***Abstract* -** The goal of the project is to create a microphone array-based FPGA-based gunshot detection and localization system. The system records audio from several microphones, passes it through an ADC, and then uses a Band-pass Filter (BPF) to separate out the frequencies that are important. A Convolutional Neural Network (CNN) is used to classify the filtered data in order to separate gunshots from other noises. Geometric equations are used to estimate Direction of Arrival (DOA) and compute Time Difference of Arrival (TDOA). The outcomes are shown on a graphical LCD panel. The technology, which provides real-time notifications on the direction of firing, is intended for use in defense and public safety applications.

Index Terms - Signal Processing,TDOA,MATLAB Simulation,Real-time signal Analysis,Graphical LED Display

I. Introduction

In modern defense and security scenarios, real-time detection and localization of gunshots can be critical in preventing casualties and enabling a swift response. This project aims to develop an FPGA-based system that uses a microphone array to detect and determine the direction of gunshots. The system employs digital signal processing techniques, machine learning algorithms, and a graphical interface to deliver accurate and timely information. Key components include an omnidirectional microphone array, ADC7768 for signal conversion, an FPGA (ZYNQ SoC), and a graphical LCD for output display. Communication between components is facilitated by the SPI interface.

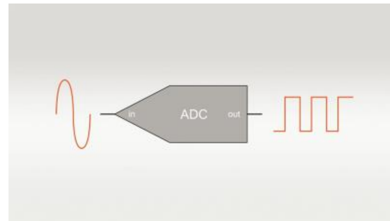
3. Iplementation Details:
3.1 Hardware Setup:

I. Microphone Array (6 Omnidirectional Microphones):

- Purpose: Capture acoustic signals from multiple directions.
- Setup: Position the microphones in a known geometric pattern (e.g., circular, linear) to aid in accurate TDOA and DOA calculations.
- Connection: Each microphone is connected to the ADC7768 via a pre-amplifier circuit to ensure signal levels are suitable for digital conversion.

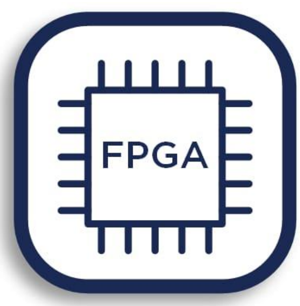
II. Analog-to-Digital Converter (ADC7768):

- Purpose: Convert analog signals from the microphones to digital format.
- SPI Interface: The ADC7768 communicates with the FPGA through the SPI (Serial Peripheral Interface) protocol, which is crucial for high-speed data transfer.
- Configuration: The ADC is configured to sample the input signals at a high rate (e.g., 48 kHz) to capture all relevant frequencies.



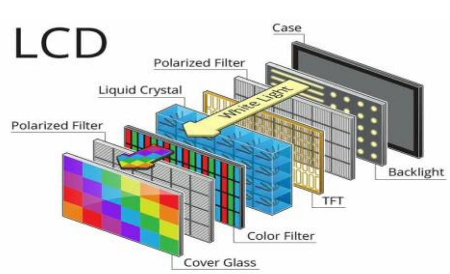
III. FPGA Board (ZYNQ SoC FPGA Development Board):

- Purpose: Perform all signal processing tasks, including filtering, classification, TDOA calculation, and DOA estimation.
- Verilog Programming: Implement each processing step as a Verilog module. Vivado Design Suite is used for development and synthesis.
- Integration with SPI: SPI is configured on the FPGA to communicate with the ADC7768, ensuring that the digitized signals are correctly received and processed.



IV. Graphical LCD Display:

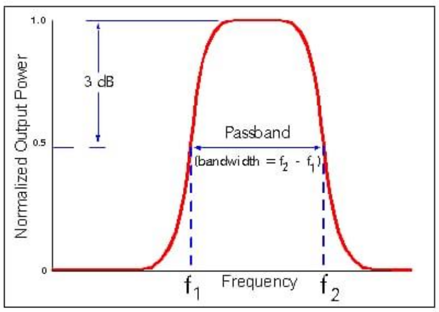
- Purpose: Visually display the direction of the detected gunshot.
- Connection: The LCD is connected to the FPGA's output pins, where the DOA results are sent for display.
- SPI Communication: In some cases, the graphical LCD might also communicate over SPI, requiring additional SPI configuration on the FPGA.



3.2 Signal Processing Steps

i. Band pass Filtering (BPF):

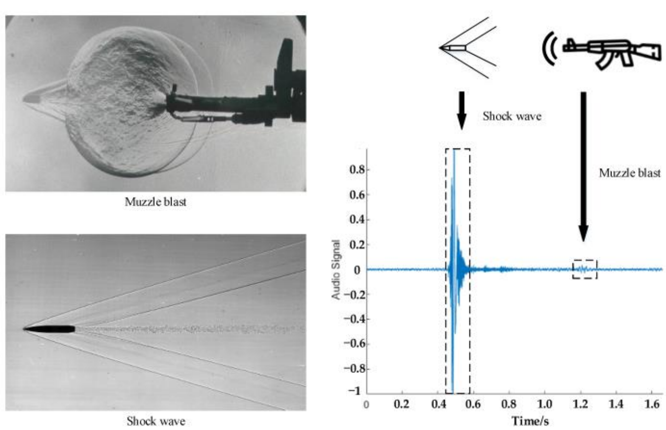
- Purpose: Isolate the frequency range (e.g., around 3 kHz) most relevant for gunshot detection.
- Verilog Implementation: Implement the BPF as a Verilog module using coefficients calculated via MATLAB or another DSP tool. The coefficients are stored in a ROM block or passed as parameters.
- Integration: The filtered digital signals are then passed to the classification stage.



bandpass filter

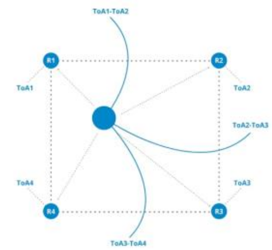
ii. Gunshot Classification (Using CNN):

- Purpose: Differentiate gunshot sounds from other noises.
- Model Training: A CNN is trained on a labeled data set of gunshot and non-gunshot sounds. The model is then quantized and converted for FPGA deployment.
- Verilog Implementation: Implement the CNN in Verilog or use Vivado's DSP or Machine Learning IP cores to accelerate computation.
- Integration: The classified signals are fed into the TDOA calculation module for further processing.



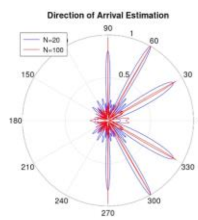
iii. TDOA Calculation (Using Cross-Correlation):

- Purpose: Calculate the time difference of arrival between signals captured by different microphones.
- Cross-Correlation: Implement cross-correlation in Verilog using FFT-based methods. The Xilinx FFT IP core can be utilized to efficiently perform this operation.
- SPI Data Handling: Ensure the data received via SPI from the ADC is properly aligned and stored for cross-correlation processing.



iv. DOA Estimation:

- Purpose: Estimate the direction of the gunshot based on the calculated TDOA.
- Geometric Equations: Solve the geometric equations (using methods like Taylor series expansion or Gaussian elimination) to convert TDOA into an angle or coordinate.
- Verilog Implementation: Implement the mathematical operations required for DOA estimation in Verilog. The computed direction is then prepared for display on the graphical LCD.



v. Visualization on Graphical LCD:

- Purpose: Display the computed direction on a graphical interface.
- Implementation: Use Verilog or a dedicated IP core to send the coordinates or angles to the graphical LCD.
- SPI Control: If the LCD uses SPI for communication, configure the FPGA to handle SPI communication for displaying results

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Working Process:

❖ Microphone Setup: Material and Functionality

In a gunshot detection system, the selection and configuration of microphones are crucial for capturing sound accurately. For this system, **omni-directional microphones** are used, as they capture sound from all directions, making them ideal for detecting impulsive sounds like gunshots, which can come from any direction in an open area.

