# ACTIVE NOISE CONTROL ON HIGH FREQUENCY NARROW BAND DENTAL DRILL NOISE: PRELIMINARY RESULTS

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## Abstract

Dental drills produce a characteristic noise that is uncomfortable for patients and is also known to be harmful to dentists under prolonged exposure. It is therefore desirable to protect the patient and dentist whilst allowing two-way communication. A solution is to use a combination of the three main noise cancellation methods, namely, Passive Noise Control, Adaptive Filtering and Active Noise Control. Dental drill noise occurs at very high frequency ranges in relation to conventional ANC, typically 2kHz to 6kHz and it has a narrow band characteristic due to the direct relation of the noise to the rotational speed of the bearing. This paper presents a design of an experimental rig where first applications of ANC on dental drill noise are executed using the standard filtered reference Least Mean Square (FXLMS) algorithm. The secondary path is kept as simple as possible, due to the high frequency range of interest, and hence is chosen as the space between headphone loudspeaker and error microphone placed in the ear (input of the headphone loudspeaker and the output of the error microphone). A standard headphone loudspeaker is used for the control source and the microphone inside of an "Ear and Cheek Simulator Type 43AG" is used as the error microphone. The secondary path transfer function is obtained and preliminary results of the application of ANC are discussed.

## 1. INTRODUCTION

Current applications of Active Noise Control (ANC) are generally in lower frequency ranges as Passive Noise Control (PNC) is very effective for higher frequency ranges. However, due to design restrictions the effectiveness of PNC cannot be exploited maximally and is therefore restricted to a certain amount of noise reduction of dental drill noise [1]. Therefore it is proposed to use a combination of ANC and PNC for the best possible noise control solution for a dental surgery. Hence this involves the application of ANC in frequency ranges above the usual application frequency range due to the dental drill noise characteristic [2]. Another reason for the current applications of ANC in low frequencies is the restriction due the fundamental physical rules of sound wave superposition [3,4]. Here it is said that the higher the frequency, the smaller the possible "zone of quiet" is.  Therefore, if dental drill noise is considered this restriction plays a huge role because the frequency range of interest is typically between 2 kHz to 6 kHz. Figure 1 shows an electromotor driven dental drill and an air turbine driven drill with the power spectral density (PSD) plots and their related noise peaks caused by the bearings of the drills. It can be seen that the electromotor driven drill has got two main noise peaks due to two rotating parts, namely intermediate shaft and the bur shaft. The air turbine drill has got only one main peak because it has only one rotating shaft, which is the bur shaft.

