A WIRELESS ULTRASONIC NDT SENOR SYSTEM

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Abstract: Ultrasonic condition monitoring technologies have been traditionally utilized in industrial and construction environments where structural integrity is of concern. Such techniques include active systems with either single or multiple transmit-receiver combinations used to obtain defect positioning and magnitude. Active sensors are implemented in two ways; in a thickness operation mode, or as an area-mapping tool operating over longer distances. In addition, passive ultrasonic receivers can be employed to detect and record acoustic emission activity. Existing equipment requires cabling for such systems leading to expensive, complicated installations.

This work describes the development and operation of a system that combines these existing ultrasonic technologies with modern wireless techniques within a miniaturized, battery-operated design. A completely wireless sensor has been designed that can independently record and analyze ultrasonic signals. Integrated into the sensor are custom ultrasonic transducers, associated analogue drive and receive electronics, and a Texas Instruments Digital Signal Processor (DSP) used to both control the system and implement the signal processing routines. BlueTooth wireless communication is used for connection to a central observation station, from where network operation can be controlled.

Extending battery life is of prime importance and the device employs several strategies to do this. Low voltage transducer excitation suffers from poor signal-to-noise ratios, which can be enhanced by signal processing routines implemented on the DSP. Routines investigated include averaging, digital filtering and pulse compression.

Introduction: Ultrasonic Non destructive Testing (NDT) and in particular structural health monitoring cover a broad application spectrum, ranging from passive acoustic emission detection to active time-of-flight structure interrogation. Generally, the results of such testing are integral to planned maintenance schedules, however circumstances can lead to immediate operator intervention.

Traditional structural health monitoring systems have two major shortcomings. Firstly they are cabled systems, using either copper wires or fiber optics. Not only does this lead to high installation and maintenance costs but can also lead to problematic network extension or reconfiguration. Furthermore, traditional cabled systems overcome the inherently large ultrasonic insertion losses with high excitation voltage and power levels. Time-of-flight pulse echo systems using pulsed excitation operate with voltage levels in the hundreds. Not only are these voltages impractical to generate and use in a battery-powered devices, but also when operating in hazardous environments they introduce health and safety implications.

This work describes the design and implementation of a wireless, miniaturized battery-powered ultrasonic test unit. The sensor is fully autonomous while complying to typical matchbox dimensions of 35x50mm. The low voltage excitation operation must be addressed as the high insertion loss can lead to an unfeasibly low signal-to-noise ratio (SNR), rendering the measurement unreliable. This paper investigates a number of techniques to improve such situations. The simplest technique would implement an averaging algorithm to resolve the low SNR. Assuming the noise is random in nature, a characteristic of electrical noise [1], averaging will reduce its effects, however this could take hundreds of cycles unnecessarily consuming battery power. A digital Finite Impulse Response (FIR) filter can remove out-of-band noise and can provide a substantial SNR enhancement. A third technique that has found widespread favor is the pulse compression or matched filtering method [2]. Correlation between the received signal and the transmission waveform can provide a significant SNR enhancement.

Matchbox Concept

The distributed matchbox concept can be illustrated as shown in Figure 1.

