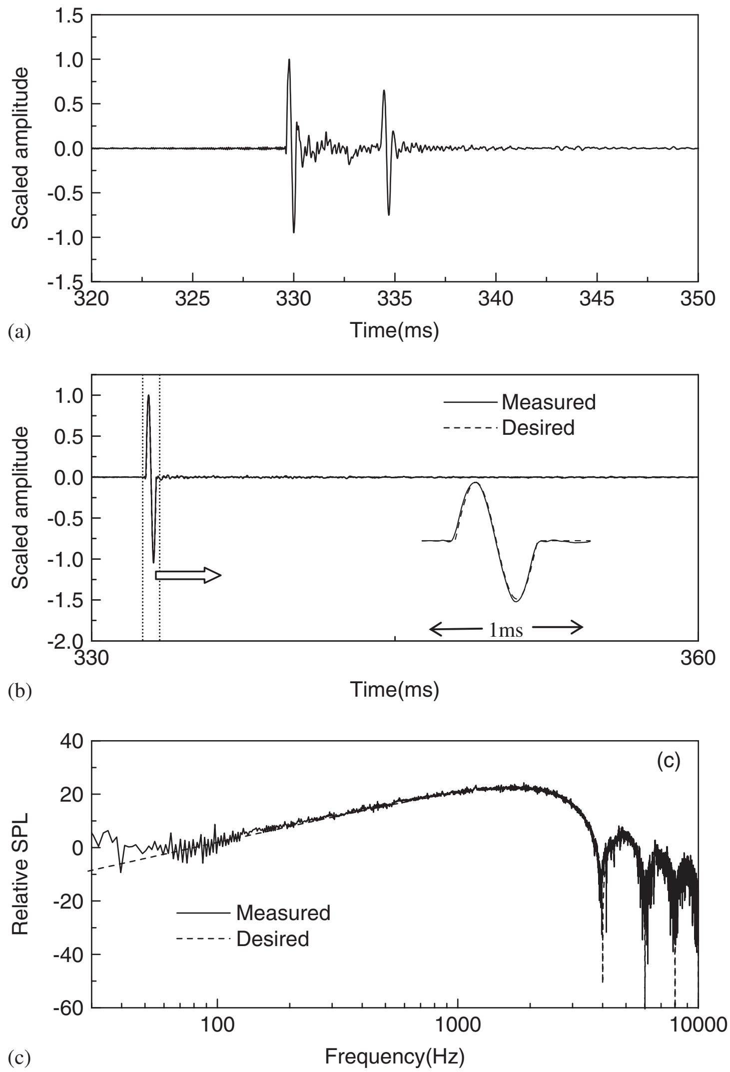

Fig. 9. Digitally controlled generation of a 0.5 ms one-cycle Sine pulse on a 0.5 m long pipe; (a) synthesized input, (b) measured pulse, and (c) measured spectrum.

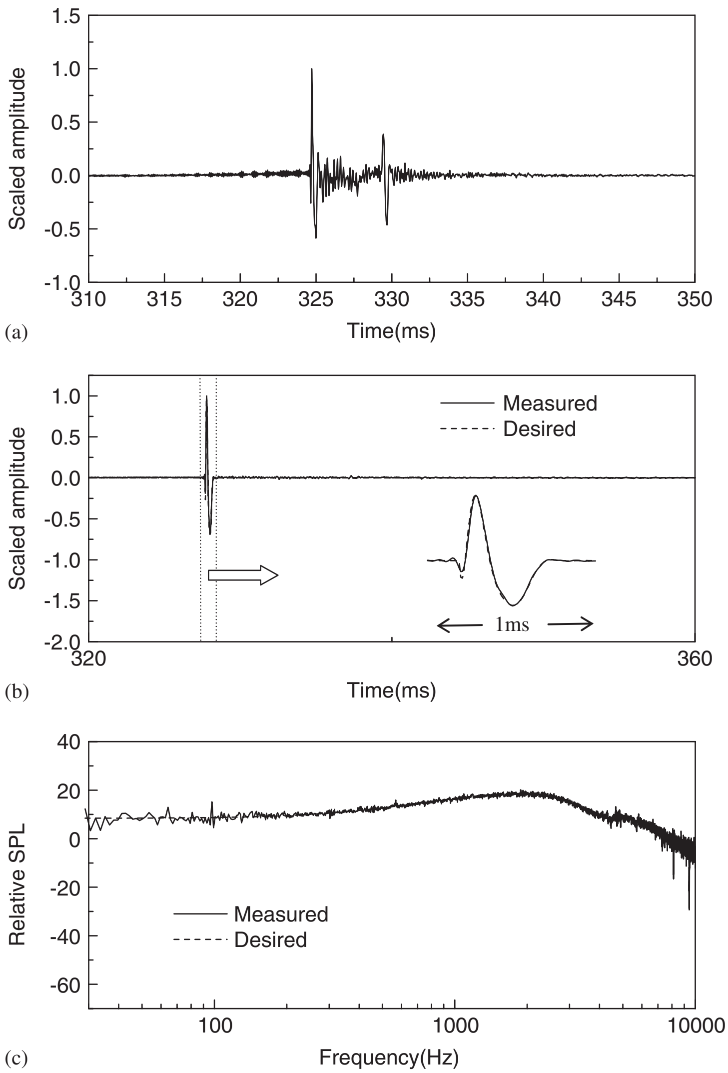

Fig. 10. Digitally controlled generation of a 0.5 ms pulse of Eq. (3) on a 0.5 m long pipe; (a) synthesized input, (b) measured pulse, and (c) measured spectrum.

the measured output signals are satisfactorily controlled except the minor differences at the very beginning of a pulse where the desired abrupt waveform has infinite time derivatives.

From Figs. 6–10, we can see that the spectra of the digitally controlled output signals follow those of the desired signals very well down to 40 Hz and up to 6000 Hz. So, the range of the frequency response of the digitally controlled sound source is even wider than the recommended operational range of the loudspeaker (40–4000 Hz). However, it is very difficult to further shorten the emitted pulse or widen its spectrum since the present digital control method is ultimately limited by the response of the loudspeaker itself.

When the measurement is carried out at a position ten times of the pipe diameter or 0.2 m from the pipe exit, the production of desired waveforms of up to 86 dB sound pressure levels can be well controlled using the present method. However, it was found in our experiment that as the sound pressure level was increased to 101.2 dB, the resulted waveform began to differ from the desired waveform by the occurrence of a trailing tail. When the sound pressure level was further increased to 107.0 dB, severe distortions were found in the output waveform and the reflection at the pipe exit was not completely cancelled. It should be noted that the sudden area change at the pipe exit reflects most of the sound wave back into the pipe. Only a small fraction of acoustic energy emits outside. When the sound emission is around 100 dB at the microphone position, the peak value of the sound pulse in the pipe reaches a level as high as 1000 Pa or 155 dB. Thus, the occurrence of nonlinear sound propagation renders the current linear method ineffective.

5. Conclusions

The non-flat response of the loudspeaker and the sound reflection at the pipe exit have hindered the employment of the loudspeaker–pipe system for generating desired short-duration pulses in acoustic measurements. It is demonstrated that the sound reflection at the pipe exit can be effectively annihilated by the controlled emission of a second sound pulse and the non-flat response compensated by the transfer function of the loudspeaker. Thus, sharp pulses of various desired waveforms and durations as short as 0.5 ms are successfully generated. It is believed that this method will find wide applications in time-domain acoustics where point pulses of desired waveform and short extents in space and time are preferred over harmonic waves of impracticably large spatial and temporal extents.

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