ISSN 1392-2114 ULTRAGARSAS, Nr.1(38). 2001.

Synthesis of acoustic impulses in a solid waveguide

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Introduction

Short acoustic signals – impulses are needed in various fields: non-destructing testing, diagnostics, measurement, etc. One possible application of these impulses is calibration of acoustic emission (AE) transducers [1].

Difficulty of measurement of these characteristics becomes obvious at low frequencies (20 … 200 kHz). In order to avoid influence of reflections it is necessary to increase the size of the object where acoustic impulses are generated. To solve this problem we may use acoustic waveguides. At the end of the waveguide a short impulse is generated, while the other end of the waveguide is used for calibration.

Widely known methods of exciting impulses (by laser, sparkle, strike) have one basic disadvantage: because of dispersive nature of a waveguide, the duration of the calibration impulse is extended and therefore is not applicable for measuring the impulse response. Also, the fwave of these signals is out of control.

The aim of the article is to investigate possibilities of a piezoelectrical excitation, and synthesis of a excitation signals by different methods, to generate an impulse at the of waveguide, suitable for calibration, to propose criteria for evaluation of a synthesized impulse and to compare methods of the synthesis.

Principle of synthesis

As it was mentioned above, we shall use piezoelectrical transducer, attached to the end of a waveguide for excitation of an acoustic impulse. The subject of further study is an acoustic system consisting of an excited piezoelectrical transducer and a waveguide. The input signal of the system $X(t)$ is voltage, the output signal $Y(t)$ is the displacement, $h(t)$ is the impulse response of the system.

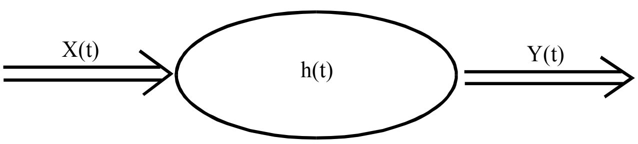

Fig. 1. **Diagram of synthesis**

The main method of synthesis is based on application of deconvolution. The system output signal *Y(t)* is equal to convolution of the system impulse response*h(t)* and the input signal *X(t):*

$$Y(f) = h(t) * X(t) = \int_{-\infty}^{\infty} h(s-t)x(t)dt \,. \qquad (1)$$

In the case of a linear system, this equation is common for the whole system and independent upon characteristics of separate parts of the system.

We solve inverse problem (deconvolution) to synthesize the signal, when impulse characteristic and output signal are known; we should find input the excitation signal.

As the solution of the deconvolution is a rather complicated problem in the time domain, we solve it in frequency domain, where multiplication is used instead of integration [2]:

$$DFT(Y) = DFT(X) * DFT(h) \,, \qquad (2)$$

where DFT is the discrete Fourier transform.

On the basis of Eq. 2, we may describe the excitation signal as:

$$X(t) = IDFT(DFT(Y) / DFT(h)) \,, \qquad (3)$$

where IDFT is the inverse Fourier transformation.

Beside the frequency method, we shall study in this article a modified frequency method and method of synthesizing signal by voltage steps.

Structure for synthesis implementation

Block diagram of the equipment used for synthesis is presented in Fig. 2.

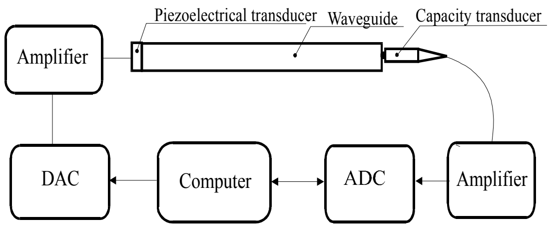

Fig. 2. **Block diagram for signal synthesis**

The signal synthesis system consists of an acoustic waveguide, piezoelectrical transducer, capacity transducer, ADC, DAC, and a computer with a special board for impulse generation. The program allows to change the board sampling frequency from 36 to 0.2815 MHz. The memory may hold 32768 samples. DAC and amplifier enable to excite a piezoelectrical transducer by the synthesized impulse, which excites mechanical vibrations in the waveguide. Propagation of the wave will create displacement, which is picked-up by the capacity transducer. Capacity transducer will detect vibration of acoustic waveguide surface, which is of the order several micrometers. Earlier measurements have shown that such a capacity transducer enables detect surface displacement of tens of nanometers. The amplified signal is fed into ADC, where the signal is sampled and transmitted to a computer.

