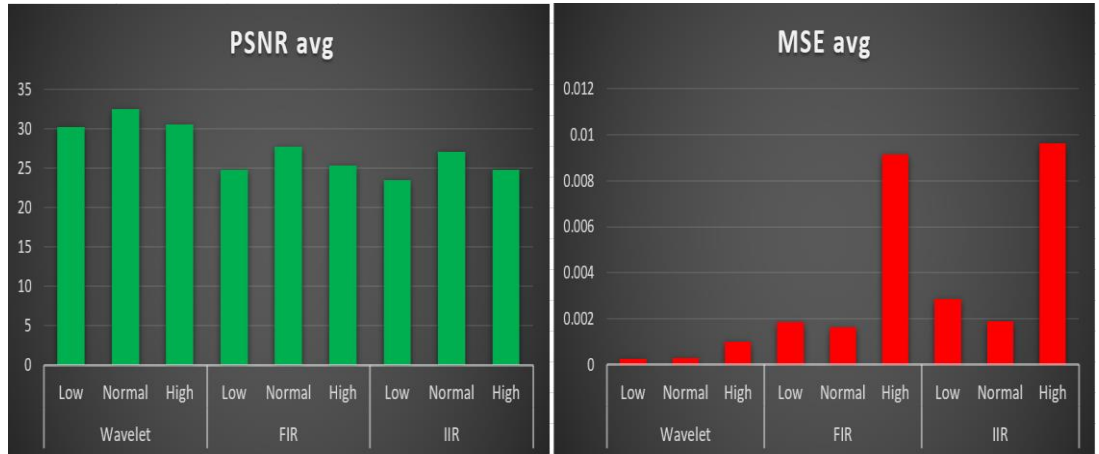


<b>Project Title</b>	<b>The 9 Sines Audio Denoising Radio</b>		
<b>Track</b>	Engineering & Applied Sciences		
<b>Supervisor</b>	Dr. Samah El-Shafiey	<b>Mentor Name</b>	Dr. Samah El-Shafiey
<b>Team Name</b>	The 9 Sines		
<b>Team Members</b>	Yousef Khaled	Khaled Hamed	Abdelrahman Ahmed
	Eslam Fathy	Abdelrahman Hatem	Text.
<b>Problem Summary</b>	<p>With the advancement of technology, the transfer of information, photos, and videos has become easier. However, during transmission, signals are exposed to various types of noise that can degrade their quality. Noise in signal processing refers to unwanted modifications during capture, storage, transmission, processing, or conversion, which may result in the loss or distortion of details. This noise can originate from both internal sources (e.g., amplifiers, transmitters, receivers) and external sources (e.g., lightning, cosmic rays, atmospheric turbulence). Effective noise removal techniques are essential to enhance the quality of audio signals, which can be achieved through noise reduction methods to prevent noise or audio filtering techniques to remove noise after it has occurred. The goal is to restore the original audio signal without any distortion.</p>		
<b>Methodology</b>	<p>Wavelet Transform (WT) is an effective method for audio denoising, particularly using the Threshold algorithm, which compresses noise in digital signals. WT consists of Continuous Wavelet Transform (CWT) and Discrete Wavelet Transform (DWT). CWT analyzes data in both time and frequency domains, using a scalable window to move across the signal. DWT is more precise, using discrete scales and translations based on powers of 2.</p> <p>Audio denoising combines Partial Differential Equations (PDEs) with wavelet thresholding. The heat equation smoothing signal while soft thresholding modifies wavelet coefficients. The process involves adding Gaussian noise to the original signal, computing Signal-to-Noise Ratio (SNR) and Root Mean Square Error (RMSE), and applying the wavelet transform to decompose the signal. Threshold values are calculated, and wavelet coefficients are adjusted using soft or hard thresholding before reconstructing the signal.</p> <p>Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters are digital filters used for signal processing. FIR filters have a finite duration impulse response, while IIR filters have an infinite duration. Fast Fourier Transform (FFT) efficiently computes the Discrete Fourier Transform (DFT), reducing complexity and speeding up spectrum analysis. Denoising with those filters involves convolving input signal with the filter's impulse response, improving SNR.</p>		
<b>Achievements and Skills Gained</b>	<ol style="list-style-type: none"> <li>1. Teamwork</li> <li>2. Leadership</li> <li>3. Time Management</li> <li>4. Problem Solving</li> <li>5. Hardware Implementation</li> <li>6. Practicing Simulation tools</li> <li>7. Writing Scientific Reports &amp; Posters</li> <li>8. Communication Skills</li> </ol>		