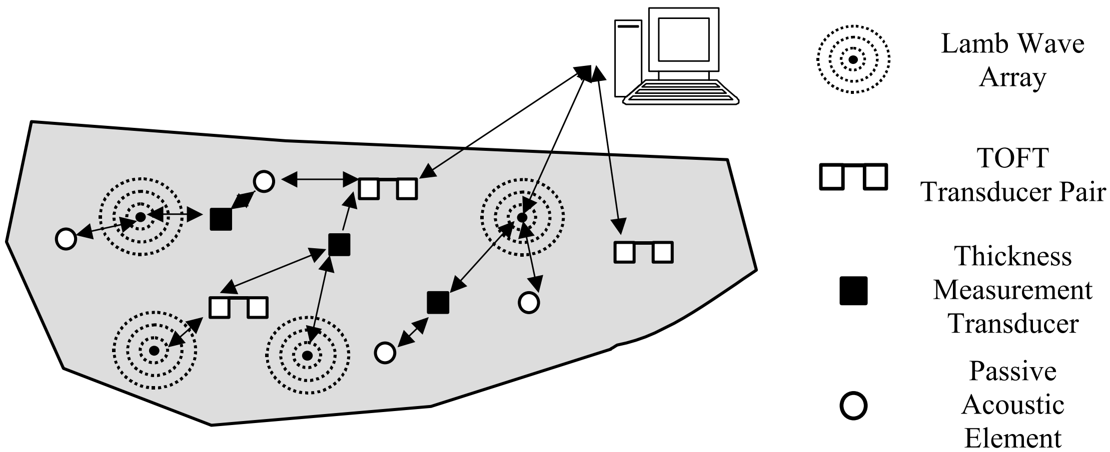


Figure 1, Thee distributed Matchbox Sensor Concept

Matchbox sensors are dispersed across a structure; typical industrial environments include pipe work, boilers or structural metal-work. Passive sensors over the structure can detect acoustic emissions, so by triangulating the event arrival timing the defect position can be calculated. Operating actively the matchbox sensor provides a versatile ultrasonic test platform, performing a number of differing ultrasonic tests. Active measurements include time-of-flight interrogation; both pulse-echo and Time of Flight Diffraction (TOFD) examinations are possible. Typical industrial applications include remaining wall thickness measurements, corrosion and erosion monitoring and crack observation. The sensors all inter-communicate using the wireless BlueTooth technology. The BlueTooth capabilities allow the network to operate adaptively, where sensor end points can convert into transmission relays. This is essential when the distance between base station and measurement point is too large for a single link.

Matchbox Design

Working with a design brief of small, easily re-configurable devices, the matchbox design concept is one of a layered modular design. Three modules are pictured in Figure 2. From left to right they are the Ultrasonic Interface Module, the control and processing module and the communications module.

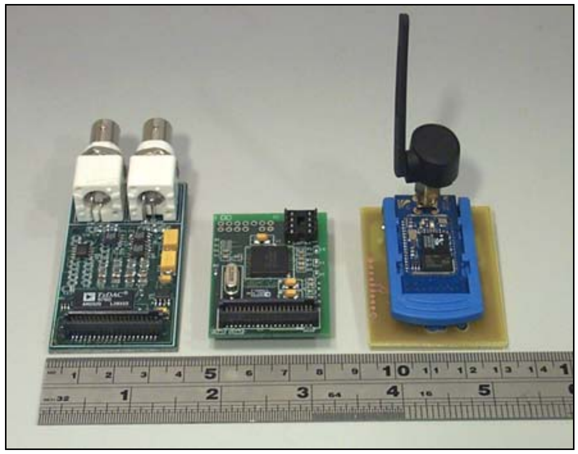


Figure 2, The modular matchbox design

Miniaturization was achieved using surface mount components and reduced footprint IC package standards; an example of this is the DSP IC. Using the fine pitch plastic ball grid array MicroStar package and a solid four-row ball array configuration at a pitch of 0.8 mm, the 144 connections can be mounted within a package size of only 12x12 mm. Routing with a standard 8 mil conductor width and 8 mil spacing can satisfy the pin pitch of almost 15 mils. Using a through hole dog-bone routing strategy, as illustrated in Figure 3, the DSP can be mounted and efficiently routed with two power planes, a ground plane and five signal layers. Although this is a relatively time consuming manufacturing process, as each hole is mechanically drilled, it minimizes cost as buried or blind vias are avoided. Another disadvantage is the top-side copper clearance; manufacturing tolerances can reduce or possibly eliminate, the copper between ball and via. The solution implemented is the teardrop via design. Instead of using a circular copper top-side via pad a tear drop design is used, effectively increasing the drill tolerance as the copper area increases only in the connection direction.

