

Real-time adaptive filtering of dental drill noise using a digital signal processor

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Abstract –The application of noise reduction methods requires the integration of acoustics engineering and digital signal processing, which is well served by a mechatronic approach as described in this paper. The Normalised Least Mean Square (NLMS) algorithm is implemented on the Texas Instruments TMS320C6713 DSK Digital Signal Processor (DSP) as an adaptive digital filter for dental drill noise. Blocks within the Matlab/Simulink Signal Processing Blockset and the Embedded Target for TI C6000 DSP family are used. A working model of the algorithm is then transferred to the Code Composer Studio (CCS), where the desired code can be linked and transferred to the target DSP. The experimental rig comprises a noise reference microphone, a microphone for the desired signal, the DSK and loudspeakers. Different load situations of the dental drill are considered as the noise characteristics change when the drill load changes. The result is that annoying drill noise peaks, which occur in a frequency range from 1.5 kHz to 10 kHz, are filtered out adaptively by the DSP. Additionally a schematic design for its implementation in a dentist's surgery will also be presented.

I. INTRODUCTION

Digital Signal Processing and Digital Signal Processors (DSP) represent a key technology in the field of Mechatronics. DSPs are used in a wide range of applications, such as in communications, controls and speech processing [1]. The increasing processing power of DSPs enables signal processing of higher frequency signals. In particular Real-Time Processing is becoming more applicable to a wide range of problems [2], and research is now addressing solutions that can only be obtained because of the increased processing power of DSPs. In dentistry, noise generated by air turbine dental drills is most prominent in a frequency range from 2 kHz to 10 kHz. Processing these frequencies with complex algorithms requires high DSP processing power.

It is well known that the main source of fear associated with dental treatment is dental drill noise [3]. There is also a risk of the dental drill noise damaging the dentist's hearing and it has been reported that dentists can suffer a hearing loss in a particular narrow frequency band [5], and also that noise control is necessary to ensure clear communication between the dentist and the patient [7]. In [8] it has been shown that the main annoying peaks can be removed with narrow band adaptive filters generated using the Matlab Signal Processing tool in a computer environment. The adaptive filter was able to reduce the main peaks in a narrow band by up to 20 dB.

This paper highlights the characteristics of different dental drills and their noise characteristics. The DSP type, the adaptive filtering algorithm and the results of its implementation on the DSP are presented.

II. DENTAL DRILL NOISE

The cutaway of the air turbine dental drill is shown in figure 1. The burr shaft is the rotating part of the drill and therefore the source of the drill noise. The speed of the drill is variable depending on the air pressure supply and can go from 200 000 rpm to a maximum speed of 400 000 rpm. The air turbine driven drill produces the unpleasant noise peak due to its high rotational speed. Figure 2 shows the FFT plot of a free running air turbine drill with an approximate speed of 250000 rpm, i.e. it is not running at maximum speed, and hence the peak frequency occurs at approximately 4.2 kHz.

III. TI TMS320C6713 DSK

The TI TMS320C6713 DSK starter kit is a low cost stand-alone DSP development platform that can be used to develop applications for the TMS320C67xx DSP family [13]. It includes the C6713 floating-point digital signal processor (DSP) and a 32 bit stereo codec (AIC23) for input and output (Fig. 3.). The AIC23 codec uses a sigma-delta technology that provides Analogue to Digital conversion (ADC) and Digital to Analogue conversion (DAC) and has got variable sampling rates from 8 kHz to 96 kHz [1]. It includes 16 MB synchronous dynamic random access memory (SDRAM) and 256 kB of flash memory. Furthermore it includes two inputs (LINE IN, MIC IN) and two output ports (LINE OUT, HEADPHONE). The DSK operates at a frequency of 225 MHz and has got a single power supply of 5 V.