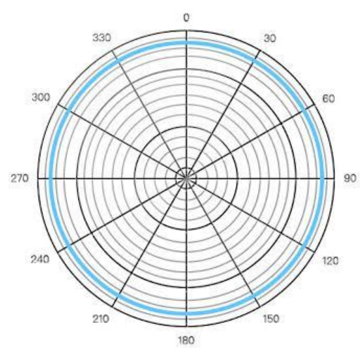
1. Material of the Microphones

The omni-directional microphones used in this project are typically composed of the following materials:

- **Diaphragm**: The diaphragm is the most sensitive part of the microphone and is usually made of a thin, flexible material such as **Mylar** (a type of polyester) or **polypropylene**. The diaphragm moves when sound waves hit it, converting acoustic energy into mechanical vibrations. This material is chosen because it is lightweight and responsive to high-speed vibrations, making it ideal for capturing high-frequency, impulsive sounds like gunshots.
- **Housing**: The external casing of the microphone, which provides protection and structural support, is typically made of materials such as **aluminum** or **steel**. These metals are chosen for their durability and ability to shield the internal components from external interference and physical damage.
- **Electret Material** (in electret microphones): Many omnidirectional microphones use an **electret condenser** element, where a permanently charged material (such as Teflon or other types of fluoropolymers) is placed near the diaphragm. This material maintains an electrostatic field, which aids in the conversion of sound waves into electrical signals without needing an external power source for polarization.
- **Acoustic Foam or Mesh**: Some microphones may include an acoustic foam or mesh layer on the outside to reduce wind noise and protect the diaphragm. This material is usually polyurethane foam, which allows sound waves to pass through while minimizing unwanted noise.

2. How Omni-directional Microphones Work

Omni-directional microphones capture sound from all directions because of their design. The key components that enable this function are the diaphragm and the transducer (which converts acoustic energy into electrical signals). Here's a breakdown of how the microphones work in this system:



Sound Wave Reception: When a gunshot occurs, sound waves propagate in all directions from the source. An omni-directional microphone, being sensitive to sound from every angle, picks up these sound waves regardless of its orientation. This makes it suitable for detecting gunshots in an open or urban environment where the origin of the shot may not be predictable.

- **Diaphragm Movement**: As the sound waves reach the microphone, they cause the diaphragm (a thin, sensitive membrane) to vibrate. The louder and sharper the sound (such as a gunshot), the more intense the diaphragm's movement.
- **Conversion to Electrical Signal**: The diaphragm is connected to an electrical circuit (in the case of electret or condenser microphones, this would include an electret material and a backplate). The movement of the diaphragm in response to the sound waves generates a varying electrical signal that corresponds to the sound's intensity and frequency.
- **Signal Transmission**: The electrical signal generated by the diaphragm's movement is then sent through shielded twisted pair (STP) cables to the signal conditioning circuit. The purpose of this is to ensure that the signal reaches the processing unit with minimal interference or noise.

3. Performance in Gunshot Detection

Omni-directional microphones are well-suited for gunshot detection for several reasons:

- **Wide Frequency Response**: The frequency range of gunshots typically falls between **20 Hz and 3 kHz**, though high-caliber guns may produce sounds at slightly lower or higher frequencies. The microphones used in this project are designed to capture this entire range with high sensitivity, ensuring that no crucial data is missed.
- **High Sensitivity to Sudden, Loud Sounds**: Gunshots are typically loud, impulsive sounds, and omnidirectional microphones are designed to capture such sudden changes in acoustic pressure. The diaphragm material, typically made from Mylar or other similar materials, is sensitive enough to accurately respond to these rapid changes.
- **Durability and Protection**: In environments where gunshots are detected (outdoors or urban areas), microphones are exposed to the elements. The metal housing (such as aluminum or steel) used in these microphones is critical for ensuring durability, protecting the sensitive diaphragm and internal components from physical damage, dust, moisture, and debris. Additionally, the housing serves as a shield against electromagnetic interference (EMI), which can degrade the quality of the captured sound.

4. Integration with the System

Once the microphones capture the sound waves and convert them to electrical signals, the **shielded twisted pair (STP) cables** transmit these signals to the **signal conditioning circuit**. This part of the system ensures that:

- The microphone signals maintain integrity over long distances.
- Electromagnetic interference is minimized through the shielding in the cables.
- The audio signals are amplified and filtered to remove noise and isolate the relevant frequencies associated with gunshot sounds.

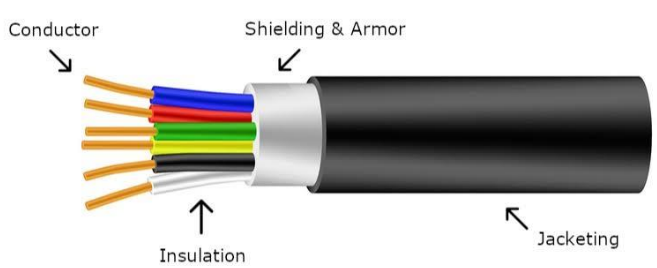
❖ Shielded Twisted Pair Cable: Material and Functionality

Shielded Twisted Pair (STP) cables play a critical role in ensuring the integrity of audio signals transmitted from the microphones to the signal conditioning circuit in a gunshot detection system. These cables are specifically designed to minimize electromagnetic interference (EMI) and reduce noise, ensuring that the signals from the microphones are transmitted cleanly, even in environments with significant electrical noise.

1. Material of Shielded Twisted Pair (STP) Cables

STP cables are composed of several layers, each of which serves a specific purpose in signal protection and transmission. The key components and materials are as follows:

- **Conductors (Signal Wires)**: The core of the STP cable consists of two insulated copper wires. Copper is the most common material used for conductors in STP cables due to its excellent electrical conductivity, malleability, and resistance to corrosion. These conductors are responsible for carrying the electrical signal generated by the microphones.
    - **Hot Conductor**: This wire carries the primary audio signal from the microphone. It is referred to as "hot" because it is the wire that transmits the active signal.
    - **Cold Conductor**: This wire carries the inverse of the audio signal, which helps in noise cancellation. It is referred to as "cold" because it works in conjunction with the hot wire to maintain the integrity of the signal.
- **Insulation**: Each of the copper conductors is coated with an insulating material, typically **polyethylene** or **PVC (Polyvinyl Chloride)**. This insulation prevents electrical contact between the conductors and the surrounding components, ensuring that the signal does not short circuit or degrade during transmission.
- **Twisting of Conductors**: The two conductors (hot and cold) are twisted together. The twisting helps to reduce **crosstalk** and **electromagnetic interference**. By twisting the conductors, the cable ensures that any noise or interference that affects one conductor equally affects the other. Since the signals are inverse, this interference can be canceled out when the signals are recombined at the receiving end (in the signal conditioning circuit).



- **Shielding**: Surrounding the twisted pair of conductors is a layer of shielding material. This is typically made from **aluminum foil** or **braided copper**.
- **Aluminum Foil Shield**: Aluminum foil is lightweight and cost-effective, and it provides excellent protection against electromagnetic interference. It creates a barrier that blocks external noise from affecting the internal conductors.
- **Braided Copper Shield**: Some higher-quality STP cables use braided copper for shielding. This material is more flexible and durable than foil shielding, making it better for applications where the cable may need to be bent or flexed frequently. Copper braiding also provides additional protection against both electromagnetic and radio frequency interference (RFI).
- **Outer Jacket**: The entire assembly is encased in an outer jacket, typically made from **PVC** or **polyethylene**. The outer jacket provides mechanical protection, shielding the internal components from physical damage, moisture, and environmental hazards. It also provides flexibility, allowing the cable to be installed in various environments.

2. How Shielded Twisted Pair Cables Work

The primary function of STP cables is to transmit signals while minimizing noise and interference. This is achieved through a combination of the twisting of the conductors and the shielding. Here’s how it works:

Signal Transmission: The two copper conductors within the STP cable carry the electrical signals generated by the microphone. The **hot** wire carries the original signal, while the **cold** wire carries an inverted version of the same signal. This differential signal setup allows for effective noise cancellation.

Twisting for Noise Reduction: Twisting the two conductors together ensures that any external electromagnetic interference affects both wires equally. Since the two wires carry opposite signals, the interference can be canceled out when the signals are processed at the receiving end. This technique is particularly effective in environments where EMI is prevalent, such as urban areas with a lot of electrical equipment.

Shielding for Additional Protection: The shield (either aluminum foil or braided copper) acts as a physical barrier that prevents external electromagnetic noise from reaching the conductors. This is especially important for long cable runs or in environments where the cable may be exposed to high levels of interference. The shield is usually grounded at one or both ends to dissipate any intercepted interference.

Grounding: The shield is connected to the ground to create a path for any unwanted electrical noise or interference to flow away from the signal-carrying conductors. This ensures that the interference does not corrupt the audio signal. Grounding the shield helps in maintaining the integrity of the signal over long distances.

- **Noise Cancellation**: At the receiving end (the signal conditioning circuit), the signals from the hot and cold wires are recombined. Any interference that affects both wires equally is canceled out, leaving only the clean audio signal. This technique, known as **common-mode rejection**, ensures that the signal reaching the signal conditioning circuit is as close to the original microphone output as possible.

