Project Notes:

Project Title: Mobile App to Allow for Multi Source Noise Cancellation in Public Spaces
Name: **Vishal Balagani**

Note Well: There are NO SHORT-cuts to reading journal articles and taking notes from them. Comprehension is paramount. You will most likely need to read it several times, so set aside enough time in your schedule.

Contents:

Knowledge Gaps:

This list provides a brief overview of the major knowledge gaps for this project, how they were resolved and where to find the information.

Knowledge Gap	Resolved By	Information is located	Date resolved
IoT Networks	Article 6	Article #6 Notes: Aerosense: A Self-Sustainable and Long-Range Bluetooth Wireless Sensor Node for Aerodynamic and Aeroacoustic Monitoring on Wind Turbines	9/25/23
What is DSP	Article 8	Article #8 Notes: Ciaramella: A Synchronous Data Flow Programming Language For Audio DSP	10/12/23
Mobile App Frameworks	Article 13	Article #13 Notes: React Native vs Flutter, cross-platform mobile application frameworks	12/13/23
Doppler Shift	Article 5	Article #5 Notes: Dual-microphone active noise cancellation paved with Doppler assimilation for TADS	9/24/23
Websocket	Article 12	Article #12 Notes: Communicating and Displaying Real-Time Data with WebSocket	12/13/23

Literature Search Parameters:

These searches were performed between 8/17/2023 and 12/15/2023.
List of keywords and databases used during this project.

Database/search engine	**Keywords**	**Summary of search**
Science.org	Generative AI ChatGPT	How has ChatGPT affected society
Science.org	Artificial Intelligence	How can we better train AI models
IEEEXplore	Assistive Glove	Options for assistive gloves for the visually impaired
Scopus	Active Noise Cancellation	Research on different setups with Active Noise Cancellation
ScienceDirect	Active Noise Cancellation	Projects using noise cancelation in multiple use cases
Internet of Things	IEEE Xplore	Different projects utilizing IoT technology and networks
DSP	Elsevier	DSP Programming languages
Flutter	ScienceDirect	Mobile App framework comparisons
Websocket	Elsevier	Projects including Websocket for data transmission
WebRTC	Scopus	Projects including WebRTC for data transmission
IoT	ArXiv	Projects using IoT to solve problems

Tags:

Tag Name	
#GenerativeAI	
#LLM	Large Language Model
#ChatGPT	
#AI	Artificial Intelligence
#AssistiveTechnology	
#VIP	Visually Impaired Person
#AssistiveGlove	
#ANC	Active Noise Cancellation
#MultiPointANC	Multi-Source ANC
#DSP	Digital Signal Processing
#LMS	Least Mean Squares Algorithm
#IoT	Internet of Things
#BLE	Bluetooth Low Energy
#Sensors	
#ProgrammingLang	Programming Language

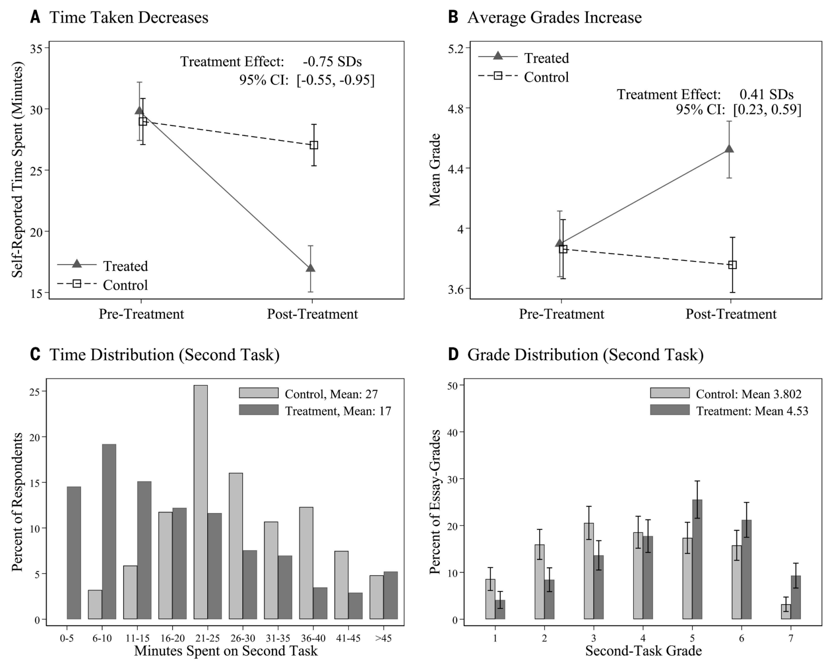
#App	
#Networks	
#Transmission	

Article # Notes:

Source Title	
Source citation (APA Format)	
Original URL	
Source type	
Keywords	
#Tags	
Summary of key points + notes (include methodology)	
Research Question/Problem/ Need	
Important Figures	
VOCAB: (w/definition)	
Cited references to follow up on	
Follow up Questions	

Article #1 Notes: Experimental evidence on the productivity effects of generative artificial intelligence

Source Title	Experimental evidence on the productivity effects of generative artificial intelligence
Source citation (APA Format)	Noy, S., & Zhang, W. (2023). Experimental evidence on the productivity effects of generative artificial intelligence. *Science*, 381(6654), 187–192. https://doi.org/10.1126/science.adh2586
Original URL	https://www.science.org/doi/10.1126/science.adh2586
Source type	Research Article
Keywords	Productivity, Distribution, ChatGPT, Enhancement
#Tags	#GenerativeAI, #LLM, #ChatGPT
Summary of key points + notes (include methodology)	An experiment was conducted in which of 453 college-educated workers, half were exposed to and given access to ChatGPT and both groups were told to complete tasks which they regularly performed. The results showed that those who were exposed 40% less time to complete their tasks and the quality of their work rose by 18%. Notes: ● Time taken for the task was decreased by 11 minutes ● Treatment groups grades were 0.45 SD above the mean ● Results were similar even when participants were incentivized to do better ● Most people submitted the direct ChatGPT output with minimal intervention ● Most people haven't used it before the experiment ● Optimism and excitement rose about AI
Research Question/Problem/ Need	When workers are exposed to generative AI systems, how will they choose to interact with them and how will it impact their productivity?

Important Figures	 Graph showing results of study and different experiments conducted. Confidence intervals were constructed and displayed.
VOCAB: (w/definition)	Attrition: the action or process of gradually reducing the strength or effectiveness of someone or something through sustained attack or pressure Robust: strong and healthy; vigorous Superficial: appearing to be true or real only until examined more closely Dissemination: the action or fact of spreading something, especially information, widely Aggregate: a whole formed by combining several (typically disparate) elements
Cited references to follow up on	T. Eloundou, S. Manning, P. Mishkin, D. Rock, GPTs are GPTs: An early look at the labor market impact potential of large language models. arXiv:2303.10130 [econ.GN] (2023). E. W. Felten, M. Raj, R. Seamans, How will language modelers like ChatGPT affect occupations and industries? SSRN [Preprint] (2023); http://dx.doi.org/10.2139/ssrn.4375268. S. Peng, E. Kalliamvakou, P. Cihon, M. Demirer, The impact of AI on developer productivity: Evidence from GitHub Copilot. arXiv:2302.06590 [cs.SE] (2023).
Follow up Questions	Will this excitement still be there a year in the future? How do people feel about other LLM's?

	If the workers were to repeat the tasks with GPT-4 how would the results change?

Article #2 Notes: Improving artificial intelligence with games

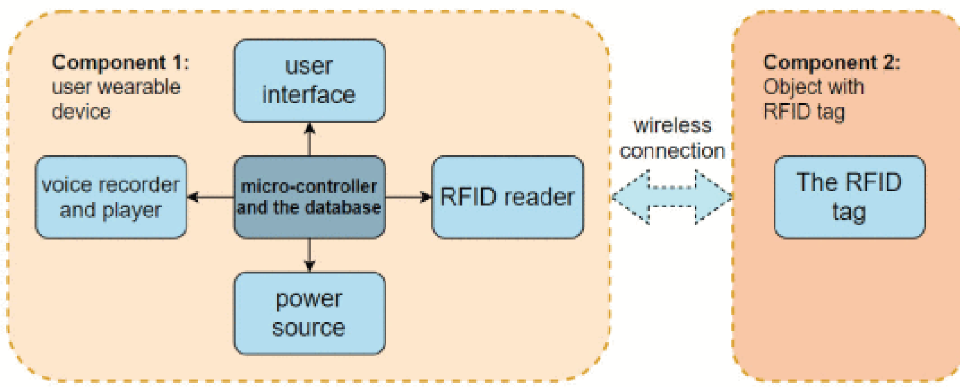
Source Title	Improving artificial intelligence with games
Source citation (APA Format)	Wurman, P. R., Stone, P., & Spranger, M. (2023). Improving artificial intelligence with games. *Science, 381*(6654), 147–148. https://doi.org/10.1126/science.adh8135
Original URL	https://www.science.org/doi/10.1126/science.adh8135
Source type	Perspective Article
Keywords	Reasoning, Optimization, Stochastic, Reinforcement Learning, State-Value Estimation
#Tags	#AI
Summary of key points + notes (include methodology)	AI systems have come a long way to beat humans in games which allowed AI research to advance with logic, predictions, and value estimation. However, in much more complex games like Forza and Minecraft AI will need to take much more complicated skills and adapt to human-like commonsense thinking. By training AI models to play simulation games like these the technology can advance more to be applied in the real world as well. Notes: ● Monte Carlo Search Trees (MCTS) was a search algorithm used to handle large state spaces ● MCTS did not account for long term value ● Strategic games need deep search and state-value estimation ● Video games require more “commonsense” ● Video games help develop real world robots
Research Question/Problem/ Need	Can we improve practical AI systems in the real world by training them on games?
Important Figures	None

VOCAB: (w/definition)	Anecdotal: not necessarily true or reliable, because based on personal accounts rather than facts or research Fidelity: faithfulness to a person Preclude: prevent from happening Stochastic: randomly determined; having a random probability distribution or pattern that may be analyzed statistically but may not be predicted precisely.
Cited references to follow up on	W. Crist et al., Ancient Egyptians at Play: Board Games Across Borders (Bloomsbury, 2016) M. Campbell, A. J. Hoane Jr., F. Hsu, Artif. Intell. 134, 57- (2002) N. Brown, T. Sandholm, Science 359, 418- (2018)
Follow up Questions	Is it possible to make games specifically to train AI models on human tasks? How can we approach applying these in game skills onto real world robots? Can we reverse this process and have AI models attempt completing games being trained on human behavior?

Article #3 Notes: An RFID-Based Assistive Glove to Help the Visually Impaired

Source Title	An RFID-Based Assistive Glove to Help the Visually Impaired
Source citation (APA Format)	Sedighi, P., Norouzi, M. H., & Delrobaei, M. (2021). An RFID-Based Assistive Glove to Help the Visually Impaired. *IEEE Transactions on Instrumentation and Measurement*, *70*, 1–9. https://doi.org/10.1109/TIM.2021.3069834
Original URL	https://ieeexplore.ieee.org/document/9389748
Source type	Journal Article
Keywords	auditory, tactile, vibrotactile, feedback, navigation, reliability, safety, simplicity,
#Tags	#AssistiveTechnology, #VIP, #AssistiveGlove

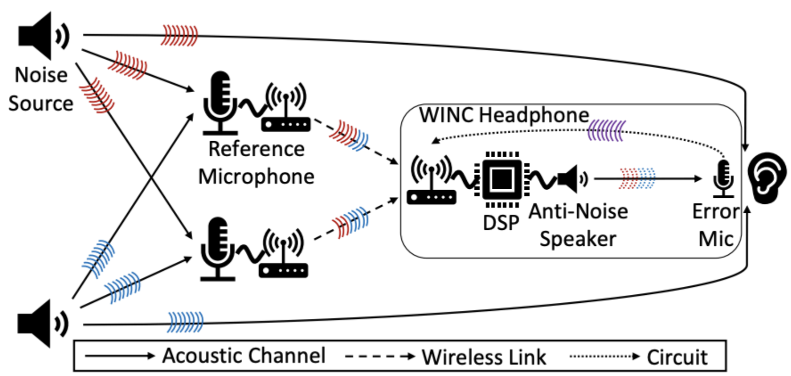
Summary of key points + notes (include methodology)	This article that I read was about an assistive glove which helps the visually impaired navigate their surroundings. In the article I read they wanted to create a wearable device that could reliably identify and relay information to blind people. The device they came up with was a glove that could identify objects and verbally communicate it to the wearer. It would do this using an RFID reader, a speaker, and a microphone all connected to a raspberry pi. The way it would identify objects would be through a passive RFID tag. The object would need to have a tag equipped to it. When the RFID reader comes in proximity with a new tag it would prompt the raspberry pi to take an audio recording where the user would speak and say what the object is. Then whenever the reader scans that same tag any other time it would play the audio recording back so the user can identify what the object is.
	
	● Visually Impaired people can not perceive their surroundings
	    ○ Device must be able to be simple, reliable, safe, and power efficient
	● Other devices focus on navigation and localization
	    ○ Many use depth perception and gps which have disadvantages

	○ RFID is decent at both ● Device main functionality is to identify objects ○ RFID reader would scan passive RFID tags and communicate object through audio ○ Could record new audio on new tag scan ○ Can replace old audio ○ Passive tags must be affixed to objects ○ Run using RPi ● Experiment was run with 17 participants ○ Given a task to identify objects a mean success rate of 96.32% was achieved ○
Research Question/Problem/ Need	Can navigation with RFID effectively be used to aid visually-impaired people?
Important Figures	
VOCAB: (w/definition)	6-DOF pose:used to determine position in 3D space
Cited references to follow up on	R. K. Katzschmann, B. Araki and D. Rus, "Safe local navigation for visually impaired users with a time-of-flight and haptic feedback device", IEEE Trans. Neural Syst. Rehabil. Eng., vol. 26, no. 3, pp. 583-593, Mar. 2018. I. Khan, S. Khusro and I. Ullah, "Technology-assisted white cane: Evaluation and future directions", PeerJ, vol. 6, pp. e6058, Dec. 2018. D. Zhou, Y. Yang and H. Yan, "A smart 'virtual eye' mobile system for the visually impaired", IEEE Potentials, vol. 35, no. 6, pp. 13-20, Dec. 2016.
Follow up Questions	Can the device be used to help people get directions rather than just identify objects? How are the users supposed to mount passive tags to prospective items? Is there another way to achieve the desired output?

