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INFLUENCE OF THE TRANSDUCER BANDWIDTH ON COMPRESSED ULTRASONIC ECHOES

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The aim of this work is to explore and analyze the influence of the transducer bandwidth on the compressed echoes resulted from the Golay complementary codes transmission. For that reason, a computer simulation and experimental verification were performed that reflected the influence of the transducer bandwidth on the distortion of a signal. This study helps to elucidate why the echoes ringing is present for narrow bandwidth transducers. As known, the shape and symmetry of the pulse waveform and its interaction with the transducer bandwidth and its tuning circuitry have a profound effect on the pulse echo performance achievable from a medical scanning probe.

The computer simulation was performed using the Matlab® software for different fractional transducer bandwidths – from wideband transducers of 100% (ideal case), 90%, 75% to narrowband ones of 50% and 25%. The 16-bits Golay complementary sequences at nominal frequency of 1 MHz were used to illustrate the transducer bandwidth influence on the resulted signal. It was shown that the decreasing of the transducer bandwidth results in a considerable drop of the amplitude of the compressed echoes from 20.1 V for the 90% fractional bandwidth down to: 17.1 V, 12.5 V and 6.6 V for 75%, 50% and 25% bandwidths, respectively. The widths of the compressed echoes were widening at the same time from 708 ns up to 2.38 μs reducing the axial resolution from about 1 mm to over 3.6 mm. In the experiments, two transducers with different fractional bandwidths of 70% and 35% and nominal frequencies of 4.8 MHz and 6 MHz, respectively, were used.

Keywords: transducer, signal analysis, distortion, filtering, Golay sequences.

1. Introduction

The distortion of electrical signals is one of the fundamental problems resulting in the final echo detection in ultrasonography. Even if the immediate problem is not an electrical one, the basic parameters of interest are often changed into electrical signals by means of different transducers [1, 5].

In our recent works [2, 3, 9, 10], we have discussed and explored the coded sequences with frequency modulation as well as the phase modulation for echo enhancement and noise cancellation in medical ultrasound systems. These coded signals evoke more and more interest in medical ultrasound.

The effect of the transducer bandwidth on the characteristics of ophthalmic ultrasound images was also explored by SILVERMAN *et al.* [6]. For that reason the two ultrasonic transducers, one with a narrow bandwidth of 35% and another one with a broad bandwidth of 77%, at a nominal centre frequency of 10 MHz, were evaluated. The comparative results were shown in a form of scans of a tissue-mimicking phantom that simulated the different organs scanned in ophthalmology.

This study helps to elucidate the influence of the transducer bandwidth on the transmitted/received signal and to discuss the experimental results showing that a short pulse does not always provide a better axial resolution than a longer cycle used by the same ultrasonic transducer. Main parameters of linear filter, i.e. the time, frequency and transient responses, are addressed. Analog and digital signals filtering are discussed, too. The spectrum analysis can be used to measure the pulse characteristics of ultrasonic transducers by expressing the amplitude of the acoustic pulse as a function of frequency. Several examples of signal distortion were obtained using computer simulation. The strengths and limitations of a narrowband single transducer in comparison with a broadband one for medical ultrasound applications are discussed.

2. Linear filter. Analysis

Many systems are reasonably designed to be linear and to meet the design specifications. This has a fortuitous side benefit when attempting to analyze filters, networks, etc. A real signal can be considered to be a sum of sine waves. Also, the response of a linear filter is the sum of responses to each component of the input. Therefore, if the response of the filter to each of the sine wave components of the input spectrum is known, the output can be predicted.

Any given ultrasonographic signal can be considered in two different domains: the time and the frequency ones. That one mostly used, is the time domain. In most cases, the Fourier transform is used to transform a signal from the time domain into the frequency one and *vice-versa*.

The frequency response of the system is defined as the ratio of the phasor output to the phasor input where the output and input may be either the voltage or the current. Most common the ratio of the phasor output voltage to the phasor input voltage.

$$H(\omega) = \frac{V_{\text{out}}(\omega)}{V_{\text{in}}(\omega)}. \tag{1}$$

$H(\omega)$ is often referred to as the voltage transfer function.

Considering the frequency response, the filters fall generally into one of the three categories: low-pass filters, high-pass filters and band-pass filters or a combination of these. As the names suggest, their frequency responses have a relatively high gain in the band of frequencies allowing these frequencies to pass through the filter. Other frequencies suffer a relatively high loss and are rejected by the filter. An ideal filter is not realizable physically, but in practice, it is possible to design a physical filter that approximates closely an ideal filter if needed.

