In-Room Noise Cancellation with Wall-Transmission Detection

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I. INTRODUCTION

Ambient noises are omnipresent and have the potential to be distracting. However, some of the most distracting noises come from construction machinery, weather, and people outside of the room. This project will use an active noise control device to measure sounds traveling through walls from outside and attenuate the sound to create a quieter environment inside. This conceptual design will evaluate all the necessary components and systems ranging from inputs, processing, and outputs. This document will be precise enough to allow the project to be duplicated by others while allowing enough flexibility to choose different hardware and software to achieve this design.

The team expects this device to work in a classroom, so several subsystems will be designed and discussed that make up the active noise control device catered towards this specific constraint. Sources that play a major role in the specifications and constraints of the device are professional engineering ethics, safety standards, and derived requirements. Verification methods will be centered on a "build a little, test a little" approach to effectively allow dynamic construction and flexible changes to design as unknowns are resolved.

A) Fully Formulated Problem

The device is expected to capture the sound or vibration coming from outside through a wall and output the negative of the captured signal to produce a perceived quieter environment. The functional expectations and constraints (listed in alphabetical order) include:

A System shall output a warning signal if sounds reaching over 80 dB are outputted for a prolonged period of time, specifically 8 hours or more.

- This standard comes from the OSHA safety regulations. These standards set a limit on prolonged noise exposure to sounds above 80 dB. If an individual is exposed to these loud noises for an extended period of time, it could cause permanent damage to one's eardrum. Therefore, in the case of a critical failure, the system will send a warning when potentially harmful noises are being outputted. Then proper measures can be taken to shut down the device.

B System shall have a primary input sensor that measures acoustic vibrations through a physical medium.

- This constraint is derived from the need of the system to acquire vibrational acoustic data through a physical medium as an input.

B.1 Primary input sensor shall output an appropriate type of signal that is compatible with connected subsystems.

- This constraint is derived since the sensor must be able to communicate with the processing block. The processing block can only operate on certain data types and signals. The output of this sensor must therefore be an appropriate signal and data type, such as a continuous-time analog voltage signal, in order to be used properly.

B.2 Primary input sensor shall output a signal with minimal external interference generated by its power supply.

- This standard originates from the potential of the power source to emit a frequency that adversely affects the input of the system. Measures will be put into place to ensure the power supply emits a clean frequency with minimal interference.

C System shall convert an analog input to a digital sample for use in processing.

- This constraint originates from the need to convert the analog signal received from the microphone into a data type that the processor can properly use. The processor will not be able to properly analyze an analog signal, therefore it must be converted into a digital one.

C.1 Analog to digital conversion shall sample at a minimum rate of 40 KHz

- In order to prevent aliasing, the minimum sampling rate must be double the highest frequency inputted into the system. This value comes from the Nyquist theorem [1]. The frequency range of human hearing is about 5 Hz to 20 kHz. The system must therefore have a sampling rate of twice the maximum frequency audible to people.

C.2 System shall use the full dynamic range of the ADC in order to minimize quantization error.

- This constraint originates from the need for accuracy when developing this system. The higher the quantization error the less noise attenuation that will be achieved.

C.3 The system shall be able to prevent unwanted distortion from input signal clipping.

- Clipping can create harsh distortion on a signal. In order to create a system that minimizes distortion the system shall implement an appropriate form of clipping.

C.4 The system shall have a minimum bit depth of 16 bits.

- As bit depth increases, quantization noise decreases. A bit depth of 16 bits allows a dynamic range and noise ratio of approximately 100 dB SPL (Sound Pressure Level). This is approximately the SPL of heavy machinery and construction [2].

D System shall have an error sensor that measures acoustic residual noise based on the primary input sensor.

- This constraint originates from the need to measure the accuracy of the emitted output from the system. If the emitted noise is inaccurate in comparison to the reference input, the error sensor will pick this up and send the data back to the processor for correction.

D.1 Error sensor shall output an appropriate type of signal that is compatible with connected subsystems.

- This is a derived constraint since the sensor must be able to communicate with the processing block. The processing block can only operate on certain data types and signals. The output of this sensor must therefore be an appropriate signal and data type, such as a continuous-time analog voltage signal, in order to be used properly.

E System shall have a primary output speaker that generates acoustic sounds based on its input from the system.

- This is an explicit device requirement since the device needs to output a sound to reduce noise from the wall.

F System shall have a primary processing subsystem that is able to receive input, process the input, and output the appropriate response.

- This constraint is an explicit requirement since the output must be processed from the input.

F.1 Current and previous input samples must be temporarily stored.

- A causal system is dependent on the current and previous inputs.
- The number of samples held is dependent on the size and type of filter used in order to insure proper convolution.
- The number of samples held is dependent on the maximum system delay allowed since group delay increases as length increases.

F.2 Current and previous error samples must be temporarily stored.

