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APPLICATION NOTES

Using Acoustic Beamforming for Pass-By Noise Source Detection

Abstract

This application note discusses a technique known as beamforming for determining noise location of passing vehicles. The technique has several challenges including simultaneous acquisition of a large sensor array, advanced signal processing and storage of extremely large data sets, and the environmental considerations required for operating outdoors on a test track. This application note recommends best practices for performing beamforming and walks through the complete system required from the test environment to the sensors to the math, analysis, and data storage with the goal of the reader being able to apply the technique.

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Introduction to Acoustic Beamforming

Traditional pass-by noise tests are useful for determining the sound level of noise sources on a vehicle during operation. These are often measured to ensure compliance with local and federal laws that govern acceptable sound emitted from a vehicle to produce minimal traffic noise. Pass-by noise source identification takes this one step further by identifying the source of the noise as well, so engineers can fine-tune vehicle design to minimize noise.

More Than Vehicles
This same technique can be applied to more than just automobiles. Boeing has used acoustic beamforming to help locate and reduce noise on its aircraft, and the Korean Railroad Research Institute has used it to reduce noise emissions on its high-speed KTX trains.

You can use several analysis techniques to identify noise sources including near-field acoustic holography (NAH) and beamforming. NAH operates in a sound's so-called "near field," within one or two wavelengths of the source, where sound waves act as circular waves emanating from the source. Beamforming, on the other hand, is more suited to the far field, located at roughly seven wavelengths from the noise source, where sound waves act as planar waves [1]. If you can think of a sound source like a pebble falling in water, then sound emanating from the source is like the waves that come from where the pebble hit. When you are close to where the pebble fell, the waves are clearly in a circle, centered on where the pebble fell. If you are standing far away on the shore, however, the waves no longer look like circles but just a straight line coming toward you. Sound waves behave in the same way. Because measurement sensors can often not be close enough to passing vehicles to be in the near field and perform NAH, beamforming is the preferred technique for noise location in such applications.

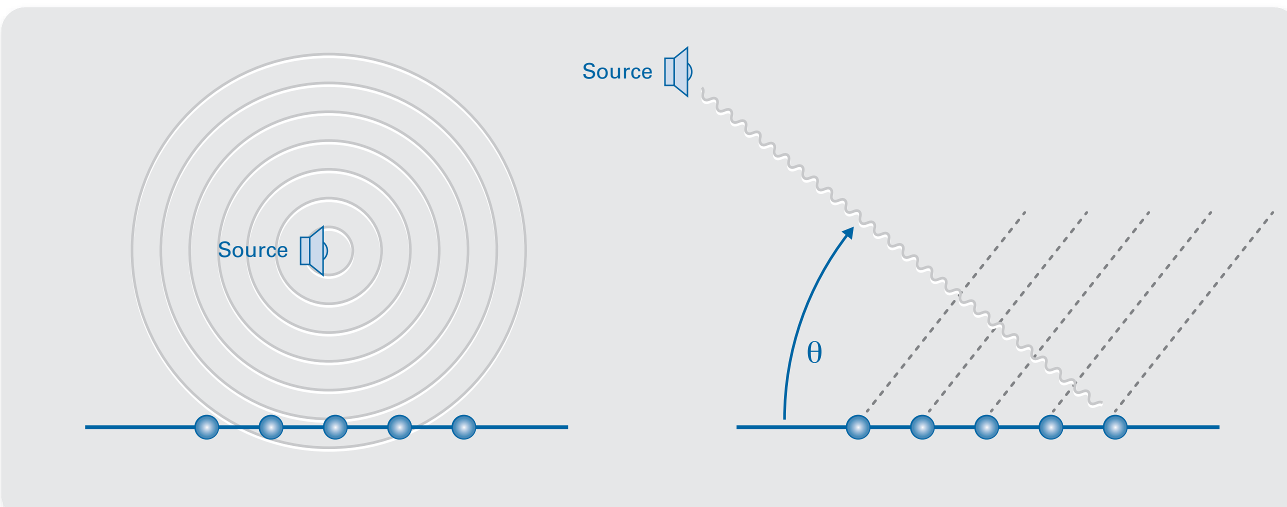

Figure 1. Sound waves in the near field (A) move away from the source in a circular pattern. In the far field (B), they are far enough away from the source that they appear to move in a straight line.

Beamforming presents several challenges including simultaneously measuring a large number of microphone sensors, transmitting and saving this large amount of acquired waveform data, and being able to quickly process large data sets to visualize the results in real time. This application note strives to offer the best techniques for performing beamforming on passing objects and provide guidance so that the end result is an accurate analysis of noise location and volume on vehicles.