The computer controls the ADC, the special impulse board and carries out calculations.

One of the main problems of synthesis is to get the desired impulse at the system output. There are several criteria by which we can create a synthesized impulse:

- the waveform of the synthesized impulse;
- spectrum of the synthesized signal;
- complexity of the excition impulse;
- amplitude of the output signal;
- the level of parasitic "ringing" in the tail of the synthesized signal.

In theory, we may model an impulse of any form, but in practice, implementation of such an impulse is limited by a noise, sampling in time and frequency domains.

On the basis of above mentioned criteria, we shall use the impulse characteristic to calculate excition impulse:

$$X(t) = h(t), \qquad \text{when } t < \tau$$

$$X(t) = h(t)e^{-(t-\tau)/\lambda}, \text{when } t > \tau \qquad (4)$$

where: λ is the damping factor, τ is the time instant, when the signal damping is started. Depending on values of τ and λ, the desired impulse can be the first whole or half period of the system impulse response.

Simulation and experiments

In experiments waveguide made from an aluminum rod (length – 1 m, diameter – 0.025m) was used.

Fig. 4 presents a system impulse response and the desired impulse, calculated from Eq. 4.

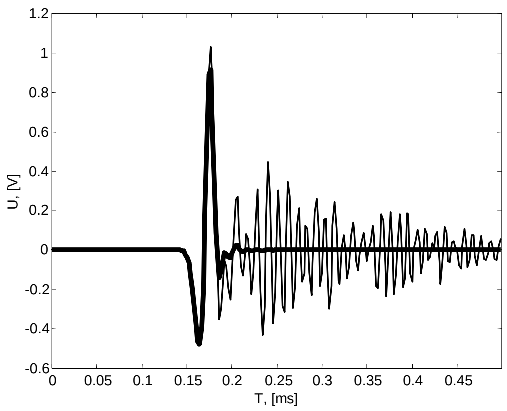

Fig 4. **System impulse response and the desired output signal**

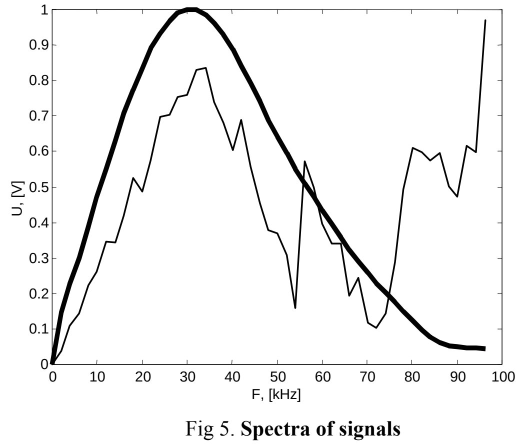

Fig 5. **Spectra of signals**

Thin black line corresponds to the system impulse response, thick solid line to the desired impulse at the system output. Spectra of both signals are presented in Fig. 5.

Using deconvolution in frequency domain, we obtained the excition impulse, which is shown in Fig 6. The fast Fourier transform is a periodic sequence, so excitation should be the sequence of several impulses, not one. Unfortunately, we are not able to do that, because of finite dimensions of the acoustic waveguide. The wave excites new impulses reflected from the walls of the waveguide. For this reason we should use the Hanning window function, which decreases influence of walls and allows synthesizing signal of one impulse to be applicable. A calculated excition impulse is depicted in Fig. 6. The experimental output signal, created by the excited system with a synthesized impulse is presented in Fig. 7.