- A causal system is dependent on the current and previous inputs.
- The number of samples held is dependent on the size and type of filter used in order to ensure proper convolution. The type of adaptive filter algorithm used will also determine the size in order to ensure proper functionality.
- The number of samples held is dependent on the maximum system delay allowed since group delay increases as length increases.

F.3 Signal shall be processed before phase shifting to account for the approximate acoustic response of the room.

F.3.a System shall adapt for small changes in the acoustic response of the room.

- System shall have an approximation of the acoustic response of the room between each input/error combination in a general environment.
- System shall adapt to small changes in the acoustic response over time.

F.3.b System shall adapt for significant changes in the acoustic response and soundscape of the room.

- System shall be able to classify the acoustic responses and sounds currently being processed by the system and make changes to processing accordingly.
- System shall have an approximation for acoustic responses for reasonably foreseen environment changes.

F.4 System shall phase shift the processed signal accordingly for proper destructive interference of the targeted noise.

- The principal mechanism of active noise control is the destructive interference of acoustic sound waves. Destructive interference occurs when two superimposed sound waves are 180° out of phase (opposite phase).

G System's primary process shall have a maximum delay of 1.4 ms from input to output.

- The system must be able to output the anti-noise as soon as the targeted sound reaches the output speaker. Constraint **N** specifies a maximum distance the sound can travel before the system must output its sound. Combined with the speed of sound (350 m/s at +30°C) gives a maximum delay of 1.4 ms.

H Device shall not reach 60°C (140°F).

- Excessive heat, either external or generated internally must be controlled since the device will be easily accessible. The "Standard Guide for Heated System Conditions that Produce Contact Burn Injuries (C 1055-92)," issued by the American Society for Testing Materials (ASTM) recommends that touchable surfaces be at or below 60°C because the average person can touch this kind of surface for up to five seconds without sustaining irreversible damage.

I System shall be powered by standard wall outlets.

- This constraint is derived from the desire for the system to be implemented long-term. By connecting the system to a standard wall outlet, it eliminates the need for temporary battery power and the system can run for a longer period of time.

J System shall have accessible power toggle.

- This constraint originates from the safety standards in place for the system and the stakeholder requests for accessibility. If the system encounters a critical failure or needs to be shut down for any other reason, there will be an easily accessible power toggle that can shut down the system.

K System shall follow OSHA standards 1910.95 for occupational noise exposure and 1910.304-305 for wiring design and protection. [3]

- This constraint originates from the need of the system to follow a proper set of safety regulations. OSHA contains many critical regulations that involve emitted sounds and wall power safety which are imperative in this project, therefore these OSHA standards will be followed in both design and in implementation.

L System shall refer to IEEE 518-1982 Guide for the Installation of Electrical Equipment to Minimize Electrical Noise Inputs to Controllers from External Sources. [4]

- This constraint originates from the need to eliminate excess electric noise from interfering with low voltage signals that the system produces. This standard helps to guide the safe implementation of a way to reduce excess electric noise.

M System shall follow standard IEC 60950-1 for information technology equipment. [5]

- This constraint originates from the use of wired and battery powered information technology within this design. This standard outlines some of the general safety requirements that go along with wired and battery powered devices so it is appropriate for this standard to be followed in the design and implementation of the device.

N System shall not impede movement and easy access throughout the room. The speaker output shall not be more than 0.5 meters away from the window or wall being measured.

- This constraint involves accessibility concerns that are strictly accommodated to in public schools. The American Disabilities Act ensures all environments across campus are made accessible to all students. Many classrooms have limited space and little room for adjustment, so leaving a device in a critical pathway of a classroom would cause many problems for individuals with limited mobility.
- The desire for a discrete system that does not limit mobility was also requested in the survey given to stakeholders. It was specifically requested that the device not be bulky or cause problems for students moving throughout the classroom.

O System shall minimize its visual distraction.

- This constraint originates from the survey given to stakeholders. Along with not impeding mobility, the device will not be a visual distraction to students and teachers. It will be implemented as discretely as possible, outside of the the direct line of sight of students.

P System shall attenuate targeted noises by at least 20 dB.

- This constraint originates from the critical measure of success for the system. While an ideal noise-cancellation system will be able to fully cancel out noises, the main goal of this project is to attenuate unwanted signals by 20 decibels.

Q System shall follow the American National Standards Institute (ANSI) for safe use of lasers rules Z136.1, Z136.5, and Z136.8. [6] [7]

- This constraint comes from the ANSI regarding the use of lasers for general use and in classrooms. These standards ensure that the correct lasers are being used and safety measures are being followed. Since they ensure safety, care should be taken to follow relevant standards ANSI set.

R System shall have a mount or multiple mounts so that it can be easily installed within the classroom in an unfrequented area.

- This constraint was derived from the need for accessibility of the device and the room in which it occupies. With a mount, the device will not have to rest on the floor or furniture and will not be an obvious distraction. This creates more options for implementation and helps to fulfill constraints **N** and **O**.