3. Performance in Gunshot Detection System

In the context of a gunshot detection system, the STP cables play a critical role in ensuring that the sound captured by the microphones is transmitted without significant degradation. Here's how it contributes to the system:

- **Signal Integrity Over Long Distances**: In many cases, the microphones are placed far from the signal conditioning circuit, especially in outdoor or large-area installations. The longer the distance, the more likely it is that external noise will affect the signal. The twisting of the conductors and the shielding in the STP cables ensure that the signal remains clean, even over long cable runs.
- **Electromagnetic Interference Protection**: Urban environments, where gunshot detection systems are often deployed, can be full of sources of electromagnetic interference—such as power lines, cellular towers, and electrical equipment. The shielding in the STP cables ensures that these external noise sources do not corrupt the audio signals.
- **Durability**: The outer jacket of the STP cables, typically made from PVC or polyethylene, ensures that the cables can withstand physical wear and tear, moisture, and temperature variations. This makes them suitable for outdoor installations where weather conditions can vary.

4. STP Cable Integration with the System

In your gunshot detection project, the STP cables are used to connect the omnidirectional microphones to the signal conditioning circuit. The process of integration can be broken down into a few key steps:

- **Connecting the Microphone to the STP Cable**: Each microphone is wired to the STP cable's hot and cold conductors. The audio signal generated by the microphone (as an electrical signal) is fed into these conductors.
- **Shielding and Grounding**: The shield of the STP cable is grounded to ensure that any intercepted electromagnetic interference is safely dissipated. Grounding can be done at the microphone end or at the signal conditioning circuit end, depending on the system design.
- **Transmission to the Signal Conditioning Circuit**: The audio signals are transmitted through the twisted pair conductors, with the shield protecting them from external noise. By the time the signals reach the signal conditioning circuit, they are still clean and accurately represent the sound captured by the microphones.
- **Noise Cancellation at the Receiving End**: Once the signals reach the signal conditioning circuit, they are recombined, and any noise introduced during transmission is canceled out. The result is a clean audio signal, ready for further amplification and filtering in the signal conditioning circuit.

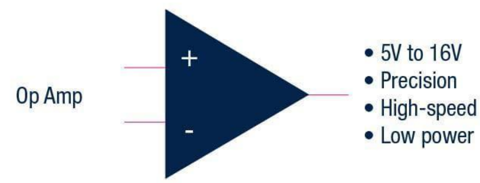
❖ **Signal Conditioning Circuit: Materials and Functionality**

A signal conditioning circuit plays a vital role in ensuring that the signals generated by microphones in your gunshot detection system are properly prepared for further processing by the analog-to-digital converter (ADC) and ultimately the FPGA. These circuits are designed to filter, amplify, and prepare the analog signals to ensure accuracy, clarity, and compatibility with downstream electronics.

1. Materials Used in Signal Conditioning Circuit Components

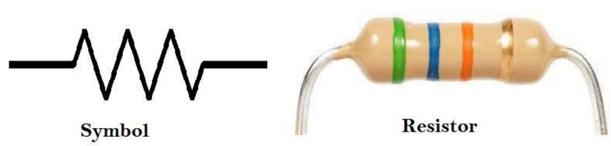
A signal conditioning circuit typically consists of several key components that perform various tasks such as amplification, filtering, and noise reduction. The materials used in these components are critical for their performance. Here are the main elements and the materials used:

- **Operational Amplifiers (Op-Amps)**: These are the heart of most signal conditioning circuits, used for amplifying weak signals from the microphones.



- **Material**: Operational amplifiers are made using **semiconductor materials**, typically silicon. The internal architecture of an op-amp is based on transistors, resistors, and capacitors, which are integrated onto a single chip.

- **Resistors**: Resistors are used to control the gain of the operational amplifiers, allowing for precise control over the amount of amplification applied to the signal.



- **Material**: Resistors in signal conditioning circuits are typically made from **carbon film**, **metal film**, or **thin film**. Metal film resistors are preferred for their precision and temperature stability.

- **Capacitors**: Capacitors are used for filtering signals and removing unwanted noise. They can also be used to create low-pass or high-pass filters in the circuit, which allow specific frequencies to pass through while blocking others.
    - **Material**: Capacitors are made from a variety of materials, depending on their function. Common

materials include **ceramic**, **polyester film**, and **electrolytic materials**. Ceramic capacitors are widely used for their stability and low cost, while electrolytic capacitors are used for higher capacitance values.

- **Inductors**: In some cases, inductors are used in conjunction with capacitors to form more complex filters, such as band-pass filters. These filters can help isolate the frequency range of interest, which is crucial in gunshot detection where specific frequencies need to be detected.
  - **Material**: Inductors are typically made from **coiled copper wire** wound around a **ferromagnetic core**. Copper is used for its excellent conductivity, while the ferromagnetic core helps in amplifying the magnetic field created by the coiled wire.
- **Diodes**: Diodes are often used for signal protection and conditioning. They can help protect the circuit from voltage spikes that could damage sensitive components.
  - **Material**: Diodes are made from **semiconductor materials**, typically silicon or germanium. These materials allow the diode to conduct electricity in one direction, making them useful for protecting against voltage surges.

Printed Circuit Board (PCB): The entire signal conditioning circuit is mounted on a PCB, which serves as the physical platform for the components and provides the electrical connections between them.

  - **Material**: PCBs are typically made from **fiberglass** or **FR4**, a flame-retardant material. The conductive traces on the PCB, which connect the various components, are made from **copper**, providing a low-resistance path for the electrical signals.

2. How the Signal Conditioning Circuit Works

The purpose of the signal conditioning circuit is to modify the raw signals from the microphones, making them suitable for processing by the ADC and FPGA. Here's a step-by-step breakdown of how it works:

a) *Input Signal from the Microphones*

- **Raw Signal**: When a gunshot occurs, the microphones capture the sound and convert it into an electrical signal. These signals are typically weak and may contain unwanted noise, interference, or distortions.
- **Initial Input**: The signal enters the signal conditioning circuit through shielded twisted pair cables, which provide protection against electromagnetic interference.

b) *Amplification (Using Operational Amplifiers)*

- **Weak Signal**: The raw signal generated by the microphone is usually too weak for direct processing by the ADC. Therefore, the first step in the signal conditioning process is amplification.
- **Op-Amps**: Operational amplifiers are used to boost the signal to a level that is appropriate for further processing. The gain of the op-amp is controlled by resistors that determine how much the signal is amplified. For example, if the microphone signal is 10mV, and the ADC requires an input of 1V, the op-amp will amplify the signal by a factor of 100.
  - **High Gain**: In gunshot detection, high gain may be necessary due to the relatively weak initial signal compared to the level of processing required. However, the gain must be carefully controlled to avoid amplifying noise along with the signal.

c) *Filtering (Using Capacitors and Inductors)*

- **Noise Reduction**: After amplification, the signal is passed through filters that remove unwanted frequencies. Gunshot detection systems are interested in specific frequency ranges (e.g., 500 Hz to 3 kHz), so any signal outside this range is considered noise.
  - **Low-Pass and High-Pass Filters**: These filters are designed using capacitors and resistors or inductors. For instance, a low-pass filter will allow signals below 3 kHz to pass through while attenuating higher frequencies that may be noise or irrelevant data.
  - **Band-Pass Filter**: A band-pass filter allows signals within the desired frequency range (500 Hz to 3 kHz) to pass through while blocking both lower and higher frequencies. This is critical in isolating the sound of a gunshot from other environmental sounds.

d) *Protection (Using Diodes and Grounding)*

- **Voltage Spikes**: Sometimes, environmental factors or sudden electrical surges can cause voltage spikes in the signal. Diodes are used to protect the circuit by allowing current to pass in only one direction, preventing damage from excessive voltage.
  - **Clipping**: Diodes can also be used to clip the signal if it exceeds a certain threshold, ensuring that the signal remains within the range required by the ADC.

e) *Output to ADC*

- **Clean Signal**: After amplification, filtering, and protection, the output signal is a clean, conditioned analog signal. This signal is now strong enough and free of noise or interference, making it suitable for conversion into digital form by the ADC.
  - **Impedance Matching**: The signal conditioning circuit may also include impedance matching elements to ensure that the signal is transferred efficiently from the circuit to the ADC. Proper impedance matching prevents signal reflection

and loss, ensuring accurate conversion by the ADC.

3. Importance of Signal Conditioning in Gunshot Detection

Signal conditioning circuits are essential in gunshot detection systems for several reasons:

- **Accuracy**: Without proper amplification and filtering, the weak microphone signals could be swamped by noise, leading to inaccurate detection or missed gunshots.
- **Noise Reduction**: The environment in which gunshot detection systems operate often contains various sources of noise (wind, traffic, background conversations). The filtering aspect of the signal conditioning circuit ensures that only the frequencies related to gunshots are processed.
- **Compatibility with ADC**: The ADC requires signals within a specific voltage range and with minimal noise for accurate digital conversion. The signal conditioning circuit ensures that the signals from the microphones meet these requirements.