For the sake of this work, we were simulating filters considering their amplitude response rather than the impulse and step responses. To preserve shape, when a signal splits into its Fourier components, all the components must pass through the system with the same gain factor and the same delay. Distortion due to frequency-dependent gain is the *amplitude distortion*, this one due to a frequency-dependent delay is the *phase distortion*.

Figure 1 shows qualitatively the transient response of a band-pass filter. If the resonance is narrow compared to its frequency, then it is said to be a high "Q" resonance, where the quality factor Q of the filter is defined as:

$$Q = \frac{\text{Center Frequency of Resonance}}{\text{Bandwidth of } -3\text{dB Points}}. \tag{2}$$

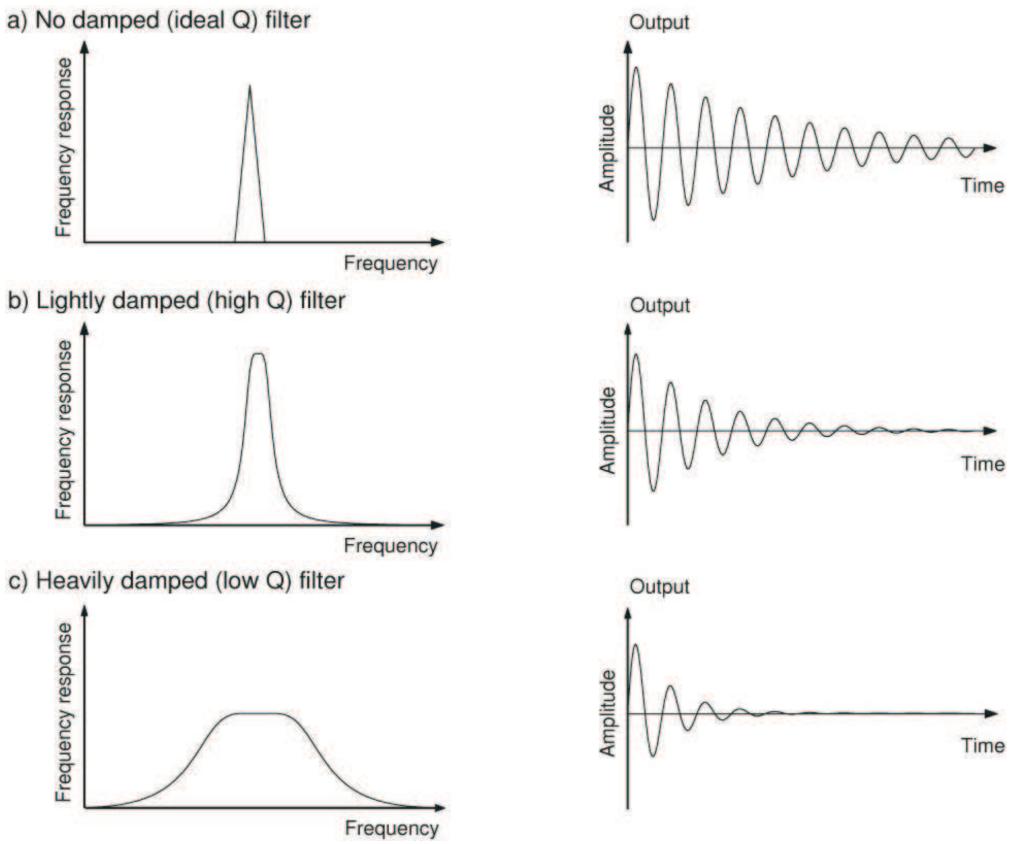

Fig. 1. Transient response of band-pass filters. Similar characteristics are obtained by ultrasonic transducers.

Figure 1a shows a filter frequency response when the filter quality Q is ideal. In this case, an impulse response is not damped. Figure 1b shows a high Q filter frequency response. It has an impulse response which dies out very slowly. A time response which decays slowly is said to be *lightly* damped. Figure 1c shows a low Q resonance. It has an impulse response which dies out quickly. In this case, a time response is *heavily* damped. This illustrates a general principle: *signals which are broad in one domain are narrow in the other one*.