S System shall not store the audio data for longer than needed to process.

- This constraint comes from ethical considerations involving privacy within a classroom. Storing audio long-term is not a requirement for the device to function properly, so it should not be included to minimize unnecessary ethical questions.
- Depending on the country or state, two-party consent laws may be a factor when limiting the device's audio storage capacity.

T Shall be easy to remove and setup in a different room.

- This constraint was derived from consideration of broader impacts: Given the adaptability of the device, it is expected to be moved around to different positions in a room and to different rooms.

II. Ethical, Professional, and Standards Considerations

With any device implemented that could have an affect on students and staff, there are broader considerations that must be taken into account. As the device is meant to be used by individuals with no technical experience, constraints are put in place to ensure the safety and intended effects for everyone to come into contact with the device.

A) Ethical

Along with the use of microphones and recording devices, there comes the ethical consideration of individual privacy. To account for this, the privacy and safety that is expected while conversing in the classroom will not be breached and accessed at a later date.

There are also ethical implications associated with the implementation of the device impeding movement within the classroom. The device should be implemented in such a way that nobody will be negatively affected by its placement and position, which could potentially lead to distraction or reduced performance for individuals within the classroom.

An ideal system would only cancel out the noise from outside and not accidentally affect other noises in the room. There is the chance that the system will cancel out unforeseen noises within the room such as voices. With this in mind there might be certain individuals that are closer to the speaker that will not be able to here other noises within the classroom as well as other students can. Extra care will be taken to try and reduce this risk as efficiently as possible.

As engineers, the safety of the device is a very important matter to consider. The system should not function in such a way that could cause harm to an individual or individuals. If the system is capable of causing harm, that will be considered a critical failure.

B) Standards

If a laser is utilized in any portion of the device, whether it be in the input or output, the design will follow the safety regulations outlined by the American National Standard for safe use of lasers. These standards constrain the types of lasers that will be used, the area in which they are implemented, and ensure safe use of light emitting devices. [6] [7]

Due to the use of acoustic recording and sound emission devices and the methods that will be used to power them, the standards outlined by the International Electrotechnical Commission will be followed. These standards help to ensure safety when using audio, video, and other powered devices [5].

Standard IEC 60950-1 deals with wired and battery powered devices below 600 V [5]. This device will make use of wall power so this standard will be followed to eliminate harm from electric shock, fire, and mechanical instability.

IEEE standard 518-1982 is a guide for the installation of electrical equipment to minimize electrical noise inputs to controllers from external Sources [4]. In this design a filter will be implemented to reduce the electric noise give off from the power source. Although this standard is no longer a requirement in devices, it is a good source to follow to ensure a safe implementation and solutions is implemented.

OSHA standard 1910.304-305 provides an in depth guide for safety and regulation of wired power [3]. Whether this be from a battery or through the wall these standards include safe regulations to follow and implement. This system will be dealing with power from a wall outlet so it is important that these standards are followed.

C) Broader Impacts

Due to the limited time and money allocated to this project and design, there are many broader impacts, both positive and negative, that must be considered.

In the event that the system does not work properly or effectively, it may have the opposite functionality of what was originally intended. The device has the potential to generate extra noise instead of anti-noise if designed incorrectly. There is also the chance that does not generate any noise at all due to some other subsystem in the device not transmitting properly.

The goal of this design is to reduce the noise that comes from outside of the designated room. This is under the assumption that the sound within the room is generally quiet. If the noise inside the room because drastic or excessive it is unclear what effects this could have on the system. Extra testing will need to be done to ensure the noise within the room does not negatively affect the capabilities of the outside-noise canceling system.

Once this design is implemented within the classroom, it will be imperative to let those who use the room know of the addition and provide them with a general understanding of how the system works. In the event that the system malfunctions and manual steps need to be taken, it will be important that individuals know what to do and how to power the device on or off or remove it. It will also be necessary to explain what not to do to the device. While the device will attempt to not be a distraction it will still be accessible to the regular student, so to ensure safety they must be aware of general regulations regarding interaction with the device.

Due to the lack of research on active noise cancellation within a general room rather than through headphones, it is unclear whether or not the generated anti-noise could cause headaches or sensory disruption among individuals. Testing will need to be done to test the effects of anti-noise on individuals and see if the noise is truly canceled out.

Ideally, the system will have high adaptability and be transferable between different areas. If the system is successful, then it can be used in multiple settings, drastically expanding its impact on the school and the potential noises that can be reduced.

Another greater impact that this system could impose is an effect on other noise cancellation devices: Whether or not the device outputs a signal that negatively affects other nearby devices should be considered. Testing will need to be done to ensure that individuals' devices are not negatively affected.

D) Constraints Derived From Broader Considerations

Constraint **S** is directly derived from the consideration of privacy ethics. This constraint emphasizes that the device will not store the audio that it records. Instead, it will analyze it and send it to the next block.

