

Ultrasonic Beamforming with Delta-Sigma Modulators

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Abstract: - Delta - Sigma ($\Delta\Sigma$) modulation allows delay resolution in ultrasound beamformers to be achieved by simple clock cycle delays applied to the undecimated bitstream, greatly reducing the complexity of the signal processing and the number of bits in the data path. The signal resampling involved in dynamic $\Delta\Sigma$ beamforming, however, disrupts synchronization between the modulators and demodulator, causing significant degradation in the signal-to-noise ratio. As a solution a beamformer architecture based on non-uniform oversampling $\Delta\Sigma$ conversion was explored. Using this approach, the echo signals received by the transducer array are sampled at time instants determined by the beamforming timing and then digitized by single-bit $\Delta\Sigma$ A/D conversion prior to the coherent beam summation. The $\Delta\Sigma$ coded beamsums obtained by adding the delayed 1-bit echo signals are then processed through a decimation filter to produce final outputs.

Key-Words: - Delta-Sigma, Beamformer, Ultrasound, Delay

1 Introduction

The newer digital ultrasound imaging systems have greatly benefited from improvements in semiconductor electronics, which have improved the speed and accuracy of the necessary data conversion. These advancements have resulted in high quality, 2-D real-time imagers. Recent advances in semiconductor electronics have made it possible to develop ultrasound beamformers that can acquire and process data at a sufficiently high rate to produce real-time 3-D images.

Because a significant fraction of a 2-D data set has to be collected and processed over the course of a single transmit event, beam formation must either occur serially with a great amount of memory and speed or in parallel using duplication of hardware.

The need for simplicity in the delay calculation forces the need for an A/D converter that samples well in excess of the Nyquist rate to reduce error in calculation of the fine delays. Although such a system is feasible, there is great potential for further hardware reduction.

An interesting method of achieving such a high data conversion rate, while making major simplifications in the signal processing hardware, is the use of a $\Delta\Sigma$ modulator as the A/D converter in the beamformer, as proposed by [1], and developed in [3]. When a $\Delta\Sigma$ modulator is operated with a sampling frequency (f_s) of 32x the center frequency (f_c) of the ultrasound signal or higher the number of bits involved in the signal processing is greatly reduced, and the need to interpolate between samples is

eliminated. It then becomes feasible to achieve parallel beam formation via duplication of hardware, which has been drastically simplified over the conventional case. The price of this simplicity is the need for a high speed $\Delta\Sigma$ modulator. Although such high speed operation of the $\Delta\Sigma$ modulator might appear to be wasteful of power, $\Delta\Sigma$ modulators use significantly less power than Nyquist-rate A/Ds operating at the same speed, because they supply only a single bit of resolution at that speed. That single bit of resolution, however, is enough to provide the required delay accuracy, saving the conversion to high resolution until after the complex beamformer hardware, where it is needed at a much lower rate. Most importantly, the speed and size of the digital hardware is reduced because of the simplified signal processing and reduced number of bits being processed.

2. $\Delta\Sigma$ Modulators

The oversampling ratio OSR is defined to be the ratio of the sampling frequency f_s to twice the signal band edge (the Nyquist rate)

$$OSR = f_s / (2f_B) = 1 / (2f_B T) \quad (1)$$

where T is the sampling period. As the oversampling ratio increases, the quantization noise can be spread over a larger range of frequencies, so the noise power in band changes.

Using (1) we can write the noise power that lies within the limited signal band ($-f_B$, $+f_B$), as

$$\sigma_{ey}^2 = \int_{-f_B}^{f_B} P_e(f)df = 2 \int_0^{f_B} P_e(f)df = \sigma_e^2(2f_B T) = \frac{\sigma_e^2}{OSR} \quad (2)$$

The signal to noise ratio (SNR) of a multi-bit A/D is found by logarithmically compressing the ratio of the signal power to the quantization noise power.

$$SNR = 10 \log_{10} \left(\frac{\sigma_x^2}{V^2} \right) + 4,77 + 6,02N \quad (3)$$

where N is the number of bits in the quantizer. The relation shows that every extra output bit from the quantizer provides an extra 6dB improvement in the SNR.

An oversampled A/D converter exploits the relation (2) so that a very high sampling rate is used with a very simple, perhaps single-bit quantizer, to provide adequate SNR. Delta sigma A/Ds are a subclass of oversampled A/Ds that filter the quantization noise to reduce in-band noise power.