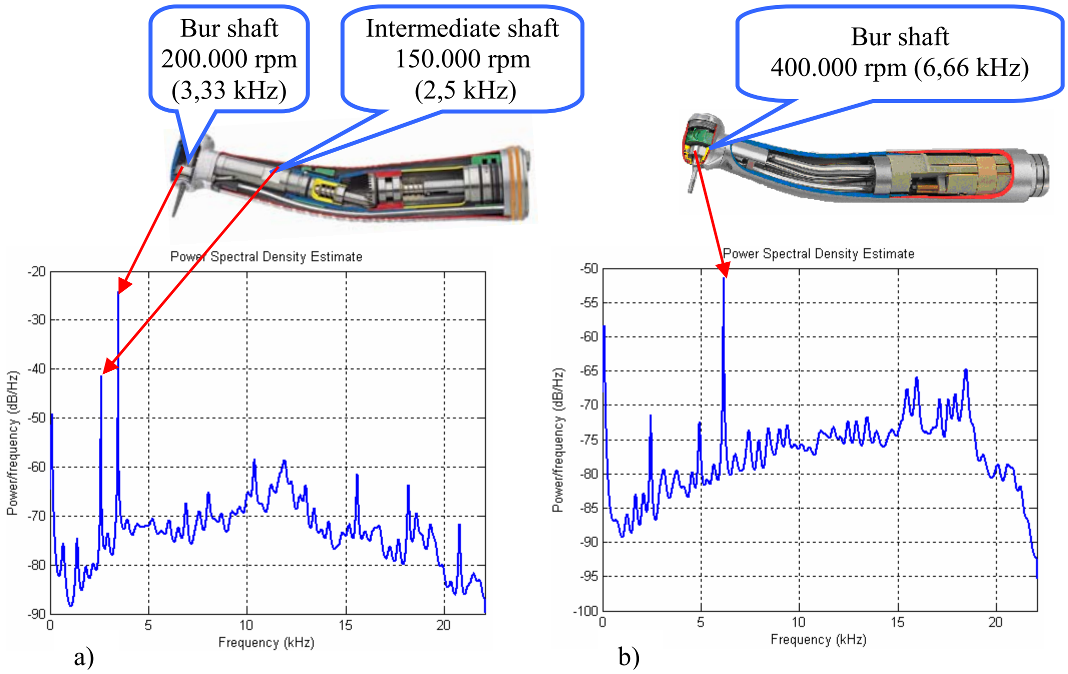


Figure 1. Power Spectral Density of the a) electromotor driven dental drill b) air turbine driven drill

As discussed in [2] due to the characteristics of dental drill noise it is proposed that ANC can be applied in higher frequencies in the special case of dental drill noise. This paper discusses the design of an experimental rig where first ANC experiments are run to find out the applicability of ANC for dental drill noise. To summarize the assumptions in [2] which are allowing high frequency ANC are i) Dental drill noise is periodic hence predictable, ii) the frequency of interest is narrow band iii) the reference signal is accessible therefore feed forward control is applicable iv) the area of interest is small (ear channel). Although the reference signal is accessible and the noise source is periodical a system such as described by Chaplin [8] cannot be used because the reference signal in this case is picked up by a microphone instead of using a synchronous pulse generator.

## 2. FILTERED REFERENCE LEAST MEAN SQUARE (FXLMS) ALGORITHM

The electro-acoustic transfer function from the loudspeaker input to the error microphone output requires a modification of the Least Mean Square (LMS) algorithm to compensate the delay caused by the transfer function. Figure 2 and equation (1) show the block diagram and the equation of the standard LMS coefficient update algorithm applied for noise cancellation (no cancellation path transfer function) purposes, respectively.

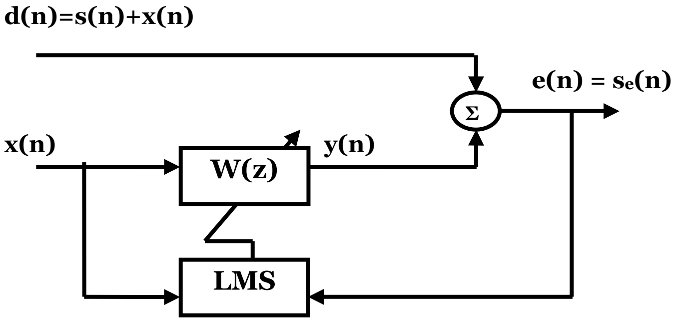


Figure 2. Block diagram of an adaptive digital filter as a noise canceller

$$w(n+1) = w(n) + 2\mu e(n)x(n) \tag{1}$$

Where w(n) are the filter coefficients, e(n) error signal or the signal estimate $s_e(n)$, x(n) is the reference signal and μ is the step size factor (convergence factor), which governs the speed of the convergence of the coefficients. The filter with the coefficients w(n) is generally a finite impulse response (FIR) filter and can be described as in equation 2.

$$y(n) = \sum_{i=0}^{N-1} w_i(n)x(n) \tag{2}$$

In ANC, the path from the primary source to the error microphone is acoustical and hence must be compensated by an additional filter in the updating algorithm [5]. Therefore, the LMS algorithm is modified by an additional filter, which is filtering the reference signal before it enters the updating LMS algorithm. This filtering of the reference signal gives also the name for the modified algorithm for ANC applications, which is the FXLMS algorithm. It was first derived by Widrow and is discussed in [6] under adaptive control systems. Figure 3 shows the block diagram of the FXLMS implementation and equation 3 shows the modified version of the LMS algorithm.