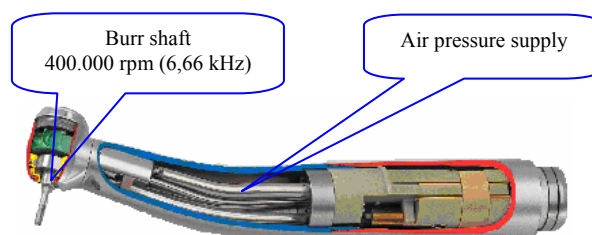


Fig. 1. Air turbine dental drill (maximum speeds)

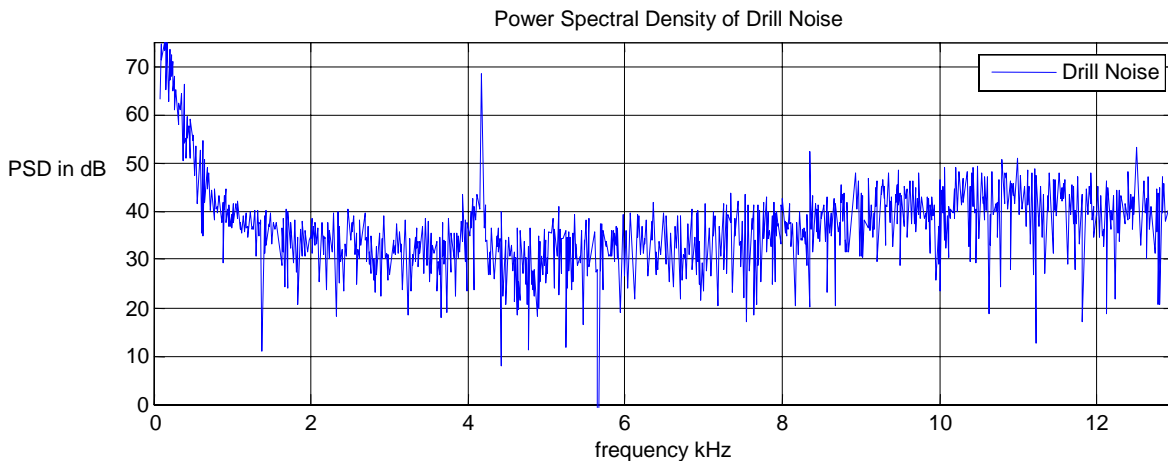


Fig. 2. PSD plot of the air turbine dental drill noise

The architecture of the TMS320C6713 is well suited to numerically intensive algorithms. The internal memory is structured so that a total of eight instructions can be called every cycle. For example with a clock rate of 225 MHz, the C6713 is capable of calling eight 32-bit instructions every $1/(225 \text{ MHz})$ or 4.44 ns [1]. The C6713 (C671, C6711) belongs to the family of floating-point processors, whereas the C62xx and C64xx belong to the family of the C6x fixed-point processors. The C6713 is also capable of fixed-point processing and it enables the developer to update the algorithm to a fixed-point calculation after running under floating point on the same DSK.

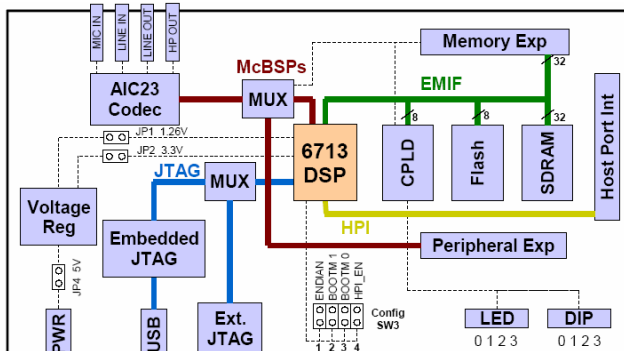
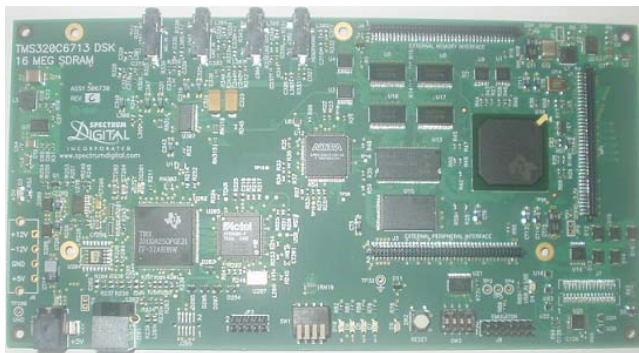