A first order $\Delta\Sigma$ modulator (figure 1) consists of an integrator, an internal flash ADC or quantizer and a DAC employed in the feedback path. The signal that is quantized is a filtered version of the difference between the input and the analog representation of the modulator output. The filter is a discrete time integrator whose transfer function is $z^{-1}/(1-z^{-1})$.

If the DAC is ideal the modulator output is given by:

$$Y(z) = X(z)z^{-1} + E_q(z)(1-z^{-1}) \quad (4)$$

so the signal transfer function is $STF(z) = z^{-1}$ and the noise transfer function is $NTF(z) = (1-z^{-1})$. The output is simply a delayed version of the input plus the quantization noise that has been shaped by a first order high pass filter.

The peak SNR is given by:

$$SNR_{peak} = 10 \cdot \log \left(\frac{3}{2} 2^{2N} \right) + 10 \cdot \log \left(\frac{3}{\pi^2} \cdot (OSR)^3 \right)$$

Thus the peak SNK increases by 9dB or 1.5 bits when OSR is doubled but we need a relative high

OSR, more than 1000 to achieve a SNR of 16 bits (100dB) with this modulator.

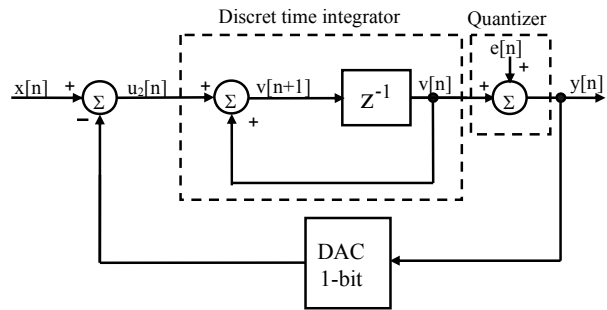


Fig. 1 A first order delta-sigma modulator

The structure of a second order $\Delta\Sigma$ modulator contains two integrators and a forward path is included in both integrators – figure 2. The output of the modulator is given by:

$$Y(z) = X(z)z^{-2} + E_Q(z)(1-z^{-1})^2 \quad (5)$$

where $STF(z) = z^{-2}$ and $NTF(z) = (1-z^{-1})^2$.

Compared with the first order NTF, the second order NTF provides more quantization noise suppression over the low frequency signal band and more amplification of the noise outside the signal band i.e. more power noise is pushed to higher frequencies. The peak SNR of the second order architecture is:

$$SNR_{peak} = 10 \cdot \log \left(\frac{3}{2} 2^{2N} \right) + 10 \cdot \log \left(\frac{5}{\pi^4} \cdot (OSR)^5 \right)$$

Therefore for every doubling of the OSR, the peak SNR increase by 15 dB or equivalent resolution by 2.5 bits. To obtain a SNR of 16 bits we need an OSR of only 116 which is much smaller than the first order case.

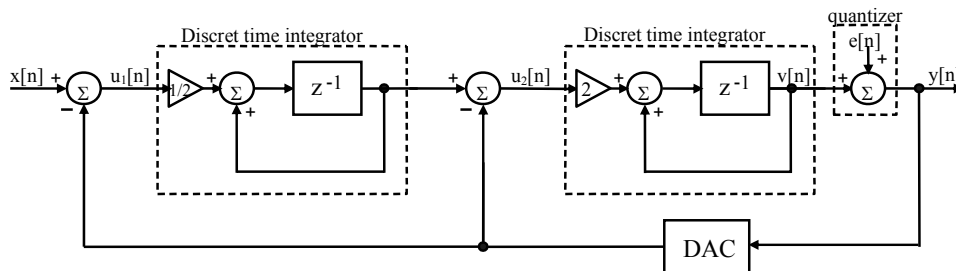


Fig. 2 A second-order delta-sigma modulator

$\Delta\Sigma$ modulators with higher-order noise shaping can achieve higher resolution by pushing more noise power outside the signal band. Unfortunately, unlike first and second order modulators it is possible for

modulators of third or higher order to go unstable especially when large input signals are applied. On the other hand multi-bit modulators architectures using multi-bit quantizers reduce the amount of in

band quantization noise and improve the SNR by 6 dB for each bit added. The drawbacks of this architecture are the requirement for full linearity of feedback DAC and the significant increase of the converter complexity