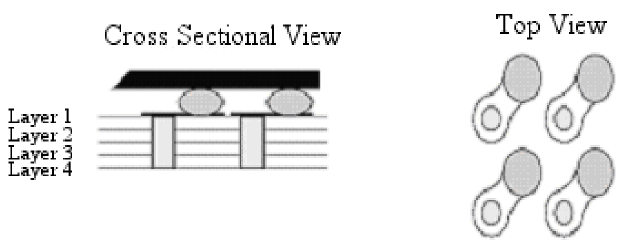


Figure 3, The dog-bone vertical routing strategy

Ultrasonic Interface Module

This is the lowest layer in the ultrasonic module. This layer has a dual role to both drive and receive from the ultrasonic transducer. Two ultrasonic drive schemes have been implemented; the pulsed drive and the Arbitrary Waveform Generator (AWG). The pulsed drive excitation uses a silicon CMOS switch to discharge a capacitor's energy into the transducer. Typical pulse waveform characteristics include; a voltage level of 16 volts and a pulse width of 150 nS. The AWG directly interfaces to the DSP busses converting parallel data into analogue drive signals. Combining digital DAC technology with high-speed low-power op-amps the AWG can accurately reproduce excitation waveforms such as tone bursts, chirps or RF modulated binary codes. Using a current output 50MSPS DAC binary data is converted into analogue signals, however the DAC output is unsuitable for transducer drive, hence it is initially amplified to the maximum rail-to-rail voltage. A high current output dual op-amp configuration then buffers the final output.

The receiver is a dual cascaded non-inverting AC coupled op-amp arrangement giving 40dB with a -3dB upper cutoff of 65MHz.

Control and Processing Module

The control and processing module hosts the necessary electronics to both capture and process the ultrasonic data, including amongst other components the DSP, the ADC and the boot EEPROM. The ADC is a 10 bit low-power device also running at 50MSPS, the captured SNR is 48dB. The DSP is a Texas Instruments TMS320VC5402 low power, fixed-point processor capable of operation up to 100 MIPS from a 100 MHz clock using a pipelined instruction architecture. The DSP incorporates 16 kB, 16 bit RAM and 4 kB, 16 bit ROM on-board. Serial data connection to the Bluetooth module is provided by the DSP's software driven UART. The software code is developed and compiled in the Code Composer Studio package using C, C++ and assembly language.

Communications Module

This layer is used to implement the networking aspects of the sensor. Using Bluetooth the module can inter-communicate with the base station and other modules. A fully embedded Bluetooth solution has been implemented illustrating the communications concept. Using Cambridge Silicon Radios (CSR) Casira development kit in conjunction with CSR's chip Bluetooth solution a single IC solution has been implemented. The single chip solution not only satisfies the hardware requirements but also due to the on-board RISC processor fulfills the software protocol requirement. The RISC processor, although limited in terms of memory and speed, is able to implement the upper BlueTooth stack layers, up to and including the application layer. This application uses a modified Serial Port Profile. Using point-to-point links each sensor can directly communicate with the surrounding modules.

Signal Processing

As noted previously the problem faced by low voltage excitation ultrasonic systems is the inherently low reception SNR. All time-of-flight measurements record echo timings, in turn these values can be used in conjunction with the sound speed in the test material to calculate depths and distances. Implemented on-board the DSP is a peak detection algorithm that simply records the echo signal. Knowing the maximum value index and the sampling frequency the echo timing can be found. Problems are encountered when the echo is masked by system noise, hence, the DSP implements SNR enhancement algorithms to extract the signal from the noise. Three techniques are presented which all improve the SNR and hence the detection reliability:

- Averaging: A technique commonly used within measurement systems is the averaging algorithm. Using two memory arrays an averaging algorithm can be easily implemented, as described by equation 1.1. If a system emits a periodic signal x where N is the period of the signal and the reflector is stationary then the noise level will reduce with l, the number of repetitions [3]:

$$y[n]=\frac{1}{l}\sum_{l=0}^{l-1} y[n-lN] \tag{1.1}$$

- FIR filtering: Using a Finite Impulse Response (FIR) digital filter the SNR can be enhanced by removing out-of-band noise [4]. The band of interest in this case is approximately 2 – 3 MHz, the -3dB pass band of the two-way transducer loss. Implementing an optimally designed pass band from equation 1.2 as a direct structure we can precisely calculate the computational effort, equation 1.3 [5], a factor directly related to the number of filter coefficients, and the gain in SNR.