3. Signal distortion during transmission

The signal distortion, which often happens in practical problems of ultrasonography, is discussed here. In a filter or device, any departure of the output signal waveform from that which should result from the input signal waveform is called *signal distortion*. Signal distortion may result from many factors. Mainly it depends on the transmitting-receiving path that consists of devices which include nonlinearities in the transfer function of the active device, such as a transistor or operational amplifier. A distortion may also be caused by a passive component such as a coaxial cable or by inhomogeneities, reflections, frequency dependent absorption etc. in the propagation path. As a result, different artefacts in the obtained ultrasound image can occur.

There are two main kinds of filters: analog and digital filters. They are quite different in their physical makeup and in how they work. Each of them has self-advantages and disadvantages.

Analog signal filtering allows:

- trend removal – high-pass filtering;
- selection of useful frequency bands – low-pass, band-pass and high-pass filtering;
- signal-to-noise ratio improvement;
- anti-aliasing filter;
- frequency analysis.

Signal filtering described here is, in a way, a continuation of the aforementioned signal processing in [9].

An analog filter uses analog electronic circuits made up of components such as resistors, capacitors and op amps to produce the required filtering effect. Such filter circuits are widely used in applications such as noise reduction, video signal enhancement, graphic equalizers in hi-fi systems and many other areas.

There are well-established standard techniques for designing an analog filter circuit for a given requirement. At all stages, the signal being filtered is an electrical voltage or current which is a direct analogue of the physical quantity (e.g. transducer output) involved. The analog input signal must first be sampled and digitized using an ADC (analog-to-digital converter). The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output through a DAC (digital-to-analog converter) to convert the signal back to the analog form. Figure 2 shows the basic setup of such a system.

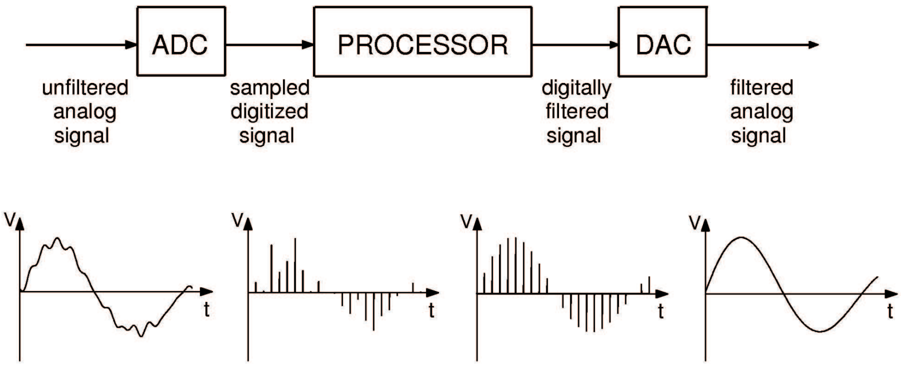

Fig. 2. Basic setup of analog signal filtering.

Digital filters that incorporate digital-signal-processing techniques have received a great deal of attention in technical literature in recent years. Digital filters offer features that have no counterparts in other filter technologies. These things can been done by utilizing digital filters (unlimited flexibility) that are not possible in the analog world.

The main advantages of digital filters in comparison with the analog ones are:

- programmable, i.e. they can be easily changed without affecting the circuitry (hardware);
- easily designed, tested and implemented on a general-purpose computer or workstation;
- extremely stable with respect to both time and temperature;
- much more versatile in their ability to process signals in a variety of ways; this includes the ability of some types of digital filters to adapt to changes in the characteristics of the signal;
- allows the extraction of useful frequency bands, noise reduction, frequency analysis.

4. Transducer band-pass filtering. Computer simulation

As already mentioned above, signal filtering has a potential effect on signal distortion. Here the influence of the filter bandwidth on the amplitude of the compressed signal using computer simulation is considered. The influence of the filter, amplifier or ultrasonic transducer frequency bandwidth on the transmitted/received signal can be examined by applying the band-path filtering. The computer calculations were performed using the Matlab® software. The algorithm looks as follows: at first two Golay complementary sequences are numerically synthesized. Next their Fourier transforms are evaluated by means of the FFT algorithm (**fft** routine in Matlab®). The band-pass filtering is performed by the Chebyshev filter of second order with different bandwidth (**cheby1** routine in Matlab®), followed next by the inverse Fourier transform evaluation by means of the FFT algorithm (**ifft** routine in Matlab®). The correlation of the filtered signal with the transmitted one is performed using a correlation function (**xcorr** routine in Matlab®). The results obtained are plotted using the **plot** function in Matlab®. Experimental results, which take into account the influence of both the bandwidth of the transmitting-receiving path and the ultrasonic transducer are considered in the second part of this section.

