Final Year Project(BEng)

2016

Digital Active Noise Control System

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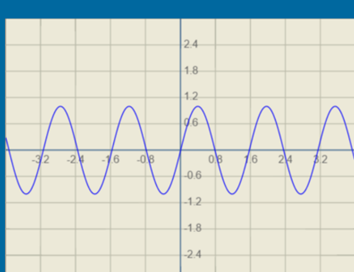
Location: Anglesea A0.07

Noise Cancelling

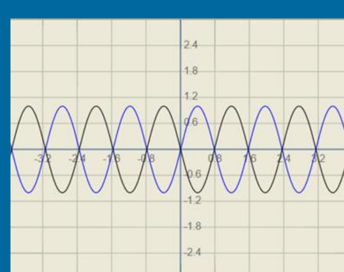
In many of todays industries, machinery, equipment and vehicles produce high volume sounds. These noises can have serious safety implications. Engines from airplanes, rotors from helicopters and diesel generators on ships require maintenance and servicing. This requires technicians and engineers to come in close proximity with the potential to damage hearing and hinder adequate communication. Ear defenders are commonly used but active noise cancellation can greatly reduce the level of unwanted sound from harming the operator.

This project uses Ear Defenders combined with an ANC system provides duel protection from unwanted noise. Using a DsPIC microprocessor the environmental noise is phase shifted by 180º to create a sound of opposite phase but equal frequency. This is then fed to speakers in the ear defenders.

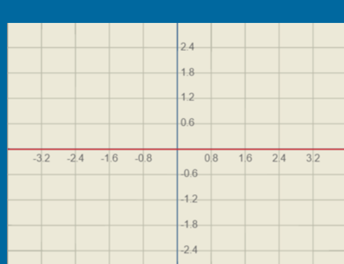
The system uses two microphones; one reference mic, to pick up the unwanted noise and the other to provide feed back from the speakers. This feed back signal should ultimately be zero as this is what the user will hear.



Unwanted noise



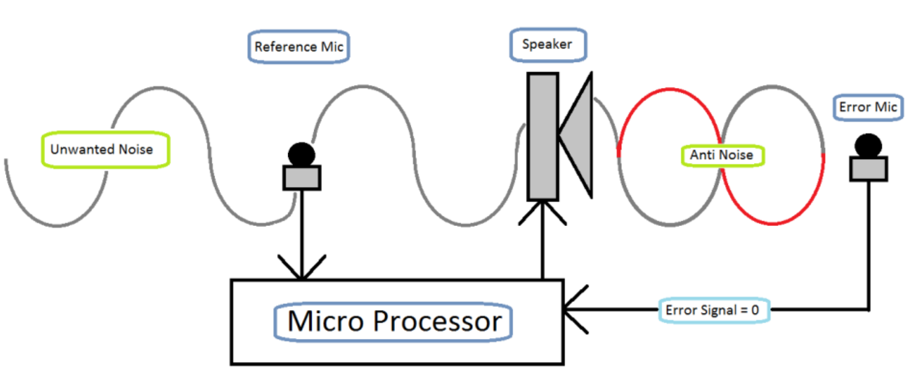
Anti Noise & unwanted

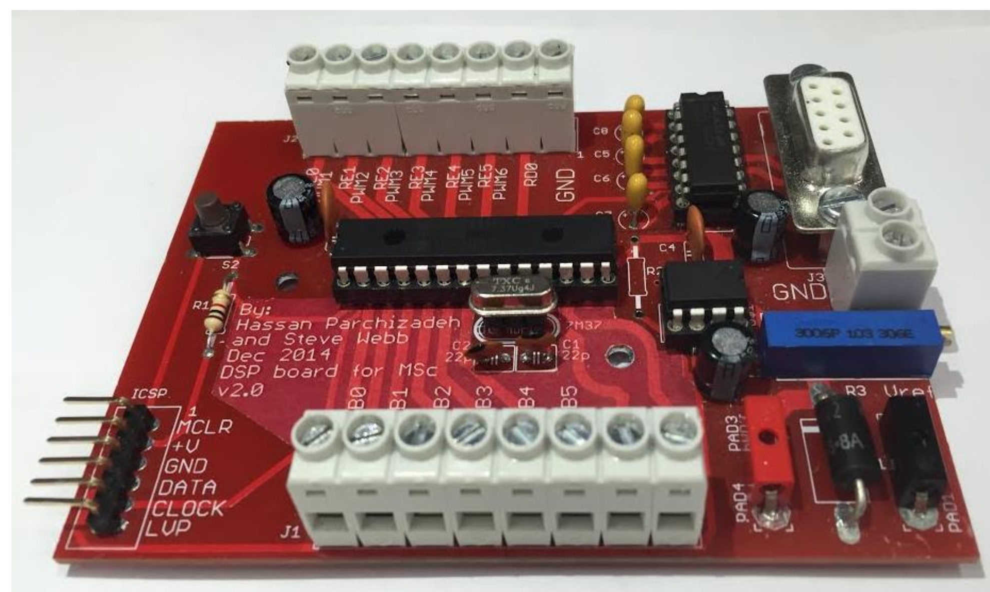


Zero output

This zero output signal is used to continually update the digital processor to ensure the correct sound is being produced.

ANC System Diagram





Tinnitus and hearing loss are the top two attributed disabilities which US veterans suffer from. The ear defenders used in this project are military grade for use on helicopters. Upgrading the sound protection can help reduce the number of personnel affected by loud noise.

System Components

This project uses the **DsPic 30F4012 201-SP** which is used to process the sound waves. It uses an **Adaptive Least Mean Square Adaptive Notch Filter** to produce an anti-phase signal. It operates at 100Hz with a sampling frequency of 4000Hz.

The **LMS** generates two weights, updated by the error mic. These are as follows

W0 =-W0 - 0.000001 * e * X2;

W1 =-W1 - 0.000001 * e * X1;

These weights, W0 and W1, are updated every sample by 'e' (error mic), 'X2' (reference sample) and 'X1' (reference sample with 90º delay).

The processor is programmed in C and uses 2 inbuilt Analogue to Digital converters to process the input signals. The processor then outputs the anti phase signal through a Digital to analogue converter.

The output is fed to headphone speakers where the user should hear a much attenuated sound.

School of Engineering

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