One of the key benefits of $\Delta\Sigma$ modulators over multi-bit flash ADCs is the relative simplicity of the $\Delta\Sigma$ ADC implementation. Based on this the modulator architecture for ultrasound applications will be chose in order to ensure the simplicity and stability requirements and a resolution equivalent to a 10 bits multi-bit converter which correspond to a 60dB dynamic range. An OSR of 32 for a third order modulator assure the desired dynamic range. This means that for a transducer array operating at 3.5 MHz the modulators and the digital section of the beamformer will be clocked at a frequency around 200 MHz.

3 The beamformer architecture.

3.1 Steering and focus delay

A ultrasonic beam generated by a phased array can be both focused and steered by properly delaying the transmitted and received signals. Consider a phased array with N elements and inter-element spacing $d = \lambda/2$, Fig. 2. It is to be focused and steered at point (r, θ) , where θ is the steering angle and r is the range.

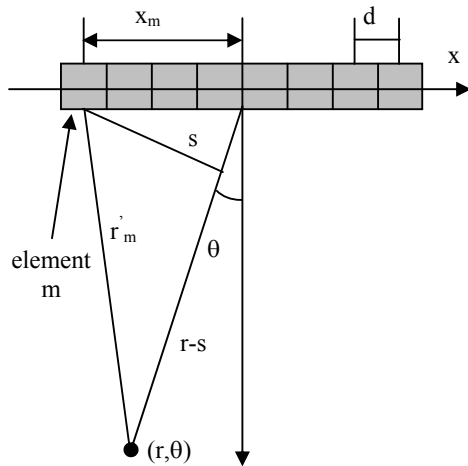


Fig. 3 The geometry of an array focused to point (r, θ)

The one way delay for element m is given by: $\tau_m = (r - r'_m)/c$ (5) where r'_m is distance between element m and point (r, θ) , $0 \leq m \leq N-1$. According to Law of Cosine, the distance r'_m , is

$$r'_m = \sqrt{x_m^2 + r^2 - 2x_m r \sin \theta} \quad (6)$$

where x_m is the position of the m'th element. The one way delay is

$$\tau_m = \frac{1}{c} \left(r - \sqrt{x_m^2 + r^2 - 2x_m r \sin \theta} \right) = \tau_m^s + \tau_m^f \quad (7)$$

$$\tau_m^s = \frac{s}{c} = \frac{x_m \sin \theta}{c} - \text{steering delay} \quad (8)$$

$$\tau_m^f = \frac{r - s - \sqrt{x_m^2 + r^2 - 2x_m r \sin \theta}}{c} - \text{focusing delay} \quad (9)$$

The relations 7-9 show a nonlinear dependence of the delays from the focal point coordinates. Only focusing delay τ^f is dependent on the point's range r. Focusing can be fixed, dynamic, or composite.

Fixed focusing synthesizes a focus at only one particular range R, typical in the middle of the interrogated region. In *dynamic focusing*, the focusing delay is adjusted as function of range r. Dynamic focus on reception is commonly used so that depth of field is extended without any reduction of frame rate.

3.2 Delta-Sigma beamformer

Beamformers based on $\Delta\Sigma$ have two major advantages over traditional multi-bit methods.

I. Simplifying the ADC structure and the possibility of integrating on the same chip together with the digital processing blocks. The necessary circuits for implementing $\Delta\Sigma$ ADC are much simpler than those used in multi-bit flash ADC (several operational amplifiers and a comparator in contrast to the 255 comparators used for a 8 bit flash ADC), and this will reduce the resources used for interconnection, the size and energy usage.

II. Precise simple delays can be ensured by manipulating the samples taken with a high sampling rate. The complicated circuits needed for interpolating/decimating are replaced with a conventional FIFO memory that can ensure the delay with the desired resolution before summing. Further, the whole area summing is simplified because the first summing level is built with one-bit adders.