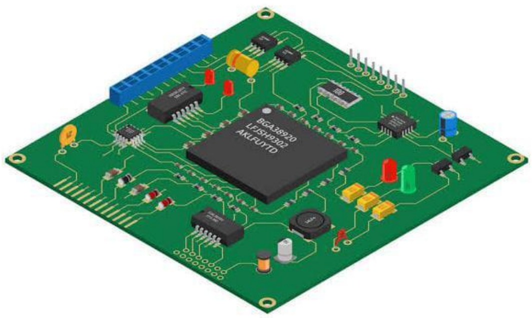
❖ **ADC (Analog-to-Digital Converter): Materials and Functionality**

The Analog-to-Digital Converter (ADC) is a key component in your gunshot detection system, tasked with converting the conditioned analog signals from the microphones into digital data that can be processed by the FPGA. The ADC is vital for translating real-world sounds, such as gunshots, into numerical values that can be analyzed by digital systems. Below is a detailed description of the materials used in ADCs and how they function in your application.

1. Materials Used in ADC Components

The ADC is an integrated circuit (IC) made up of various electronic components, each made from specific materials to perform its function efficiently. Here's a breakdown of the key components and the materials used:

- **Semiconductors (Silicon, Gallium Arsenide)**: Like most integrated circuits, ADCs are primarily built from **semiconductors**, typically **silicon**. The silicon forms the foundation for the transistors that control the flow of current through the ADC.
  - **Alternative Materials**: In high-performance applications, materials like **gallium arsenide (GaAs)** may be used instead of silicon due to their superior electron mobility. However, silicon remains the most common material due to its cost-effectiveness and widespread use in CMOS (complementary metal-oxide-semiconductor) technology.
- **Resistors and Capacitors**: ADCs also contain precise resistors and capacitors to form the necessary voltage dividers, filters, and sample-and-hold circuits.
  - **Material**: Resistors in ADCs are often made from **metal films** such as **nickel-chromium (NiCr)** for accuracy, while capacitors may be made from **ceramic** or **polymer** materials for their stability and high-frequency performance.
- **Transistors (MOSFETs)**: The internal logic and switching mechanisms of an ADC rely heavily on **MOSFET (Metal-Oxide-Semiconductor Field-Effect Transistors)**.
  - **Material**: MOSFETs are typically made from silicon or, in advanced ADCs, from more exotic semiconductors such as gallium arsenide.
- **PCB (Printed Circuit Board)**: The ADC IC is mounted on a **printed circuit board (PCB)**, which provides the electrical connections to the surrounding components like the signal conditioning circuit and FPGA.
  - **Material**: The PCB is generally made from **FR4 fiberglass** with **copper traces** to route electrical signals between components.



2. How the ADC Works

The ADC's role is to take the conditioned analog signals, which are continuous in time and amplitude, and convert them into discrete digital values. This conversion process involves several steps: sampling, quantization, and encoding. Here's a simplified breakdown of the ADC's operation.

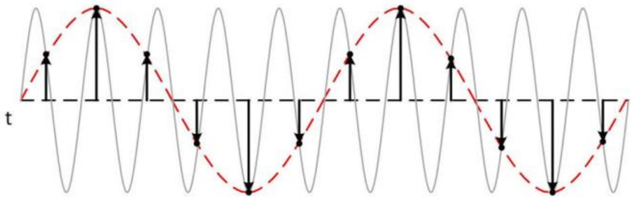
a) *Analog Signal Input*

- **Input from Signal Conditioning Circuit**: The analog signal from the signal conditioning circuit, representing the sound of a gunshot, is fed into the ADC. This signal is typically within a voltage range specified by the ADC, and it should be free of noise or distortion, thanks to the conditioning process.

b) *Sampling*

- **What Is Sampling?**: The ADC samples the incoming analog signal at regular intervals. Each sample represents the signal's amplitude at a specific moment in time. This process is governed by the **sampling rate**, which is the number of samples taken per second (measured in Hz). For gunshot detection, a high sampling rate is important to capture the rapid changes in the sound wave.
  - **Nyquist-Shannon Theorem**: The sampling rate should be at least twice the highest frequency present in the analog signal to avoid **aliasing**, which is the distortion that occurs when the signal is under-sampled. For a gunshot frequency

range of 20 Hz to 3 kHz, a sampling rate of 6 kHz or higher is recommended.



c) *Sample-and-Hold Circuit*

- **Capturing the Signal**: Within the ADC, a **sample-and-hold circuit** captures the instantaneous voltage of the analog signal at the moment of sampling. This ensures that the ADC has a stable signal to work with during conversion.
  - **Materials**: The sample-and-hold circuit uses **MOSFETs** and **capacitors** to capture and hold the signal. The capacitors store the charge representing the analog signal's voltage during the hold phase.

d) *Quantization*

- **What Is Quantization?**: After the signal is sampled, the ADC quantizes it, which means it maps the continuous range of the analog signal's amplitude to a finite set of discrete levels. The number of levels depends on the **resolution** of the ADC, measured in bits.
  - **Resolution**: A 10-bit ADC, for example, can represent the input signal with 1024 discrete levels ($2^{10} = 1024$). Higher resolution provides more precise digital representations of the analog signal.
  - **Material Involvement**: This process is managed by digital logic within the ADC, built from transistors and capacitors.

e) *Encoding*

- **Digital Representation**: Once the signal is quantized, it is encoded into a binary number. For example, if the signal amplitude falls into the 300th level of a 10-bit ADC, the ADC will output the binary equivalent of 300 (100101100). This binary data is then transmitted to the FPGA for further processing.
  - **Internal Logic**: The encoding process is managed by **combinational logic circuits** within the ADC, made from MOSFET transistors.

f) *Transfer of Data to FPGA*

- **Data Transfer**: The digital values are sent from the ADC to the FPGA through a data bus. The FPGA then processes the digital data to analyze the sound and determine the direction of the gunshot. The data bus typically consists of multiple data lines (equal to the ADC's resolution), which transmit the binary data in parallel.
  - **Connection**: These lines are typically made from **copper** on the PCB, ensuring efficient electrical connections between the ADC and the FPGA.

3. How the ADC Is Used in Gunshot Detection Systems

In your gunshot detection project, the ADC plays a pivotal role in converting the conditioned analog signals (representing gunshot sounds) into digital data that can be processed by the FPGA. Here's why the ADC is so important in this context:

a) *Accuracy of Sound Data*

The precision of the gunshot detection system depends largely on the accuracy of the ADC. A higher-resolution ADC will result in more accurate digital representations of the analog gunshot sounds, which improves the system's ability to distinguish between different types of sounds.

b) *Real-Time Processing*

The ADC must operate quickly to handle the high sampling rate required for real-time gunshot detection. If the ADC's sampling rate is too low, important details of the gunshot sound may be missed, reducing the accuracy of the system in determining the direction and nature of the shot.

c) *Compatibility with FPGA*

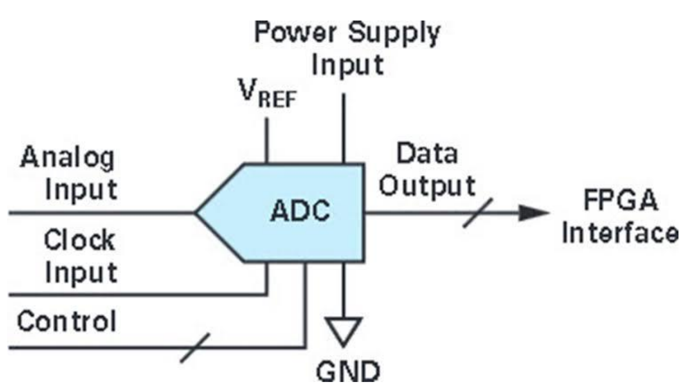
The ADC must output data in a format that can be easily read and processed by the FPGA. This typically involves ensuring that the digital data lines are properly connected to the FPGA's input pins and that the ADC's output format is compatible with the FPGA's processing algorithms.

❖ ADC to FPGA Input pins:

The output of the ADC module is connected to the FPGA input pins for further processing in your gunshot detection system.

1. ADC Output: Digital Data

Once the ADC converts the analog signal into a digital format, it outputs this data in parallel across multiple lines, known as a data bus. The number of lines in this bus corresponds to the **resolution** of the ADC. For example, if your ADC is 10-bit, there will be 10 output lines, each representing one bit of the digital data.



2. Data Bus: Transferring the Data

The digital data (binary values) generated by the ADC is sent out through the **data bus**. The data bus consists of several **parallel lines**, where each line carries one bit of the binary output from the ADC. These lines are typically **copper traces** on a printed circuit board (PCB), which are electrically connected to the FPGA input pins.