$$r[j] = \sum_{k=0}^{nh} h[k]x[j-k] \quad 0 \le j \le nx \tag{1.2}$$

$$cycles = 4 + nx * (4 + nh) \tag{1.3}$$

where *r* is the resultant signal
h is the input signal array
x is the filter coefficient array
nx is the number of input samples
nh is the number of filter coefficients

- Pulse Compression: Essentially pulse compression is an estimation algorithm; hence the suitability for echo detection is apparent [5]. Implementing from a cross-correlation technique, equation 1.4, the computational effort can be calculated, equation (1.5) [6].

$$r[j] = \sum_{k=0}^{nr-j-1} x[j+k] * y[k] \quad 0 \le j \le nr = nx + ny - 1 \tag{1.4}$$

$$cycles = (16 + (nx-3)(nx-2) + 17*(nx-3)) + (14 + (ny - nx + 1)(nx - 2 + 8)) + 41 \tag{1.5}$$

Where *nx* is the number of elements in array *x*, the reception array
ny is the number of elements in array *y*, the comparison array

Results: To verify and analyze the three different SNR enhancement algorithms, a single point test setup was used. The test set up operates a dual-probe transducer in pitch catch mode with a 70mm deep steel block, of which the two-way insertion loss is 50dB at the 2.5 MHz center frequency. The test is intended to capture the back wall echo and hence the depth of the material.

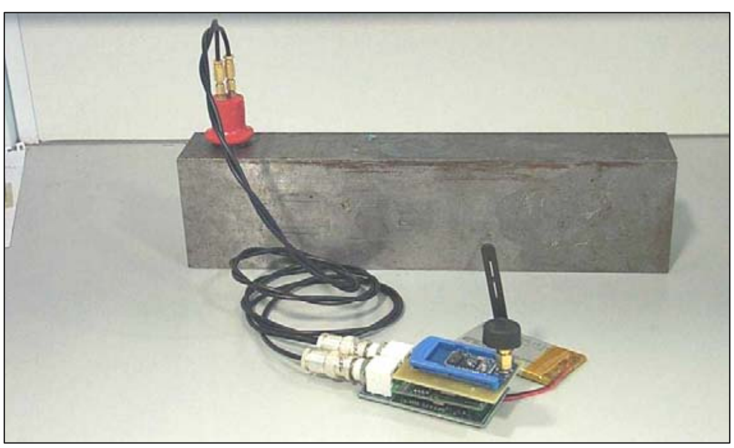


Figure 4, The experimental Set-Up

Averaging

Averaging over 50 periods we can plot both the noise rms voltage and the echo SNR.