Because of the wide dynamic field required for ultrasound applications, ADC $\Delta\Sigma$ must be a second or third level modulator in order to keep an adequate signal versus noise ratio. To ensure both an adequate signal/noise ratio and an adequately accurate delay we chose a sampling frequency of at least 32 times higher than the bearing frequency. This will give an over sampling factor $OSR = 16$. In the uniform sampling beamformer, figure 1, the digital output of the modulator is applied to the delay lines. These are of variable length, and are implemented as FIFO memory or as shift registers and dynamically delay

the signals. These shift registers must operate at the sampling frequency and contain several hundred cells in order to host the dynamic delays during each beam. The length of each delay line depends on the next parameters: *sampling rate, array geometry, steering angle, maximum imaging depth, carrier frequency*. When the length of the delay line changes in order to create a new focal zone, the samples must be repeated, inserted or eliminated at the input, output or even in the middle of the shift register.

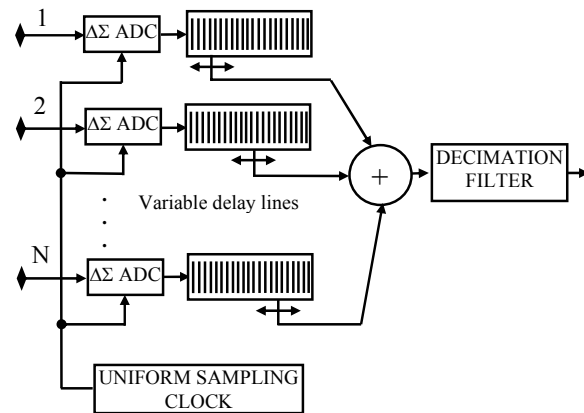


Fig. 4 Block diagram for an uniform $\Delta\Sigma$ beamformer

The delayed samples from all the array elements are then digitally summed to form a beam. This summing must be done at the sampling frequency and this requires a set of digital pipeline adders for the whole area. Once the beam has been generated by summing the signal it is applied to a low pass filter with to eliminate the quantization noise. This superior order filter can be long and complicated. It also has certain power consumption, but only one such filter is required in order to generate a beam. In order to use the traditional processing blocks (envelope detection, scan conversion) it is necessary to reduce the sampling rate through the decimation of the filtered sample string.

The $\Delta\Sigma$ ADC together with the low pass reconstruction filter constitutes a modulator – demodulator pair. Between the modulator and demodulator can be introduced blocks that executes linear operation without this interfering with SNR.

The process of creating the ultrasonic beam by the delay-sum algorithm can be divided into two distinct stages:

- **Delay** – the time lag of the information provided by the area elements.
- **Sum** - Summing of the delayed elements.

Focusing the acoustic waves in a close field requires the delay after a parabolic formula – see (7).

Therefore, the beam forming from ultrasonic signals is done by a non-linear delay operation, followed by the linear summing operation.

Using the sigma-delta with uniform modulation converters for beam formation requires the intercalation between the modulator and demodulator of the non-linear delay process followed by the linear summing process – fig. 5

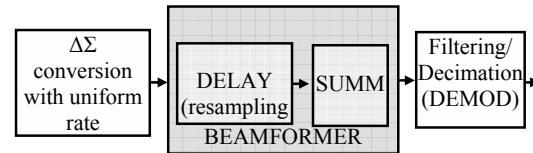


Fig. 5 Uniform $\Delta\Sigma$ beam formation

The non-linearity of the delay leads to the resampling of the signals delivered by the array elements that lead to the repeated use of some samples and some samples are never used at all. The repetition or elimination of samples from the sigma-delta modulators outputs leads to the asynchrony of the modulator / demodulator ensemble. The preponderance of one or the other mechanisms is a function of the ratio between the sampling frequency (f_M) and the forming frequency (f_{BF}).

The sigma-delta beamforming with non-uniform rate consists of the extraction of the non-linear delay process from the modulator / demodulator ensemble and placement of this before the modulator. In this way the synchrony of the modulator / demodulator couple is ensured, because between these two blocks only the linear summing operation is intercalated – figure 6.

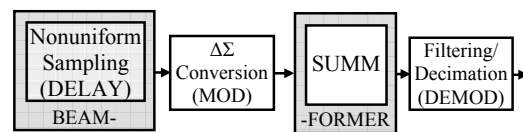


Fig. 6 Non uniform $\Delta\Sigma$ beam formation

The digital beam formers that are based on non-uniform sampling create the dynamic focusing by sampling the echo samples at specific moments that ensure the coherent summing operation at the reception. The clock for the non-uniform sampling is generated by the delay set stored in a digital memory operated with a uniform clock f_M . The sampling clock generator creates different non-uniform clock for each area element.