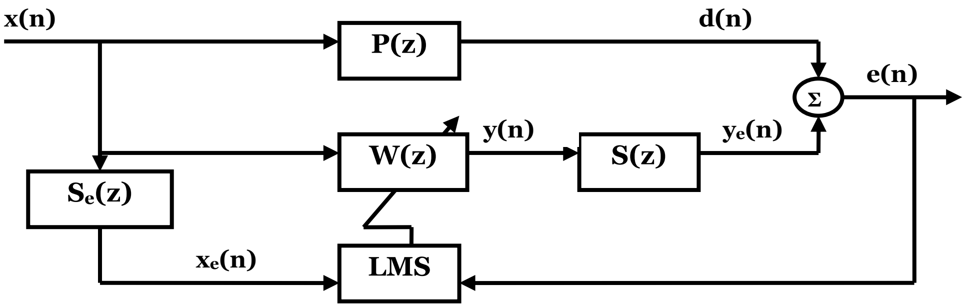


Figure 3. Block diagram of an FXLMS algorithm applied in ANC

$$w(n+1) = w(n) + 2\mu e(n)x_e(n) \tag{3}$$

As it can be seen from equation 3 that the only difference to the LMS algorithm is instead of entering the reference signal directly to the updating algorithm, it is filtered before by an FIR filter, $S_e(z)$, described by equation 4, where $s_i(n)$ are the coefficients of the FIR filter.

$$x_e(n) = \sum_{i=0}^{N-1} s_i(n)x(n) \tag{4}$$

## 3. OFF-LINE CANCELLATION TRANSFER FUNTION MODELLING

The electro-acoustic transfer function from the loudspeaker input to the error microphone output (Fig. 4) denoted as S(z) in figure 3 must be obtained and implemented as a filter (here denoted as $S_e(z)$) in the algorithm to compensate the introduced delays. The offline modelling method proposed in [5] was considered.

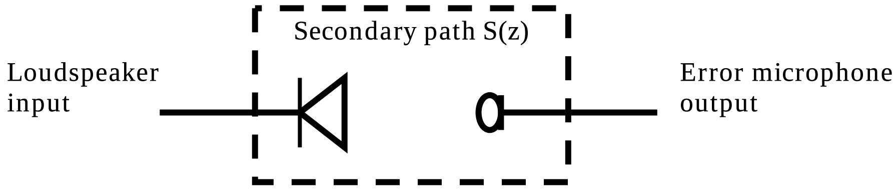


Figure 4. Electro-acoustic transfer function S(z)

The off-line modelling technique includes the following steps and as it is shown in figure 5:

1. Generate sampled white noise signal x(n)
2. Obtain desired signal d(n) from the error sensor
3. Apply adaptive filter algorithm as follows
    a. Compute adaptive filter output using an FIR filter (Equation 2)
    b. Compute error signal

$$e(n) = d(n) - y(n) \tag{5}$$

    c. Update coefficients using the LMS algorithm (Equation 1)
4. Go to 1 for next iteration until adaptive filter $S_e(z)$ converges until the power of e(n) is minimised