Article #4 Notes: A Wireless IoT Network for Multi-Noise Source Cancellation

Source Title	A Wireless IoT Network for Multi-Noise Source Cancellation
Source citation (APA Format)	Janveja, I., Wang, J., Guan, J., Jog, S., & Hassanieh, H. (2023). WINC: A Wireless IoT Network for Multi-Noise Source Cancellation. *The 22nd International Conference on Information Processing in Sensor Networks*, 110–122. https://doi.org/10.1145/3583120.3586964
Original URL	https://dl-acm-org.ezpv7-web-p-u01.wpi.edu/doi/abs/10.1145/3583120.3586964
Source type	Journal Article
Keywords	Internet of Things, Active Noise Cancellation, Reference Mic, Digital Signal Processing, Adaptive Filtering
#Tags	#ActiveNoiseCancellation, #MultiPointANC, #DSP, #LMS, #IoT

Summary of key points + notes (include methodology)	Currently, Noise Canceling Headphones work by capturing noise from a reference mic and processing an anti-noise signal which is then played to the user. However, this performance is inefficient because they have very little time to produce the anti-noise signal and they capture noise from only one source. A proposed solution is to have multiple reference mics setup in an area and send recorded noise signals through an IoT network, which travels faster than sound, to the processing unit to produce anti-noise signals. This solution is optimal because it allows the headphones more time to produce the anti-noise signal and allows the headphones to see into the future almost.
	
	● Problems with current ANC
	    ○ Too close to eardrum
	    ○ Mixes noises coming from different sources
	● WINC adds extra reference mics into the environment
	    ○ Connected through IoT network which is faster than sound
	    ○ Splits audios from different sources to make computation easier

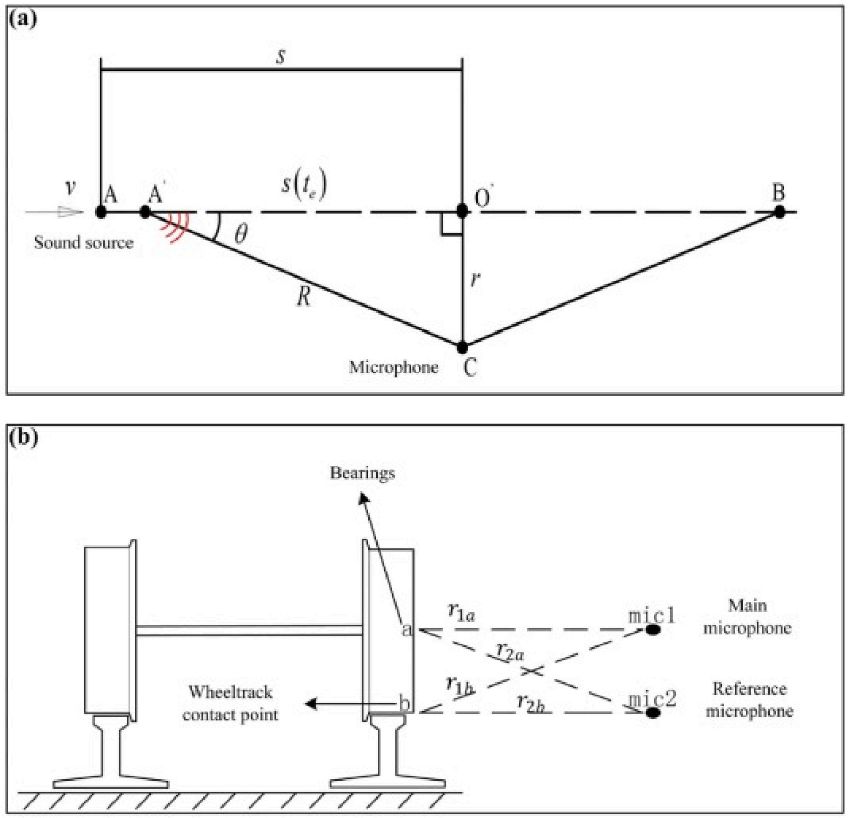
	■ Even though sources are split, the same feedback error is applied to all sources ○ Since IoT is faster than sound the microphones allow the headphones to look ahead in future and know what sounds the headphones will capture ● The WINC system canceled noise by around 9db above the industry standard at the time (Bose QC35) ○ However in terms of pure Noise Canceling the QC35 was worse by around 28db the extra 19db came from passive noise canceling on the QC35
Research Question/Problem/ Need	How can we make Active Noise Canceling on headphones more effective?
Important Figures	 **Figure 2: System Overview of WINC's Multi-Noise Source Multi-Reference ANC Over Wireless IoT Networks.**
VOCAB: (w/definition)	Propagation: transmission of motion, light, sound, etc. in a particular direction or through a medium Frequency Domain: the analysis of mathematical functions or signals with respect to frequency
Cited references to follow up on	Masaichi Akiho, Miki Haseyama, and Hideo Kitajima. 1999. A practical method to reduce a number of reference signals for the ANC system. In 1999 IEEE International Conference on Acoustics, Speech, and Signal Processing. Proceedings. ICASSP99 (Cat. No. 99CH36258), Vol. 4. IEEE, 2387–2390. Jordan Cheer, Vinal Patel, and Simone Fontana. 2019. The application of a multireference control strategy to noise cancelling headphones. The Journal of the Acoustical Society of America 145, 5 (2019), 3095–3103. WB Mikhael and PD Hill. 1988. Acoustic noise cancellation in a multiple noise source environment. In 1988., IEEE International Symposium on Circuits and

	Systems. IEEE, 2399–2402.
Follow up Questions	Is there any way to make this mobile? Can we make this less computationally demanding?

Article #5 Notes: Dual-microphone active noise cancellation paved with Doppler assimilation for TADS

Source Title	Dual-microphone active noise cancellation paved with Doppler assimilation for TADS
Source citation (APA Format)	Liu, F., Zhao, X., Zhu, Z., Zhai, Z., & Liu, Y. (2023). Dual-microphone active noise cancellation paved with Doppler assimilation for TADS. *Mechanical Systems and Signal Processing*, *184*, 109727. https://doi.org/10.1016/j.ymssp.2022.109727
Original URL	https://www-sciencedirect-com.ezpv7-web-p-u01.wpi.edu/science/article/pii/S0888327022007993#f0040
Source type	Journal Article
Keywords	Train bearings, Track-side Acoustic Detection System, Doppler effect, Least Mean Squares Algorithm, Inverse Modulation, Re-Modulation
#Tags	#ActiveNoiseCancellation, #MultiPointANC, #DSP, #LMS
Summary of key points + notes (include methodology)	The article focuses on the application of a Train Acoustic Diagnostic System to monitor the health of train bearings through noise. However this system is flawed because wheel-track contact noise overpowers the bearing noise. The proposed solution is to use IM(Inverse Modulation)-RM(Re-Modulation) to mitigate Doppler distortion and applying LMS (Least Mean Square) for Active Noise Cancellation (ANC). Since the two microphones are in 2 different positions Doppler distortion affects the frequencies observed by the mics. The IM processing effectively eliminates Doppler distortion. RM is then used as a check to ensure consistency in Doppler distortion correction. The LMS algorithm is employed in an adaptive filter to cancel noise. The algorithm is adjusted iteratively based on the differences in error during each cycle. They found in simulation studies, the combination of IM-RM-LMS successfully eliminated wheel-track noise. Similar results were achieved in the experimental setup. ● TADS is used to monitor health of bearings on trains by noise

	○ Wheel-Track contact noise overpowers noise from the bearings ○ Goal is to effectively cancel wheel-track noise to isolate bearing noise ● When using multiple microphones Doppler Effect messes with the input frequencies ● Proposed Solution: Use IM-RM to solve the Doppler distortion and use LMS for ANC ● IM-RM processing is added before the reference input of the filter ○ IM processing is first processed on the input from the reference mic this effectively eliminates doppler distortion ○ RM is then used on this signal as a check to make sure the doppler distortion is consistent. ● LMS algorithm is used in the adaptive filter for noise cancellation ○ The adaptive filter applies weight coefficients and filter inputs to the input signal ○ These weights and inputs are decided based on the least mean square of the error ○ Weights are adjusted by differences in error each iteration of the cycle ● Within the simulation study IM-RM-LMS effectively eliminated wheel-track noise ● Within the experimental setup similar results were achieved
Research Question/Problem/ Need	How can we eliminate Doppler distortion to effectively cancel wheel-track noise in TADS?

<table>
<tr><td>Important Figures</td><td>


Microphone Setup, Diagram to derive modulation equations</td></tr>
<tr><td>VOCAB: (w/definition)</td><td>Amplitude Modulation: a process by which the wave signal is transmitted by modulating the amplitude of the signal

Doppler Effect: an increase (or decrease) in the frequency of sound, light, or other waves as the source and observer move toward (or away from) each other.

Adaptive Filter: digital filters whose coefficients change with an objective to make the filter converge to an optimal state

Interpolation Fitting: to use interpolation to smooth observed data, fill in missing data, and make predictions.</td></tr>
<tr><td>Cited references to follow up on</td><td>D. Hou, H. Qi, C. Wang
High-speed train wheel set bearing fault diagnosis and prognostics: Fingerprint feature recognition method based on acoustic emission

S. Wang, J. Zhang
An Intelligent Process Fault Diagnosis System based on Andrews Plot and Convolutional Neural Network

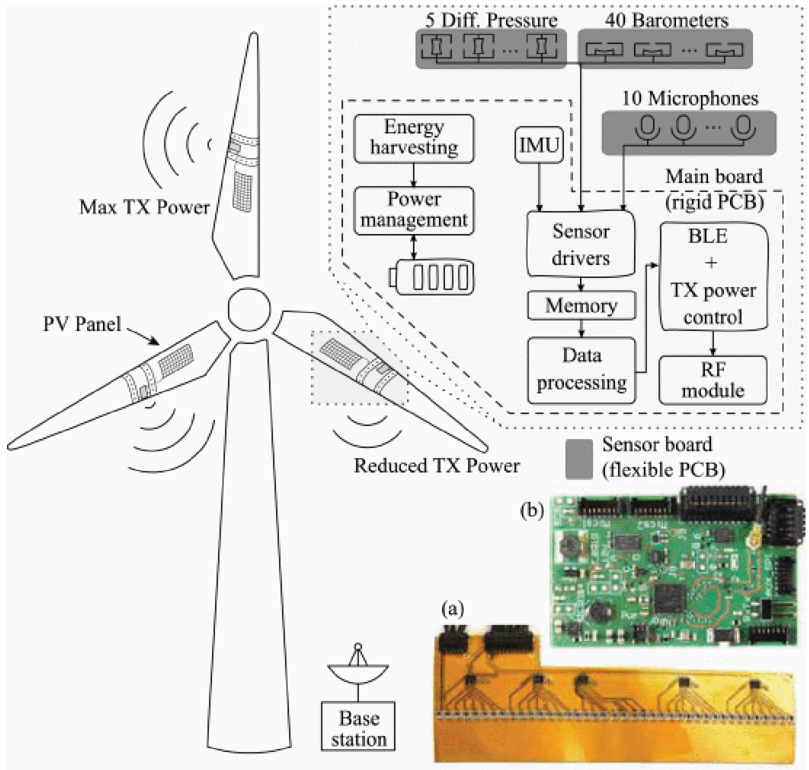
J. Ye, E. Stewart, D. Zhang
Integration of Multiple Sensors for Noncontact Rail Profile Measurement and</td></tr>
</table>

	Inspection
Follow up Questions	Is doppler distortion still a factor we must take into account for when the sound source isn't spinning? Would this solution scale if we were to add additional microphones? Are there any other clear applications of this besides TADS?