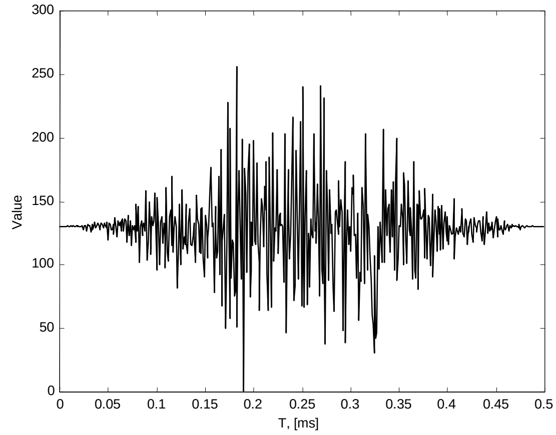

Fig. 6. **Calculated excitation signal**

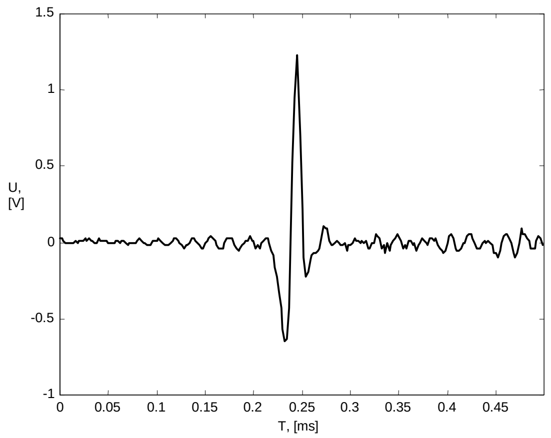

Fig 7. **Measured signal**

The spectrum of the output signal shown in Fig. 8 does not fully correspond to the desired signal spectrum. The main reason is the insufficient number of bits in DAC and influence of high frequency components. These problems will be discussed at the end of the article.

The main advantages of the synthesis method are high speed and universal calculation algorithm, applicability to different systems. However, the method has disadvantages

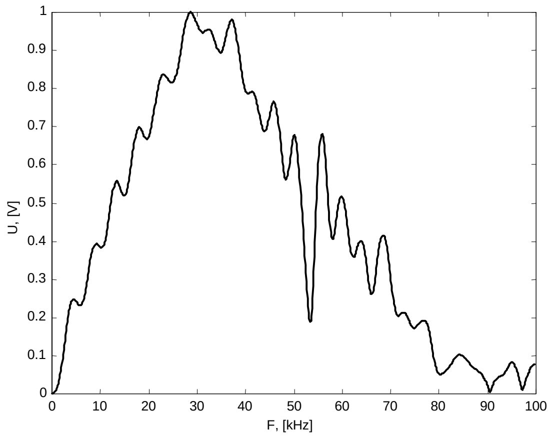

Fig. 8. **Output signal spectrum**

which are evident in Eq. 3. If frequency response of the system approaches to zero, that means that synthesized signal will approach to infinity. To avoid this disadvantage, filtering of those high frequencies was proposed. Signal filtering is performed digitally. As the system is linear, filtering of the system output signal may be evaluated by filtering the input signal given by:

$$Y(e^{jw}) = X(e^{jw}) * (h(e^{jw}) * f(e^{jw})) = (X(e^{jw}) * f(e^{jw})) * h(e^{jw}), \quad (5)$$

where $f(e^{jw})$ is the filter response and w is the angular frequency.

This filtering allows us to avoid synthesizing high frequency components, which are unnecessary for calibration of low frequency AE transducers.

This method was used for a solid aluminum rod. A system impulse response shown in Fig. 4, was filtered by the Butterworth filter of the 10th order and cut-off frequency 45 KHz. The filtered response is shown in Fig.9.