Figure 3 demonstrates the sequence of the filtering procedure for a Golay complementary pair of 16 bits length at a centre frequency of 1 MHz. This procedure is shown for the ideal case, in other words when the full spectrum is passed completely through the filter (transducer).

At first two Golay sequences of length of 16 bits are shown (Fig. 3a). The time duration of such sequences is equal to 16 μs for the frequency of 1 MHz. Then, the power spectrum of each sequence obtained using the FFT algorithm is shown in Fig. 3b. The power spectra of such sequences are not as narrow as for the sine burst of compatible duration since any changing of phase leads to a spectrum widening. Since Fig. 3 represents an ideal case, the full spectrum is passed through. Next, the power spectrum is transformed into the time domain using the inverse Fourier transform (Fig. 3c). In the case presented, the filtered sequences are identical with the transmitted ones, without any distortions. The filtered signals obtained are correlated with original ones, respectively (Fig. 3d). Note: the time duration of the resulted signal equals $2T$, where T is the time duration of the sequences. In this case it is equal to 32 μs. The amplitudes of these signals are equal to N, where N is the sequences length; in our case it is equal to 16. It should be noted that each of the correlated output signals has a considerable amount of side-lobes. Adding two resulting correlated signals, which have side-lobes equal in amplitude but opposite in sign, results in a final output, the side-lobes' amplitude of which is equal to zero (Fig. 4). It is a specific property of the Golay codes, particularly suitable for the unambiguous range detects ability.

The procedure described above reflects the real case when the filtering is realized twice; the first time when the signal is transmitted and the second time when the signal is received.

The same procedure was executed for different fractional bandwidths, namely for 90%, 75%, 50%, and 25%. All these computer simulations were calculated at the centre frequency of 1 MHz.

Figure 5 illustrates the computer simulation of the RF signals after a double passing through the transducer during transmission and reception. This computer simulation is similar to the situation that occurs in a real case except the potential influence of the electronic and acoustical noise. However, in our case the noise influence is not so important since we have concentrated here on the signal distortion, i.e. on the echo's shape and its amplitude after filtering that takes place when the spectrum of the coded sequences is wider than bandwidth of the transducer.