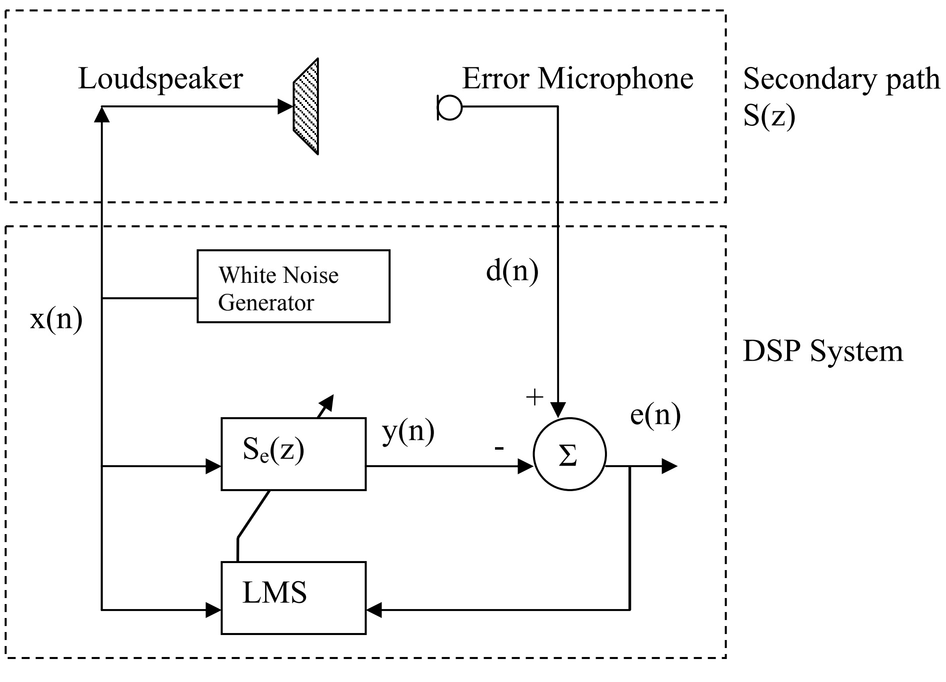


Figure 5. Block diagram of the off-line secondary-path modeling technique (adapted from [5])

## 4. EXPERIMETNAL SETUP

The TI TMS320C6713 DSK was used as the DSP using the DUAL3006 [9] audio daughter card for having additional channels. As described in figure 5 an internal white noise is generated by the DSP that is the reference signal for the modelling system. Figure 6 shows the power spectral density plot of the white noise that has a flat power distribution over the frequency range of interest.

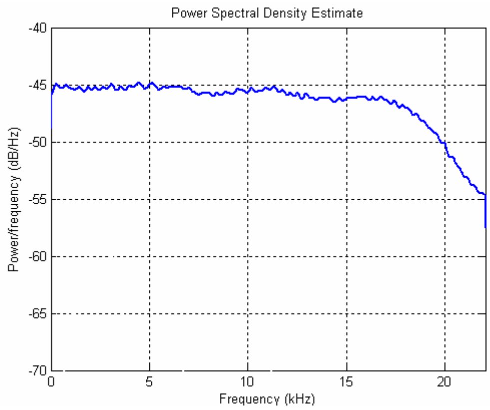


Figure 6. Power Spectral Density of white noise

The G.R.A.S. Ear and Cheek Simulator Type 45AG is reproducing the acoustic properties of the ear of an average human head [7]. It has a build in ½ inch pressure microphone type 40AG and a ¼ preamplifier type 26AC. The microphone is also powered by a power module type 12AD (all from G.R.A.S. Sound and Vibration). The loudspeaker is provided inside a circum-aural, open Sennheiser headphone (HD 555). The secondary path explained in the previous section is represented by the microphone and loudspeaker explained here. Figure 7 shows the experimental setup and the components.