- For example, if the ADC reads a gunshot sound and converts it into a 10-bit binary number, say 1101101010, this number is transferred through the 10 parallel lines to the FPGA.

3. FPGA Input Pins

The ADC output lines (data bus) are physically connected to the **input pins** on the FPGA. Each pin on the FPGA receives one bit of the digital data from the ADC.

- If your ADC is 10-bit, you need 10 input pins on the FPGA, where each pin is assigned to one of the 10 bits of the ADC output.
- The pins are generally labeled, such as DATA_0, DATA_1, ..., DATA_9, with DATA_0 being the least significant bit (LSB) and DATA_9 being the most significant bit (MSB).

4. FPGA Clock Synchronization

FPGA input pins are synchronized with a **clock signal**. This clock ensures that the FPGA reads the digital data at the correct times. When the FPGA clock is in sync with the ADC's output, the FPGA captures the digital values from the ADC at regular intervals.

- The clock signal can be connected separately or come from the ADC, ensuring that the FPGA knows exactly when to read the incoming data.

5. Processing the Digital Data Inside the FPGA

Once the ADC data is input into the FPGA, the FPGA processes this data according to its programmed logic. For a gunshot detection system, the FPGA would use algorithms such as:

- **Digital Signal Processing (DSP)** to analyze the audio signal,
- **Direction of Arrival (DOA)** algorithms to determine the location of the gunshot,
- **FFT (Fast Fourier Transform)** to break the sound signal into frequency components.

6. Verilog or VHDL Code in FPGA

The FPGA is programmed with a hardware description language (such as **Verilog** or **VHDL**) to specify how the input data should be processed. Inside the FPGA, the digital signals are processed using various **logic blocks**, such as:

- **Adders, multipliers**, and other digital operations to process the sound data,
- **Filters**, such as bandpass filters, to isolate the gunshot frequency range (e.g., 500 Hz to 3 kHz),
- **Algorithms** that estimate the location and direction of the gunshot based on the data.

7. Storing or Displaying the Processed Data

After processing the input data, the FPGA can store the results in memory, send them to other modules for further processing, or display the data (such as showing the direction of the bullet on a graphical display).

❖ Detailed Description of Bandpass Filter

1. Purpose of Bandpass Filtering

A bandpass filter is a signal processing tool that allows signals within a certain frequency range to pass through while attenuating frequencies outside this range. In the context of a gunshot detection system, the primary goal of the bandpass filter is to isolate the frequency band that is most likely to contain gunshot sounds, while minimizing background noise and irrelevant frequencies.

Gunshots typically produce sounds that have significant energy within a specific frequency range, usually between *800 Hz and 3000 Hz*. This range encompasses most of the acoustic signatures of gunshots. By using a bandpass filter to focus on this range, the system can effectively differentiate gunshots from other sounds like voices, traffic noise, and environmental sounds that generally fall outside this range.

2. Types of Bandpass Filters

There are several types of bandpass filters that can be used in signal processing, each with its own characteristics in terms of performance, complexity, and suitability for specific applications:

- ❖ Butterworth Filter: This type of filter is known for its smooth frequency response within the passband, which means it does not introduce ripples (fluctuations in amplitude) within the frequency range it is designed to pass. The Butterworth filter provides a relatively gentle roll-off, meaning the transition from passband to stopband (the range of frequencies that are blocked) is not very sharp. This makes it ideal for applications where a smooth response is required.
- ❖ Chebyshev Filter: This filter is characterized by a faster roll-off compared to the Butterworth filter, which allows it to more quickly attenuate frequencies outside the passband. However, it introduces ripples in the passband or stopband, which can slightly distort the signal. Chebyshev filters are useful when a sharper cutoff is needed, but some ripples in the signal are acceptable.

- ❖ Elliptic Filter (Cauer Filter): The elliptic filter offers the steepest roll-off among all standard filters, making it highly effective in separating frequencies that are very close together. It achieves this at the cost of introducing ripples in both the passband and stopband. Elliptic filters are often used in applications where the separation of closely spaced frequencies is critical, but it is less preferred for applications where maintaining a clean signal is important.
- ❖ Bessel Filter: This filter type is known for its excellent phase response, meaning it preserves the shape of the signal waveform within the passband. While the Bessel filter does not have a particularly sharp roll-off compared to the other types, its phase characteristics make it useful in applications where signal timing is critical.

Each filter type can be designed with different "orders," which refers to the steepness of the roll-off. Higher-order filters provide sharper cutoffs but increase the computational complexity and the resources required for implementation.

3. Design Considerations for Bandpass Filters:

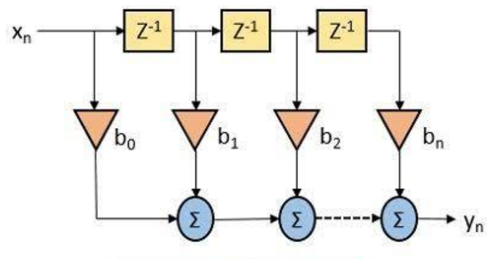
When designing a bandpass filter for a gunshot detection system, several factors need to be considered to ensure optimal performance:

- Cutoff Frequencies: The choice of lower and upper cutoff frequencies determines the range of frequencies that the filter will allow to pass. For gunshot detection, these are typically set around 800 Hz (lower cutoff) and 3000 Hz (upper cutoff). Frequencies below 800 Hz may include low-frequency noise, while frequencies above 3000 Hz may contain irrelevant high-frequency noise.
- Filter Order: The order of the filter affects how sharply the filter transitions from passband to stopband. A higher-order filter will have a steeper roll-off, allowing for more precise isolation of the desired frequency range. However, higher-order filters are more complex and require more processing power, which may impact real-time performance.
- Sampling Rate: The sampling rate of the system's analog-to-digital converter (ADC) plays a crucial role in the design of the bandpass filter. The sampling rate must be at least twice the highest frequency of interest (according to the Nyquist theorem) to avoid aliasing, where higher frequencies are incorrectly mapped to lower frequencies. For a filter with an upper cutoff of 3000 Hz, a minimum sampling rate of 6000 Hz is needed, but higher rates (such as 16 kHz) are often used for more accurate results.
- Real-Time Processing Requirements: Since the gunshot detection system operates in real-time, the filter design must consider the computational efficiency of the chosen filter type and order. Filters that require complex calculations might need to be simplified or approximated to ensure the system can process signals quickly enough to provide timely detection.

4. Implementation of Bandpass Filters in FPGA

Implementing a bandpass filter on an FPGA (Field Programmable Gate Array) involves converting the digital filter design into a format that can be executed by the FPGA's hardware. The FPGA is ideal for this task due to its parallel processing capabilities and low latency, which are critical for real-time signal processing.

Digital Filter Design: The bandpass filter is typically implemented as a digital filter using techniques like Direct Form I or Direct Form II. These forms represent the filter as a set of difference equations that can be easily mapped to hardware. The coefficients for these equations are precomputed based on the desired filter specifications (e.g., cutoff frequencies, filter order).



FIR Filter

Finite Impulse Response (FIR) vs. Infinite Impulse Response (IIR) Filters: Digital filters can be categorized into FIR and IIR types. FIR filters are inherently stable and have a linear phase response, making them easier to implement in hardware but often requiring more coefficients for a given performance level. IIR filters, on the other hand, can achieve a similar performance with fewer coefficients but require careful design to ensure stability. For bandpass filtering in a gunshot detection system, IIR filters like Butterworth or Chebyshev are often chosen due to their efficiency.

- ○ FPGA Resource Utilization: When implementing the filter, considerations must be made for the FPGA's resources, such as Look-Up Tables (LUTs), Digital Signal Processing (DSP) blocks, and memory. Efficient use of these resources ensures that the filter operates correctly without consuming excessive power or causing delays in other parts of the system.

5. Practical Considerations and Challenges

- ✓ Noise and Interference: In real-world scenarios, noise and interference can impact the effectiveness of the bandpass filter. Environmental factors like wind, rain, and background urban noise can introduce signals within or near the passband. The filter design must account for these possibilities by possibly widening the passband or implementing adaptive filtering techniques.
- ✓ Adaptive Filtering: In dynamic environments where the noise characteristics change rapidly, an adaptive bandpass filter that adjusts its cutoff frequencies and gain based on real-time analysis of the incoming signal might be necessary. Such adaptive filters can be more complex to implement but provide better performance in varied conditions.
- ✓ Latency: The filter must process signals quickly to ensure that the overall system response is timely. Excessive filtering delays can reduce the effectiveness of gunshot

detection, especially in high-security applications where rapid response is essential.