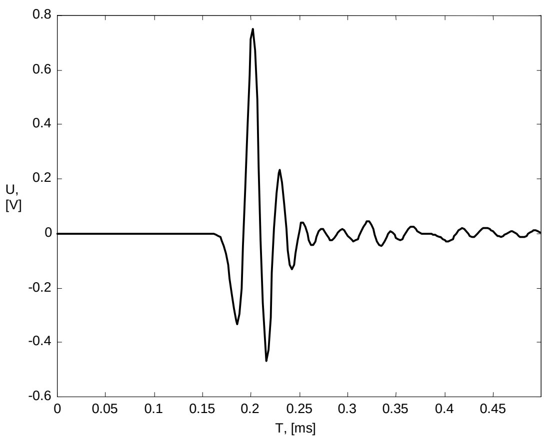

Fig. 9. **Filtered system impulse response**

Because the system impulse characteristic has changed, we should filter the desired impulse so that according to Eq. 3 values would not be too large. The desired filtered impulse is shown in Fig. 10.

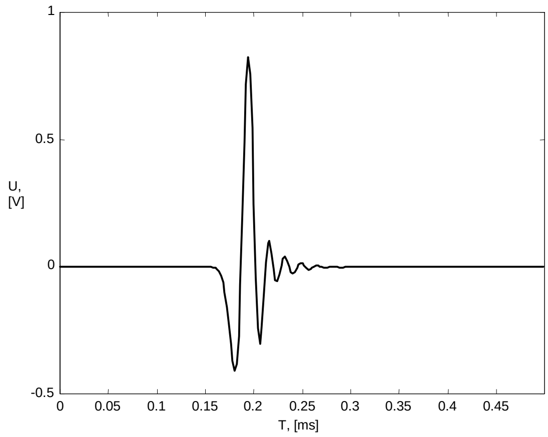

Fig. 10. **Desired output impulse**

The excited impulse is filtered by the same Butterworth filter. The result is illustrated in Fig. 11

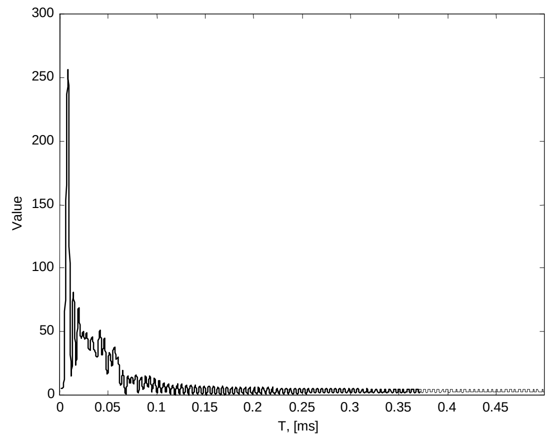

Fig.. 11. **Filtered excited impulse**

This impulse excites a system, which generates an acoustic signal at the output (Fig. 12).

The main advantage of the method is that the software carries out filtering and there is no need to change the structure of the system.

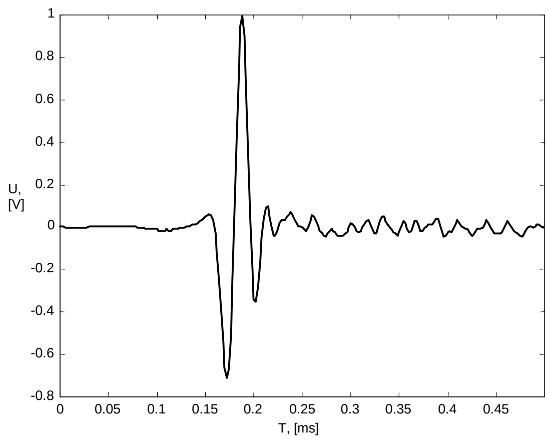

Fig. 12. **Experimental measurement of the output signal**

In the literature [3, 4] there is described a method of synthesizing signal by steps. The method is based on the sum of positive and negative leaps that reject unnecessary components of output signal. This method was tested in the discussed system. Unfortunately, it is very complicated to apply for complex signals, because it is necessary to solve optimization problem:

$$\Delta U = \arg\min_{\tau,K} F(U_{des} - K * U_{step}(t-\tau)), \qquad (6)$$

where ΔU is the minimization result, U_{desr} is the desired signal at the system output, U_{step} is the signal synthesized by leaps, K is the amplitude factor, τ is the delay.