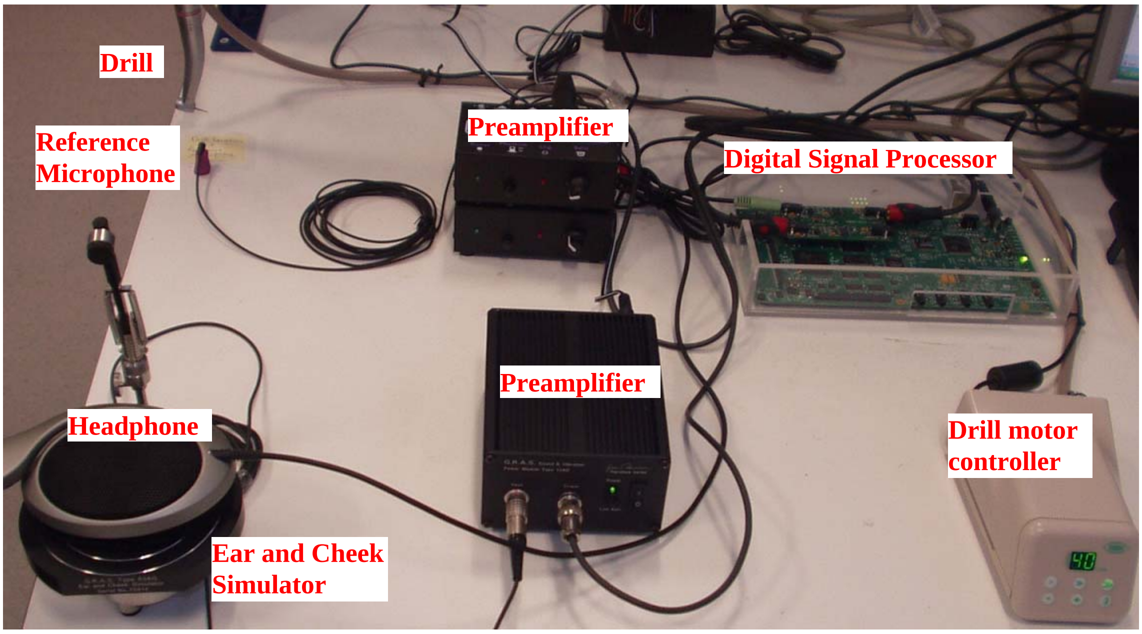


Figure 7. Experimental Setup

## 5. PRELIMINARY RESULTS

Figure 8 shows the convergence of the error signal e(n) and the filter output y(n) of the off-line secondary path modelling technique. It can be seen that the control signal y(n) is converging to the value of the microphone output signal d(n) so that the error signal is converging to a minimum value according to equation 5.

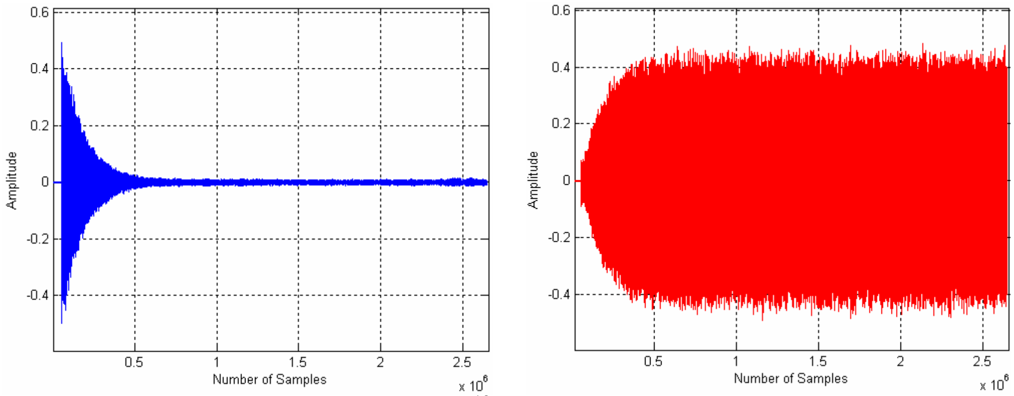


Figure 8. Filter length N = 512 and $\mu = 10^{-12}$ a) Error e(n) convergence b) Filter output y(n)

Figure 9 shows the frequency responses of the microphone output d(n) and the filter output y(n), which are expected to be the same after conversion. Figure 9 a) shows the frequency responses before convergence and b) after convergence of the control signal to the optimum value so that the difference between the filter output and the microphone output is ideally zero.