Constraint **T** is derived from the broader considerations of the future use of the design. By constraining the design to be relatively easy to remove and setup, the overall design and structure of the system is influenced.

Constraints **J** and **A** are derived from the consideration for the safety of the device in mind. In the case that the device experiences a critical failure and is outputting a potentially harmful signal, it is imperative that the system can be shut off or limit the time it emits such signals.

III. BLOCK DIAGRAM

A) Input Channel

The input sensor will be the primary source of data for the rest of the system. This is where acoustic sound signals will be measured and prepared so that they can be sent to processing for further manipulation. This subsystem block will consist of a sound sensor and a method of amplification.

Input: Acoustic sound through wall

Output: Digital signal sample

The input sensor will follow Constraints **B**, **B.1**, and **B.2** outlined in the in depth constraints section. These specify both the way in which the signal will be acquired and the state the signal should be in once it is transmitted. Proper functionality can be verified during design and testing by confirming that an acoustic signal is being received by the input sensor and the waveform has little to no interference.

1) Sound Sensor

The sound sensor will be responsible for acquiring the raw acoustic data through the wall and turning it into a waveform that can then be filtered and amplified by the rest of the system. This can be verified by sending the signal to a speaker with a flat input response, to determine if correct frequencies are being obtained.

Input: Acoustic sound from physical medium

Output: Analog voltage signal

This part of the system will follow the specifications outlined by constraint **B**. Which ensures that that the method used to acquire an acoustic noise signal is operating properly.

2) Pre-Amplification

The signal will be amplified before being sent to the analog to digital converter in order to make use of the full dynamic range of the ADC. The level of amplification is determined by minimization of quantization error. This is hardware specific and depends on the dynamic range of the analog to digital converter.

Input: Analog voltage signal

Output: Amplified analog voltage signal

The constraint that this part of the system will follow is **C.2** and **B.1** which states that the full dynamic range of the ADC will be utilized so that the system is as accurate as possible. It is imperative that this part of the system functions properly so that the processing block has an appropriate signal to utilize.

3) Analog to Digital Conversion

In order for the signal to be processed it needs to be digital, so the analog signal will be converted into a digital samples. To achieve this an analog-to-digital converter(ADC) will be utilized. There will be small sources of error that the system needs to be able to acknowledge and minimize. This can be verified by confirming that a digital signal output has the same properties as a predetermined input.

Input: Analog voltage signal

Output: Digital signal sample

The constraints that will be followed by the ADC converter are **C**, **C.1**, **C.2**, **C.3**, and **C.4**. This part of the system must be able to take in an analog input and convert it to a digital signal. This part of the system must also be sampled at at least 40 kHz in order to ensure the outputted samples properly represent the analog input signals. The quantization error must also be minimized to make sure the desired attenuation of 20 dB is achieved. In order for the digital signal to not experience distortion a soft clipping scheme will be utilized. The digital samples will also require a minimum bit depth of 16 bits. This will give the system enough samples to properly counteract noise in the processing section. These constraints can be tested by measuring and observing the signals, and resourcing an ADC with a bit depth of at least 16.