Article #6 Notes: Aerosense: A Self-Sustainable and Long-Range Bluetooth Wireless Sensor Node for Aerodynamic and Aeroacoustic Monitoring on Wind Turbines

Source Title	Aerosense: A Self-Sustainable and Long-Range Bluetooth Wireless Sensor Node for Aerodynamic and Aeroacoustic Monitoring on Wind Turbines
Source citation (APA Format)	T. Polonelli, H. Müller, W. Kong, R. Fischer, L. Benini, & M. Magno, "Aerosense: A Self-Sustainable and Long-Range Bluetooth Wireless Sensor Node for Aerodynamic and Aeroacoustic Monitoring on Wind Turbines," in *IEEE Sensors*

	Journal, vol. 23, no. 1, pp. 715-723, 1 Jan.1, 2023, doi: 10.1109/JSEN.2022.3224307.
Original URL	https://ieeexplore-ieee-org.ezpv7-web-p-u01.wpi.edu/document/9967940
Source type	Journal Article
Keywords	Wind turbines, IoT, Sensors, Monitoring, Wireless sensor networks
#Tags	#IoT, #BLE, #Sensors
Summary of key points + notes (include methodology)	This article provides a solution to a self-sustaining metric monitoring system for wind turbines to improve generation in the future. They chose to achieve this using several IoT sensors connected to a base station through BLE which would upload data to the cloud. Sensors were chosen with specific constraints in mind to preserve energy consumption and, in turn, battery life. When collecting data, important data, like barometric, would undergo lossless compression and less important, like audio, would undergo lossy compression to better transfer data over BLE. Overall they found BLE was successful in transferring up to 1.2MB/s of data within closer ranges which would successfully carry the compressed data. The device also successfully had a runtime of at least 110 days before recharging. ● To increase electricity generation in wind turbines more data must be collected ○ Currently, there is no effective wind turbine metric monitoring system ○ Proposed solution: create a self-sustaining monitoring system for wind turbines through IoT sensors ● Specifications for each sensor were set so the device would be sustainable with little maintenance when set up ○ When choosing pressure sensors, sampling rate and thermal capability were considered ○ Accuracy and water-resistance were considered when choosing a barometer ○ Bandwidth and power consumption were considered when choosing a mic ○ When choosing an IMU(vibration) sensor, active and passive power consumption were considered ● The device is powered by a solar panel and a rechargeable battery which adaptively supplies power based on need ● When necessary, lossless compression is performed on the pressure sensor data ● When necessary, lossy compression is performed on the audio from the microphones ● Compression saved power for audio but for pressure sensor data the effect was negligible

	• Expected lifetime on normal usage was at least 110 days
Research Question/Problem/ Need	Is it possible to create a sustainable and efficient wind turbine monitoring system through current IoT technology?
Important Figures	 Device technical diagram
VOCAB: (w/definition)	Airfoil: shaped surface, such as an airplane wing, tail, or propeller blade, that produces lift and drag when moved through the air Aeroacoustic: branch of acoustics that studies noise generation via either turbulent fluid motion or aerodynamic forces interacting with surfaces Entropy: a logarithmic measure of the rate of transfer of information in a particular message or language Quantization: the process of mapping continuous infinite values to a smaller set of discrete finite values
Cited references to follow up on	R. Fischer, H. Mueller, T. Polonelli, L. Benini and M. Magno, "WindNode: A long-lasting and long-range Bluetooth wireless sensor node for pressure and acoustic monitoring on wind turbines", Proc. 4th IEEE Int. Conf. Ind. Cyber-Phys. Syst. (ICPS), pp. 393-399, May 2021. F. Di Nuzzo, D. Brunelli, T. Polonelli and L. Benini, "Structural health monitoring

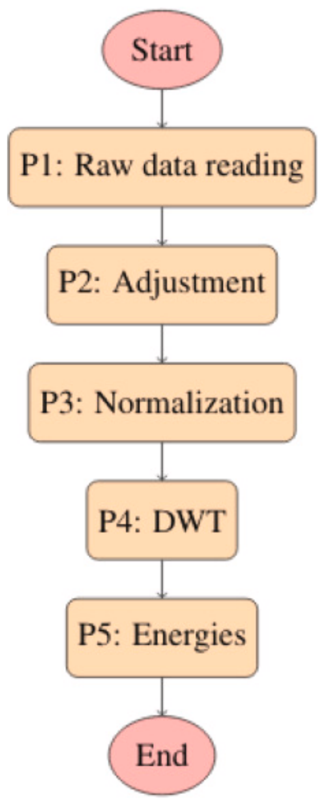
	system with narrowband IoT and MEMS sensors", IEEE Sensors J., vol. 21, no. 14, pp. 16371-16380, Jul. 2021. X. Jin, Z. Xu and W. Qiao, "Condition monitoring of wind turbine generators using SCADA data analysis", IEEE Trans. Sustain. Energy, vol. 12, no. 1, pp. 202-210, Jan. 2021.
Follow up Questions	At closer ranges would we be able to transfer audio data through BLE without compression? Is it viable to make the device completely self-sustainable with a larger power source and solar panel? After analyzing the collected data, what can we change about wind turbines to increase power generation?

Article #7 Notes: Discrete Wavelet Transform in digital audio signal processing: A case study of programming languages performance analysis

Source Title	Discrete Wavelet Transform in digital audio signal processing: A case study of programming languages performance analysis
Source citation (APA Format)	Escola, J. P. L., Souza, U. B. de, & Brito, L. da C. (2022). Discrete Wavelet Transform in digital audio signal processing: A case study of programming languages performance analysis. *Computers and Electrical Engineering*, *104*, 108439. https://doi.org/10.1016/j.compeleceng.2022.108439
Original URL	https://www-sciencedirect-com.ezpv7-web-p-u01.wpi.edu/science/article/pii/S0045790622006541
Source type	Journal Article
Keywords	Time Complexity, Discrete Wavelet Transform, Computational Cost, Performance
#Tags	#ProgrammingLanguage, #DSP

Summary of key points + notes (include methodology)	Digital Signal Processing (DSP) plays a critical role in various domains, but it is often associated with high computational demands. Its algorithms often have a runtime complexity of O(n^2) which is very complex. This article aimed to determine whether the choice of programming language in developing DSP algorithms significantly impacted performance. They decided to run the tests using a Discrete Wavelet Transform algorithm which is known for its efficient time-frequency analysis. The algorithm consisted of reading data, adjusting the vector, normalizing data, and performing the DWT analysis. They wrote this algorithm in C, C++, Java, Matlab, Python, Octave, R, and Scilab and compared the runtimes of all. They found that C, C++, and Java were the most efficient, finishing with times under 0.5s. Python was next with around 15s. The rest were much worse, taking up to an hour. It was decided that language does matter very much and depending on your criteria you should choose which is the best for your situation.
	
	● DSP is usually very computationally heavy
	    ○ O(n^2)
	    ○ Big data, usually use gpu cuda cores

	• Can choice of programming language make a big impact on performance when implementing DSP algorithms
	• Decide to test using Discrete Wavelet Transform algorithm
	    ○ Very efficient time-frequency analysis
	• Write the same algorithm in C, C++, Java, Matlab, Python, Octave, R, and Scilab
	• Steps of Alg
	    ○ Read Data
	    ○ Adjust vector or array, fill with zeroes for purpose of DWT
	    ○ Normalize data, remove 0 hz frequencies
	    ○ Apply DWT
	• Differences in language structure
	    ○ Parentheses, data reading, vectors/arrays, math operations, delimiting, special functions
	• C, C++, Java were the most efficient all with times under 0.5s
	• Python is also pretty efficient at times under 20s
	• The rest are pretty horrendous with times up to an hour
	• C++ is best for larger data inputs as well

Research Question/Problem/ Need	Can the choice of programming language influence the research process when writing DSP algorithms?
Important Figures	 Algorithm Steps
VOCAB: (w/definition)	Frequency Resolution: Frequency resolution refers to the ability of an algorithm to

	distinguish between two closely spaced peaks in the frequency domain Frequency Domain: the analytic space in which mathematical functions or signals are conveyed in terms of frequency, rather than time Genetic Algorithms: method for solving both constrained and unconstrained optimization problems that is based on natural selection Binarization: the process of dividing data into two groups and assigning one out of two values to all the members of the same group
Cited references to follow up on	Guido R.C. Effectively interpreting discrete wavelet transformed signals [Lecture notes] IEEE Signal Process Mag, 34 (2017) Owens J.D., Houston M., Luebke D., Green S., Stone J.E., Phillips J.C. GPU computing Proc IEEE, 96 (2008) Hong-tul Z., Dong-mei X. Analysis on algorithm of wavelet transform and its realization in C language Proceedings of the third international symposium on electronic commerce and security workshops, Citeseer (2010), pp. 336-338
Follow up Questions	What criteria could we use to determine which language to use? Why was numpy slower than python VA? Why was matlab so much slower than python?

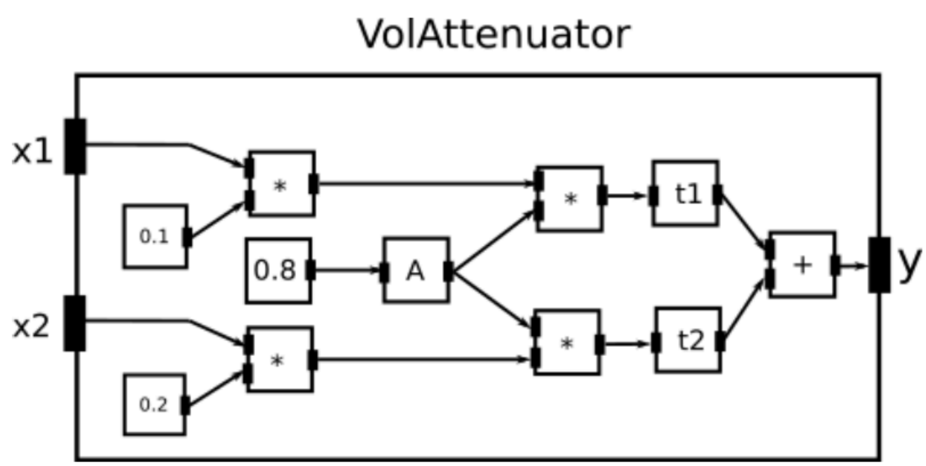
Article #8 Notes: Ciaramella: A Synchronous Data Flow Programming Language For Audio DSP

Source Title	Ciaramella: A Synchronous Data Flow Programming Language For Audio DSP
Source citation (APA Format)	Marrone, P., D'Angelo, S., Fontana, F., Costagliola, G., & Puppis, G. (2022, June 7). Ciaramella: A Synchronous Data Flow Programming Language For Audio DSP. https://doi.org/10.5281/zenodo.6573430
Original URL	https://zenodo.org/record/6573430
Source type	Conference Article
Keywords	DSP, SDF, WDF, Compile, Semantic, Scheduling
#Tags	#ProgrammingLanguage, #DSP

Summary of key points + notes (include methodology)	The journal article discusses the challenges associated with programming digital signal processing (DSP) within traditional languages, emphasizing the complexity and frequency of errors due to intricate language details. It notes the existence of past DSP languages, which were not modular and lacked support for complex functions and delay-free feedback. The proposed solution is a more intuitive DSP programming language, designed in a block-based fashion, focusing on semantics and syntax. This language operates under the Synchronous Data Flow (SDF) model and can compile code in C++ or MATLAB. It employs a network of independent processes (actors) communicating via FIFO queues and token-based input and output data handling. Static scheduling is utilized to ensure all actor firings are feasible before execution, improving accuracy. The article outlines the goals of simplicity, modularity, composability, and structural representation in this new language, featuring blocks (actors), ports for input and output, and connections to dictate the flow of data. Various types of blocks are introduced, and a compiler named Zampogna, implemented in JavaScript, is employed to parse and validate syntax, generating a tree and an instruction graph (IG) illustrating the network structure. Static scheduling is employed, and the article delves into details of code generation and optimizations, which accommodate updates during runtime. The article concludes by mentioning the implementation of a test case based on the Wave Digital Filter (WDF) model.
	● DSP programming is very complex to do within normal languages
	    ○ Intricate details of these languages can cause errors
	    ○ DSP languages have been made in the past

- They are not very modular
- Complex functions and methods
- Produces platform locked code
- Can not implement delay-free feedbacks

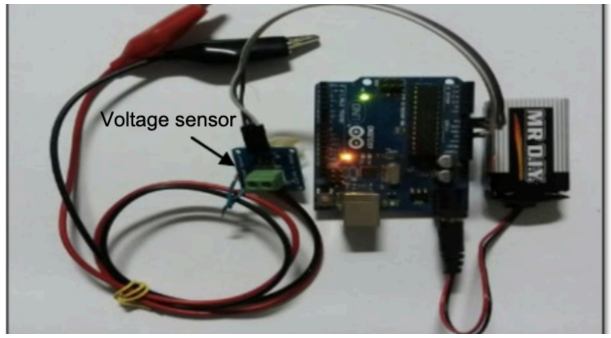
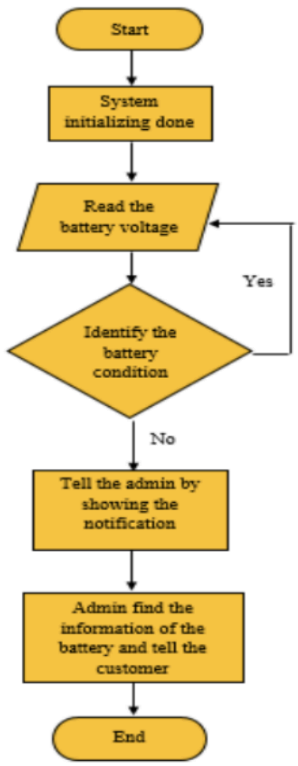
- Proposed solution is much more intuitive for DSP
  - Block based
  - Semantic vs Syntactic
  - Flatten before computation
  - SDF(Synchronous Data Flow)
  - Compiles C++ or MATLAB code
- SDF
  - Network of Independent Processes(actors)
  - Comm. through FIFO queues
  - Queues input and output data called tokens
  - STATIC SCHEDULING
    - Ensures all actor firings are possible before executing
    - Improves accuracy
- Goals
  - Simple and Unconstrained syntax
  - Modularity
  - Composability
- Structure
  - Blocks: represents an actor, performs an action
  - Ports: Input output
  - Connection: direction of flow of data
- Blocks
  - Elementary Operation Blocks: 2 in 1 out, add, sub, div, mult, etc
  - Var block: 1 in 1 out constant data source, data flows
  - Unit Delay Block: 1 in 1 out, recursion
  - Composite Block: subsystems, main method
- Zampogna: Compiler
  - Built in js
  - Parse and validate syntax, this generates a tree
  - Tree is used to generate a graph (IG) which shows the network
  - Static Scheduling
  - Generates code
- Optimization, all values update as code runs, make sure compiler accounts for this
  - Define update class
  - Constants class, sample rates class, control rates class, audio rates class
- IG Static Scheduling
  - Every node/block in the IG establishes dependency except delay, defined on initialization
  - If output port of A is connected to input of B, B has instantaneous dependency in A
  - Schedule is generated if this all dependencies are valid

	■ Check by putting in stack and working backwards ○ If stack doesn't work error is thrown ● Implement WDF as test
Research Question/Problem/ Need	Is it possible to make an effective programming language for audio DSP engineers?
Important Figures	 Shows IG of a voltage attenuator block
VOCAB: (w/definition)	Static Scheduling: memory accesses are scheduled at compile-time, such that there is no memory conflict during the execution Semantic: clearly describes its meaning to the developer Synchronous: existing or occurring at the same time Composite: made up of various parts or elements
Cited references to follow up on	"Jison," https://github.com/zaach/jison, accessed: 2022-02-07. A. Fettweis, "Wave digital filters: Theory and practice," Proceedings of the *IEEE*, vol. 74, no. 2, pp. 270-327, 1986. "dot - the fastest + concise javascript template engine for node.js and browsers." https://olado.github.io/, accessed: 2022-02-07.
Follow up Questions	Has the language been kept up to date since then if so what new functionalities have been added? Has the language been used by others since then, why or why not? Is it possible to completely get rid of the coding aspect and create a GUI to make

	this process more accessible?