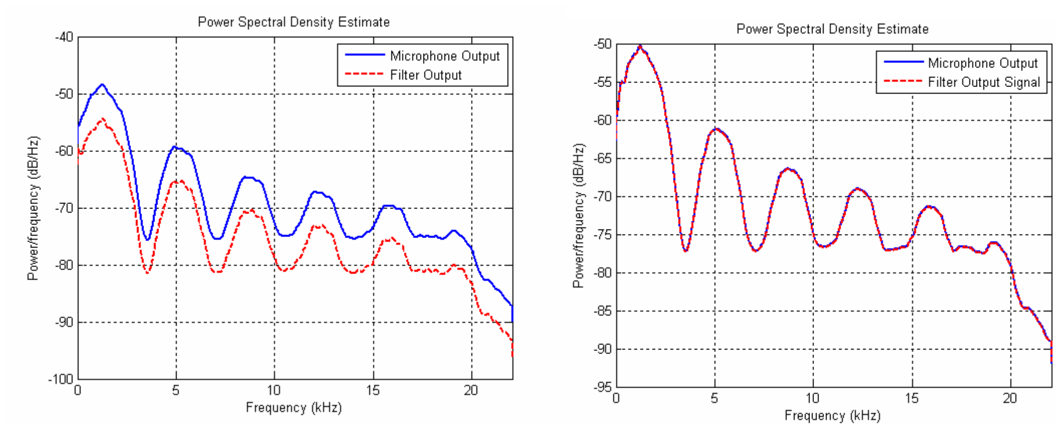


Figure 9. Frequency response of the microphone output d(n) and filter output y(n) a) before conversion b) after conversion

As it can be seen from the figures 8 and 9 the algorithm shows the expected convergence for the given filter length and convergence rate. Faster convergence times can be achieved with higher convergence rates, but they also introduce distortions into the signal.

## 5. CONCLUSIONS AND FURTHER WORK

An experimental rig for ANC experiments was constructed and described for implementing ANC for dental drill noise. The off-line cancellation path transfer function modelling technique was used to obtain the cancellation path transfer function and optimum values for the related parameters such as the filter length and convergence rate. Optimum values for the filter length and convergence rate were chosen as 512 and of $10^{-12}$, respectively. Although we

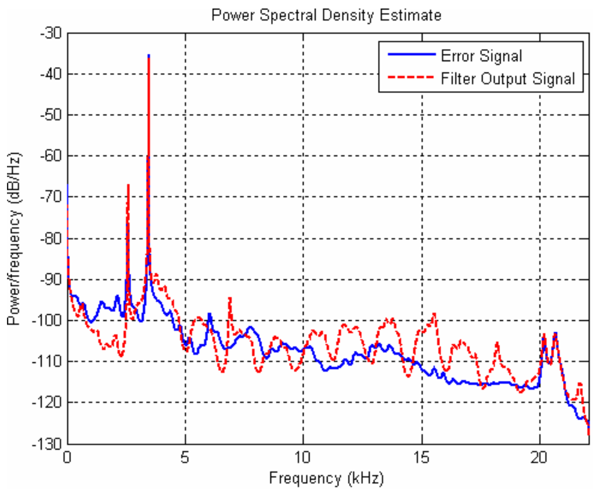


Figure 10. Error microphone signal and filter output signal of the

obtain good results in terms of cancellation path transfer function, the first ANC results do not show satisfactory reductions of the peaks. Figure 10 shows the error microphone signal and the filter output signal generated by the adaptive algorithm. The peaks show very good match but there is almost no peak reduction of the error signal. This may be due to several reasons such as i) the off-line modelling technique does not consider any changes in the cancellation path transfer function that are sensitive to changes in the room temperature and ii) small changes in the positioning of the loudspeaker to the error microphone to each other, iii) due to the fixed high sampling frequency of 48 kHz of the audio daughter card, the chosen filter length might not be the optimum and hence can introduce severe delays. Therefore work must be done in optimizing the sampling frequency and the filter length and work towards the implementation of an on-line cancellation path transfer function modelling technique to overcome the problems with the off-line modelling technique.

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