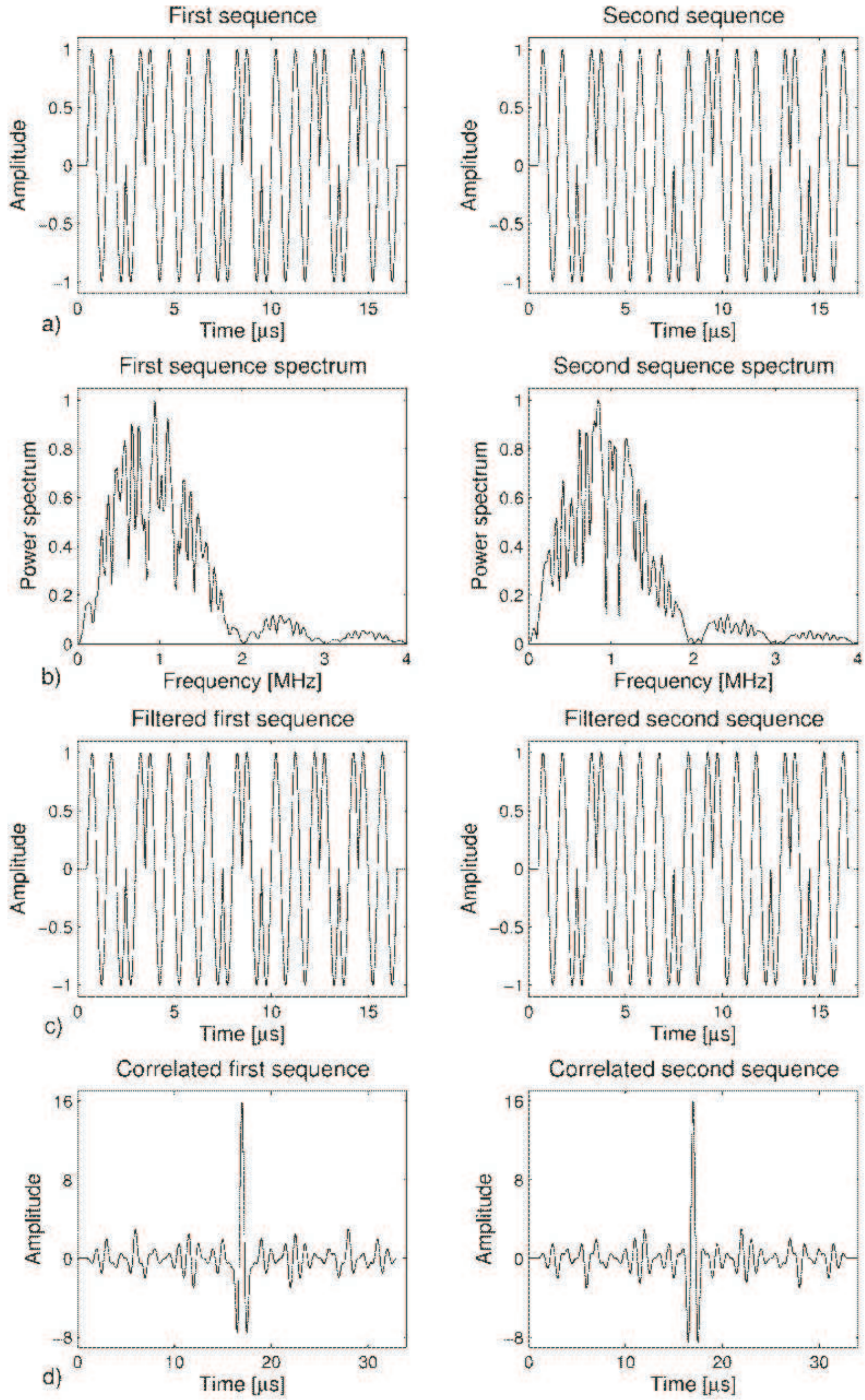

Fig. 3. Procedure of filtering Golay sequences of length of 16 bits at a centre frequency 1 MHz – 100% bandwidth.

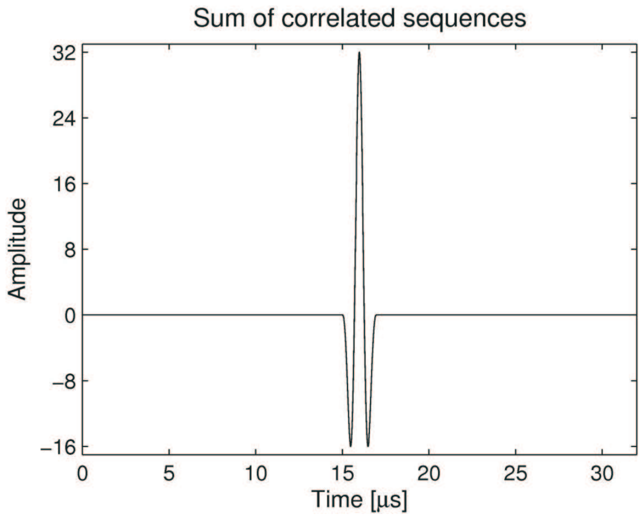

Fig. 4. Sum of the two correlated Golay sequences.

Figure 5 shows clearly the influence of the transducer bandwidth on the echoes' amplitudes and their distortions. For the very narrow transducer bandwidth the signal distortion rearranges the shape of the signal that leads to a considerable widening of the compressed signal. Also the amplitude of the signal decreases proportionally to the narrowing of the transducer bandwidth.

The compressed signals for the fractional bandwidth of 90%, 75%, 50%, and 25% are shown in Fig. 6. For an objective comparison, the amplitude of the every compressed signal is given.

From the results described above two things should be noted: the shapes of the signals obtained and their amplitudes. Similar shapes and, consequently, comparable resolutions are obtained transmitting the coded signals through transducers with fractional bandwidths 90% and 75%. In the case of transducers with the narrower fractional bandwidths, i.e. 50% and 25%, the signal width is wider and in a worse case can lead to axial resolution ambiguity in the ultrasonic image. It should be also noted that the amplitude depends on the fractional transducer bandwidth and is lower when the transducer bandwidth is narrower and therefore the resulting penetration in the tissue decreases.

Comparing the amplitude of the compressed echoes for different bandwidths shows that for a 100% bandwidth, the relative amplitude is equal to 32, which is in agreement with the theoretical gain for Golay codes of 16 bits length. By decreasing the transducer bandwidth, the relative amplitude of the compressed echoes drops; for the 90% fractional bandwidth it is equal to 20.1 volts, i.e. it is 37.2% less than for the full bandwidth, or the 75% fractional bandwidth, the amplitude drops by 45% to 17.6 volts, for the 50% bandwidth the amplitude drops correspondingly by 61% to 12.5 volts, and for the worst considered case, where fractional bandwidth of the transducer is 25%, the amplitude of the compressed echo is 6.6 volts that corresponds to a 79% amplitude drop.