Article #9 Notes: IoT-Based Battery Monitoring System for Electric Vehicle

Source Title	IoT-Based Battery Monitoring System for Electric Vehicle
Source citation (APA Format)	Helmy, M., Abd Wahab, M. H., Imanina, N., Anuar, M., Ambar, R., Baharum, A., Shanta, S., Sulaiman, M., Sanim, S., & Hanafi, H. (2018). IoT-Based Battery Monitoring System for Electric Vehicle. *International Journal of Engineering & Technology*, *7*, 505–510. https://doi.org/10.14419/ijet.v7i4.31.25472
Original URL	https://www.researchgate.net/publication/343786968_IoT-Based_Battery_Monitoring_System_for_Electric_Vehicle
Source type	Conference Article
Keywords	Arduino Uno, SIM808 GSM Shield, BMS, Lithium ion batteries
#Tags	#IoT, #Arduino
Summary of key points + notes (include methodology)	With the increasing popularity of EV's there is a need for a better battery health monitoring system. Currently, health monitoring systems notify the user of poor battery health but with the help of IoT, manufacturers can also use this as a statistic and to help customers. The proposed solution is an IoT battery health monitoring device which can notify the manufacturer of location upon degraded battery health. The device uses an Arduino with a GSM/GPS module and a voltage sensor to measure and report battery health. The voltage sensor and GPS module were both tested to check accuracy and they both returned readings more than 99% accurate. They then defined what is considered a degraded battery by whether the battery's discharge rate was 30% below normal. Once these steps were complete they developed a user interface. the interface was developed for both the user and manufacturer where they could check battery health and location of the battery.
Research Question/Problem/	Can we make battery health statistics available to manufacturers using IoT?

Need	
Important Figures	 Shows setup of the voltage hardware and testing  **Fig. 2:** System flowchart Flowchart of system
VOCAB: (w/definition)	GPRS: General Packet Radio Service is a packet oriented mobile data standard on the 2G cellular communication network's global system ZigBee: a standards-based wireless technology developed to enable low-cost, low-power wireless machine-to-machine (M2M) and internet of things (IoT) networks Li-MN: lithium ion manganese oxide battery
Cited references to follow up on	S. Yonghua, Y. Yuexi, H. Zechun, "Present Status and Development Trend of

	Batteries for Electric Vehicles," Power System Technology, Vol. 35, No. 4, pp. 1-7, 2011. L. Xiaokang, Z. Qionghua, H. Kui, S. Yuehong, "Battery management system for electric vehicles," J.Huazhong Univ. Of Sci. & Tech. (Nature Science Edition). Vol. 35, No. 8, pp. 83-86, 2007. C. Piao, Q. Liu, Z. Huang, C. Cho, and X. Shu, "VRLA Battery Management System Based on LIN Bus for Electric Vehicle," Advanced Technology in Teaching, AISC163, pp. 753-763, 2011.
Follow up Questions	Why was GPRS chosen as the standard, given that it is quite slow? Has this technology been tested on a real car battery? Why not build other metrics into the device like temperature?

Article #10: Audio signal noise cancellation with adaptive filter techniques

Source Title	Audio signal noise cancellation with adaptive filter techniques
Source citation (APA Format)	Venkata Sudhakar, M., Prabhu Charan, M., Naga Pranai, G., Harika, L., & Yamini, P. (2023). Audio signal noise cancellation with adaptive filter techniques. *Materials Today: Proceedings*, *80*, 2956–2963. https://doi.org/10.1016/j.matpr.2021.07.080
Original URL	https://www-sciencedirect-com.ezpv7-web-p-u01.wpi.edu/science/article/pii/S2214785321049324
Source type	Journal Article
Keywords	Infinite Impulse Response Filter (IIR), FInite Impulse Response Filter (FIR), optimization algorithms, Least mean squares filter (LMS), JAYA algorithm, PSO algorithm, Adaptive Noise Cancellation System (ASNC)
#Tags	#DSP, #ANC #LMS
Summary of key points + notes (include methodology)	The article delves into the application of adaptive filtering systems for noise reduction while preserving desired signals, particularly in the context of voice signal processing. It highlights the importance of adaptive filters, which can adapt to the surrounding conditions and adjust parameters like coefficients, step size, and length to optimize their performance. The paper introduces three algorithms, namely the Least Mean Square (LMS), JAYA, and Particle Swarm Optimization (PSO), which are employed to enhance noise cancellation. The LMS algorithm employs gradient descent for noise reduction, while JAYA and PSO use bio-inspired techniques to find optimal weights for noise cancellation. The results demonstrate that these algorithms effectively reduce noise and amplify signal quality in various scenarios, making them not only valuable for speech noise cancellation but also applicable in fields like the military and medical sectors. This work underscores the importance of adaptive noise cancellation mechanisms in achieving high-quality signal processing and communication. ● Adaptive filters are needed to filter noise and preserve certain signals

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Digital filters like IIR and FIR are used
Adaptive filters update and adapt to situation using parameters

Coefficients
Step Size
Length

LMS

Gradient Descent, step function, iteration
Iteratively adjusts filter coefficients
Refines weights iteratively
Improves signal clarity in NC

JAYA

Bio-inspired optimization
Effectively makes weights compete against each other
Identifies “winner” of the match and employs it

PSO (Particle Swarm Optimization)

Bio-inspired

Inspiration from swarm behavior

Weights are treated as particles navigating a solution space
Particles communicate to converge towards optimal weights

Results:

LMS

Reduction in noise gradually
Clear captured signal

JAYA

Substantial noise reduction
Amplification of desired signal

PSO

Substantial Noise Cancellation
Amplification of signal strength

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<tr>
<td>Research Question/Problem/ Need</td>
<td>How can algorithms, ranging from PSO, LSO, and JAYA, be better utilized to promote an Adaptive Noise Cancellation System (ASNC) to optimize for the intended functionality?</td>
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<td>Important Figures</td>
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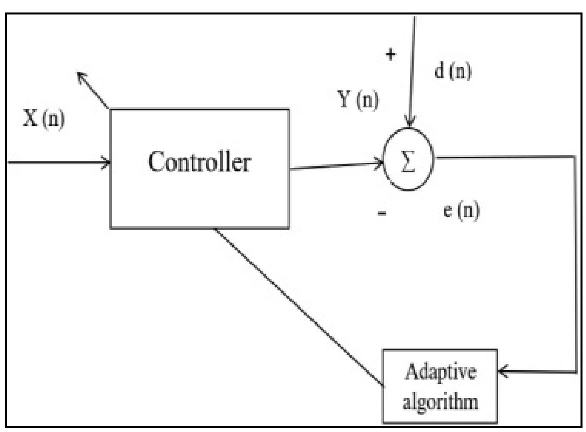

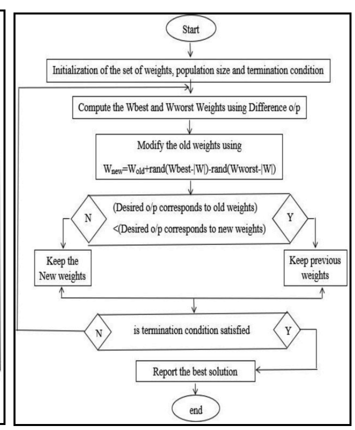

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	- Figure 1: Describes the necessity of having an algorithm that adapts to its environment to ensure that it can be optimized. Following the Flow Diagram, the adaptive algorithm must be accessed in special cases that have not been adapted for, to grow the adaptive system - Figure 4: This is a different form of a flow diagram, as there are 3 different variants, this particular one, focuses on the last PSO Algorithm. The system cruciates on the different yes or no, intermediate steps, or gaining the overall understanding of the system to understand the expected, improved output.
VOCAB: (w/definition)	Infinite Impulse Response Filter (IIR): a type of digital filter that reacts to abrupt changes or impulses in a signal. It is widely used in signal processing in various fields such as audio and image processing, telecommunications, and control systems. Finite Impulse Filter: a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time JAYA Algorithm (JAYA): a new metaheuristic algorithm, which has a very simple structure and only requires population size and terminal condition for optimization.
Cited references to follow up on	R.V. Rao Jaya: A simple and new optimization algorithm for solving constrained and unconstrained optimization problems Simon Haykin. Adaptive signal processing: next-generation solutions. Wiley-IEEE Press. ISBN: 978-0-470-19517-8. 2010. No 3rd cited source because there is a limited amount of information that is geared toward this field of optimization
Follow up Questions	Three specific algorithms were analyzed , what was the thought process behind choosing these systems composed to others? These three specific programs, all combine the capability to manage the finite and infinite impulse response filter to map and deviate the sounds produced, are there any structural fallacies? The graphs and figures review the data collated through these models, but do not specify the accuracy, what were specific percentages?

Article #11 Notes: MUTE: Bringing IoT to Noise Cancellation

Source Title	MUTE: Bringing IoT to Noise Cancellation
Source citation (APA Format)	Shen, S., Roy, N., Guan, J., Hassanieh, H., & Choudhury, R. R. (2018). MUTE: 2018 Conference of the ACM Special Interest Group on Data Communication, ACM SIGCOMM 2018. *SIGCOMM 2018 - Proceedings of the 2018 Conference of the ACM Special Interest Group on Data Communication*, 282–296. https://doi.org/10.1145/3230543.3230550
Original URL	https://dl.acm.org/doi/10.1145/3230543.3230550
Source type	Conference Article
Keywords	Noise Cancellation, Acoustics, Internet of Things, Wearables, Adaptive Filter
#Tags	#ANC, #MultiPointANC, #DSP, #IoT, #LMS
Summary of key points + notes (include methodology)	In today's consumer headphones true ANC has yet to be achieved. This is because there are only a few inches between the reference microphones and the ear, so the antinoise signal cannot be produced in time to cancel out the incoming noise. MUTE solves this issue by using an external microphone that is placed away from the headphones. This microphone is connected to the headphones via an IoT relay. This allows the headphones to lookahead and produce antinoise signals for noise that has not reached the headphones yet. MUTE also uses an updated LMS algorithm called Lookahead Aware Noise Cancelling (LANC). LANC uses a gradient descent to update the values of an adaptive filter in a direction that minimizes the residual error. This allows LANC to accurately predict the values of certain noise channels and to switch filter values based on drastic changes in noise behavior. Tests on humans have shown that MUTE cancels noise, on average, up to 9 dB more than Bose QC35's. This is a significant improvement to ANC and allows for true cancellation. ● True ANC can not be achieved because of the minimal amount of time between noise reaching a reference microphone and the users ears ○ Usually there is a time delay between when the accurate antinoise is produced for a noise

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- ○ With enough time overhead this can be avoided and true ANC can be achieved.
- This was done by using an external microphone set up a distance away from the antinoise speaker.
- This microphone would be connected via an IoT relay which would relay noise faster than sound travels to give enough time gap to produce antinoise
- The new algorithm designed was called Lookahead Aware Noise Cancelling which was a form of LMS that used gradient descent to update filter values to properly predict the values of certain noise channels.
- Lookahead was also used to predict change in sound profile to more quickly switch filter values based on drastic changes in noise behavior.
- After testing on humans and measuring values it was found that MUTE canceled noise up 9db more than Bose qc35 on average.

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<tr>
<td>Research Question/Problem/ Need</td>
<td>Is it possible to achieve true noise cancellation with the use of lookahead?</td>
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<td>Important Figures</td>
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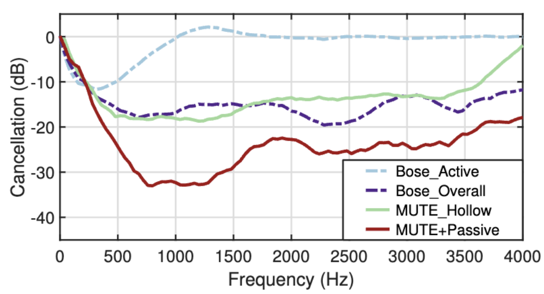
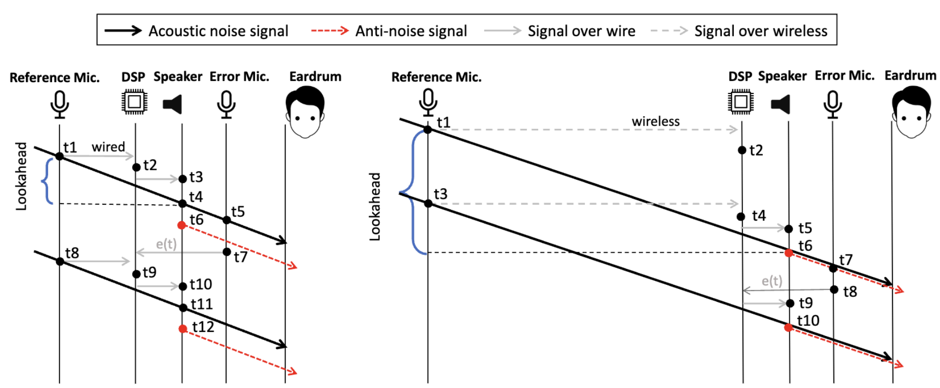