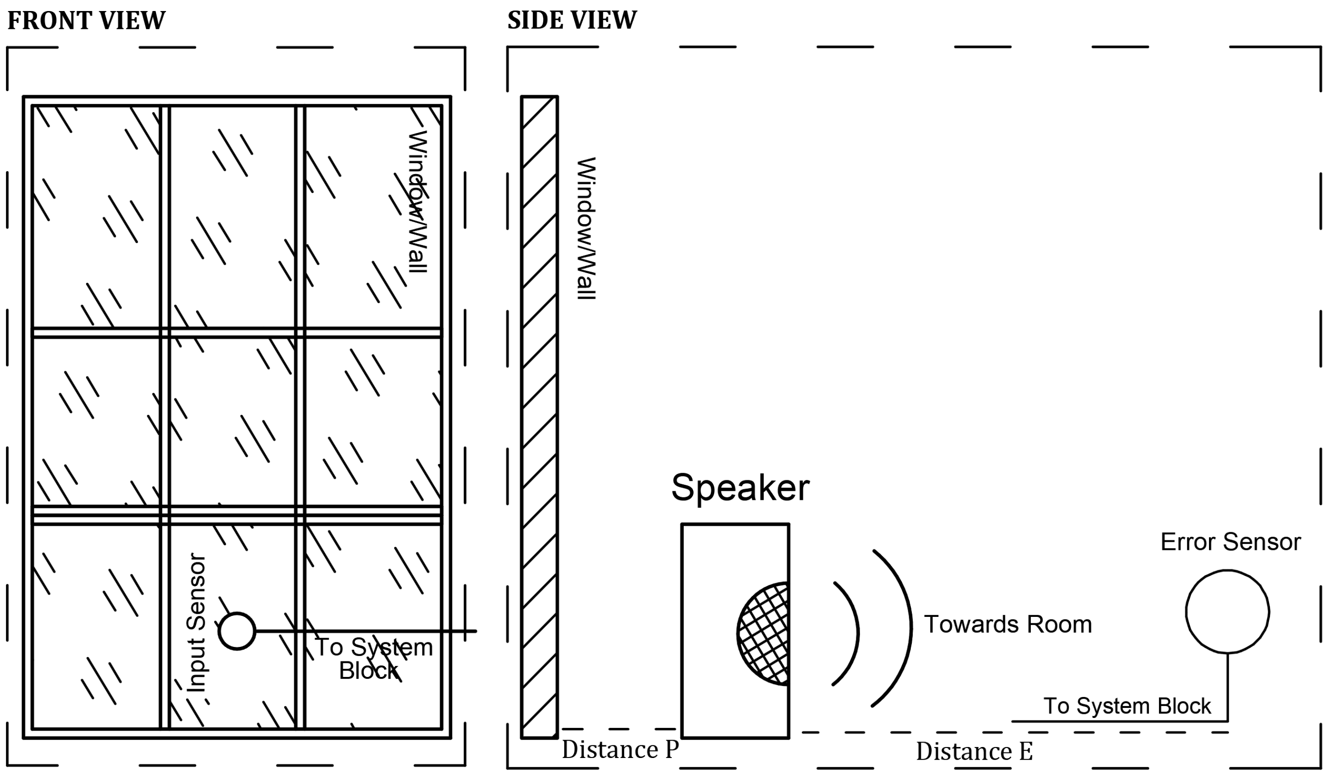

Fig. 1. System Layout In Relation To The Wall

B) Processing

The main functionality of the processing subsystem is to do the processing needed to achieve a properly phase shifted signal that will create a quite zone within the room. If the noise from outside is not properly attenuated then it is likely a problem with the processing subsystem, this information will be used to verify correct operation.

Input: Digital signal samples from error and input channels

Output: Digital signal samples to output channel

The processing subsystem will conform to all constraints falling under **F**. These constraints ensure that the processing is done correctly, while limiting distortion, properly storing usable data, adapting to changes large or small in the environment, and properly canceling out the noise that is received.

1) Input and Error Delay Line

Holds current and previous inputs to be used in the processing of the system in an array. The values in the array will be checked to confirm that they are the same values that have been inputted into the subsystem. The input delay line will need to correctly hold the previous and current inputs, this could be tested by observing the data stored in the input delay line. The maximum amount of information allowed to be stored must also be calculated in order to ensure the delay line holds enough data for optimal accuracy. This can be done by calculating the maximum delay of the system since the maximum information that can be stored is a function of the maximum delay.

Input: Digital signal samples

Output: Digital signal samples

This part of the system must follow constraints **F** and **F.2** The first constraint is required for this part of the system because the delay line must output the response needed for correct functionality. The previous and current samples must also be stored. The maximum system delay must also be considered in order to get an accurate understanding of how many samples can be stored in the input error delay line.

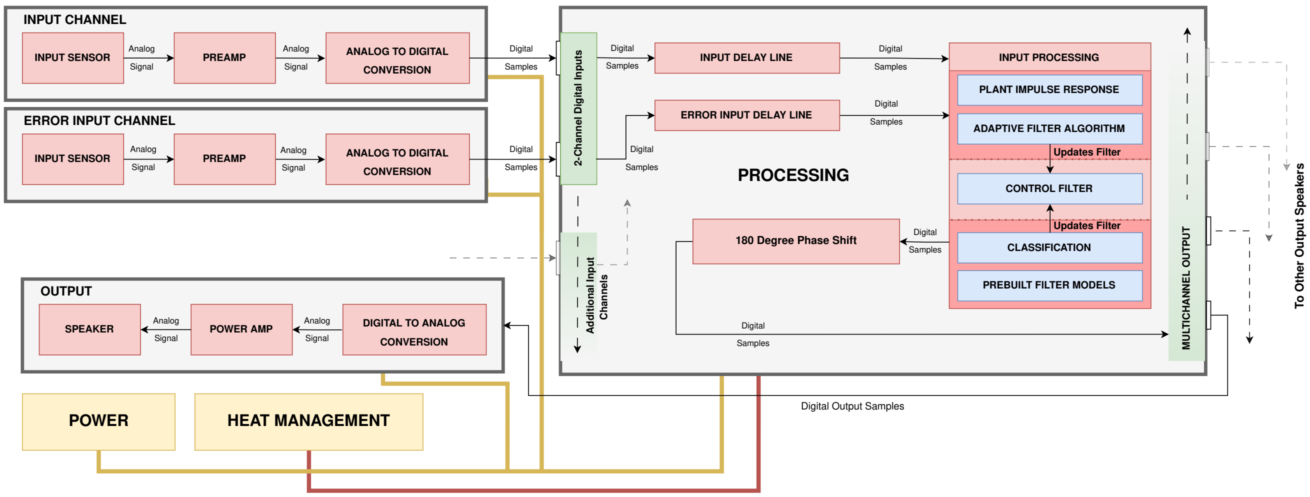

Fig. 2. Function Diagram for System Block

2) Input Processing

The Input processing block accounts for the acoustic response of the room through the use of simulated results and adaptive algorithms. The functionality will be verified by checking the output of the system and confirming that the error is decreasing. This block is the main block responsible for attenuating the noise signal, so if this block has an error then the overall functionality of the system will decrease.