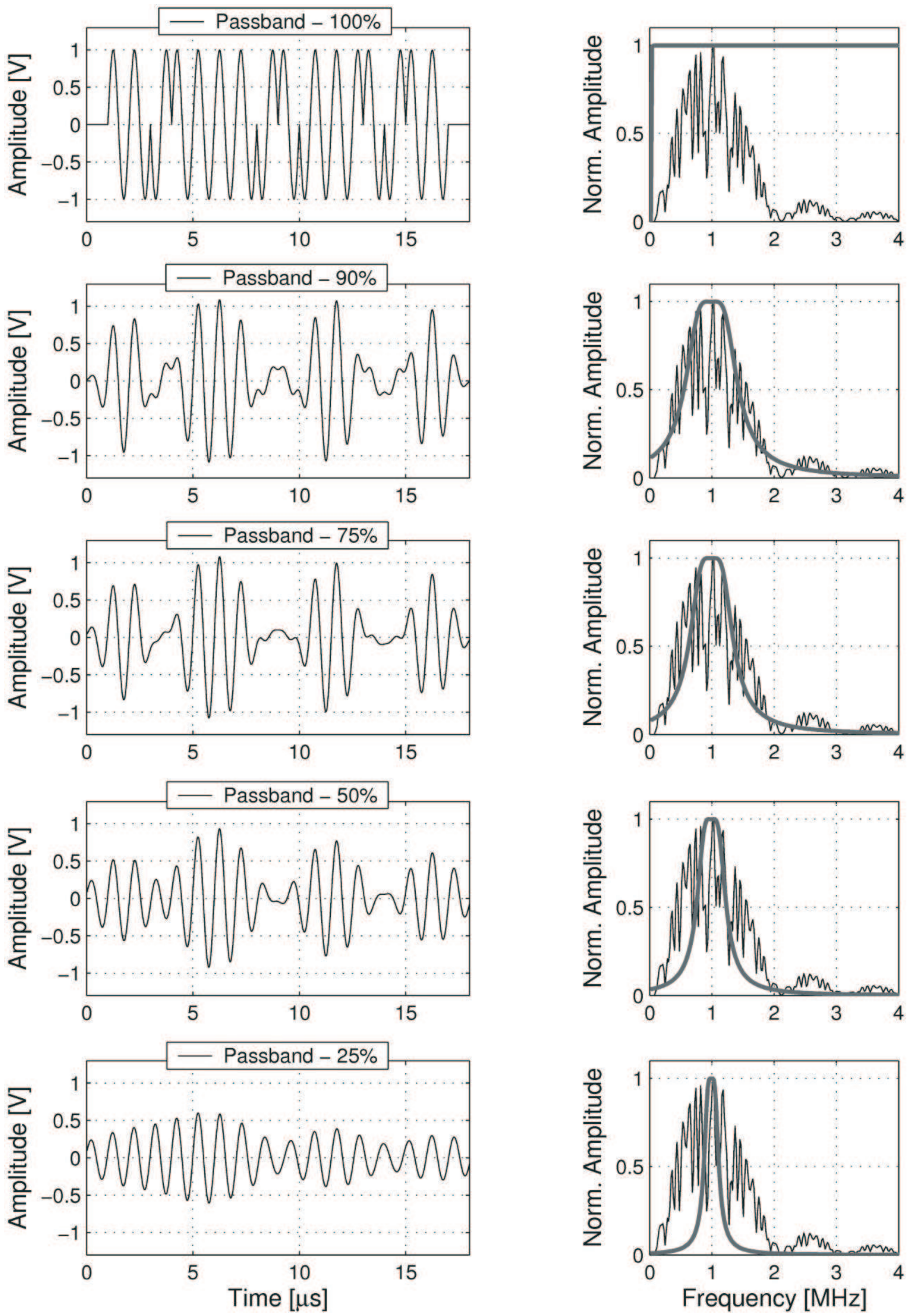

Fig. 5. RF signals passed through a transducer with different fractional bandwidths (left) and bandwidth of the Golay code and transducer (bold grey line) (right).