Global minimum may be found after the full sorting. So this method requires large computation resources. In our case, any additional step will increase calculations by 10^5 times, changing amplitude and time position.

Considering the mentioned disadvantages, before application of the method to a solid aluminum rod, we shall introduce several additional conditions:

1. Initial damping of high frequency oscillations;
2. Further damping of impulse oscillations, left by little steps.

Thus minimization problem is being solved (6), but only one step is used for calculation. A minimization criterion is a level of high frequency components in an output signal. For calculations we use the experimentally measured system transient response. The next leap equals a half of main impulse duration and is of the opposite polarity because of a need to minimize distinction of the main impulse. Further steps depend upon theoretical output signal. Small distinctions are eliminated by opposite direction leaps. The excited impulse for the solid aluminum rod created by this method is presented in Fig.13.

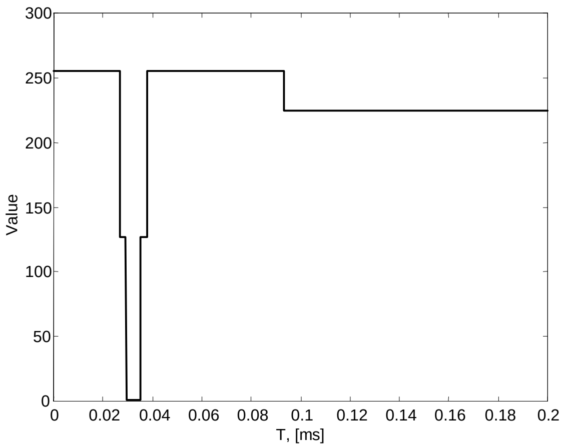

Fig. 13. **Excited impulse**

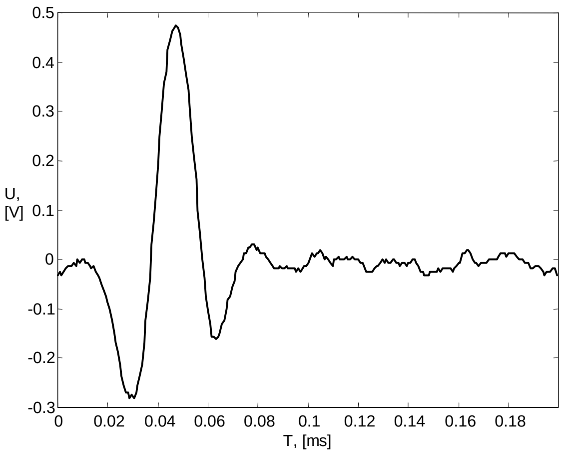

Fig. 14. **Output signal**

The signal at the system output is shown in Fig. 14. Fig. 15 presents a spectrum of the output signal.

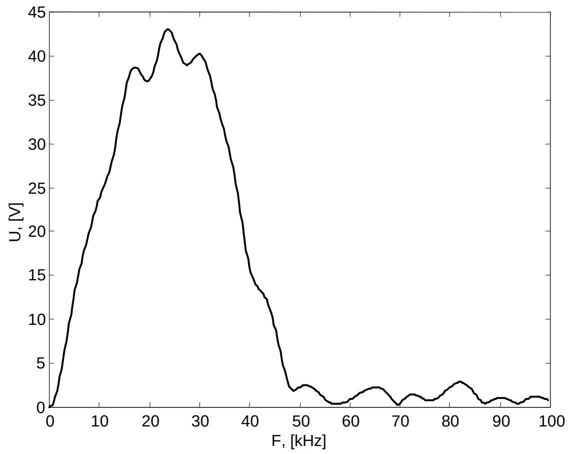

Fig. 15. **Spectrum of the output signal**

Synthesis errors

Synthesis error is the difference between experimentally measured and calculated exitation signals. There are several criteria to evaluate this inadequacy: maximum difference between measured and calculated signals, correlation coefficients, standard deviation. We choose minimum square deviation, because maximal difference does not hold information about the entire signal and correlation coefficients is equal to one if the difference is small.