- Integration with Other System Components: The bandpass filter must be seamlessly integrated with other parts of the system, such as the analog-to-digital converter (ADC), gunshot classification module, and localization algorithms (TDOA and DOA). This requires careful design of data flow and communication between components to maintain synchronization and minimize data loss.

❖ Description of Gunshot Classification

Gunshot classification is a critical process in a gunshot detection system, responsible for distinguishing between gunshot sounds and other types of noises (e.g., fireworks, car backfires, or ambient noise) to accurately detect gunshots. The classification process involves analyzing acoustic features of the detected sound and applying machine learning algorithms to identify whether the sound is indeed a gunshot. Below is a comprehensive explanation of the gunshot classification process, including the methodologies, feature extraction techniques, classification algorithms, and formulas used.

1. Overview of Gunshot Classification

Gunshot classification in a gunshot detection system involves several key steps:

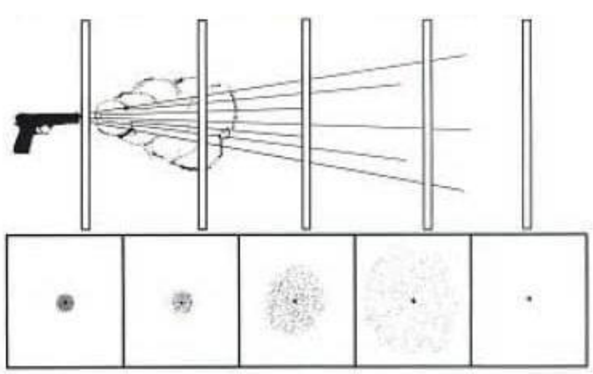
i. Preprocessing: Preparing the raw audio signal for analysis by removing noise and enhancing relevant features using filtering techniques.

ii. Feature Extraction: Extracting meaningful features from the preprocessed audio signal that can help in differentiating gunshots from other sounds.

iii. Feature Selection: Choosing the most relevant features for classification to improve accuracy and reduce computational complexity.

iv. Classification Algorithms: Applying machine learning algorithms to classify the sound as a gunshot or non-gunshot based on the extracted features.

v. Post-Processing: Refining the classification results by considering contextual information or using ensemble methods to reduce false positives and false negatives.



2. Preprocessing of Audio Signals

Before any classification can be performed, the raw audio signals captured by the microphones must be preprocessed to enhance the quality and relevance of the data:

- ❖ Noise Reduction: This step involves removing background noise that may interfere with the classification process. Common techniques include spectral subtraction and Wiener filtering, which suppress noise by analyzing the power spectrum of the audio signal.
- ❖ Normalization: The amplitude of the audio signals is normalized to ensure that all sounds are on the same scale. This helps in making the features extracted from different recordings comparable.
- ❖ Segmentation: The continuous audio stream is divided into smaller segments or frames (e.g., 20 ms to 100 ms), which are then analyzed individually. This allows for the detection of short-duration events like gunshots and helps in focusing on specific time intervals where a potential gunshot occurred.

3. Feature Extraction

The success of gunshot classification heavily relies on the ability to extract discriminative features from the audio signal. Features represent the characteristics of the sound and can be broadly categorized into time-domain, frequency-domain, and time-frequency domain features.

3.1 Time-Domain Features

- ✓ Zero Crossing Rate (ZCR): The rate at which the audio signal changes sign (i.e., from positive to negative or vice versa) is a measure of the signal's frequency content. Gunshots generally have a high ZCR due to their impulsive nature.

$$ZCR = \frac{1}{N-1} \sum_{n=1}^{N-1} \mathbb{I}\{x[n] \cdot x[n-1] \cdot$$

where (x[n]) is the audio signal and (mathbb{I} { cdot }) is an indicator function that is 1 when its condition is met and 0 otherwise.

- ✓ Energy: The energy of an audio segment represents the sum of the squared amplitudes of the signal, indicating the signal's loudness. Gunshots are typically characterized by high energy.

$$E = \sum_{n=1}^{N} |x[n]|^2$$

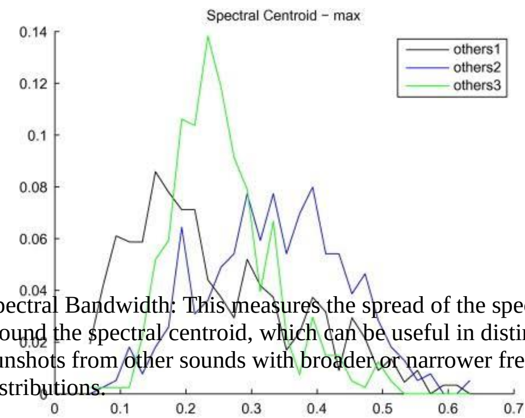
- ✓ Entropy of Energy: This measures the abruptness or unpredictability in the energy distribution of the signal, which is often higher in gunshots due to their sudden onset and decay.

3.2 Frequency-Domain Features:

- Spectral Centroid: Represents the "center of mass" of the spectrum and indicates where the majority of the signal's energy is located in the frequency domain. Gunshots often have a spectral centroid in a specific range due to their characteristic frequency components.

$$\text{Spectral Centroid} = \frac{\sum_{k=1}^{K} f[k] \cdot X[k]}{\sum_{k=1}^{K} X[k]}$$

where (f[k]) is the frequency of the (k)-th bin and (X[k]) is the magnitude of the Fourier Transform at that bin.



Spectral Bandwidth: This measures the spread of the spectrum around the spectral centroid, which can be useful in distinguishing gunshots from other sounds with broader or narrower frequency distributions.

$$\text{Spectral Bandwidth} = \sqrt{\frac{\sum_{k=1}^{K} (f[k] - \text{Spectral Centroid})^2 \cdot X[k]}{\sum_{k=1}^{K} X[k]}}$$

- Mel-Frequency Cepstral Coefficients (MFCCs): MFCCs are a popular set of features used in audio classification. They capture the power spectrum of the signal on a perceptual scale (the Mel scale) and are particularly effective for distinguishing between different types of sounds, including gunshots.

3.3 Time-Frequency Domain Features

- Spectrogram: A visual representation of the spectrum of frequencies in a signal as it varies with time. It provides a more comprehensive view of how the energy in different frequency bands evolves over time, which is useful for identifying gunshots with distinct patterns.
- Wavelet Transform: This method analyzes the signal at different scales or resolutions, providing both time and frequency information. Wavelet-based features can capture transient characteristics of gunshots effectively.

4. Feature Selection

After extracting a large set of features, it is crucial to select the most relevant ones to improve classification performance and reduce computational overhead. Common feature selection techniques include:

- ❖ Principal Component Analysis (PCA): Reduces the dimensionality of the feature space by transforming the original features into a set of linearly uncorrelated components.
- ❖ Recursive Feature Elimination (RFE): A technique that recursively removes the least important features based on their contribution to a given classifier's performance.
- ❖ Correlation-Based Feature Selection (CFS): Selects features that are highly correlated with the class but have low inter-correlation, ensuring that redundant features are excluded.

Gunshot Classification Using Convolutional Neural Network (CNN):

After reducing the dimensionality of the MFCC features using PCA, the processed features are fed into a Convolutional Neural Network (CNN) for classification.

CNN Architecture for Gunshot Classification:

1.Input Layer: The input layer receives the reduced MFCC feature set from PCA

2.Convolutional Layers: Multiple convolutional layers are used to learn spatial hierarchies in the data. Each layer applies filters (kernels) to the input, detecting patterns such as edges, textures, or specific frequency components that characterize gunshots.

3.Activation Function: Non-linear activation functions like ReLU (Rectified Linear Unit) are applied after each convolutional layer to introduce non-linearity and enable the network to learn complex patterns.

4.Pooling Layers: Pooling layers (e.g., max-pooling) are added to reduce the spatial dimensions of the feature maps, making the network more computationally efficient and reducing the risk of overfitting.

5.Fully Connected Layers: The high-level features extracted by the convolutional layers are flattened and fed into fully connected layers that combine these features for final classification.

6.Output Layer: The output layer consists of a softmax activation function that provides the probability of the input being a gunshot or non-gunshot.

- Training the CNN Model: The CNN model is trained using a labeled dataset of audio recordings containing gunshots and non-gunshots. The model learns to minimize a loss function (e.g., cross-entropy loss) using optimization algorithms like Adam or SGD (Stochastic Gradient Descent).

✓ Evaluation Metrics: The model's performance is evaluated using metrics such as accuracy, precision, recall, F1-score, and the Area Under the Receiver Operating Characteristic Curve (AUC-ROC). These metrics help in assessing the model's ability to distinguish between gunshots and non-gunshots effectively.

