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Ultrasonic data acquisition: sampling frequency versus bandwidth

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Introduction

Digital ultrasonic systems offer variety of advantages over conventional systems. Signal conversion into digital form allows for easy and flexible signal processing organization. Choice of a sampling frequency usually is the first question arising during design of such systems. When a bandwidth of the system is already known, the sampling frequency can be easily determined using the Nyquist sampling criteria. Usually it is desired to have a lower sampling rate than necessary. The sampling rate reduction allows for a computation time increase and a computer memory requirement decrement. If at the beginning of digital systems implementation the first obstacle met was a capacity of computer and A/D converter buffer memory, with introduction of real-time systems the computation time became the dominating factor. Because of the factors mentioned there were several attempts to reduce the amount of acquisited data. Simplest way, is removal of the modulation using the envelope detection. However, conventional envelope detectors are not capable to restore the envelope with a sufficient accuracy. The Hilbert transform can be applied for an accurate envelope detection, but the sampling rate can not be reduced here before the A/D conversion [1]. Another usually used approach is the acquisition just of the desired time intervals. With an introduction of the digital synchronization, this approach can be realized quite easily. The powerful tool in digital signal processing is offered by [2], where it is suggested the random sampling frequency application in order to reduce the amount of data. Unfortunately, conventional processing procedures must be modified for such sampled signals. If the signal frequency band can be limited without the significant loss of an information, the easiest way to reduce the sampling rate is to apply the anti-aliasing filtering and to sample at the corresponding lower frequency [3]. It must be noted that in such a case a sufficient filtering must be applied in order to avoid the aliasing or circular convolution results have to be taken into account.

A/D conversion

Using the A/D converter the signals are sampled in the time domain and the amplitude is quantized. The quantization influence was discussed earlier in [4], therefore we will focus on a sampling. Analog signal sampling can be presented as multiplication with *shah* function III [5] which actually is a train of impulses

$$\mathrm{III}\left(\tfrac{t}{T}\right) = |T| \sum_{k=-\infty}^{\infty} \delta(t - kT),$$

where δ is the Dirac function and T is the sampling period. The Fourier transform of this function is also *shah* function

$$F\left[III\left(\tfrac{t}{T}\right)\right] = |T| \cdot III(T\omega).$$

Bearing in mind that multiplication in the time domain corresponds to a convolution in the frequency domain, one can see, that the sampled signal spectra will be periodical with a period of the sampling frequency harmonics. Furthermore, because of a discrete presentation of the spectra in a computer, the signal investigated using the Fourier transform the discrete (DFT) is also interpreted as periodical. The periodical character can turn out when performing a signal processing. For instance, multiplication in the frequency domain with a filter function corresponds to the convolution in the time domain, so an unwanted influence of the signal tail will occur in the signal head, which might cause confusing results. And opposite - the first glance a non-harmful multiplication with some sort of a compensation function in the time domain might cause the frequencies mix in the frequency domain, because of the circular convolution. For further processing, in order to avoid the circular convolution influence, in the time or the frequency domain, an artificial zero- padding can be added.

Choice of the sampling frequency is the question always arising during a signal acquisition. One can always direct use the Nyquist criteria [6] or the Kotelnikov sampling theorem [7]. These are using the maximal frequency still present in the signal spectra. Usually, the transducer central frequency is best known, because this signal easy detectable or can be calculated even from the time domain signal presentation. Because of presence of a noise and a slow decay of the signal frequency response, of the maximal frequency is complicated. Therefore, various maximal frequency determination criteria were developed. The easiest way to determine the maximal frequency is to use the term of active spectra, which is described as the frequency band containing the frequencies which are essential and sufficient to describe the signal with a required accuracy [7]. Decision criteria are based on:

- sufficient signal energy inclusion in the active spectra;
- the particular spectral density decay level;
- the first spectral zero position.