Compares MUTE performance to Bose QC35



(a) Today's ANC processing timeline

(b) *MUTE* processing timeline with large lookahead

Timeline of standard ANC process vs MUTE

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<td>VOCAB: (w/definition)</td>
<td>ISM band: portions of the radio spectrum reserved internationally for industrial, scientific, and medical purposes.</td>
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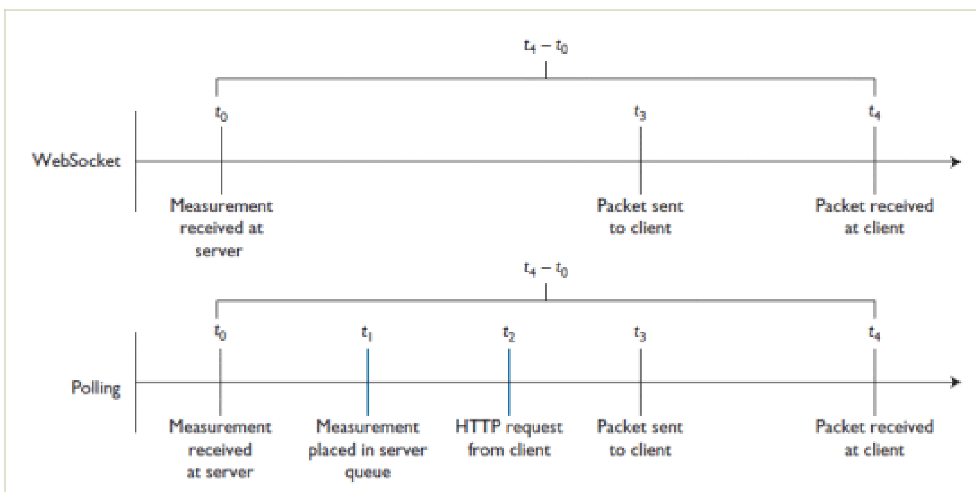
	Non-Causal: a noncausal system's output depends on the future inputs. Carrier-Frequency Offset: refers to the mismatch between the frequency of the received signal and the frequency of the local oscillator at the receiver.
Cited references to follow up on	2014. Honda's Active Noise Cancellation. Retrieved January 29, 2018 from https://www.honda.co.nz/technology/driving/anc/ 2017. Review and Measurements: Bose QC25 Noise-Cancelling Headphone. Retrieved January 30, 2018 from https://www.lifewire.com/bose-qc25-review-specs-3134560 2018. Bose QuietControl 30 Wireless Noise Cancelling Earphone. Retrieved January 29, 2018 from https://www.bose.com/en_us/products/headphones/earphones/quietcontrol-30.html
Follow up Questions	How does the LANC change when extra microphones are introduced? Could the DSP board be replaced with mobile hardware? What is this lower bound of lookahead time for cancellation to be effective?

Article #12 Notes: Communicating and Displaying Real-Time Data with WebSocket

Source Title	Communicating and Displaying Real-Time Data with WebSocket
Source citation (APA Format)	Pimentel, V., & Nickerson, B. G. (2012). Communicating and displaying real-time data with websocket. *IEEE Internet Computing*, *16*(4), 45–53. https://doi.org/10.1109/mic.2012.64
Original URL	https://ieeexplore.ieee.org/document/6197172?denied=
Source type	Journal Article
Keywords	HTTP polling, WebSocket protocol, Long polling
#Tags	#Protocols, #IoT

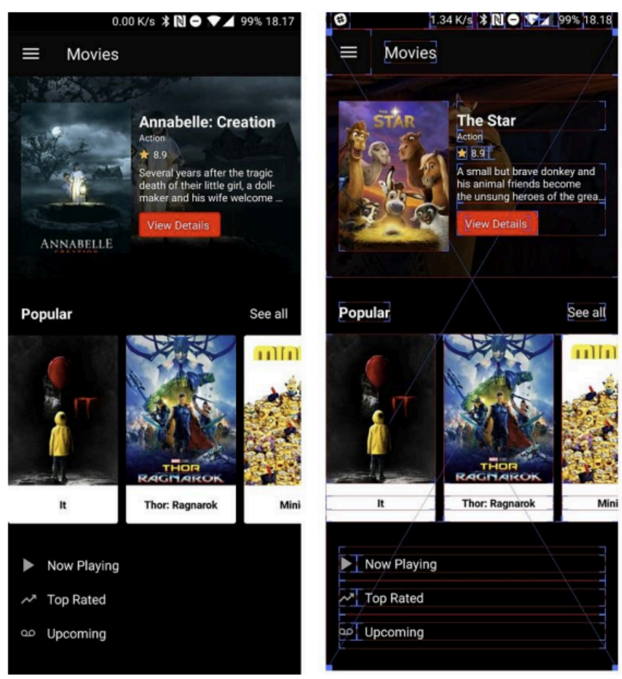
Summary of key points + notes (include methodology)	● Latency is an issue when sending real-time data over the internet
	● HTTP polling is a solution which is used when the message delivery interval is known
	    ○ Works by requesting and receiving response from a server, the time between requests is the polling interval
	    ○ When polling rate increases, overhead increases, increasing latency
	● Websocket protocol enables full duplex communication over TCP socket
	● HTTP Long polling improves by stopping empty responses
	● Websocket provides full bidirectional that operates through a single socket.
	● A Web App was developed to test these three protocols
	    ○ Used a wind sensor connected to a base station
	    ○ Wind direction and speed were polled
	● Latency was calculated by subtracting the final time(client receives packet) by the initial time(measurement is uploaded at the server)
	● Test was run using each of three protocols to receive data from the web app
	    ○ Tested in Edmonton, Canada; Caracas, Venezuela; Lund, Sweden; and Nagaoka, Japan
	● Paired t-test was conducted using resulting data which suggested there was convincing evidence mean latency is lower for websocket than http

	polling at the 99% significance level ○ These results were the same for all the locations and even when comparing to long polling Within this article a study evaluating the latency of real-time data transmission over the internet was conducted. Several different protocols including HTTP polling, WebSocket, and HTTP Long polling, were all tested for latency and performance in transmitting data. HTTP polling which is used when the message delivery interval is known, operates by requesting and receiving responses from a server at specified intervals. This can lead to increased overhead and latency as the polling rate rises. HTTP Long Polling is similar but limits the usage of empty server responses which decreases overhead. WebSocket, facilitating full duplex communication over a single TCP socket, offers bidirectional communication. The study was conducted through a web app testing retrieval of wind sensor data. Latency was measured by subtracting the time the client received the packet from the initial measurement upload time. Tests conducted in different locations - Edmonton, Canada; Caracas, Venezuela; Lund, Sweden; and Nagaoka, Japan - indicated that WebSocket consistently demonstrated lower mean latency compared to HTTP polling and HTTP Long polling at a 99% significance level, as confirmed by a paired t-test. These findings remained consistent across all tested locations, proving Websocket to be the quickest and most reliable way to transmit data over the Internet.
Research Question/Problem/ Need	How does the WebSocket protocol compare to HTTP polling and long polling for real-time data transfer?
Important Figures	 Figure showing the difference in communication by WebSocket and Polling as well as the calculation of latency for each respective protocol. Polling is shown to have an extra step.
VOCAB: (w/definition)	Jetty - An open-source Java HTTP server and servlet container for scaling web apps NTP - Network Time Protocol which synchronizes clocks for computers around the

	world UDP - User Datagram Protocol is one of the core communication protocols of the Internet protocol suite used to send messages to other hosts on an Internet Protocol network
Cited references to follow up on	J. Åkerbcrg, M.M. Gidlund and M. Björkman, "Future Research Challenges in Wireless Sensor and Actuator Networks Targeting Industrial Automation", Proc. 9th IEEE Int'l Conf. Industrial Informatics (INDIN 11), pp. 410-415, 2011. P. Lubbers and F. Greco, "HTML5 Web Sockets: A Quantum Leap in Scalability for the Web", SOA World Magazine, Mar. 2010, [online] Available: http://soa.sys-con.com/node/1315473. W. Hu, G.-P. Liu and D. Rees, "Networked Predictive Control over the Internet Using Round-Trip Delay Measurement", IEEE Trans. Instrumentation and Measurement, vol. 57, no. 10, pp. 2231-2241, 2008.
Follow up Questions	Is this latency benefit still significant given the host and client devices are within the same IP network? Are there different network protocols to be explored given the devices are connected to the same IP network? At what distance does the WebSocket connection become significant?

Article #13 Notes: React Native vs Flutter, cross-platform mobile application frameworks

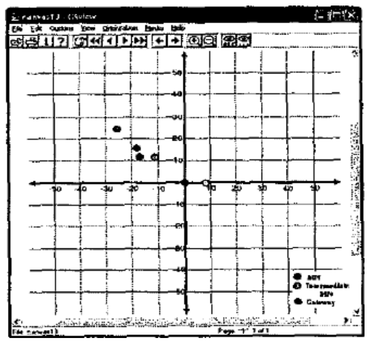
Source Title	React Native vs Flutter, cross-platform mobile application frameworks
Source citation (APA Format)	W. Wu. (2018). React Native vs Flutter, cross-platform mobile application frameworks, B.Eng. thesis, Helsinki Metropolia Univ. Appl. Sci., Helsinki, Vantaa, and Espoo, Finland, Mar. 2018.
Original URL	https://www.theseus.fi/bitstream/handle/10024/146232/thesis.pdf
Source type	Thesis Paper
Keywords	Cross-platform development, JavaScript, Dart, JSX, Widgets, Virtual DOM
#Tags	#App, #ProgrammingLang
Summary of key points + notes (include methodology)	● 2 main platforms for cross-platform development, React Native and Flutter both are very good but there are advantages to each ○ React Native is maintained by Facebook ○ Flutter is maintained by Google ● React Native is built on Javascript, Flutter is a Dart SDk ● React ○ Uses a special extension of Javascript, JSX, to build out components ○ Compiles and renders these components depending in the platform ○ Uses Virtual DOM used to re-render components upon update and provides significant performance ○ Uses Props and State which are 2 ways to store and manipulate data within the app ● Flutter ○ Hot-Reload helps developers while actively making updates ○ Built on “Widgets ” which is comparable to JSX in how you build components ○ The concept of State exists, managed by whether you need updatable information within the widget ● Study ○ An netflix clone app was built in both frameworks to compare experience ○ Flutter is more efficient and less resource demanding

	○ React Native has a much larger community and more support by developers React Native and Flutter are two popular cross-platform frameworks for developing mobile applications, but both frameworks have strengths and weaknesses. React Native was developed by Facebook and uses JavaScript to render UI components using JSX. React Native uses a virtual DOM to re-render and update components and uses props and states to manage data. Flutter was developed by Google and uses Dart to render UI components structured as Widgets. Dart is a statically typed language that is similar to Java. Dart uses hot-reload to update components while actively developing. It also uses states to manage different widget components and update them. To test the performance of these frameworks a study was conducted in which a Netflix mock app was built in each framework. It was concluded that Flutter is more efficient and less resource-demanding so it is the overall winner. Although it is stressed both platforms are very viable and React has a much bigger developer community so it may be easier to use.
Research Question/Problem/ Need	Which of the two most popular mobile app development frameworks, Flutter and React Native, provide the best experience?
Important Figures	 Figure showing the different components using the layout inspector within the Flutter mock app.
VOCAB: (w/definition)	SDK: Software Development Kit, collection of tools essentially a framework DOM: Document Object Model, tree structure diagram representing the structure of the app or document
Cited references to follow up on	Desktop vs Mobile vs Tablet Market Share Worldwide

	URL: http://gs.statcounter.com/platform-market-share/desktop-mobile-tablet. Accessed July 13, 2017 ReactJS: An Open Source JavaScript Library for Front-end Development URL:https://reactjs.org/ Accessed July 13, 2017 Official documentation of Flutter Navigator URL:https://docs.flutter.io/flutter/widgets/Navigator-class.html Accessed November 03, 2017
Follow up Questions	Compared to native languages per platform how do these perform? How do users rate apps built in each language, what seems to be the consensus? Was the mock app that was built reviewed by anyone else besides the author?

Article #14 Notes: GPS-free indoor location tracking in mobile ad hoc network (MANET) using RSSI