The following reasons causes deviations of experimentally measured and calculated signals from the theoretically found signal:

1. Noises of measured impulse response of the system and noise of the output signal;
2. Amplitude quantization;
3. Sampling in time and frequency domains.

We shall discuss all these items in detail. Let us assume that the measured system impulse response may be described as $\widetilde{h} = h + e$ [2], where h is real impulse response, e is the additional white noise. The spectrum of the impulse characteristic may be described as:

$$DFT(\widetilde{h}) = DFT(h) + w \qquad (7)$$

where w is the spectrum of a white noise

After the substitution of Eq. 7 to Eq. 3, we obtain:

$$\begin{aligned} DFT(\widetilde{X}) &= DFT(Y)/DFT(h) + DFT(Y)/w \\ &= DFT(X) + DFT(Y)/w \end{aligned} \qquad (8)$$

It is obvious from Eq. 8 that at these points where system impulse-frequency response equal to a noise, values of synthesized signal are distorted.

The calculated signal error, caused by a noise influence is shown in Fig. 16.

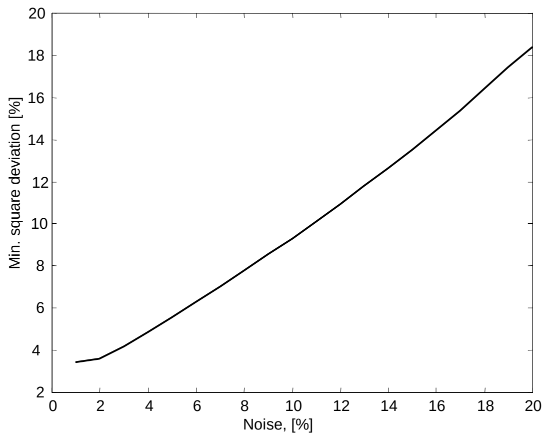

Fig. 16. **Influence of a noise**

Errors of synthesized signal are caused by finite frequency resolution using the Fast Fourier transform. The frequency resolution depends upon the number of samples. To increase the frequency resolution we should add zeros at the end of the signal. Fig 17 presents theoretically modeled deviation between the synthesized and the desired signals.