Most popular is the signal energy criterion, when the signal energy left beyond the maximal frequency is less than 5%, i.e. more than 95% of a full signal energy is used. The second criterion, the particular spectral density decay level, is criticized because the choice of a decay level for the maximal frequency has no evident relation to the signal

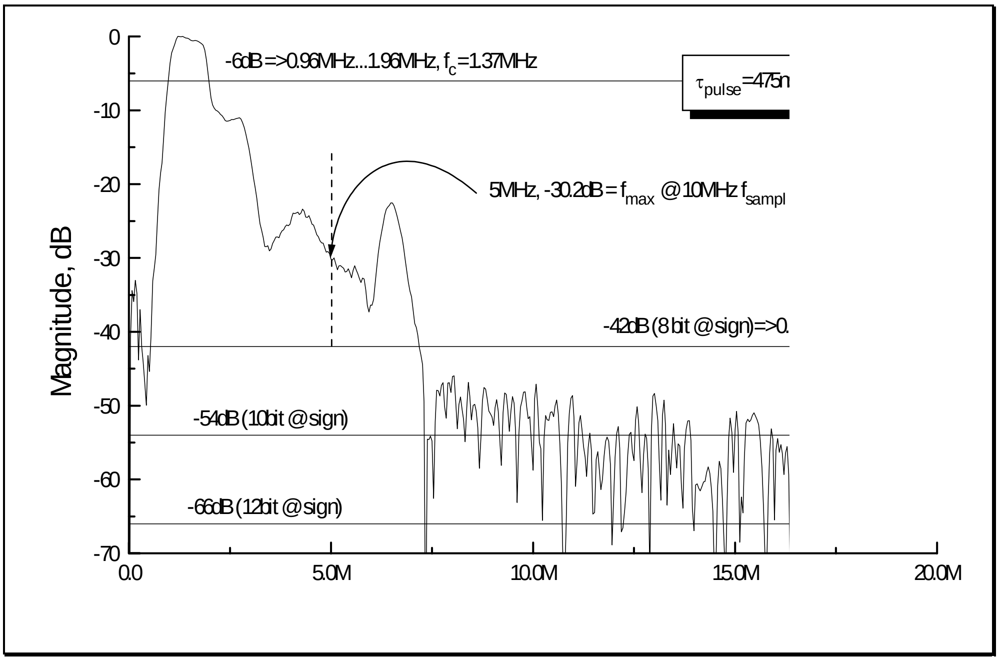

Fig. 1. **Amplitude spectra of 2 MHz transducer reflection from a steel reflector (@τ_{pulse}=500ns-5%).**

parameters. Therefore usually the level used (for instance -20dB) is just a matter of an agreement. The first spectral zero criterion is attractive because of evident relation to signal parameters, but is not easy applicable for each signal and there is still an obscure part of the signal left behind this limit, which depends on the signal shape. We have started from such concepts:

1. Since the signal is quantized and sampled, errors introduced by those two process should have same order.

2. Despite a wide frequency range, virtually every ultrasonic transducer signal is decreasing as frequency increase.

3. Random noise influence can be eliminated using data averaging, despite it will be aliased into active signal bandwidth.

4. Once the reflection containing theoretically widest spectra has been established and the sampling frequency choice has been made using this signal, the sampling criterion will be satisfied in all subsequent measurements except the noise.

5. The noise amplitude can be kept at the level satisfying the point 1 requirements by applying appropriate filtering or averaging.

The experiments were carried out with the 2 MHz wideband transducer in pulse-echo mode, which was receiving pulses in water reflected by the polished steel surface. The pulse duration was chosen in such a way, that the first spectral zero was at 2 MHz. In this case transducer