6. Post-Processing

After the initial classification, further steps can be taken to refine the results:

- Ensemble Methods: Combining multiple classifiers (e.g., Random Forest and SVM) can improve accuracy by leveraging the strengths of each model. Voting, bagging, and boosting are common ensemble techniques used in gunshot classification.
- Contextual Analysis: Considering contextual information such as the location of the sound, time of day, and correlation with other detected events can help reduce false positives (e.g., fireworks on New Year's Eve).
- Threshold Tuning: Adjusting the decision threshold of the classifier can help balance the trade-off between sensitivity (true positive rate) and specificity (true negative rate) to achieve optimal performance based on the application.

❖ Time Difference of Arrival (TDOA) for Gunshot Detection and Localization

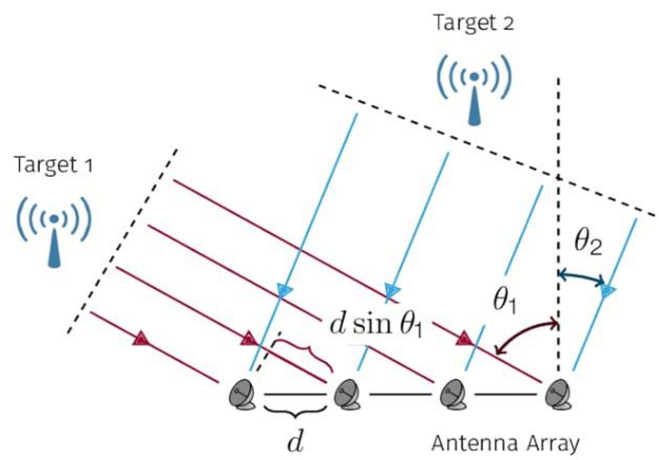
Time Difference of Arrival (TDOA) is a key technique used in sound source localization, particularly in systems designed to detect and locate gunshots. TDOA measures the difference in the arrival times of a sound signal at multiple spatially separated sensors. By analyzing these differences, the location of the sound source can be determined. This technique is crucial in applications like gunshot detection systems, where accurate and real-time localization of gunshots is necessary for effective response.

1. Overview of TDOA:

TDOA relies on the principle that a sound wave propagates through the air at a finite speed (approximately 343 meters per second in air at room temperature). When a gunshot occurs, the sound wave reaches different microphones (sensors) at slightly different times, depending on the distances of the microphones from the source.

The key idea behind TDOA is to calculate these time differences between pairs of sensors and use them to triangulate the position of the gunshot. This process typically involves:

- Measuring the Time of Arrival (TOA) at each sensor.
- Calculating the Time Differences (Δt) between pairs of sensors.
- Triangulating the Position of the gunshot using the calculated TDOA values.



2. Measuring Time of Arrival (TOA):

The first step in TDOA is to accurately measure the Time of Arrival (TOA) of the gunshot sound at each sensor. TOA is the exact time at which the sound wave reaches a sensor, and it is usually recorded relative to a reference time (e.g., the start of the recording).

- Synchronization: To ensure accurate TOA measurements, all sensors must be synchronized with a common clock. Any clock drift or synchronization error can introduce inaccuracies in the TDOA calculation.
- Sampling Rate: The sampling rate of the audio recording equipment plays a crucial role in determining the precision of TOA measurements. A higher sampling rate allows for more precise timing information, which is vital for accurate TDOA calculation.
- Signal Detection: The gunshot sound is detected by each sensor, and the TOA is recorded at the moment the sound's amplitude exceeds a certain threshold, indicating the arrival of the gunshot sound at the sensor.

3. Calculating Time Difference of Arrival (TDOA):

Once the TOA has been measured at each sensor, the next step is to calculate the Time Difference of Arrival (TDOA) between pairs of sensors. The TDOA between two sensors \(i \) and \(j \) is defined as:

$$\Delta t_{ij} = TOA_j - TOA_i$$

Here:

- (Delta t_{ij}) is the TDOA between sensors (i) and (j).
- (TOA_j) is the time of arrival of the sound at sensor (j).
- (TOA_i) is the time of arrival of the sound at sensor (i).

The TDOA value reflects the difference in the distances from the sound source to each of the two sensors.

4. Geometry of TDOA and Hyperbolic Localization:

The TDOA values can be used to derive hyperbolic equations that describe the possible locations of the sound source. For two sensors, the TDOA equation forms a hyperbola, with the sensors positioned at the foci. The sound source lies somewhere on this hyperbola.

- ✓ Hyperbolic Equation: For a pair of sensors, the locus of all possible points that satisfy the TDOA equation forms a hyperbola. The equation of the hyperbola can be expressed as:

$$\sqrt{(x-x_1)^2+(y-y_1)^2}-\sqrt{(x-x_2)^2+(y-y_2)^2}=c\cdot\Delta t_{12}$$

Here:

- ((x, y)) are the coordinates of the sound source.
- ((x_1, y_1)) and ((x_2, y_2)) are the coordinates of the two sensors.
- (c) is the speed of sound in air.
- (Delta t_{12}) is the TDOA between the two sensors.

This equation is solved for multiple pairs of sensors to obtain multiple hyperbolas. The intersection of these hyperbolas gives the estimated location of the sound source.

- ❖ Triangulation: With three or more sensors, each pair of sensors will produce a hyperbola, and the intersection of these hyperbolas provides a more precise estimate of the sound source's location. This process is known as hyperbolic triangulation.

5. Practical Considerations in TDOA Calculation:

- Array Geometry: The accuracy of TDOA-based localization is highly dependent on the geometry of the sensor array. The sensors should be placed such that they form a configuration (e.g., a triangle or a more complex array) that maximizes the coverage area and minimizes ambiguity in localization.
- Environmental Factors: Factors like temperature, wind, humidity, and obstacles can affect the speed of sound and the propagation of the gunshot sound waves. These factors must be accounted for to avoid errors in TDOA calculations.
- Multipath Effects: In urban environments, sound waves can reflect off buildings and other structures, leading to multipath propagation. This can cause errors in TOA measurements, as reflected waves may arrive later than the direct wave.
- Noise and Interference: Background noise and other interfering sounds can affect the detection of the gunshot sound and the accuracy of TOA measurements. Advanced signal processing techniques, such as cross-correlation or matched filtering, can be used to mitigate these effects and improve TDOA accuracy.

6. TDOA Estimation Techniques:

Several techniques can be used to estimate TDOA from the recorded signals:

- Cross-Correlation: Cross-correlation is a widely used method to estimate TDOA. The cross-correlation function measures the similarity between the signals received by two sensors as a function of the time lag between them. The time lag that maximizes the cross-correlation function corresponds to the TDOA:

$$\Delta t_{ij}=\arg\max\int s_i(t)\cdot s_j(t+\tau)\,dt$$

Here:

- (s_i(t)) and (s_j(t)) are the signals received by sensors (i) and (j), respectively.
- (tau) is the time lag.
- (Delta t_{ij}) is the time lag that maximizes the cross-correlation, which corresponds to the TDOA.

- ✓ Generalized Cross-Correlation (GCC): The Generalized Cross-Correlation (GCC) method is an extension of cross-correlation that incorporates weighting functions to improve TDOA estimation in noisy environments. The GCC-PHAT (Phase Transform) is a common variant used in TDOA estimation:

$$\Delta t_{ij}=\arg\max\int \frac{S_i(f)\cdot S_j^*(f)}{|S_i(f)\cdot S_j^*(f)|}\cdot e^{j2\pi f\tau}\,df$$

Here:

- (S_i(f)) and (S_j(f)) are the Fourier transforms of the signals (s_i(t)) and (s_j(t)).
- (S_j^*(f)) is the complex conjugate of (S_j(f)).

The denominator normalizes the product, emphasizing phase differences between the signals.

The GCC-PHAT method is particularly effective in reducing the effects of noise and reverberation, making it a popular choice in real-world gunshot detection systems.

❖ DOA Estimation Using Beamforming with a Triangular Microphone Array

When using a triangular microphone array and beamforming algorithms for Direction of Arrival (DOA) estimation, the goal is to determine the direction from which a sound, such as a gunshot, originates. A triangular array provides good spatial resolution and allows for 2D direction estimation (azimuth angle), making it suitable for applications like gunshot detection.

1. Overview of DOA Estimation with Beamforming:

Beamforming is a signal processing technique used to focus the sensitivity of a microphone array in a particular direction. It involves adjusting the time delays or phase shifts of signals received by each microphone so that the signals from a desired direction add constructively, while signals from other directions are suppressed. This process creates a "beam" of sensitivity in a specific direction.