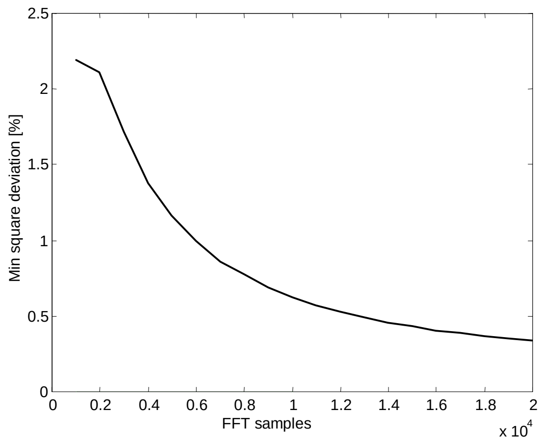

Fig. 17. **Minimum square deviation versus number of the FFT samples**

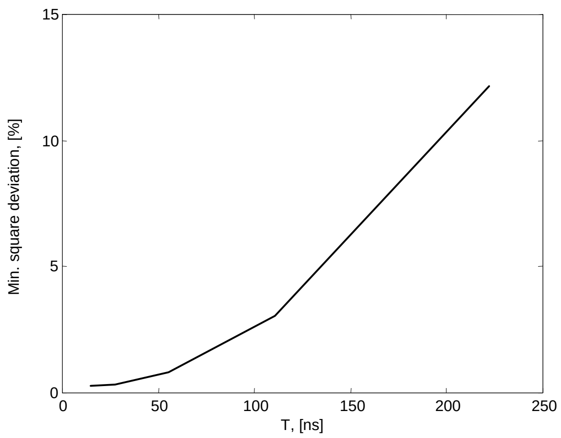

Fig. 18. **Minimum square deviation versus time sampling interval**

In this case, reading moments in time axis of the special impulse generator differ from discrete time readings of the calculated excitation signal. We should resample the signal, but this will cause additional errors. Deviation depends on complexity of the excition signal. The calculated error for a solid aluminum rod is illustrated in Fig. 18.

When the excitation signal is transmitted to the DAC, its amplitude is digitized. Fig 19 shows dependence of error upon the number of bits used for digitizing of the excitation signal amplitude. Errors caused by amplitude digitizing are proportional to the difference between desired and calculated signals: the larger is the difference, the more bits are needed to synthesize the desired signal.

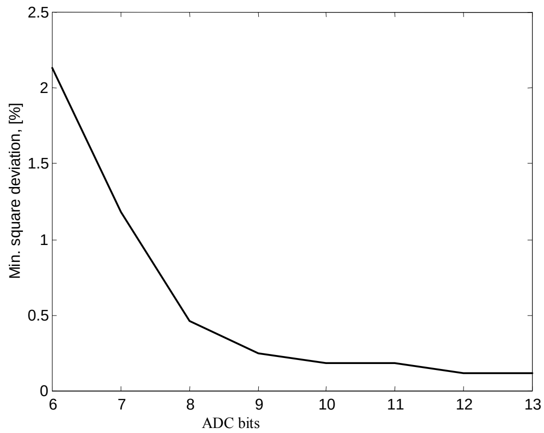

Fig. 19. **Minimum square deviation versus the number of the ADC bits**

Conclusions

Signal synthesis is a complex and individual problem. There is no universal practical solution of the problem, but several recommendations for the signal synthesis may be presented:

1. The desired signal should be “reasonable”. We cannot synthesize a signal in frequency range where values of a system frequency response are very small, as we will have to damp present resonance frequencies of the system and to excite system in frequencies different from its own resonance frequencies. This will require high amplitude resolution. Besides, the influence of noise is stronger in high frequencies, where spectrum amplitude is very small.
2. Exciting by steps are useful if there are two or more different resonance frequencies of the system. In this case, properly selected leaps will help to damp one resonance without influencing another.
3. If the frequency range is precisely known, it is recommended to apply filtering. By this way we shall avoid the influence of noise in a high frequency range.

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Akustinių impulsų kietojo kūno bangolaidyje sintezė

Reziume

Žemojo dažnio akustinės emisijos keitikliams kalibruoti reikalingas žinomas paviršiaus poslinkis su atitinkamomis laikinėmis ir dažninėmis charakteristikomis. Straipsnyje nagrinėtos sintezuoto pjezoelektrinio žadinimo akustiniame bangolaidyje galimybės. Eksperimentiškai palyginti trys skirtingi sintezavimo vientisame aliuminio strype metodai: du iš jų paremti dekonvoliucijos principu, trečias - “šuoliukais”. Apskaičiuotos paklaidos, atsirandančios dėl diskretizacijos, kvantavimo ir triukšmų. Nustatyta, kad lemiamą įtaką sintezavimo kokybei turi pageidaujamo impulso parinkimas. Parodytas žadinimo signalo skaitmeninio filtravimo efektyvumas, siekiant pageidaujamo paviršiaus virpėjimo žinomame dažnių ruože. Kad sumažėtų paklaidos, laiko mastelio perskaičiavimą siūloma sutapdinti su dažnių mastelio perskaičiavimu.

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