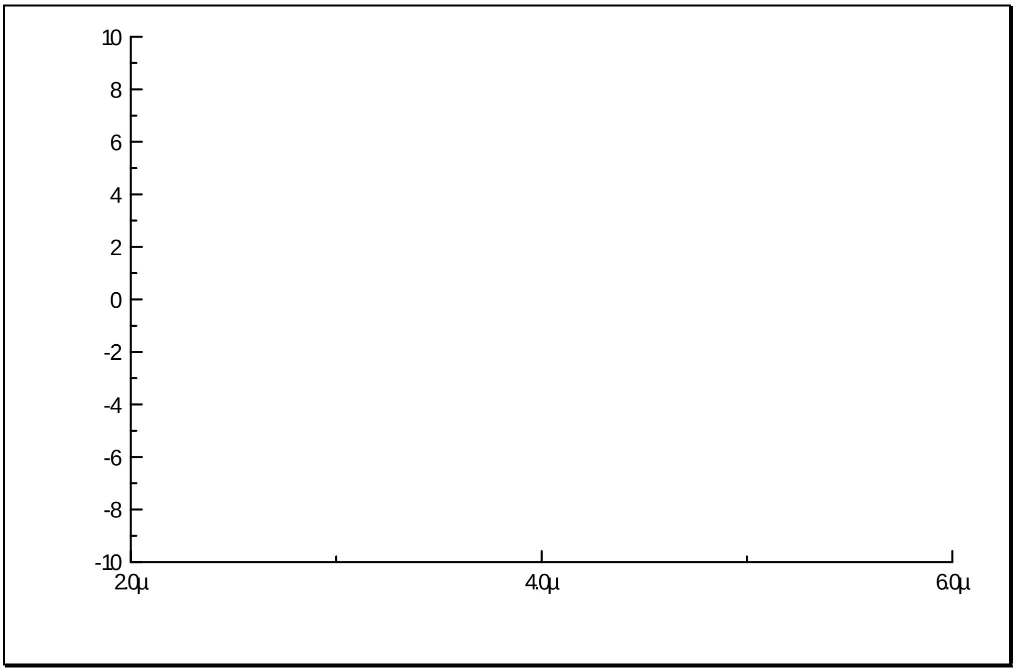

Fig. 4. **Time domain presentation of error signal (zoomed to the highest deviation area).**

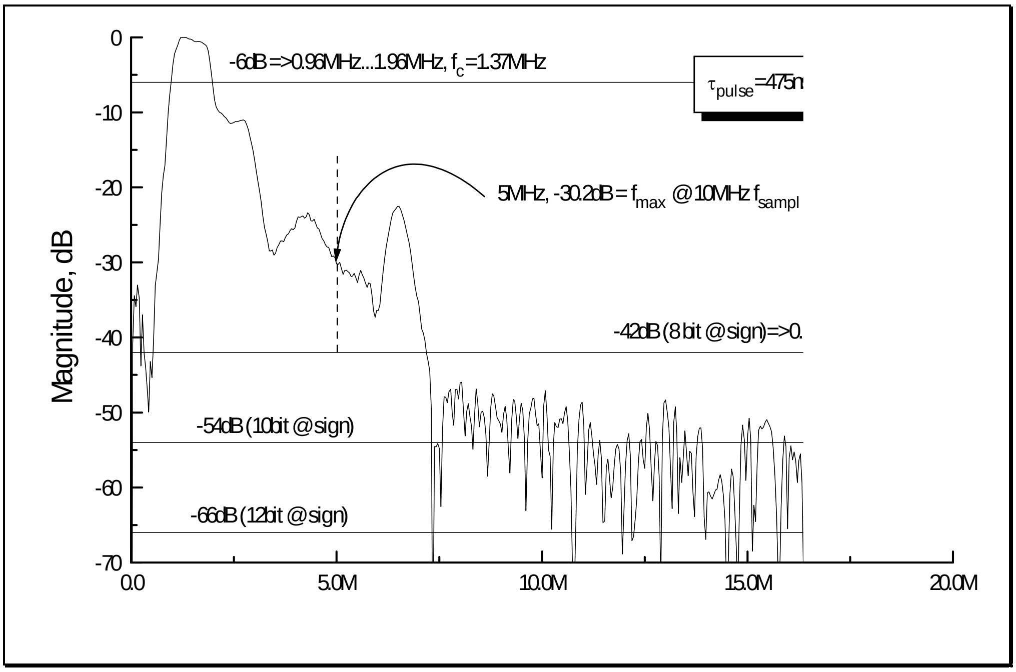

Fig. 2. **Amplitude spectra of 2 MHz transducer reflection from a steel mirror (@τ_{pulse}=500ns).**