The basic steps for DOA estimation using beamforming are:

i. Signal Acquisition: Capture sound signals at multiple microphones.

ii. Delay Calculation: Calculate the time delays or phase shifts required to align the signals from a specific direction.

iii. Beamforming: Combine the delayed signals to focus on the desired direction.

iv. DOA Estimation: Estimate the direction that maximizes the output power or minimizes noise and interference.

2. Triangular Microphone Array Configuration:

A triangular array consists of three microphones placed at the vertices of an equilateral triangle or a right-angled triangle. The triangular pattern is advantageous because it provides good coverage in all directions and allows for accurate DOA estimation in two dimensions.

- Equilateral Triangle: All three sides are of equal length. This configuration provides uniform sensitivity in all directions.
- Right-Angled Triangle: One angle is 90 degrees, and the other two angles are 45 degrees. This configuration may offer computational simplicity and can still provide good coverage.

The microphones are separated by a distance \(d\), which is chosen based on the frequency range of interest. The spacing should generally be less than half the wavelength of the highest frequency to avoid spatial aliasing.

3. Signal Model for Beamforming:

The received signal at each microphone in the array can be modeled as:

$$x_m(t) = s(t - \tau_m) + n_m(t)$$

Here:

- (x_m(t)) is the signal received at the (m)-th microphone.
- (s(t)) is the original source signal (e.g., a gunshot sound).
- (tau_m) is the time delay of arrival for the signal at the (m)-th microphone.
- (n_m(t)) is the noise at the (m)-th microphone.

The time delay (tau_m) depends on the angle of arrival (theta) of the sound wave and the distance between the microphones.

4. Delay-and-Sum Beamforming:

- ❖ Delay-and-Sum Beamforming is the simplest and most widely used beamforming method for DOA estimation. It works by delaying the signals received by each microphone so that the signals from a specific direction are aligned in time. These delayed signals are then summed up to produce the beamformed output.
- ❖ Delay Calculation: The time delay (tau_m) for the (m)-th microphone is given by:

$$\tau_m = \frac{d_m \cdot \sin(\theta)}{c}$$

Here:

- (d_m) is the distance of the (m)-th microphone from a reference point (e.g., the array center).
- (theta) is the angle of arrival of the sound wave.
- (c) is the speed of sound in air (approximately 343 m/s).

- ❖ Beamformed Output: The beamformed output for a given direction (theta) is obtained by summing the delayed signals:

$$y(t, \theta) = \sum_{m=1}^{M} x_m(t + \tau_m(\theta))$$

Here:

- (y(t, theta)) is the beam formed output for direction (theta).
- (M) is the number of microphones in the array.

❖ Beam forming Power Spectrum: The power of the beam formed output as a function of direction (theta) is used to estimate the DOA. The direction that maximizes the power corresponds to the estimated DOA:

$$P(\theta) = \left| \sum_{m=1}^{M} x_m(t + \tau_m(\theta)) \right|^2$$

The DOA is estimated by finding the angle (theta) that maximizes (P(theta)).

5. Implementing Beam forming with a Triangular Array:

To implement beam forming for DOA estimation with a triangular microphone array:

i. Microphone Placement: Position three microphones at the vertices of an equilateral or right-angled triangle. Ensure the spacing \(d \) between microphones is appropriate for the frequency range of interest to avoid spatial aliasing.

ii. Signal Acquisition: Continuously monitor the environment and acquire signals from all three microphones. The signals should be digitized and stored for processing.

iii. Pre-processing: Apply pre-processing steps such as filtering to remove background noise and normalization to ensure uniform signal levels.

iv. Calculate Delays: For each possible direction (theta), calculate the time delays (tau_m(theta)) for each microphone using the geometry of the triangular array.

v. Beam forming and Power Calculation: For each possible direction (theta), delay the signals accordingly and sum them up to compute the beam formed output. Calculate the beam forming power spectrum (P(theta)).

vi. DOA Estimation: Determine the angle (theta) that maximizes the beam forming power spectrum. This angle represents the estimated DOA of the gunshot.

vii. Visualization: Display the estimated DOA on a user interface, such as a polar plot or a map, to show the direction of the detected gunshot.

❖ Connecting FPGA to LCD Display

After the FPGA has processed the audio data and calculated the DOA, it must present this information to the user. This is done via an LCD display.

i.Transmission Protocols:

To send the DOA data from the FPGA to the LCD, a communication protocol is used. Two common protocols are:

- ✓ SPI (Serial Peripheral Interface): A fast and simple protocol where the FPGA sends the data to the LCD one bit at a time in a serial fashion. This protocol is highly efficient and suitable for fast communication between the FPGA and peripheral devices like an LCD.
- ✓ I2C (Inter-Integrated Circuit): Another popular communication protocol, I2C allows multiple devices (like sensors or displays) to communicate with the FPGA over the same bus, but at a slower speed compared to SPI. For simple LCD displays, I2C can also be used.

ii.LCD Visualization

The LCD can display the DOA data in different formats, depending on the complexity of the display and what's required for the project.

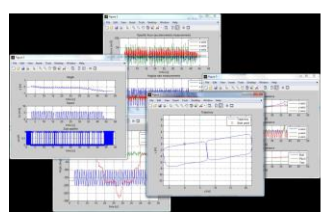
- ➢ Textual Representation: For a basic LCD screen, the FPGA can output the DOA as text, indicating the angle at which the gunshot was detected. For example, the display could show something like "45° from center," which informs the user about the direction from which the gunshot came relative to the microphone array.
- ➢ Graphical Representation: If a graphical LCD display is used, the FPGA can generate a visual representation of the DOA. For example, the display could show a compass or polar plot with an arrow indicating the direction of the gunshot. The FPGA would send graphical data (pixel positions) to the LCD, which then draws the visual representation of the detected direction.

iii. MATLAB Integration for Simulation and Testing:

Before the system is implemented in hardware, it's useful to simulate the entire process in MATLAB to ensure that the algorithms and visualization methods work as expected.

- ➢ Simulating DOA in MATLAB

MATLAB



MATLAB is a powerful platform for simulating complex systems. In this project, you can use MATLAB to simulate how the microphone array detects a gunshot and how the FPGA processes the signals to calculate the DOA.

- Phased Array System Toolbox: MATLAB has specialized toolboxes like the Phased Array System Toolbox, which allow you to simulate the behavior of a microphone array. You can input gunshot sound data and simulate how the microphones detect the sound, how the time differences are calculated, and how the DOA is estimated.
- Signal Processing: MATLAB also provides signal processing tools to help simulate filtering, cross-correlation, and TDOA calculations. This allows you to fine-tune your algorithms before implementing them on the FPGA.

➢ Visualization Testing

Once the DOA is calculated in MATLAB, you can simulate how this information would be displayed on the LCD screen.

- Visual Plots: MATLAB allows you to generate various types of visual plots, such as polar plots or compass plots. These plots can represent the direction of the gunshot visually, just as you would expect to see on the final LCD display. By simulating the visualization in MATLAB, you can ensure that the final output is clear and accurate before moving to hardware.

iv. Real-Time Gunshot Detection and Display:

Once the system is set up, the entire process works in real time, from detecting the sound to displaying the gunshot direction on the LCD.

➢ Gunshot Sound Detection:

- The microphone array picks up the sound of the gunshot, and the signal is conditioned and amplified by the signal conditioning circuit. The analog signal is converted into digital form by the ADC and sent to the FPGA.

➢ DOA Estimation:

- The FPGA processes the digital audio data using the TDOA and cross-correlation algorithms to calculate the direction of the gunshot. This estimation happens in real-time, almost instantaneously after the gunshot is detected.

➢ LCD Display:

- The FPGA sends the calculated DOA to the LCD for visualization. Depending on the type of display used, this can be a simple textual readout or a more sophisticated graphical representation that shows the direction of the gunshot relative to the microphone array.

CONCLUION:

- This project successfully demonstrated the design and implementation of a real-time gunshot detection system using only FPGA technology. By leveraging the FPGA's inherent parallel processing capabilities, the system efficiently handled multiple tasks such as signal conditioning, time difference of arrival (TDOA) calculations, direction of arrival (DOA) estimation, and gunshot classification. The FPGA-based design allowed for high-speed data processing with minimal latency, making it well-suited for time-sensitive applications like gunshot detection.
- Through the use of omnidirectional microphones and ADCs, the analog sound signals were digitized and processed on the FPGA in real-time. The system's performance highlighted the FPGA’s capability to handle complex algorithms, including digital signal filtering and machine learning-based classification, without the need for an external processor.

- This approach provides several advantages, including low power consumption, compact hardware architecture, and the potential for large-scale deployment in various environments, such as public spaces or military applications. Future work could focus on optimizing the machine learning algorithms on FPGA, expanding the system's detection range, and enhancing its robustness against noise or environmental disturbances.