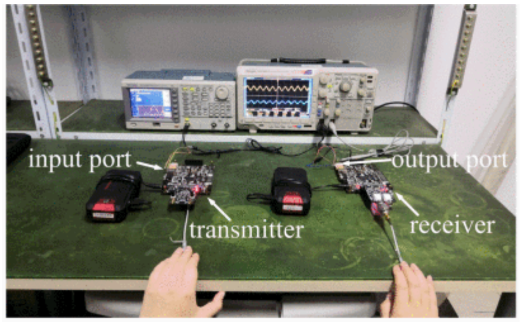
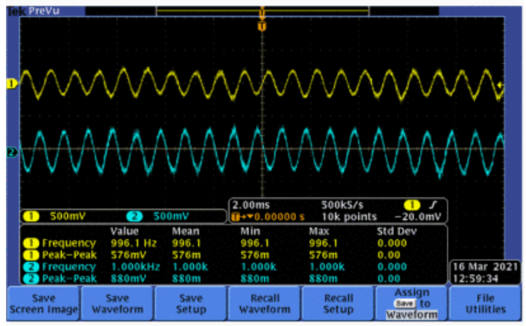
Source Title	GPS-free indoor location tracking in mobile ad hoc network (MANET) using RSSI
Source citation (APA Format)	Ali, A., Latiff, L. A., & Fisal, N. (2004) GPS-free indoor location tracking in mobile ad hoc network (MANET) using RSSI, 2004 RF and Microwave Conference (IEEE Cat. No.04EX924), Selangor, Malaysia, 2004, pp. 251-255, doi: 10.1109/RFM.2004.1411119.
Original URL	https://ieeexplore.ieee.org/document/1411119
Source type	Conference Article
Keywords	Mobile ad hoc networks, Global Positioning System, IP networks, Routing
#Tags	#Networks, #Sensors, #Transmission
Summary of key points + notes (include methodology)	● Ad hoc networks are mobile networks without infrastructure ● Mobile ad hoc networks (MANET) have nodes which are allowed to move ● Goal is to find a way to track these nodes ● 2 ways to track location, hardware and software ● Most popular hardware solution is GPS ○ Accurate up to 3-5 meters ● Active Badge uses IR signals ● Bat uses RF and ultrasound signals ● Cricket is the same as Bat but it is decentralized, no base station ● Radar is used as a software system ● The chosen solution by this article is RSSI (Received Signal Strength Indicator) ● Within the MANET network the nodes are equipped with network cards (NIC) which provides the signal strength data ● Based on RSSI distance was calculated with multiple formulas considering multiple factors like orientation and signal propagation This paper proposes a GPS-free indoor location tracking system for mobile ad hoc networks (MANETs) using RSSI. The system uses the received signal strength indicator (RSSI) to calculate the distance and the geometric position of the mobile node within the MANET network. The system consists of a gateway and mobile

	nodes (MN). The gateway is the center of the network and the GUI program resides here. The NIC card driver from each of the nodes provides information on Signal Strength (SS). The system uses the distance power law to calculate the distance between two nodes. The path loss exponent is determined by measuring the signal strength at a known reference distance. The system also uses the triangulation method to determine the location of a node. The system was tested in a test bed network. The results showed that the system can accurately determine the location of nodes in the network.
Research Question/Problem/ Need	Can RSSI be used to develop a GPS-free location tracking system for MANETs?
Important Figures	 Output of the GUI program of different MANET nodes relative to the gateway.
VOCAB: (w/definition)	Triangulation - The process of determining the location of a point by measuring angles to it from known points at either end of a fixed baseline Beacon - A type of network packet used for detection of neighboring nodes Routing protocol - The technical method by which nodes communicate and route packets between each other in a network
Cited references to follow up on	Z.Guang, A. Seneviratne, R. Chan and P. Chumchu," A Software Based Indoor Relative Location Management System", Proceedings of Wireless and Optical Communications, Canada, 2002. P. Enge, and P. Misra, "Special Issue on GPS: The Global positioning System," Proc. of the IEEE, pp. 3-172, January 1999. R. Want, A. Hopper, V. Falcao, and J. Gibbons, "The Active Badge location system", ACM Transactions on Information Systems, pp. 91-102, Jan. 1992.
Follow up Questions	How does this solution scale as the number of nodes increases? Were the produced distance measurements compared with the true values, how accurate were they? How would you the equations change for outdoor environments?

Article #15 Notes: An audio transmission system based on capacitive coupling intra-body communication

Source Title	An audio transmission system based on capacitive coupling intra-body communication
Source citation (APA Format)	Zhu, W., Zhou, T., Zhou, Y., Li, M., Chen, Y., Zhao, Y., & Song, Y. (2021). An audio transmission system based on capacitive coupling intra-body communication. *ISO/IEC Standard for Information Technology : Microprocessor Systems-Futurebus+, Profile M (Military).*, 183–187. https://doi.org/10.1109/CISCE52179.2021.9445949
Original URL	https://ieeexplore-ieee-org.ezpv7-web-p-u01.wpi.edu/document/9445949
Source type	Conference Article
Keywords	Couplings, Wireless communication, Transmitters, Receivers, Logic gates
#Tags	#Networks, #Sensors, #Transmission
Summary of key points + notes (include methodology)	● Intra-body communication (IBC) is a new form of digital audio transmission that is lower consumption and higher security, which uses the human body as a medium to transfer data ● Within the paper, a 1536kbps IBC system for audio transmission with narrowband modulation is developed ● The IBC consists of a transmitter and a receiver ○ Input microphone data is modulated via on-off keying (OOK) converted to an analog signal via a DAC and passed to the IBC electrode ○ Within the receiver, this signal from the IBC electrode is converted back to digital via ADC and then demodulated via OOK and processed by an audio chip, FPGA ● Demodulation works by taking 12-bit unsigned integers as input and returning 30-bit ints via a pass filter if the output is higher than a certain threshold the symbol is 1, otherwise 0 ● The system also makes use of a lead-lag digital phase-locked loop to eliminate signal drift

	• An experiment was conducted to test this technology where a human touched the electrodes of both the transmitter and receiver and each of the individual parts was tested • An oscilloscope was used to verify whether modulation was taking place as it should and it was found to be normal • A signal generator was used to generate a sine wave and send it through the transmitter and an oscilloscope was connected to the receiver to see whether the signal passed through properly • It was found the signal is almost lossless after passing through the system This paper introduces a novel intra-body communication (IBC) system for digital audio transmission. It uses the human body as a transmission medium, allowing for audio signal transfer between two a transmitter and receiver. The proposed IBC system operates at a data rate of 1536kbps and employs narrowband modulation for efficient transmission. It uses a transmitter and a receiver, each powered by its own battery. Audio data captured by a microphone is modulated using on-off keying (OOK) and converted to an analog signal through a Digital-to-Analog Converter (DAC). This signal is then transmitted via an IBC electrode attached to the body. On the receiving end, the signal is converted back to digital format by an Analog-to-Digital Converter (ADC) and demodulated using OOK. The demodulated signal is processed by an audio chip and Field Programmable Gate Array (FPGA) to reconstruct the original audio data. To account for signal drift, the system utilizes a lead-lag digital phase-locked loop. The proposed system was tested through experiments involving a human subject touching both the transmitter and receiver electrodes. Individual components were also tested to ensure functionality. Ansignql generator was used to generate a sine wave and send it through the system and an oscilloscope verified lossless signal transmission through the system.
Research Question/Problem/ Need	How can an IBC system for audio data transmission be implemented?

Important Figures	 (a) Diagram of a sin signal transmission experiment  (b)The comparison of input sine signal (yellow) and output signal (blue) Image showing the experimental setup for the IBC with the transmitter, receiver, and human. The bottom image shows the original vs the output signal from the IBC.
VOCAB: (w/definition)	DAC - Digital-to-analog converter, a device that converts a digital signal into an analog signal ADC - Analog-to-digital converter, a device that converts an analog signal into a digital signal PLL - Phase-locked loop, s control system that generates an output signal whose phase is related to the phase of an input signal
Cited references to follow up on	T. G. Zimmerman, "Personal Area Networks: Near-field intrabody communication", IBM Systems Journal, vol. 35, no. 3.4, pp. 609-617, 1996. Ž. Lučev, I. Krois and M. Cifrek, "A capacitive intrabody communication channel from 100 kHz to 100 MHz", 2011 IEEE International Instrumentation and Measurement Technology Conference, pp. 1-4, 2011. M. Seyedi, B. Kibret, D. T. H. Lai and M. Faulkner, "A Survey on Intrabody Communications for Body Area Network Applications", IEEE Transactions on Biomedical Engineering, vol. 60, no. 8, pp. 2067-2079, Aug. 2013.
Follow up Questions	In what contexts could this technology be implemented in industry? What are the limitations to using the human body as a medium for data transmission? Are there limitations to the data transfer rate or bandwidth with IBC technology?

Article #16 Notes: WebRTC-Based Multi-View Video and Audio Transmission and its QoE

Source Title	WebRTC-Based Multi-View Video and Audio Transmission and its QoE
Source citation (APA Format)	Maehara, Y., & Nunome, T. (2019). WebRTC-Based Multi-View Video and Audio Transmission and its QoE. *2019 International Conference on Information Networking : 9-11 January 2019, Kuala Lumpur, Malaysia /, 2019*, 911–186. https://doi.org/10.1109/ICOIN.2019.8718109
Original URL	https://ieeexplore.ieee.org/document/8718109
Source type	Conference Article
Keywords	WebRTC, MVV-A, QoE, UDP , MPEG-DASH
#Tags	#Networks, #Transmission
Summary of key points + notes (include methodology)	● Dynamic Adaptive Streaming over HTTP (DASH) is used to stream audio and video over the internet through web browsers ● This is usually employed through HTTP/TCP connections ● WebRTC is a new way to implement DASH by streaming Multi-View Video (MVV) ● This study aims to implement WebRTC for MVV-A (MVV Audio) transmission and optimizing Quality of Experience (QoE) and Quality of Service (QoS) ● The WebRTC media channel is used to transmit data via a Secure Real-time Transport Protocol (SRTTP) ● Audio and Video data are stored on a server and can be retrieved by any of the viewpoints ● The Media Client was built on the Electron web framework ● Client requests from viewpoints to the server and the server sends it back through the viewpoint to the client ● The audio encoding format is VP8 ● A study was conducted comparing MVV-A and MPEG-DASH ● Results howed MVV-A had a higher QoE than the MPEG-DASH ○ WebRTC uses UDP which is much more resilient to delay ● WebRTC overall shows more resistance to delay and better performance under load

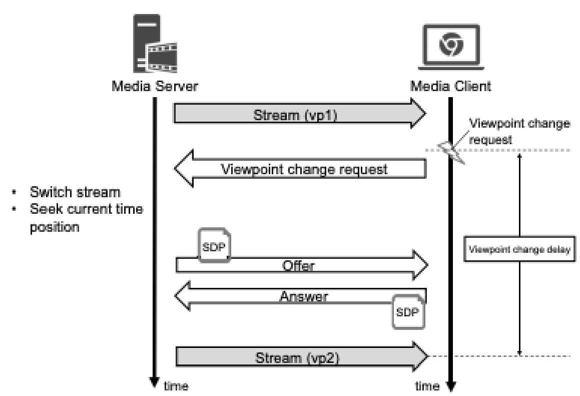
<table>
<tr><td></td><td>The article titled "WebRTC-based Multi-View Video and Audio Transmission and Its QoE" proposes a WebRTC-based MVV-A (Multi-View Video and Audio) transmission system to enhance QoE (Quality of Experience) under large delay conditions. The system employs SRTP (Secure Real-time Transport Protocol)/UDP for audio and video transmission. It allows users to watch contents from four viewpoints while selecting a viewpoint arbitrarily. Audio and video data are stored on the server beforehand. The client requests the chosen viewpoint by the user to the server, and the server transmits the requested viewpoint data to the client. The experiment compares the QoE of the WebRTC-based MVV-A transmission with that of MPEG-DASH. The results show that the MVV-A transmission achieves higher QoE than MPEG-DASH under large delay. This is because WebRTC employs UDP, which is more resilient to delay than TCP. The authors of the article also discuss the effect of network delay on QoE. They find that as network delay increases, the QoE of both decreases. However, the QoE of the MVV-A transmission decreases at a slower rate than that of MPEG-DASH. This is because WebRTC can adapt to network conditions more effectively than MPEG-DASH. Overall, the article provides a valuable contribution to the field of MVV-A transmission.</td></tr>
<tr><td>Research Question/Problem/ Need</td><td>How much more effective is WebRTC based MVV-A data transmission than MPEG-DASH</td></tr>
<tr><td>Important Figures</td><td>


Fig. 2. Sequence of viewpoint change

Diagram representing the conncetion between the media client and and the media server.</td></tr>
<tr><td>VOCAB: (w/definition)</td><td>SRTP - Secure Real-time Transport Protocol, which is a secure version of UDP that is used for transmitting audio and video data.

Network Delay - The time it takes for data to travel from one point to another in a network

MVV-A - Multi-View Video and Audio, which refers to a video streaming</td></tr>
</table>

	technology that allows users to watch a video from multiple viewpoints simultaneously
Cited references to follow up on	ISO/IEC 23009-1, "Dynamic adaptive streaming over HTTP (DASH) Part1: Media presentation description and segment formats," May 2014. A. C. Begen, T. Akgul and M. Baugher, "Watching video over the web, part I: Streaming protocols," IEEE Internet Computing, vol. 15, no. 2, Mar./Apr. 2011. E. Jimenez Rodriguez, T. Nunome and S. Tasaka, "QoE assessment of multi-view video and audio IP transmission," IEICE Trans. Commun., vol. E92-B, no. 6, pp. 1373-1383, June 2010
Follow up Questions	How would the performance of the system change by different network conditions? How would this system be scaled when increasing the number of viewpoints? Are there any security concerns with WebRTC?

Article #17 Notes: JSPatcher, a Visual Programming Environment for Building High-Performance Web Audio Applications

Source Title	JSPatcher, a Visual Programming Environment for Building High-Performance Web Audio Applications
Source citation (APA Format)	Ren, S., Pottier, L., Buffa, M., & Yu, Y. (2022). JSPatcher, a Visual Programming Environment for Building High-Performance Web Audio Applications. *Journal of the Audio Engineering Society*, *70*(11), 938–950. https://doi.org/10.17743/jaes.2022.0056

Original URL	https://wpi-illiad-oclc-org.ezpv7-web-p-u01.wpi.edu/illiad/illiad.dll?Action=10&Form=75&Value=135180
Source type	Journal Article
Keywords	JSPatcher, web-based VPL, Web Audio AP, DSP algorithms, FAUST
#Tags	#DSP, #ProgrammingLang, #Network, #Transmission, #App
Summary of key points + notes (include methodology)	● Visual based programming language to be more user friendly for non-coders ● With more support for Web Audio API, DSP languages can be compiled to WebAssembly at runtime increasing potential for web app VPLs ● The program is used to create integrations with other JS plugins ● JSPatcher user uses 3 interpretation layers ○ First layer represents low-level JS functions ○ Second layer handles WebAudio nodes ○ Third layer executes compiled code in DSP language ● First 2 layers are “imperiative” for processing the data stream in real time JSPatcher is a web-based VPL (Visual Programming Language) that allows you to build audio graphs using the Web Audio API. Users can use a web browser to graphically design and run DSP algorithms using domain-specific languages (DSLs) for audio processing such as FAUST or Gen. The application can also be used to create interactive programs and shareable artworks online with other JavaScript language built-ins, Web APIs, web-based audio plugins or external JavaScript modules. JSPatcher includes three different interpretation layers. The first layer represents low-level JavaScript features, such as variables, getters, setters and functions. The second layer, called the WebAudio layer, handles the WebAudio nodes. These two layers are “imperative” (the code is interpreted at run time) as the patcher is intended for interactive operation through user interface components as well as for processing the data stream in real-time. The third layer is different as it will execute “compiled” code written using the FAUST DSL, when custom low-level DSP is required. This can be compared to Max/Gen: Max is primarily an imperative VPL but can also include Gen 8 patchers, which are compiled to Max’s DSP modules. When JSPatcher runs a patcher that contains FAUST compiled DSPs, the corresponding WebAssembly code will be executed in an AudioWorklet node, whose processor runs in a dedicated high priority thread. The generated DSP is thus an AudioNode and can still be used as a sub-patcher of an imperative patcher.
Research Question/Problem/ Need	How can a web-based VPL like JSPatcher effectively execute audio processing using domain specific languages?