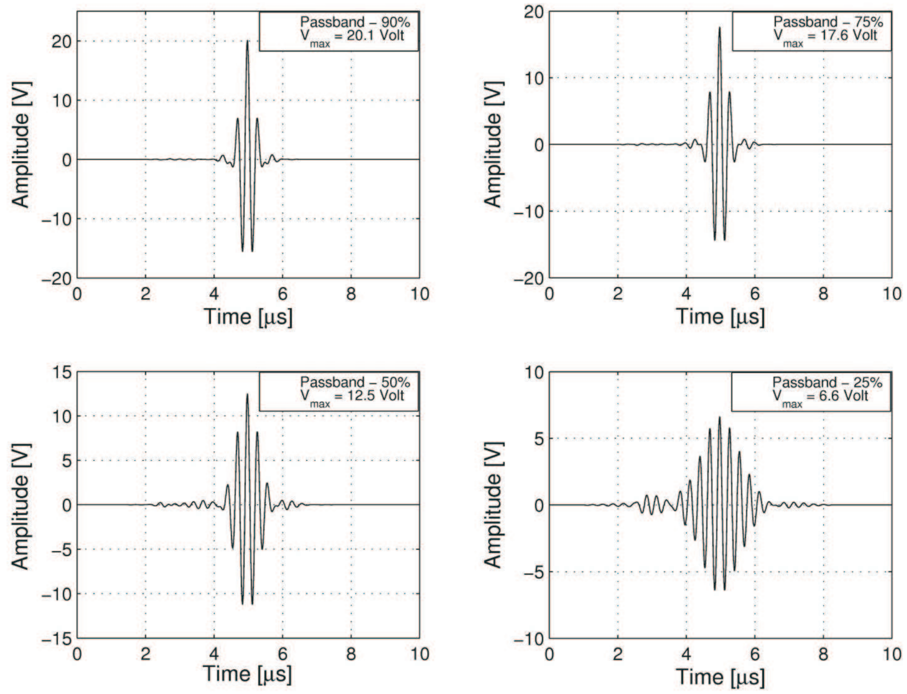

Fig. 6. Compressed signals for different fractional transducer bandwidths.

The width of the compressed signals, as mentioned before, is widening as the fractional bandwidth is narrowing. For the full bandwidth, the pulse width on the –20 dB level for a signal at the nominal frequency of 1 MHz is equal to 147 ns that corresponds to the resolution 0.2 mm. For the 90% fractional bandwidth, the pulse width is widening and is equal to 708 ns (1.090 mm). Respectively, for the examined cases of 75%, 50%, and 25% fractional transducer bandwidths, the pulse widths are equal to 713 ns (1.098 mm), 1.25 μs (1.9 mm), and 2.38 μs (3.67 mm), respectively. Obviously, the resolution is proportional to the nominal frequency of the burst signal and is improving with increasing frequency.

Analysing the results obtained, the difference in the compressed signals with fractional bandwidths of 90% and 75% is not noticeable and transducers with similar fractional bandwidths are often used in practice. However, this difference is evident when the fractional bandwidth is narrowing to 50% or 25%. In this case, the resolution drops even by 18 times in relation to the ideal case and transducers with such fractional are not acceptable in ultrasonography.

5. Experimental verification

The validity of the computer simulation was verified experimentally. To this end, the transducers with different fractional bandwidths of 70% and 35% and nominal frequencies of 4.8 MHz and 6 MHz, respectively, were used. The plexiglass plate of a thickness of 1.3 mm located in a water tank oriented normally to the ultrasonic beam was used as a reflector.

Figure 7 shows the computer simulation and compressed echo signal when the transducer with fractional bandwidth of 70% at a nominal centre frequency of 4.8 MHz was used.

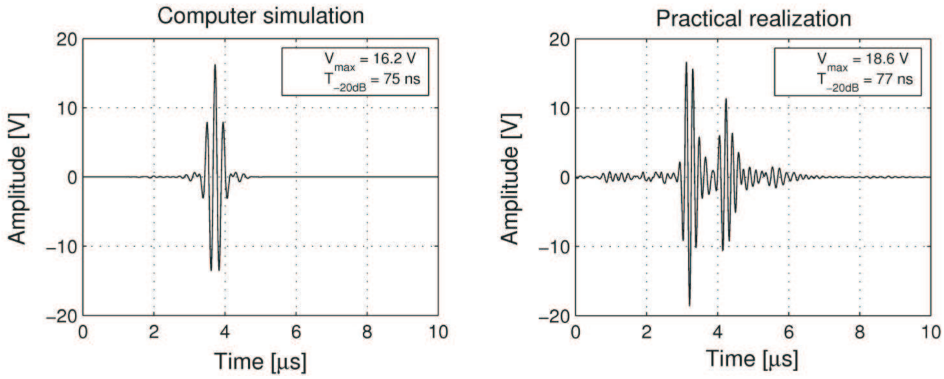

Fig. 7. Computer simulation of the compressed echo from the perfect reflector (left) and compressed echo signal reflected from the plexiglass plate of thickness of 1.3 mm (right) using a transducer at nominal frequency of 4.8 MHz and fractional bandwidth of 70%. The experimental data present two adjacent echoes from the anterior and posterior surfaces of the plexiglas plate.