**\*\* Filling all fields in the forms are mandatory. Leaving empty fields may affect in reviewing/ shortlisting your project.**

**\*\* Supervisor/Mentor could be your course professor/teaching assistant/tutor or parent.**

(Cont.)

Project Title	The 9 Sines Audio Denoising Radio																																			
Main Results	<p>Figure 1 PSNR-MSE Test Results</p>  <p>The figure consists of two bar charts. The left chart, titled 'PSNR avg', shows PSNR values for Wavelet, FIR, and IIR methods at Low, Normal, and High frequencies. The right chart, titled 'MSE avg', shows MSE values for the same methods and frequencies. In both charts, Wavelet consistently shows the highest PSNR and lowest MSE, followed by FIR, and then IIR.</p> <table><tr><th>Method</th><th>Frequency</th><th>PSNR avg</th><th>MSE avg</th></tr><tr><td rowspan="3">Wavelet</td><td>Low</td><td>30</td><td>0.0005</td></tr><tr><td>Normal</td><td>32</td><td>0.0005</td></tr><tr><td>High</td><td>30</td><td>0.0010</td></tr><tr><td rowspan="3">FIR</td><td>Low</td><td>25</td><td>0.0015</td></tr><tr><td>Normal</td><td>28</td><td>0.0015</td></tr><tr><td>High</td><td>25</td><td>0.0090</td></tr><tr><td rowspan="3">IIR</td><td>Low</td><td>24</td><td>0.0025</td></tr><tr><td>Normal</td><td>27</td><td>0.0018</td></tr><tr><td>High</td><td>25</td><td>0.0095</td></tr></table>		Method	Frequency	PSNR avg	MSE avg	Wavelet	Low	30	0.0005	Normal	32	0.0005	High	30	0.0010	FIR	Low	25	0.0015	Normal	28	0.0015	High	25	0.0090	IIR	Low	24	0.0025	Normal	27	0.0018	High	25	0.0095
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Discussion and Conclusion	<p>After doing the experiment with the data set and analyzing the results in the main results, we concluded that:</p> <ol style="list-style-type: none"><li>1.The Wavelet method is obviously the best method.</li><li>2.The difference between FIR and IIR is not as much, but the FIR is better.</li><li>3.Highest performance is achieved at normal frequency that ranges from (40-60) kHz.</li></ol> <p>So, we recommend to use the Wavelet frequency with normal-frequency audio files to get the best performance</p>																																			
References	<ul style="list-style-type: none"><li>• Kumar, Nishant. (2013). Optimal Design of FIR and IIR Filters using some Evolutionary Algorithms</li><li>• J. Jebastine, B. S. Rani (2012), “Design and implementation of noise free Audio speech signal using fast block least Mean square algorithm”, Signal &amp; Image Processing: An International Journal</li><li>• <a href="https://github.com/youefkh05/The_9Sines">https://github.com/youefkh05/The_9Sines</a></li></ul>																																			
Future Work and Suggestions	<p>If we have more time, we will test some other IIR filter techniques that may be more accurate. We have already tested one design technique, the Bilinear Transform.</p> <p>We also have two others: Impulse Invariance and Step Invariance.</p> <p>Additionally, we could test another algorithm, the Adaptive LMS</p>																																			
Group Photo	