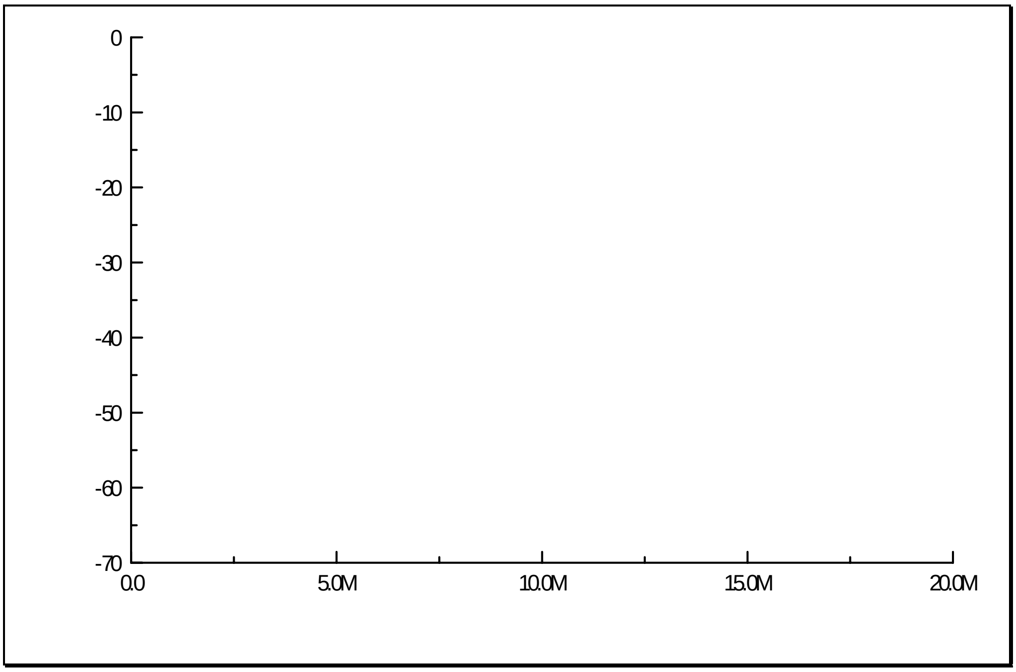

Fig. 3. **Amplitude spectra of 2 MHz transducer reflection from a steel mirror (@τ_{pulse}=500ns+5%).**

bandwidth was artificially increased. The matching coil was removed, so we also had the second and third harmonics present. A large number (1024) of A-scans were obtained in order to have good noise reduction and anti-aliasing filter with -6dB/octave was applied at the 5 MHz cut-off frequency. Every A-scan was stored on a disk separately. After collection, samples at the same time position were averaged. Such an approach allowed us to increase the signal to noise ratio, keeping the quantization noise level low. The averaged A-scan was Fourier transformed and amplitude spectra calculated in dB with respect to the central frequency amplitude. From these calculations the signal bandwidth at -6dB level was evaluated and the 8 bit A/D converter dynamic range levels based maximal frequency extracted. By investigating the results obtained (Fig. 1, Fig. 2 and Fig. 3 are presenting the spectra obtained with a slightly different pulse duration), one can see that despite bandwidth variations, the maximal frequency in spectra has not changed and is 7.2 MHz. Using the assumptions made above, the 8 bit converter should use 14.8 MHz sampling frequency (2x7.2 MHz).

Usually, the sampling frequency is chosen to be about 4 times higher than the transducer central frequency. We have chosen 10 MHz sampling frequency and resampled the same original A-scan with this frequency by taking just every 4-th sample. This sampling frequency choice satisfies the 95% of the signal energy inclusion in the active spectra criteria. In some processing algorithms (e.g. Wiener deconvolution, autoregresive spectral extrapolation, L1 norm deconvolution) the higher sampling rate is required in order to avoid interpolation between samples. We decided to verify experimentally the errors introduced by zero padding of the slightly undersampled signal. Such an interpolation method (convolution with the *sinc* function in the time domain) is the stated reconstruction method in sampling theorem definition [6]. Since the chosen sampling frequency was lower than required, the 3-rd harmonic was aliased into the signal spectra. The signal, sampled at 10 MHz, was transformed into the frequency domain, zero padded at higher frequencies in order to obtain 40MHz equivalent sampling and then transformed back into the time domain. After DC removal and normalization of both the original (signal $s_o(nt_{s40})$) and the resampled signal (signal $s_r(nt_{s40})$), the new signal $s_r(nt_{s40})$) was examined from the point of view of differences from the original $s_o(nt_{s40})$. Fig. 4. is presenting the error signal ($s_{err}(nt_{s40})=s_o(nt_{s40})-s_r(nt_{s40})$) in percents from the original ($s_{err}(nt_{s40})/max(abs(s_o(nt_{s40})))$. For a better comparison, the original signal $s_o(nt_{s40})$ is presented on the same picture.

Two relative error evaluation criteria were chosen:

1. Normalized average absolute difference, the L1-norm:

$$\|L1\| = \frac{\sum_{n=1}^{N_{samp}} \left| s_o(nt_{s40}) - s_r(nt_{s40}) \right|}{\sum_{n=1}^{N_{samp}} \left| s_o(nt_{s40}) \right|} \cdot 100\% ;$$

2. The normalized standard deviation, L2-norm:

$$\|L2\| = \frac{\sqrt{\sum_{n=1}^{N_{sampl}} \left[s_o(nt_{s40}) - s_r(nt_{s40}) \right]^2}}{\sqrt{\sum_{n=1}^{N_{sampl}} \left[s_o(nt_{s40}) - \bar{s}_o \right]^2}} \cdot 100\%$$