The dynamically delayed samples are then digitized with a set of one-bit sigma-delta modulators, before the coherent summing. Each modulator is activated

with his non-uniform clock created by the generator – figure 7. Each modulator's exit is a string of one-bit data that includes the delayed original signal plus the high frequency quantifying noise.

These one-bit representations for the echo samples appear at various moments in time on reception channels because there is no regularity in the signal arrival time. If the time between two consecutive focal points $1/f_{BF}$ is greater than the maximum delay between elements ($\max \{\tau_n(r, \theta)\}$), then for every focal point the requested sample in the forming process appears before the sample required for the next focal point. The one-bit coded samples that appear at different moments in time at the modulators' exits are aligned by the FIFO buffers on each channel and will be simultaneously sent to the adders that generate the rough version for the fascicle that will contain the sum of the delayed echoes plus the quantifying noise. The reconstruction filter will eliminate the quantifying noise and will reduce the sampling rate close to the Nyquist rate.

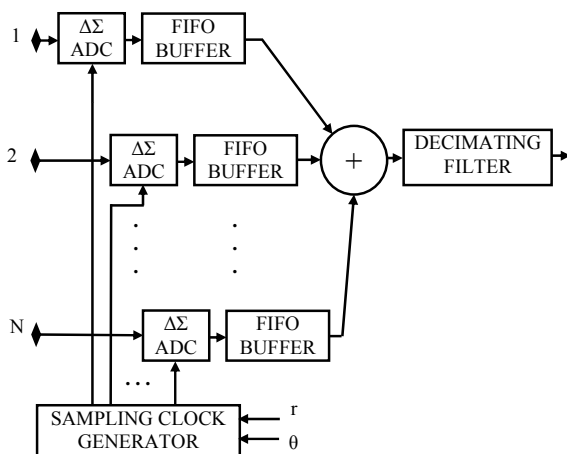


Fig. 7 Structure of a non uniform $\Delta\Sigma$ beamformer

From a structural point-of-view, the digital beam former with a non-uniform sigma delta modulation is made up of two sections:

- An analogue section - the sigma-delta modulators
- The digital section which contains the following entities: Digital controller, FIFO memory, adder and decimating filter.

4 Simulation results

The $\Delta\Sigma$ based beamformers presented above was emulated in software using real ultrasound data available over the www. [4].

Four beamforming methods and the corresponding structures were emulated using Matlab scripts:

- Conventional multi-bit beamformer with fixed focus on transmission and dynamic focus on reception – which was the reference (C-FFT_DFR)
- Uniform $\Delta\Sigma$ beamformer with fixed focus on transmission and dynamic focus on reception ($U\Delta\Sigma$ -FFT_DFR)
- Uniform $\Delta\Sigma$ beamformer with fixed focus on transmission and fixed focus on reception ($U\Delta\Sigma$ -FFT_FFR)
- Non-uniform $\Delta\Sigma$ beamformer with fixed focus on transmission and dynamic focus on reception ($N\Delta\Sigma$ -FFT_DFR)

For performance evaluation was used the spectral analysis at the beam level. The figures 8 – 11 show the emulation results.

Analyzing the diagrams we can see the presence of an unexpected noise in the frequency range 0-1.7MHz for $U\Delta\Sigma$ -FFT_DFR. The noise level is 30 dB greater than for C-FFT_DFR. The noise appears from the resampling during dynamic focusing of the delta sigma modulated signals. The hypothesis is confirmed by the noise absence from the diagram of $U\Delta\Sigma$ -FFT_FFR. In this case due to the fixed focus the noise difference to conventional beamformation is less than 10dB. It is obvious that the dynamic focusing for uniform sampling disrupts the synchronization for $\Delta\Sigma$ modulator / demodulator pair.

The same noise with 10 dB greater than for C-FFT_DFR is present in the $N\Delta\Sigma$ -FFT_DFR diagram This suggest that the $\Delta\Sigma$ with nonuniform sampling beamformer is a good solution which use the $\Delta\Sigma$ modulation benefits and provide for dynamic focusing a SNR performance comparable with the conventional multi-bit beamformer.