Important Figures	 Node generated by FAUS
VOCAB: (w/definition)	FAUST - A functional language for audio programming WebAssembly: A portable binary format that allows running code written in various languages on the web AudioWorklet node: A special type of Web Audio node that runs code in a separate thread for improved performance
Cited references to follow up on	V. Lazzarini, S. Yi, J. Heintz, Ø. Brandtsegg, I. McCurdy, and others, Csound: a sound and music computing system. (Springer Publishing Company, Incorporated, 2016). S. Yi, V. Lazzarini, and E. Costello, “WebAssembly AudioWorklet Csound,” in Proceedings of the International Web Audio Conference (Berlin, Germany) (2018 Sep.). T.-C. Hui and S. Ren, “Urban Sound Tales: The Invisible Landscapes – the Sonic Past of the Two Cities,” in Proceedings of the International Web Audio Conference (Cannes, France) (2022 Jul.). https://doi.org/10.5281/zenodo.6769891
Follow up Questions	Would there be advantages of using other languages besides FAUSt for JSPatcher? How much time do people save using JSPatcher vs hard-coding in a DSP Language? What are some possible applications of JSPatcher in the real world?

Article #18 Notes: IoT-based Analysis for Smart Energy Management

Source Title	IoT-based Analysis for Smart Energy Management
Source citation (APA Format)	Huang, G.-L., Anwar, A., Loke, S. W., Zaslavsky, A., & Choi, J. (2022). IOT-based analysis for Smart Energy Management. 2022 IEEE 95th Vehicular Technology Conference: (VTC2022-Spring). https://doi.org/10.1109/vtc2022-spring54318.2022.9860601
Original URL	https://ieeexplore.ieee.org/document/9860601
Source type	Conference Articles
Keywords	Internet of Things, Energy management, Power demand, Wireless communication
#Tags	#AI, #IoT, #Sensors, #Networks
Summary of key points + notes (include methodology)	● Demand for energy is constantly increasing ○ It would be better if we could track energy consumption for users ● To analyze power usage patterns in devices Non-intrusive Load Monitoring (NILM) was developed ● IoT devices can be used to connect devices and sensors over the internet to collect data ● The main issue with monitoring energy consumption is energy disaggregation ○ Energy Disaggregation is used to separate the energy consumption of each appliance via the total consumption readings ● To solve energy disaggregation NILM and ILM is used ○ ILM uses a low scale metering device on multiple appliances ○ NILM uses a smart meter of the whole house and approximates appliances power ● Based on this IoT devices can run and train ML models to analyze users energy usage patterns ● It is also mentioned for best accuracy th emodels could be trained on existing datasets ● Multiple data sets were found and were compared, seen in Important Figures ● Some major possible use cases from this technology was identified, energy

saving recommendation, occupancy monitoring, device diagnosis, etc. The article discusses the use of the Internet of Things (IoT) for smart energy management. IoT networks can be used to collect data from appliances and devices at homes and offices to understand power consumption patterns by individuals and guide users for smart energy usage. This involves collecting real-time power consumption data from appliances and devices and training machine learning (ML) models to analyze energy usage patterns. The document also discusses energy disaggregation, a source separation problem that aims to separate the energy consumption of individual appliances from the total consumption readings of multiple appliances. There are two types of methods for energy disaggregation: Non-intrusive Load Monitoring (NILM) and Intrusive Load Monitoring (ILM). NILM utilizes a smart meter and works based on the total sum of power consumption data where the ILM attempts to approximate the individual appliance characteristics vis lower scale meters per device. The document summarizes the existing public datasets for energy disaggregation, including REDD, BLUED, Smart*, AMPds, iAWE, SustData, UK-DALE, etc.These datasets are compared in Table 1 of the document. Finally, the document proposes potential use cases for smart energy management based on IoT networks, including accurate energy billing, occupancy monitoring, appliance classification, faulty appliance detection, building efficiency, and demand-side management.

Research Question/Problem/ Need

How can IoT smart devices be used to analyze user energy usage patterns?

Important Figures

TABLE I
TABLE 1: COMPARISON OF SEVERAL DISAGGREGATION DATASETS [8]

Dataset	Location	No.of units	Duration	Aggregate Rate	Appliance Rate	Labeled	Year	Avalibility
REDD [1]	USA	6	3-19 days	1Hz&15kHz	1/3 Hz	No	2011	http://redd.csail.mit.edu
BLUED [22]	USA	1	8 days	12kHz	-	Yes	2012	http://portoalegre.andrew.cmu.edu:88/BLUED
Smart* [23]	USA	3	3 months	1Hz	1Hz	No	2012	http://traces.cs.umass.edu/index.php/Smart/Smart
AMPds [24]	Canada	1	1 year	1/60 Hz	1/60 Hz	No	2013	http://ampds.org/
iAWE [25]	India	1	73 Days	1Hz	1Hz	No	2013	http://iawe.github.io/
SustData [26]	Portugal	50	4-17 months	50 Hz	2s/3s	No	2014	https://osf.io/2ac8q/
UK-DALE [27]	UK	4	3-17 months	16 kHz	1/6 Hz	Yes	2015	https://data.ukedc.rl.ac.uk
SmartSim [28]	-	1	7 Days	-	1 Hz	-	2016	https://github.com/klemenjak/smartsim
REFIT [29]	UK	20	2 years	-	8-10s	Yes	2017	https://www.refitsmarthomes.org/datasets/
EMBED [8]	USA	3	14-27 Days	12 kHz	1 Hz	Yes	2018	http://embed-dataset.org/
SynD [30]	-	1	180 days	-	5 Hz	-	2020	https://github.com/klemenjak/synd/

Table containing the list of organized datasets containing power usage data. Each dataset is organized by location, number of appliances, Duration, Year, and appliance power usages

VOCAB: (w/definition)

Blind source separation (BSS) - A problem in signal processing that involves separating multiple sources from a mixed signal.

Disaggregation - The opposite of aggregation, disaggregation is separating or analyzing aggregated data to obtain individual component values.

Load signature - The unique pattern of electricity use over time that can identify an individual appliance.

Cited references to follow up on

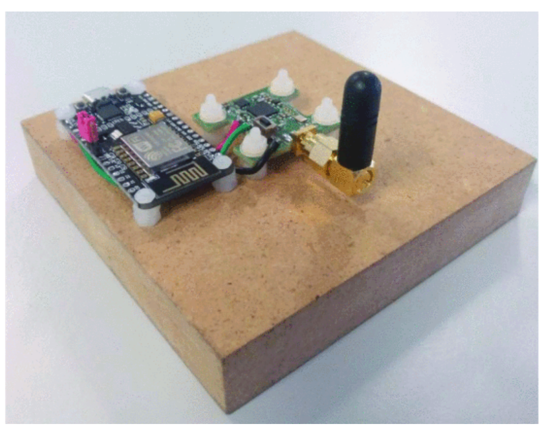
J. Z. Kolter and M. J. Johnson, "Redd: A public data set for energy disaggregation research", Workshop on data mining applications in sustainability (SIGKDD), vol.

	25, pp. 59-62, 2011. S. Henriet, U. Şimşekli, B. Fuentes and G. Richard, "A generative model for non-intrusive load monitoring in commercial buildings", Energy and Buildings, vol. 177, pp. 268-278, 2018. P. Asghari, A. M. Rahmani and H. H. S. Javadi, "Internet of things applications: A systematic review", Computer Networks, vol. 148, pp. 241-261, 2019.
Follow up Questions	What are the potential cost savings to homeowners implementing this technology? How can we address potential security concerns by consumers? Is this technology future proof, might there be better ways that NILM and ILM to monitor power usage?

Article #19 Notes: Design and Implementation of an RSSI-Based Bluetooth Low Energy Indoor Localization System

Source Title	Design and Implementation of an RSSI-Based Bluetooth Low Energy Indoor Localization System
Source citation (APA Format)	Cortesi, S., Dreher, M., & Magno, M. (2021). Design and implementation of an RSSI-based Bluetooth low energy indoor localization system. 2021 17th International Conference on Wireless and Mobile Computing, Networking and Communications (WiMob). https://doi.org/10.1109/wimob52687.2021.9606272
Original URL	https://ieeexplore.ieee.org/document/9606272
Source type	Conference Article
Keywords	Bluetooth Low Energy, Localization, Indoor Localization, Low Power Design, kNN
#Tags	#BLE, #Networks, #AI, #Transmission, #IoT
Summary of key points + notes (include methodology)	● Indoor Positioning System(IPS) is used within hospitals to locate people within the building ● There is a need for a laocalization system accurate to a few meters, for this IoT devices could be used ● BLE was considered to be the best technology to use for this scenario where accuracy, range, energy consumption, and availability are important ● The use of MCU's were also necessary boosting the performance of all the previously stated criteria ● The proposed solution is to use a BLE SoC for indoor localization within hospitals ● BLE iBeacon was chosen as the protocol for RSSI communication between beacons and receivers ○ BLE uses GATT to exchange data between connected devices, as opposed to this iBeacon exchanges data between all possible devices

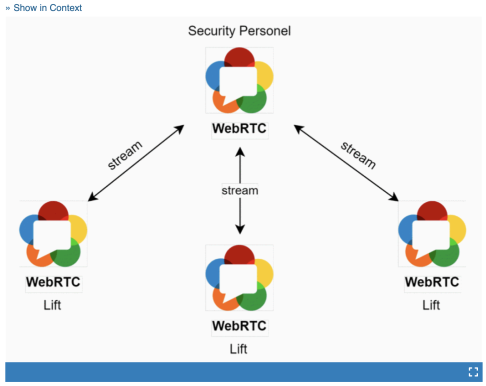
	• A logarithmic distance loss model is used to calculate distance by RSSI and a kNN algorithm is used to more accurately tell distance based on past data and predictions
	• An experiment was conducted in a classroom using 4 beacons to get distances
	    ○ Best case error of 0.27m
	    ○ Worst case error of 2.5m
	    ○
	The Indoor Positioning System (IPS) serves a crucial role in hospitals by enabling the accurate localization of individuals within the building. Recognizing the necessity for a localization system with precision to a few meters, the integration of Internet of Things (IoT) devices emerged as a viable solution. Among the various technologies evaluated for this scenario, Bluetooth Low Energy (BLE) stood out as the optimal choice, addressing key considerations such as accuracy, range, energy consumption, and availability. The incorporation of microcontroller units (MCUs) further enhanced performance across these criteria. The proposed solution advocates for the utilization of a BLE System-on-a-Chip (SoC) specifically tailored for indoor localization within hospital environments. The BLE iBeacon protocol was selected to facilitate Received Signal Strength Indicator (RSSI) communication between beacons and receivers. Unlike traditional data exchange methods, BLE utilizes Generic Attribute Profile (GATT) for connected devices, while iBeacon broadcasts data to all possible devices. To calculate distance based on RSSI, a logarithmic distance loss model is applied, complemented by a k-Nearest Neighbors (kNN) algorithm for more accurate distance predictions based on previous data. Experimental validation conducted in a classroom environment, employing four beacons, demonstrated a remarkable best-case error of 0.27 meters and a worst-case error of 2.5 meters. This study showcases a comprehensive approach to indoor localization in healthcare settings, emphasizing the efficacy of BLE technology and MCU integration for enhanced performance.

Research Question/Problem/ Need	Can a viable indoor positioning system be bult using BLE for use in hospitals?
Important Figures	 Prototype of the receiver including the antenna and WLAN module.

VOCAB: (w/definition)	Edge computing - Performing computation at the edge of the network near the source of data rather than in the cloud. k-Nearest Neighbors (kNN) algorithm - A classification algorithm that assumes the instance belongs to the class of its k closest training examples in feature space. Beacon - Fixed transmitters that broadcast Bluetooth signals with identifiers.
Cited references to follow up on	T. V. Haute, E. D. Poorter, P. Crombez, F. Lemic, V. Handziski, N. Wirstrom, et al., "Performance analysis of multiple indoor positioning systems in a healthcare environment", International Journal of Health Geographics, vol. 15, no. 1, pp. 7, 2016, [online] Available: https://doi.org/10.1186/s12942-016-0034-z. F. Furfari, A. Crivello, P. Baronti, P. Barsocchi, M. Girolami, F. Palumbo, et al., "Discovering location based services: a unified approach for heterogeneous indoor localization systems", Internet of Things, vol. 13, no. nil, pp. 100334, 2021, [online] Available: https://doi.org/10.1016/j.iot.2020.100334. Y. A. Kim, H. Lee and K. Lee, "Contamination of the hospital environmental by pathogenic bacteria and infection control", Korean Journal of Nosocomial Infection Control, vol. 20, no. 1, pp. 1, 2015, [online] Available: https://doi.org/10.14192/kjnic.2015.20.1.1.
Follow up Questions	Is this solution compliant with all HIPAA laws? How durable are these devices, can they withstand the hospital environment? Will the accuracy decrease as the number of beacons scales?