Input: Digital signal sample from error and input delay lines

Output: Processed digital signal samples to the phase shifter

This part of the system will follow constraints **F.3**, **F.3.a**, and **F.3.b**. These state that the system must be able to deal with changes in the acoustic response of the room in order to properly cancel out unwanted noise. In order for this part of the system to properly function, the processing block must be able to quickly and efficiently adapt to changes in the environment.

I. Plant Impulse Response:

The Plant impulse response holds a simulated approximation for the acoustic response for the room. This will be used as a baseline acoustic response that the adapting algorithms can use as a starting point to minimize optimization time.

Input: Filter coefficients from simulated results

Output: Filter coefficients

The plant impulse response will follow constraints **F.3** and **F.3.a**. The main purpose of a plant impulse response is to set an ideal response for the system to be able to model itself off. This helps the system deal with small changes in the environment and remain consistent. The other components of input processing will aid in the adaptation of the plant impulse response.

II. Adaptive Filter Algorithm:

The goal of the adaptive algorithms is to minimize error overtime through the iterative updating of the plant impulse response. This updates the control filter so that attenuation can be maximized. If the system is reducing error and updating itself then that means that adaptive filter algorithms are working properly.

Input: Plant Impulse Response, Input Delay Line, Error Delay Line, Control Filter

Output: Updated Control Filter

The constraints that pertain to the adaptive filter algorithm are **F.3.a** and **F.3.b**. These constraints ensure that the system will be able to adapt in the event that there are small or big changes in the acoustic response of the room.

III. *Control Filter:*

Is used to account for the acoustic response of the room. The control filter is applied to the input delay line and is used to reduce and smooth out high-frequency associated noises generated inside the room. Using this control filter will help achieve a more clear output. Verification will be checked by finding out if different filters are being utilized for different acoustic responses.

Input: Filter updates from algorithms

Output: Processed digital input signal

The constraint this pertains to is constraint **F.3.a** and **F.3.b**. The control filter will assist in our system adapting to small changes in the acoustic response of the room. It will also assist in our system adapting to significant changes in the acoustics of the room. Combining both will allow the system to be as accurate and efficient as possible.

IV. *Classification:*

The classification block classifies different sounds and infers changes in the plant impulse response. It then updates the plant impulse response used by the adapting algorithms. Classification can be verified by checking to see if different filters are used at different times. The correct operation would mean that the system has inferred a change in response of the room and adapted accordingly.

Input: Input delay line, Prebuilt Filter Models

Output: Update to the plant impulse response

The constraint derived from the classification system is **F.3.b**. Over time the classification system will be able to infer what type of response is needed for changes in the acoustic environment, this helps aid the system in responding to significant changes in the acoustic response of the room.

V. *Prebuilt Filter Models:*

A set of plant impulse responses that are used to minimize delay. This allows the system to have multiple baselines and minimize the number of iterations needed for a large change in the plant impulse response. This can be verified through simulation and testing to see if different prebuilt filter models are utilized during different inputs.

Input: Array of simulated plant impulse responses

Output: Filter models

This part of the system complies with **F.3.bF.3.aF** This part of the system must be able to properly deal with small changes in the overall acoustic response from the room in a timely, efficient manner. This part of the system must also be able to deal with large changes in the acoustic response of the room in a timely, efficient manner. In the event of environmental sounds that are known to be common occurrences, the system will have approximations in place for these situations.

3) 180 degree phase shift:

After the input has been properly processed into a accurate and usable signal, the signal must then be phase shifted so that it can cancel out the noise of the room. The 180 phase shift block will take the signal and flip it 180 degrees so that it is opposite of the signal that was inputted. To verify the correct functionality of this block, the signal will need to be measured. Upon correct functionality the measured signal will not only be equivalent in magnitude to the inputted signal, but also be opposite so that when the signals are added together they cancel out. Ideally the shifted signal will be exactly opposite but for the case of this design the goal is for the ratio of the output to input will be 0.1.(Equating to 20 decibel attenuation) Meaning the output signal covers at least 90 percent of the input signal.

Input: Digital sample

Output: Shifted digital sample

The derived constraint of the phase shift block is **F.4**. This constraint explains the main idea behind why an opposite phase signal will cancel out the other signal. When opposite waves are combined it creates a destructive interference that effectively cancels out the acoustic wave, effectively eliminating the change in air pressure.

C) Output

The output block will turn the digitally processed signal into a usable sound that destructively interferes with the targeted input sound. If the noise from outside is being attenuated to any degree then that confirms correct operation of the output.