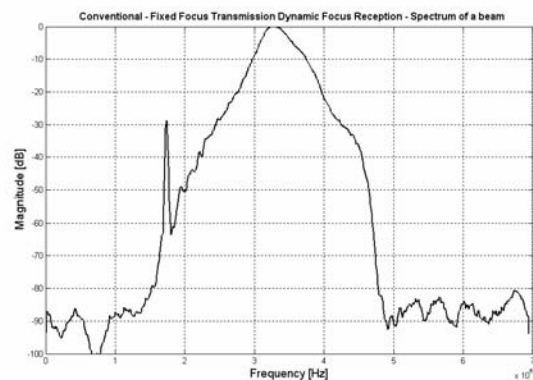


Fig. 8 Spectrum of a beam - C-FFT_DFR

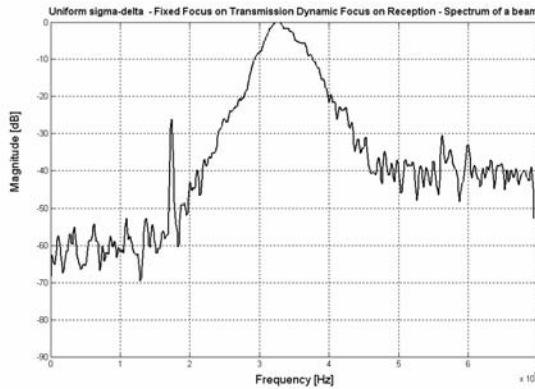


Fig 9 Spectrum of a beam - $U\Delta\Sigma$ -FFT_DFR

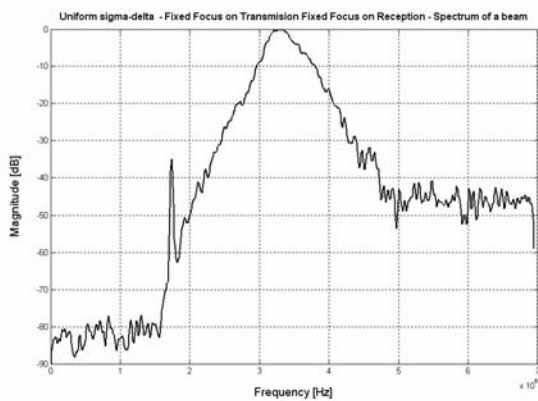


Fig. 10 Spectrum of a beam - $U\Delta\Sigma$ -FFT_FFR

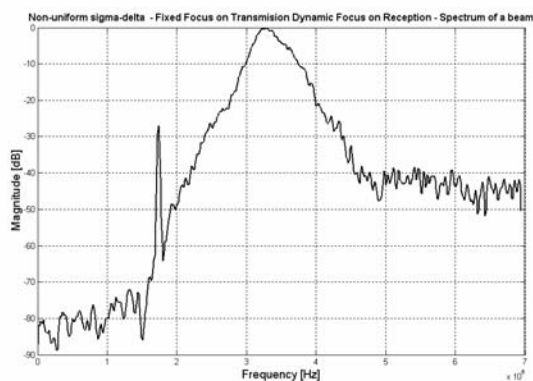


Fig. 11 - Spectrum of a beam - $N\Delta\Sigma$ -FFT_DFR

5 Conclusion

The signal resampling involved in dynamic $\Delta\Sigma$ beamforming disrupts synchronization between the modulators and demodulator, causing significant degradation in the signal-to-noise ratio. As a

solution a beamformer architecture based on non-uniform oversampling $\Delta\Sigma$ conversion was explored. Using this approach, the echo signals received by the transducer array are sampled at time instants determined by the beamforming timing and then digitized by single-bit $\Delta\Sigma$ conversion prior to the coherent summation. The performance of the proposed beamforming solution was evaluated by MATLAB emulations using experimental data. The $\Delta\Sigma$ coded beamsums obtained by adding the delayed 1-bit echo signals are then processed through a decimation filter to produce final outputs. Delta sigma ($\Delta\Sigma$) modulation allows delay resolution in ultrasound beamformers to be achieved by simple clock cycle delays, greatly reducing the complexity of the signal processing and the number of bits in the data path.

The simplifications offered by this technique have the potential for low power and portable operation in advanced ultrasonic imaging systems

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