Fig. 3. TMS320C6723 based DSK board: a) board b) diagram

IV. IMPLEMENTATION METHOD

A. Procedure

The implementation of the algorithm is eased significantly using Matlab Simulink and the Embedded Target for a TI C6000 DSP. The software tools not only shorten the time for the implementation but also generate the necessary C and Assembler code, i.e. no professional knowledge of the programming languages is necessary unless there is a need for further code optimisation. Figure 4 shows the implementation procedure from the Simulink model to the target DSP.

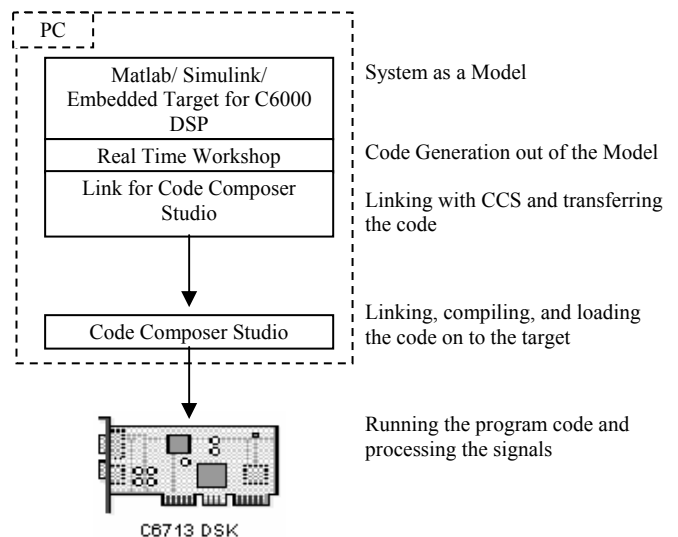


Fig. 4. Flow diagram of implementation on the target

The model of the intended filtering algorithm is generated using the Matlab Simulink Signal Processing Blockset. The next step is to replace the input blocks and output blocks of the adaptive filter algorithm using the blocks from the Embedded Target for the TI C6000 DSP, which links automatically to the input and output slots of

the DSK. Figure 5 shows the input and output blocks which are attached to the Normalised LMS algorithm (NLMS) block. They represent two microphone inputs and one signal output on the TI TMS320C6713 DSP. Here the reference signal is connected to the input port and the desired signal is connected to the desired port.

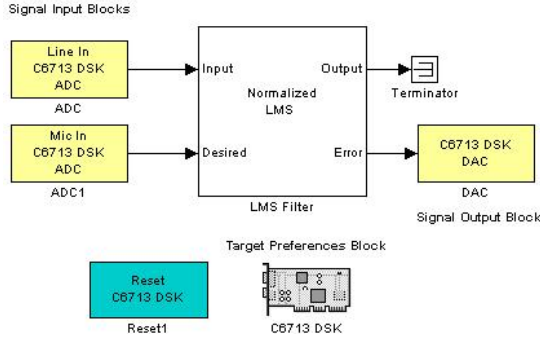


Fig. 5. Embedded Target C6000 input and output blocks in Matlab/Simulink

The Real Time Workshop tool in Matlab Simulink is used to generate the code from this block. The code generated is then linked and transferred to the Code Composer Studio (CCS) by the Link for Code Composer Studio. The code is then linked, compiled and loaded to the target by the CCS and finally the DSK processes the signal with the algorithm.