In creating this application note, the author used NI cDAQ-9178 and NI 9234 C Series hardware to perform the data acquisition and NI LabVIEW software with the NI Sound and Vibration Measurement Suite for the signal processing and graphical display.

Measurement Standards for Vehicle Pass-By Tests

The purpose of the measurement dictates which of the many international standards detailing the required measurement methods you should adhere to. For example, ISO 362 stipulates a method for measuring the pass-by noise of a vehicle. Though the beamforming technique discussed here has a lot in common with ISO 362 methods, there are some notable differences, especially with measurement sensors. For example, ISO 362 uses single microphones for simple noise-level determination, while beamforming requires microphone arrays for source identification. You also may be interested in these ISO measurement standards:

- ISO 362—Pass-by noise
- ISO 13325—Tire noise
- ISO 5130—Exhaust noise

A standard that is certainly applicable to beamforming is ISO 10844, which specifies the test track for pass-by noise tests. The track a vehicle is tested on can affect not only the noise produced by the vehicle but also how sound waves propagate away from it. By controlling the track geometry and surface properties, the standard minimizes variation in tests performed at different testing locations [2]. Though not specifically required for beamforming by any international standards, it is recommended that you test your vehicles on an ISO 10844-compliant test track for consistent and meaningful results.

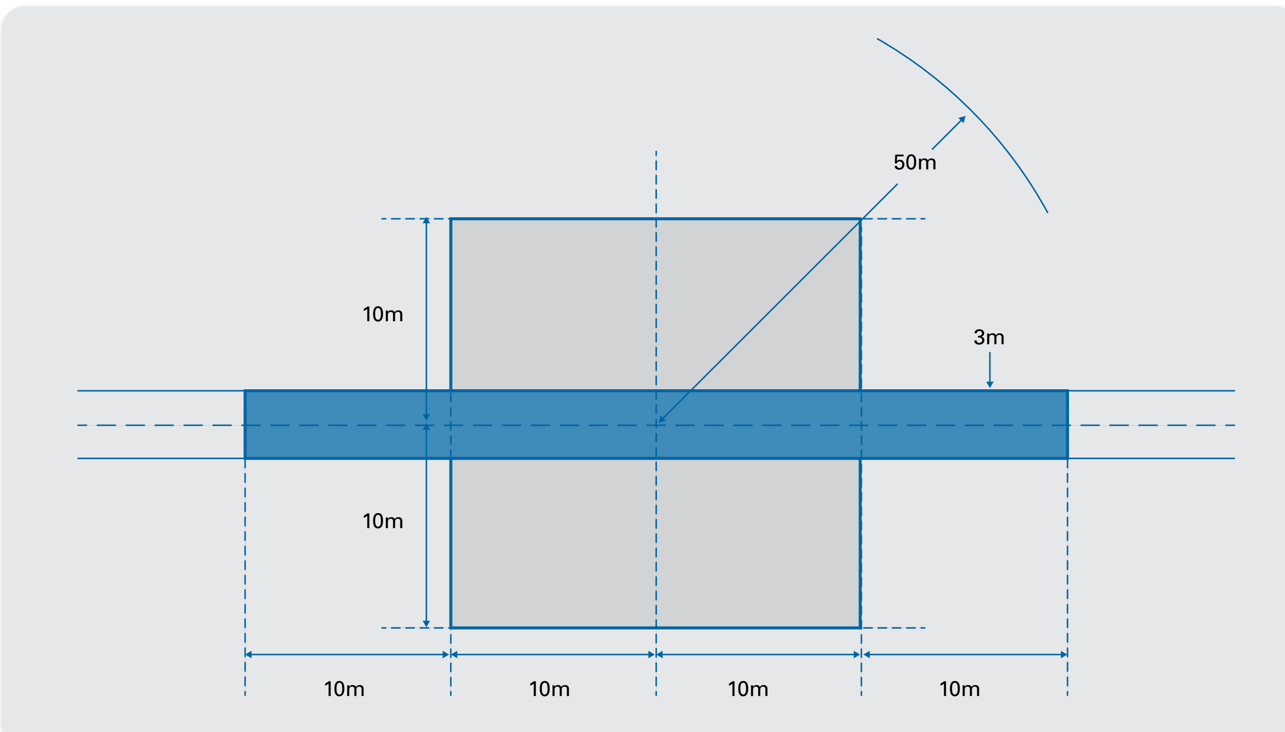

Figure 2. ISO 10844 specifies test tracks for vehicle noise emissions tests. You can use these same standards for noise source location using acoustic beamforming to reduce echoing and ensure repeatability between tests.

Measurements Resolution and Dynamic Range

Two of the most important aspects of any sound measurement, including beamforming, are the resolution and dynamic range of the measurement. Both of these aspects are dependent on the hardware in the system making the sound measurements, including the microphone array and the data acquisition hardware. Throughout this paper, explore recommendations on how to choose appropriate system components to maximize resolution and dynamic range.