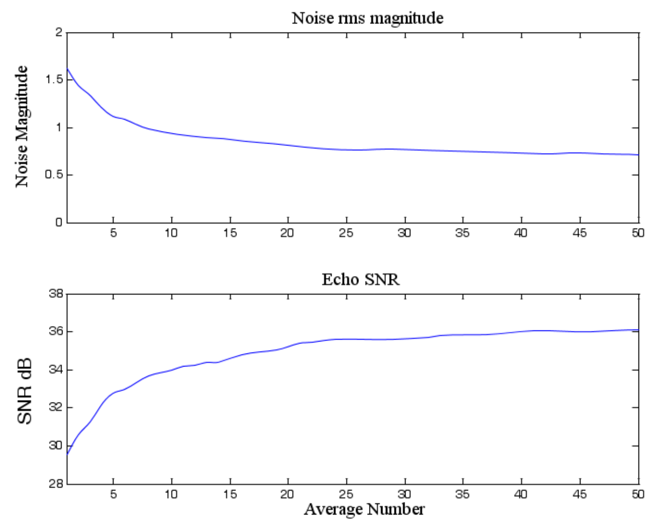
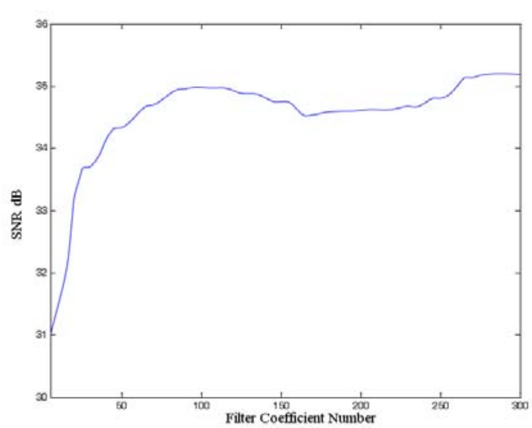


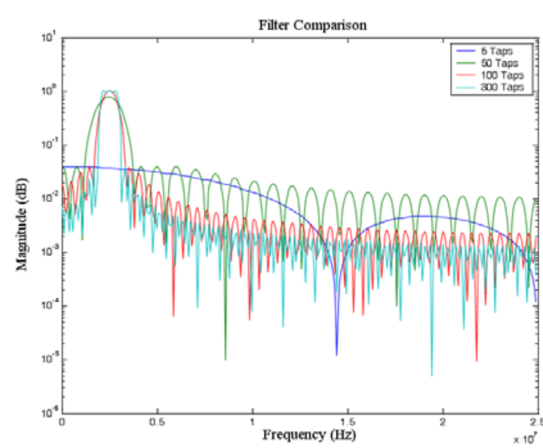
Figure 4 A comparison of SNR enhancement while averaging

Filtering

Using MatLab [7] we can design the optimal FIR filter. The filter coefficient values are transmitted to the DSP where the DSP implements the filter design. Using a constant sample number for reception the computational effort is solely dependent on the filter length, equation (1.3). The optimal design is one that minimizes the filter length while maximizing the system SNR. Figure 5.a shows the SNR plotted against the filter length, while Figure 5.b shows a frequency magnitude plot of selected filter responses.



(a)



(b)

Figure 5.a, comparison of SNR against filter coefficients and, 5.b, example filter magnitude responses

Pulse Compression

Using a constant reception length the computational effort is related directly to the excitation signal length. Figure 6.c shows a plot of system SNR against excitation length, which is varied in length from 10 to 200 samples. Figures 6.a and 6.b show plots of noise rms level and echo rms level against excitation length.