B. Adaptive Digital Filtering and the Normalised Least Mean Square Algorithm (NLMS)

The reason for applying an adaptive digital filter is in order to cope with the frequency of the drill noise changing with time. An adaptive filter has two distinct parts, a digital filter with adjustable coefficients and an adaptive algorithm, which is used to adjust the coefficients of the digital filter [12]. Figure 6 shows the block diagram of an adaptive filter as a noise canceller. The aim of the system is to find the optimal signal (noise) estimate, $n_e[k]$.

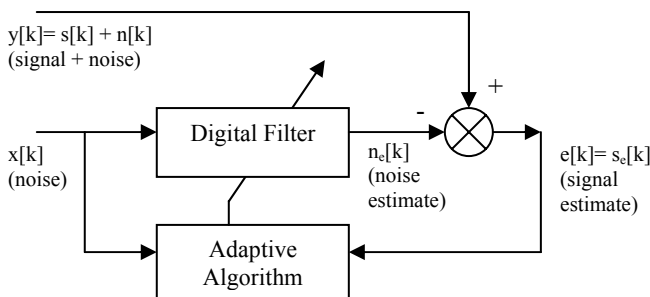


Fig. 6. Block Diagram of an adaptive digital filter as a noise canceller

The adaptive algorithm applied is the NLMS, which is a modified version of the LMS algorithm. The adaptive filter block in Matlab implements a Finite Impulse Response (FIR) filter using the stochastic gradient algorithm known as the Least Mean Square (LMS) algorithm [11]. The LMS algorithm is widely used in adaptive filtering applications,

due to its effectiveness in terms of computation and storage requirements [12]. Equation 1 shows the LMS used to upgrade the adjustable coefficients, $w[k]$, of the FIR filter.

$$w[k + 1] = w[k] + \mu e[k]x[k] \quad (1)$$

$w[k]$: Filter coefficient

$e[k]$: Error signal

$x[k]$: Reference signal

μ : Step size factor (convergence factor)

The step size factor μ governs the stability and the convergence of the adaptation process. Therefore by modifying this value it is aimed to optimise the speed of the convergence while maintaining the independence of the signal power. The NLMS algorithm is then written as follows [9]:

$$w[k + 1] = w[k] + \mu[k]e[k]x[k] \quad (2)$$

$$\mu[k] = \frac{\alpha}{N\hat{P}_x[k]} \quad (3)$$

$\mu[k]$: Time varying step size factor normalised by N

N: Filter length

α : Constant ($0 < \alpha < 2$)

$\hat{P}_x[k]$: Power estimate of $x[k]$ at time k.

The NLMS algorithm is already provided in the Simulink Signal Processing Blockset therefore it is not necessary to program the code (see Fig. 5.), which eases and shortens the implementation of the intended model.

B. Experimental implementation on the TMS320C6713 DSK

Figure 6 shows the experimental setup for processing/filtering the drill noise. The signal processing is performed on air turbine dental drill noise. Two standard microphones are used to pick up the reference signal and the desired signal and let to the input ports of the DSK. The signals are processed through the DSP. After processing the signal is let through the output port to a headphone or other interface such as a minidisk recorder, to record the filtered signal and analyse it as a wave file on the PC. The distance between the microphone for the desired sound and the microphone for the drill noise has to be at least 45 cm according the test standards [14].

V. RESULTS

Several tests have been conducted to find out the optimal number of the filter length N ($N=32$) and the optimal value for the step size factor μ ($\mu = 0.9$). For obtaining these values the steps shown in figure 4 were done recursively. Figure 7 shows the power spectral density plot of the original signal compared with the filtered signal. It can be seen that the peak at approximately 4.2 kHz is