Article #20 Notes: Internet of Things Intercommunication Using SocketIO and WebSocket with WebRTC in Local Area Network as Emergency Communication Devices

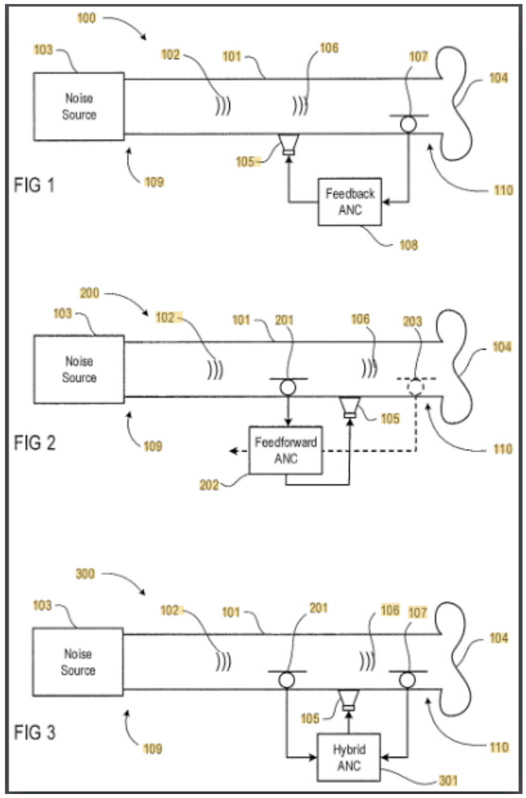
Source Title	Internet of Things Intercommunication Using SocketIO and WebSocket with WebRTC in Local Area Network as Emergency Communication Devices
Source citation (APA Format)	N. I. Ariffin, M. A. S. Hamdan and S. F. Kamarulzaman, "Internet of Things Intercommunication Using SocketIO and WebSocket with WebRTC in Local Area Network as Emergency Communication Devices," *2023 IEEE 8th International Conference On Software Engineering and Computer Systems (ICSECS)*, Penang, Malaysia, 2023, pp. 268-273, doi: 10.1109/ICSECS58457.2023.10256297.
Original URL	https://ieeexplore.ieee.org/document/10256297
Source type	Conference Article
Keywords	Emergency Services, Throughput, Quality Assessment, Resource Management, Internet of Things, VIdeo recording, Local area networks
#Tags	#Networks , #Transmission, #IoT
Summary of key points + notes (include methodology)	● Emergency systems are becoming more important than ever before since people are needing adaptive technology integrated solutions ● By analyzing the technology, specifically web recording, we can begin to see how a market opens to further analyze the potential benefit ● In Malaysia and New York City, large buildings require extra safety precautions that can help preserve the security of everyone in these areas ● WebRTC is unique in the sense that it has the capability to transfer, save, and move data in a more efficient manner ● Socket.io may be utilized to help ensure that the emergency services get these notifications as soon as possible to ensure that there is an overall development. ● This will result in increased video quality and ensure that these identification mechanisms is beneficial. ● WebRTC is a simulation tool that is paired with WebConnect to mediate the proper flow of these materials from one data storage center to another over time. ● Socket.io is a fallback option for if there were intermediate difficulties with the existing software ● The LAN Device can be modified to ensure that each data set is stored accurately and precisely. ● Privacy concerns are addressed with the understanding that these video

	footage will be in limited, short systems to ensure that there is proper project development. ● Results ● 640 by 480 pixels was 30 frames per second permitting the privacy concerns ● The latency within the 40 millisecond to 120 milliseconds in varying models, so Design 3 of the systems diagram would be optimal in this particular case ● Devices 2 and 4, analyzed within figure 2, had absolutely no dropped packages signifying that data is transferred rapidly and effectively through short periods of time. ● Quality of the video may be improved in the future, but the overall length of the videos was sufficient for the purposes of this particular project
Research Question/Problem/ Need	Determining the overall efficiency of the Emergency of Systems (EOC) algorithms pertaining to the security of individuals?
Important Figures	Fig. 1. » Show in Context  Figure 1 details the ideology of how intercommunication occurs between different WebRTC, especially given their recent advancement in computational capability
VOCAB: (w/definition)	Emergency Warning and Intercommunication System ("EWIS") - type of warning system used to warn the occupants of a building in the event of a fire or other emergency. WebRTC (Web Real-Time Communication) - technology that enables Web applications and sites to capture and optionally stream audio and/or video media, as well as to exchange arbitrary data between browsers without requiring an intermediary. local area network (LAN) - network contained within a small geographic area, usually within the same building.
Cited references to follow up on	C. CTBUH, *Malaysia the Skyscraper Cen-ter*, May 2023, [online] Available: www.skyscrapercenter.com. J. Cui and Z.- Y. Lin, *Research and Implementation of WebRTC Signaling via*

	WebSocket-based for Real-time Multimedia Communications, Feb. 2016, [online] Available: https://doi.org/10.2991/iccsae-15.2016.72. N. NST, "Study Shows lift Escalator Accidents Due to Poor main-tenance Wrongful Use and Vandalism", *New Straits Times*, Oct. 2018, [online] Available: https://www.nst.com.my/news/nation/2018/10/425633/study-shows-lift-escalator-accidents-due-poor-maintenance-wrongful-use.
Follow up Questions	How can other large language models integrate the conclusions within this paper to better their algorithms detection, especially in cases that determine safety of people? How may services that provide the security aspect of an individual's life be maximized? What are potential other solutions that the researchers found but disregarded because they thought it wasn't optimal?

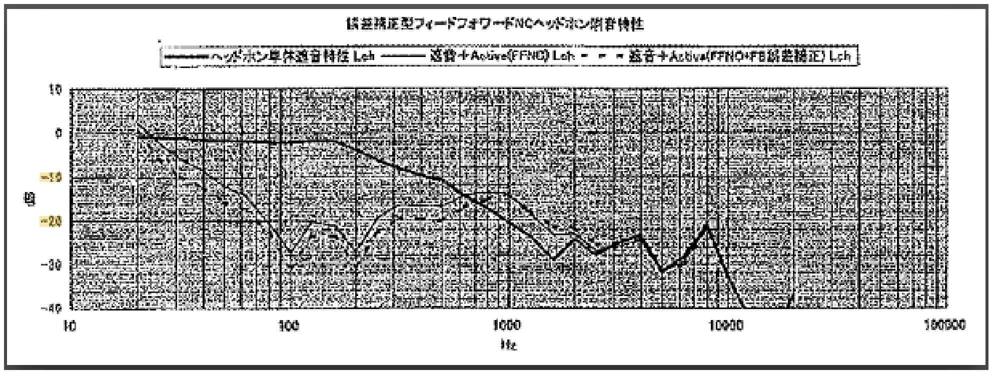
Patent #1 Notes: Active noise reduction in headphones

Source Title	Active noise reduction in headphones, US10721555B2
Source citation (APA Format)	Christoph, M. (2020, July 21). Active noise reduction in headphones.
Original URL	https://patents.google.com/patent/US10721555B2/en
Source type	Patent
Keywords	Active noise reduction, Headphones, Rigid cup-like shell, Microphones, Sound pickup, Electrical signals
#Tags	#ANC, #DSP
Summary of key points + notes (include methodology)	This patent describes an improvement in active noise reduction headphones. The conventional active noise reduction systems include feedback, feedforward, and hybrid configurations. In feedback systems, a microphone is placed in the acoustic path from the noise source to the listener's ear, and noise is reduced based on the microphone's signal. Feedforward systems position the microphone between the noise source and the speaker, while hybrid systems combine both approaches. The proposed improvement involves a headphone with a rigid cup-like shell containing a cavity. The headphone includes a microphone arrangement with at least three microphones regularly distributed over the outer surface of the shell. These microphones pick up sound and provide an electrical signal. An active noise control filter processes this signal to generate a second electrical signal. A speaker in the cavity generates sound based on the second electrical signal. The active noise control filter is designed to reduce noise that travels through the shell from outside to inside
Research Question/Problem/ Need	How can we effectively reduce noise emitted by multiple sources coming from various directions in noise canceling headphones?

Important Figures	 Shows DSP loop of different types of ANC approaches
VOCAB: (w/definition)	Transfer function - A mathematical description relating the output of a system to the input in the frequency domain Feedforward - A noise cancellation technique using a reference microphone to predict noise before it reaches the ear Feedback - A noise cancellation technique using an error microphone to measure noise that has reached the ear
Cited references to follow up on	Patent US4654546A Patent US20080159555A1 Patent CN101257729A
Follow up Questions	What are the advantages of using an areal microphone over a single microphone in terms of noise reduction performance? How does the proposed headphone design compare to existing noise-cancelling technologies on the market? Are there any considerations or challenges in terms of power consumption that users should be aware of when using these headphones?

Patent #2 Notes: Noise cancel headphone

Source Title	Noise cancel headphone, JP2012023637A
Source citation (APA Format)	Kimura, T. (2012, February 2). Noise cancel headphone.
Original URL	https://patents.google.com/patent/JP2012023637A/en?q=(noise+cancelation+headphones)&oq=noise+cancelation+headphones
Source type	Patent
Keywords	Feedforward noise cancellation, Feedback noise cancellation
#Tags	#ANC, #DSP
Summary of key points + notes (include methodology)	The patent contains a design to a noise-canceling headphone that addresses the limitations of conventional feed-forward noise-canceling headphones by incorporating a feedback-type error signal correction circuit. Conventional noise-canceling headphones analyze noise collected by a built-in microphone and generate sound waves with opposite phases to cancel out the noise heard by the user. The invention combines a feed-forward noise-canceling method with a feedback-type error correction circuit to improve performance. Feedback methods are known for their ability to reduce external noise across a broader frequency range, while feed-forward methods can theoretically eliminate noise completely if phase and level are perfectly matched. However, each method has its weaknesses. The disclosed noise-canceling headphone aims to enhance noise canceling by combining features of both methods and addressing their weaknesses. It comprises a housing, a headphone unit within the housing, and a microphone unit facing outward for collecting external noise. The headphone uses a feed-forward method to cancel noise and includes an error detection microphone unit in the front air chamber to detect noise not canceled by the feed-forward method. An error correction signal is generated using a feedback method to cancel this remaining noise.
Research Question/Problem/ Need	Is it possible do design ANC headphones that address the limitations of both a feedback and feedforward system?

Important Figures	 Graph showing noise reduction using the newly developed algorithm.
VOCAB: (w/definition)	Front air chamber - Space between the earpad, ear canal, and headphone housing where noise can still exist after feedforward cancellation Error noise - Noise that the feedforward system fails to fully eliminate in the front air chamber Error correction signal - Signal generated by the feedback system to cancel out residual error noise
Cited references to follow up on	Patent CA2107316C Patent SG106582A1 Patent EP1688910B1
Follow up Questions	Does the error noise affect sound quality of the headphones? How was the error correction signal calculated? How would this methodology change if increasing the number of microphones?

Patent #3 Notes: Noise reduction headphones testing apparatus and method

Source Title	Noise reduction headphones testing apparatus and method, JP5589156B2
Source citation (APA Format)	ヤン ホアソン リウ. (2014, September 17). Noise reduction headphones testing apparatus and method.
Original URL	https://patents.google.com/patent/JP5589156B2/en?q=(headphones)&oq=headphones
Source type	Patent
Keywords	Test apparatus, Sealed chamber, Noise source, Test panel
#Tags	#DSP, #ANC
Summary of key points + notes (include methodology)	The invention addresses the challenge in testing noise reduction headphones, where high-power external noise sources are typically required for accurate testing. The use of such sources can lead to noise pollution in the testing environment, particularly in production plants with high background noise. The invention provides a noise reduction headphone test apparatus that includes a sealed chamber, a noise source, a test panel, a measurement microphone, and a measurement comparison module. The sealed chamber isolates the sound emitted from the noise source. During testing, the test panel can be combined with a noise reduction headphone to create a coupling chamber. The test panel has a sound introduction hole for transmitting noise source sound into the coupling chamber. The measurement microphone records noise signals before and after activating the noise reduction function, and the measurement comparison module calculates the noise reduction amount.
Research Question/Problem/ Need	How can we better test the efficiency of noise cancellation headphones and their true noise reduction?

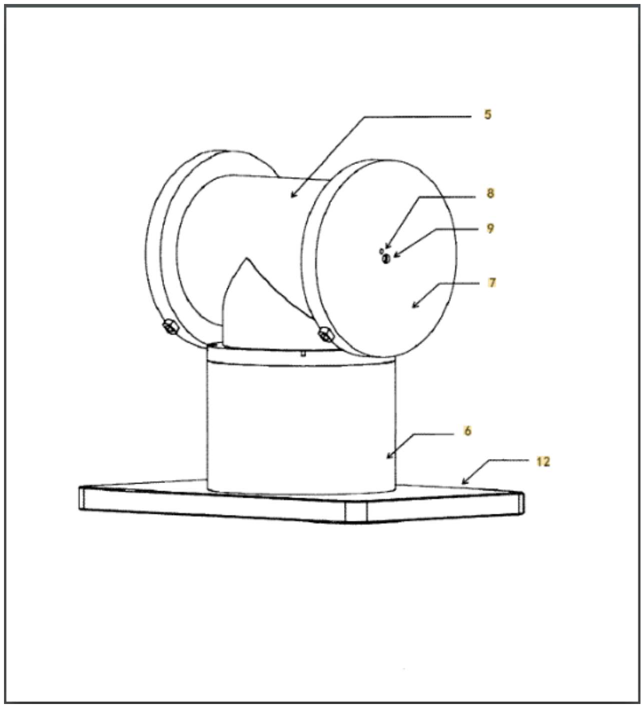
<table>
<tr><td>Important Figures</td><td>

Image of the sealed chamber.</td></tr>
<tr><td>VOCAB: (w/definition)</td><td>Transfer Function (G, H) - A mathematical representation of how a system, in this case, the headphone, responds to input signals

Coupling Chamber - A space formed when the test panel is combined with a noise reduction headphone, used for testing noise reduction

Cavity: A hollow space, such as the sealed chamber</td></tr>
<tr><td>Cited references to follow up on</td><td>Patent US6086541A
Patent WO2001026272A2
Patent WO2001052737A1</td></tr>
<tr><td>Follow up Questions</td><td>How does this compare to existing measurement models?

How could they convey the results to make it intuitive?

Does this cost more to manufacture than current methods?</td></tr>
</table>