Input: Digitally processed shifted signal

Output: Anti-noise

The output channel for the system will conform to constraints **A**, **E**, and **P**. These constraints state include the correct operation of the output and constrain what the output emits. This is to ensure that the output of the system emits anti-noise but also is safe.

1) Speaker

The speaker will be responsible for taking an analog voltage signal and converting to an audible sound. Correct operation can be verified by checking to see if the speaker is outputting.

Input: Analog voltage signal

Output: Acoustic anti-noise

The system speaker will conform to constraint **E**, which states that the speaker will generate an acoustic sound based on the input of the system. It is imperative that the speaker operates properly or else the anti-noise will not be effective.

2) Power Amplifier

Provides an appropriately powered signal to the speaker

Input: Analog voltage signal

Output: Amplified analog voltage signal

Constraint Derived: Ideally, the system would always reduce noise, but this may not always be achievable. As described in **A**, loud noises reaching 80 dB could cause damage to an individual's ears. If a loud sound reaching 80 dB is detected in the input and error channel, a warning signal shall be displayed. If this sound continues for a total of eight hours within a twenty-four-hour period, the device will stop making sound for the 80+ dB sound until reset.

3) Digital to Analog Converter

The digital-to-analog converter will convert the digital signal output samples into an analog output signal to be played on the speaker. This will be verified by checking if the signal has been converted to a digital form.

Input: Digital signal output samples

Output: Analog output signal

This system complies with constraints **F** and **C.3**. The correct functionality of this part of the system can be tested by measuring the output signal, proving it is indeed an analog representation of the output, as needed in constraint **F** and examining the signal to make sure minimal or no clipping has occurred on the analog output as needed in constraint **??**.

D) Error Channel

The goal of the error sensor will be to read the noise from the room after output and measure the residual noise. This data will then be sent back to the error input channel to be used for processing. This system can be verified by measuring the signal that is outputted and comparing to the input signal and the plant impulse response. The error channel should be able to recognize discrepancies in the input and output and send the data to processing.

Input: Acoustic sound from speaker

Output: Digital signal sample

The error channel will follow Constraints **D** and **D.1**. Similar to the input channel the error channel will need to be able to output a signal that can be used by the processing subsystem. The error channel is further constrained by the need to compare itself to the primary input and measure residual sound.

1) Sound Sensor

The sound sensor will be responsible for acquiring the acoustic data from the room and output speaker and turning it into a waveform that can then be filtered and amplified by the rest of the system. This can be verified by sending the signal to a speaker with a flat input response, to determine if correct frequencies are being obtained.

Input: Acoustic sound from room

Output: Analog voltage signal

This part of the system will follow the specifications outlined by constraint **D**. Which ensures that that the method used to acquire an acoustic noise signal is operating properly, and the error in the room is properly being measured.

2) Pre-Amplification

The signal will be amplified before being sent to the analog-to-digital converter in order to make use of the full dynamic range of the ADC. The level of amplification is determined by the minimization of quantization error. This is hardware specific and depends on the dynamic range of the analog to digital converter.

Input: Analog voltage signal

Output: Amplified analog voltage signal

The constraint that this part of the system will follow is **C.2** and **D.1** which states that the full dynamic range of the ADC will be utilized so that the system is as accurate as possible. It is imperative that this part of the system functions properly so that the processing block has the necessary data to correct error in the output.

3) Analog to Digital Conversion

In order for the signal to be processed it needs to be digital, so the analog signal will be converted into digital samples. To achieve this an analog-to-digital converter(ADC) will be utilized. There will be small sources of error that the system needs to be able to acknowledge and minimize. This can be verified by confirming that a digital signal output has the same properties as a predetermined input.

Input: Analog voltage signal

Output: Digital signal sample

The constraints that will be followed by the ADC converter are **C**, **C.1**, **C.2**, **C.3**, and **C.4**. This part of the system must be able to take in an analog input and convert it to a digital signal. This part of the system must also be sampled at least 40 kHz in order to ensure the outputted samples properly represent the analog input signals. The quantization error must also be minimized to make sure the desired attenuation of 20 dB is achieved. In order for the digital signal to not experience distortion a soft clipping scheme will be utilized. The digital samples will also require a minimum bit depth of 16 bits. This will give the system enough samples to properly counteract noise in the processing section. These constraints can be tested by measuring and observing the signals, an

E) Power

Power is an implied subsystem. Commercially available power adapters will pull power from the classroom's wall outlets for use by the subsystems.

Input: 120 V, 60 Hz wall outlet power

Output: Appropriate Power for each subsystem

Constraint Derived: Derived by constraint **I**, **J**, **M** and **L**. These constraints are in place to ensure that any power utilized in this design is safely used. The system shall be able to power off and proper measures will be taken to ensure safety in implementation.