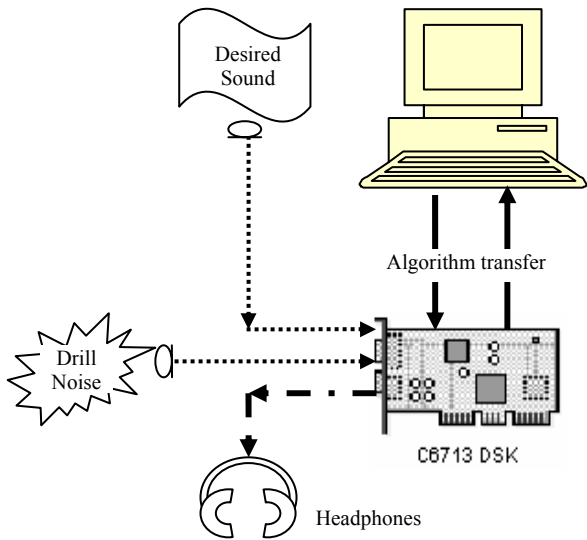


Fig. 6. Schematic diagram of the experimental setup

reduced by 20 dB. This reduction removes the highly unpleasant noise peak produced by the dental drill. The background noise is still existent but does not have the same uncomfortable effect as the main drill noise peak, and a person who wears for example headphones will hear just the background noise.

The results of the implementation of the adaptive algorithm on the DSP can be used in a headphone system

as is already done in some analogue applications. Additionally this headphone system can be implemented in a dental chair where the patient would be able to use headphones to avoid hearing the unpleasant noise.

VI. CONCLUSIONS

The characteristics of the air turbine dental drill noise were discussed and its related origin of the noise was shown. The implementation method was explained showing the procedures from the Matlab Simulink environment to the final target Digital Signal Processor (DSP). The Normalised Least Mean Square (NLMS) Algorithm was discussed for the usage as an adaptive digital filter algorithm and the experimental implementation on the Texas Instruments (TI) TMS320C6713 DSK was shown.

The results of the experiments show a significant noise reduction of the noise peak by 20 dB. However there is still a potential to optimize the combination of the digital filter length N and the step size factor μ . But this paper suggests that a headphone system can be implemented in a dental surgery in to reduce the dental drill noise effectively and ensuring improved communication between the dentist and the patient.

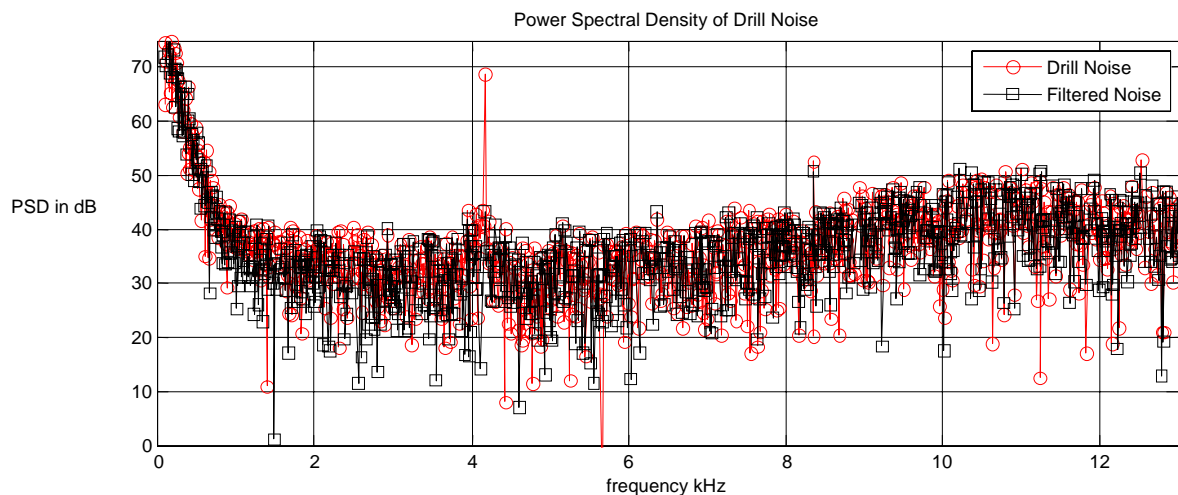


Fig. 7. PSD plot of the original signal and filtered signal

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