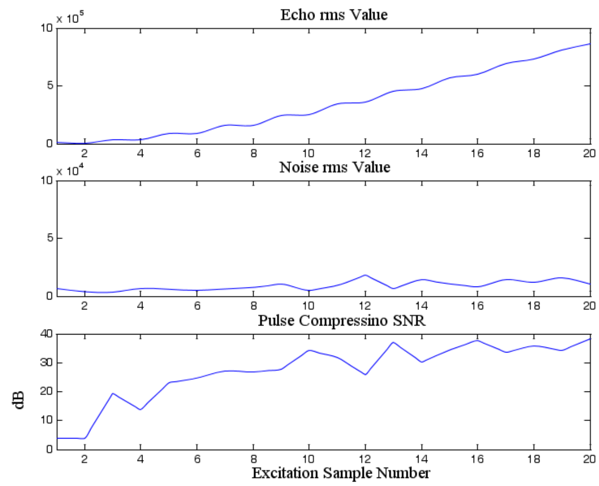


Figure 6, the noise, echo rms values and the pulse compression SNR

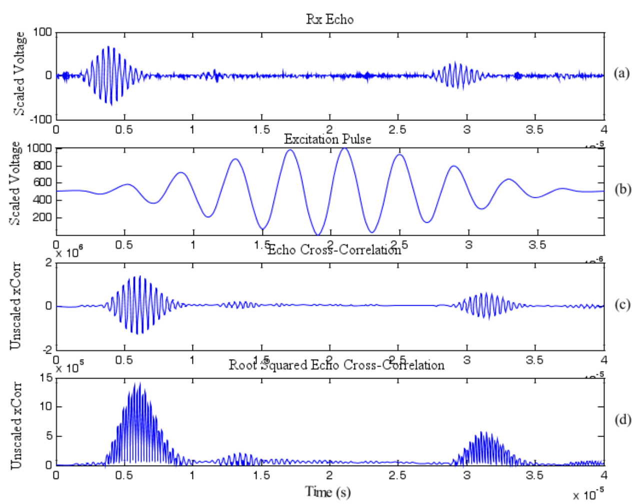


Figure 7, a time plot of the received echo, the excitation wave, cross-correlation and root squared cross-correlation

Discussion: The directly sampled data leads to a SNR of 24dB, each of the three proposed techniques increases this figure to values of 36dB, 35dB and 38dB for averaging, filtering and matched filtering respectively. The similar increases in SNR are however not comparable when considering the power consumption. Inspection of Figure 4 reveals an optimal of ten averages, or ten pitch-catch acquisitions. Ten cycles yields a SNR of 34dB however current consumption must be considered, as prolonging battery life is a major goal. When firing the transducer the ultrasonic interface module consumes approximately 45 mA, this is a combination of both the analogue amplifiers and the digital electronics running at the full speed of 50 MHz. When the ultrasonic electronics are disabled and the DSP operates at a lower clock frequency the current drain drops to 2mA, a significant decrease. Thus the averaging technique becomes less desirable and the single shot FIR filtering and matched filtering techniques become more so. The FIR filtering reaches an optimal value at one hundred filter coefficients, prior to this value the frequency response is not fully defined resulting in pass band attenuation. This point is illustrated in Figure 5b as the five coefficient filter response is attenuating both pass and stop bands. By one hundred coefficients the filter response is well defined with a stop band attenuation of 60dB. Further coefficients reduce the figures but not significantly. The pulse compression also used a windowed tone burst centered on the transducers center frequency of 2.5 MHz. As the computation is directly related to the transmitted pulse length, it is also related to the received signal length however this is kept constant throughout. The experiment analyses the SNR against transmission length. Figure 6 illustrates the SNR increasing with sample length. The test ran from sample length 10 to 200. The SNR increases proportionally with the sample count, this is of course not an indefinite relationship as the sample count is time restricted. An overly long sample count would coincide with the start of the returning echo, degrading the systems operation. Time plots of the pulse compression technique are shown in Figure 7. This depicts four time plots; (a) the received echo with two back-wall echoes present; (b) the excitation pulse, a two hundred sample pulse is used; (c) the cross-correlated signals and finally (d) the root squared cross-correlated signals. It is interesting to note that although the pulse compression technique increases our SNR the time resolution of the system has not improved a great deal. The pulses are smearing and ringing in time, a characteristic that persists through the pulse compression filter. A solution would be to use more complex excitation wave-forms, linear modulated FM signals are commonly used and theoretically Golay code pairs eliminate any side lobes [8]. Such techniques are currently under investigation.

Conclusions: A low power wireless coupled ultrasonic sensor has been developed for structural health monitoring applications. The sensor integrates all the necessary components to sustain a fully operable sensor network system. The device includes a fully functional AWG capable of ultrasonic transducer drive, operating at speeds up to 50 MHz. An on-board DSP is also included, the main function of which is implementation of the data processing algorithms. Three SNR enhancement algorithms have been studied, averaging, band pass filtering and pulse compression. The computational ability and versatility of the system make it suited to a wide range of applications ranging from time of flight thickness gauging to large area mapping using arrays. Moreover the low power consumption and reduced physical size of the device allow operation through out industrial environments. The feasibility of the health monitoring system and SNR algorithms have been demonstrated and presented.

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