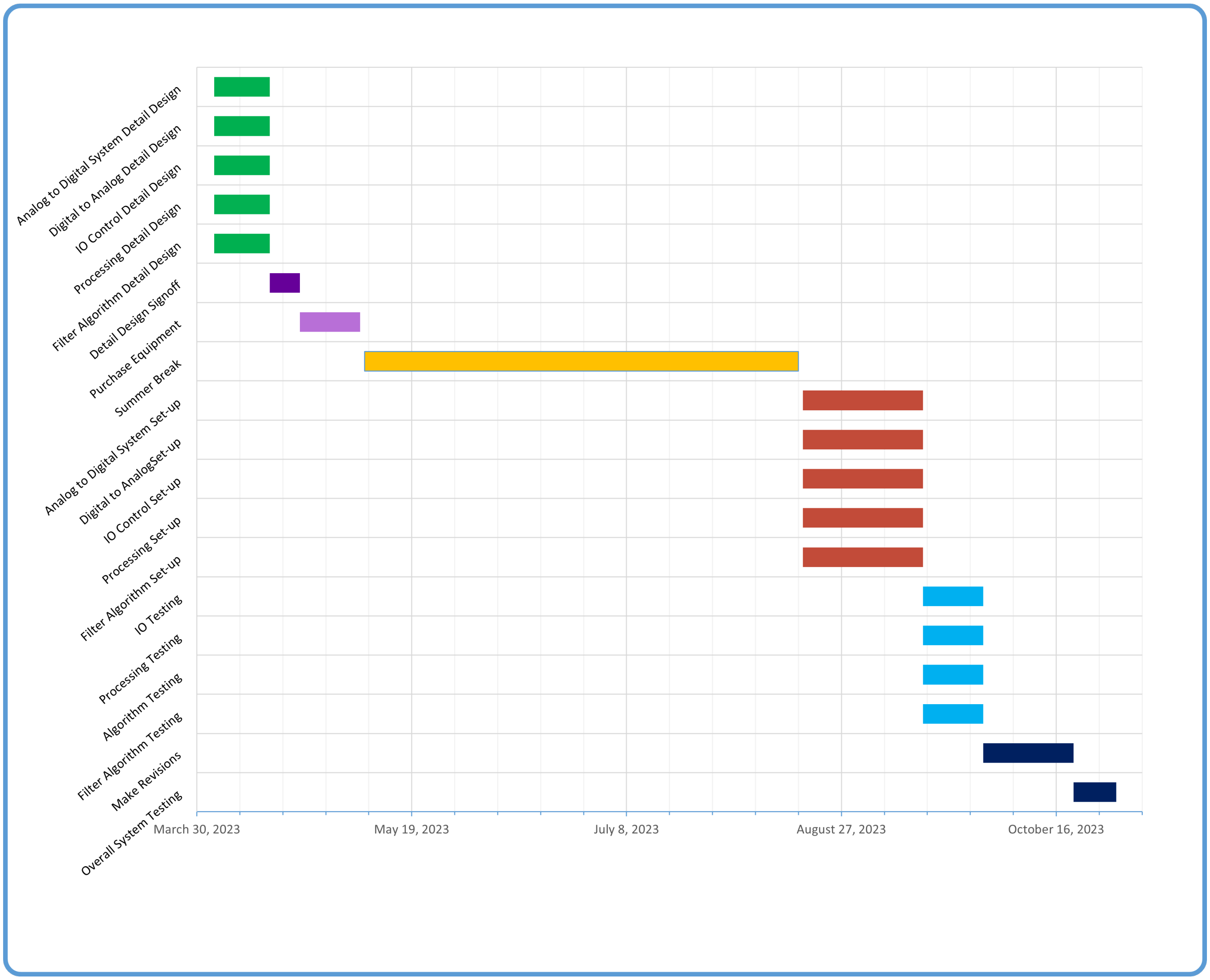

Fig. 3. Timeline

F) Heat Management

Heat management is an implied subsystem. Depending on the detailed specifications of the device hardware, more or less heat management will be needed. The device is not expected to reach high temperatures, so verification can be done with physical touch to ensure the device stays at a safe surface temperature.

Input: System heat

Output: Dispersed heat

Constraint **H** defines an upper limit of temperature to be 60°C (140°F), while many hot surfaces are designed to reach up to 49°C (120°F) to further increase safety.

IV. TIMELINE

The gantt chart in figure 3 characterizes each task in the detail design. The tasks are separated into specific parts that can be individually followed. This flexibility allows the team members to work on parts of the design they are best suited for and creates small checkpoints to follow progress. The timeline is made to be general-purpose and can change due to unforeseen circumstances, but the end product is expected to be produced near the same end date.

V. CONCLUSION

The conceptual design of in-room noise cancelling with wall-transmission detection involves three major subsystems: noise detection, signal processor, and noise output. Each of these major subsystems is divided into individual components that fully make up the device. This conceptual design displays a transparent and specific overview of the device while allowing flexibility in hardware and software to recreate the device.

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