Figure 8 shows the computer simulation and compressed echo signal when the transducer with fractional bandwidth 35% at nominal centre frequency 6 MHz was used.

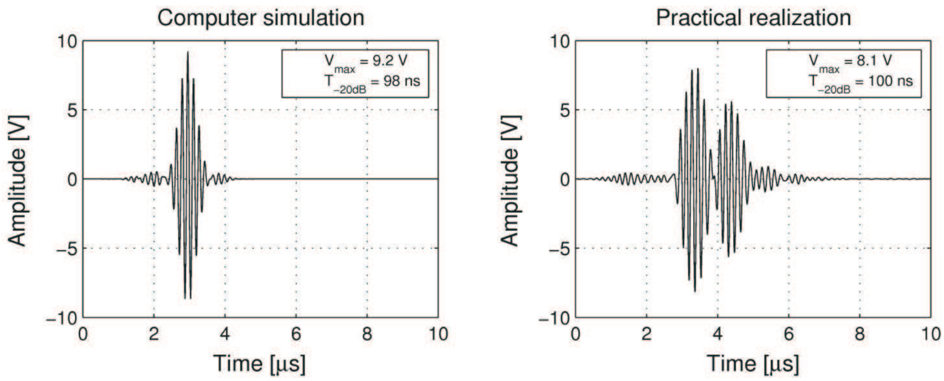

Fig. 8. Computer simulation of the compressed echo from the perfect reflector (left) and compressed echo signal reflected from the plexiglas plate of thickness of 1.3 mm (right) using a transducer at nominal frequency of 6 MHz and fractional bandwidth of 35%.

The difference in the amplitudes between the compressed signals obtained by computer simulation and those obtained from the experiments shown in Fig. 7 and Fig. 8 is caused by a different sensitivity of the ultrasonic transducers and amplifying coefficients of the transmitting-receiving path. All these factors lead to a certain inaccuracy of the obtained results.

The pulse widths of the compressed echoes at the –20 dB level for both the computer simulations and experiments are very close. It seems that the pulse width, and thereby the resolution, should be better in the case of transducer with the nominal frequency of 6 MHz than for the transducer of 4.8 MHz. However, the fractional bandwidth of the transducer has a decisive influence on the resolution of the ultrasonic system. For the transducer of 4.8 MHz and 70% fractional bandwidth the resolution was equal to 75 ns (0.11 mm) (see. Fig. 7), i.e. it was 1.3 times shorter than for the 6 MHz, 35% fractional bandwidth transducer, for which the resolution was equal to 98 ns (0.15 mm) (Fig. 8).

6. Discussion and conclusion

This work is addressed to the problem of transmitting coded signals through a transducer and its influence on the filtered/compressed echoes. To solve this problem the computer simulation was performed using the Matlab® software for different fractional transducer bandwidths. The 16-bits Golay complementary sequences at nominal frequency of 1 MHz were used as the burst signal.

The analysis performed with the help of computer simulation entirely represents the behaviour of the burst coded signals in a real case when the noise influence is not considerable.

The results of this work illustrate how the width of the compressed signal and the resulting axial resolution depend on the ultrasonic transducer with a fractional bandwidth narrower than the bandwidth of the burst signal. This dependence is proportional to the ultrasonic transducer bandwidth. Also, it has been noted that the amplitude of the filtered signal depends on the bandwidth of the transducer. It is very important in case of phase modulated coded signals, in our case 16-bits Golay sequences, since the one-cycle bit length has a wider fractional bandwidth and the energy of this signal is often attenuated by the ultrasonic transducer. As the result, the broad signal can lead to a wrong visualization of the examined organs in ultrasonography. Also, the decreasing signal amplitude limits in turn the deep penetration in the abdominal organs/tissue. This suggests that a transducer with a narrow fractional bandwidth limits the penetration in the abdominal organs/tissue that is very important in ultrasound diagnostic.

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