For the signal presented in Fig. 1. the L1-norm value was 11% and the L2-norm was 9.2%. The dashed line in Fig. 1 indicates the place from which the circular aliasing occurred and which, consequently, caused the errors mentioned. Here can be noted, that the place for the “maximal” frequency in such a case was -30 dB below the central frequency amplitude. Still, in some cases such a ratio of the sampling frequency can be acceptable, just one has to bear in mind possible artifacts caused by the circular aliasing. Of course, one would have had much lower errors if the sampling frequency was chosen for the limit below -42 dB, that is, at ratio of 8 or even 10 of the central frequency. The sampling frequency of 20 MHz satisfies the criteria we offer to use. In such a case L1-norm is 1.02 % and L2-norm is 4.4%, which fairly matches the accuracy obtained with the 8-bit A/D converter.

The algorithm

We suggest the following algorithm for an automated choice of the sampling frequency. In case the acquisition channel of an ultrasonic system has the possibility of the equivalent time sampling (ETS), it’s easy to obtain a signal acquisition at a sufficiently high sampling rate. Of course, the sampling time will grow sufficiently, especially if to take into account that a large number of the A-scans will be necessary in order to obtain a noise reduction. The reflected signal here must have as high as possible frequency content. The reflecting surface must be adjusted to be in focus and perpendicular to the ultrasonic beam. When averaging, the signal data should not be truncated to A/D converter data representation length, but the obtained improvement in dynamic range should be pertained, using the real numbers data presentation. When the sampled signal with a sufficient averaging applied is obtained, it should be transformed into the frequency domain for closer examination. The data should be presented in dB with respect to the central frequency amplitude. Then, using on the A/D converter dynamic range corresponding the maximal frequency is obtained. The sampling frequency should be twice of this frequency. In such a case errors will be reduced to a minimum because of aliasing in frequency domain and the sampling frequency will be optimal. If the system is designed bearing everything said in mind, ETS can be combined with the real-time (single-shot) sampling. Then, the ETS can be used just for a real-time sampling frequency choice and for applications with extreme accuracy requirements. The real-time sampling then can be used for conventional acquisition modes.

Conclusions

Care must be taken when choosing the A/D conversion rate. If the A/D sampling rate is chosen slightly lower than the limit required, care must be taken for possible artifacts. Here, we have proven, that -6dB /-3 dB bandwidth calculation is not enough for an optimal choice of the sampling frequency, unless a sufficient oversampling is chosen. The signal frequency characteristics have to be measured in a wide dynamic range. Then, based one of criteria for sampling frequency choice, the sampling frequency can be chosen. We suggest to use the A/D converter dynamic range for a maximal spectra frequency calculation.

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Ultragarso signalų diskretizavimas - juostos santykis su diskretizavimo dažniu

Reziumė

Skaitmeninis valdymas ultragarsinėms sistemoms suteikia papildomų privalumų. Signalo konvertavimas į skaitmeninę formą leidþia paprastai ir laksčiai organizuoti signalo apdorojimą. Projektuojant tokias sistemas, pirmiausia reikia pasirinkti diskretizavimo dažnį. Kai sistemos dažnių juosta yra žinoma iš anksto, diskretizavimo dažnis parenkamas remiantis Naikvisto (Nyquist) kriterijumi. Paprastai naudojamas kiek įmanoma žemesnis diskretizavimo dažnis. Diskretizavimo dažnis pažeminamas skaitmeninio apdorojimo trukmei sutrumpinti ir atminties talpai sumažinti. Reikia pasakyti, jog tokiu atveju būtinas pakankamas filtravimas, kad ciklinės sąsūkos (konvoliucijos) rezultatas neturėtų įtakos. Jei diskretizavimo dažnis parenkamas truputi žemesnis už reikalingą įvertinant Naikvisto kriterijų, reikia atsižvelgti ir į galimą iškraipymų įtaką. Šiame straipnyje parodėme, kad -6dB arba -3dB signalo juostos įvertinimas nėra pakankamas optimaliam diskretizavimo dažniui parinkti. Signalo dažninės charakteristikos tam tikslui turi būti ištirtos kur kas platesniame dinaminiame diapazone. Tada, remiantis straipsnyje aptartais kriterijais, optimalus diskretizavimo dažnis gali būti parinktas. Maksimaliam spektro dažniui nustatyti siūlome naudoti analoginio kodinio keitiklio dinaminio diapazono reikšmę.