Resolution in sound localization is more specifically spatial resolution, and refers to the ability of the system to resolve two separate sound sources from each other. The higher the spatial resolution, the closer the sound sources can be to each other and still register as two separate sound sources. The resolution of the system is dependent on the distance the sensors are from the source and the size of the microphone array.

$$spatial\ resolution = \frac{distance\ from\ source}{diameter\ of\ the\ array} \times sound\ wavelength$$

Dynamic range refers to the ability of the system to distinguish a sound from the noise in the environment around it. A measurement system with high dynamic range can more readily identify a quiet sound source from any noise or artifacts around it, whereas a system with poor dynamic range can identify only sound sources that are much louder than noise and artifacts around it. Dynamic range depends on the number and sensitivity of the microphones in your array, as well as the cables and data acquisition hardware you use.

Acoustic Beamforming System Overview

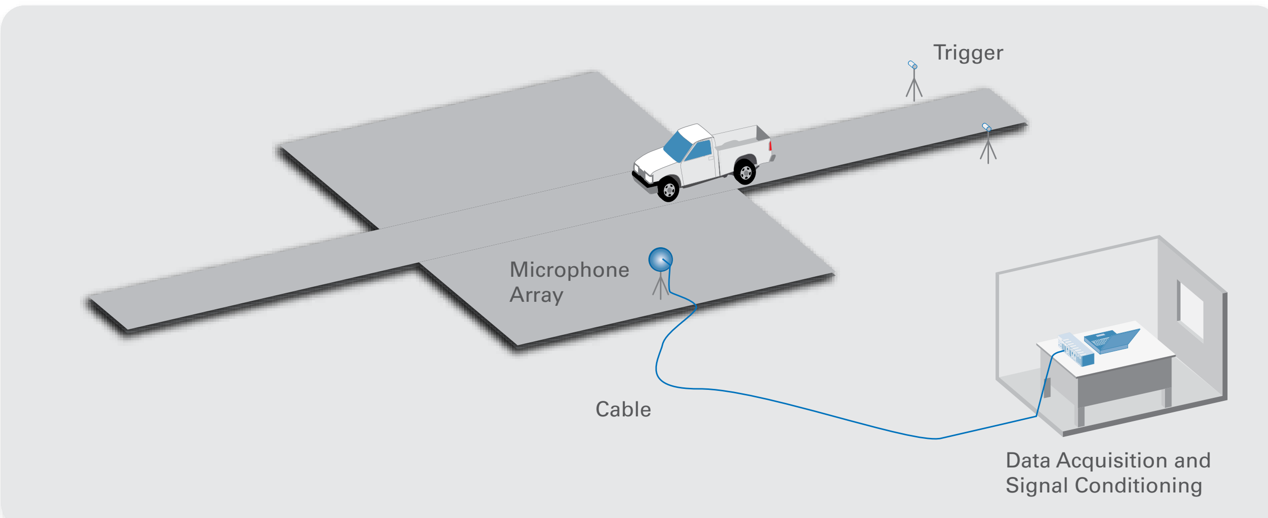

Figure 3. This complete system identifies the location of noise emissions from a passing vehicle.

Environmental Considerations on Test Tracks

Pass-by noise tests are performed exclusively outdoors because they need ample room for a vehicle to travel at high speeds by the microphone sensors. You can test indoors on a dynamometer, but the expense of building an anechoic noise dampening chamber with a full-size dynamometer far outweighs the cost of running the test outdoors. Though these tests are performed outdoors, your test system should be portable enough to be stored inside away from the elements.

Along with portability, the ability to run the system on DC or battery power is often helpful, improving the portability and reducing the cost of having to run power out to the system. Often portable data acquisition systems can also provide excitation to the microphones themselves without external circuitry to completely eliminate the need for AC power.

In addition to portability needs, your test system must handle certain temperature and moisture requirements. A test track in the middle of summer can be a hot and humid place, and a system designed solely to perform in indoor laboratory conditions may not fare well. Selecting a more rugged sensor and data acquisition system can help you ensure you maintain measurement quality in the worst outdoor conditions.

Microphone Arrays for Acoustic Beamforming

Several sensor companies provide out-of-the-box microphone arrays for noise source localization applications, and they are more than willing to help you select the proper array. The following section presents general guidelines for beamforming microphone array selection.

Beamforming microphone arrays are always circular and come in three main patterns: random, spiral, and ring. (In contrast, NAH microphone arrays are always rectangular.) The ring array performs well when the exact distance to the source is unknown, but it lacks dynamic range. The spiral array gives better results, but you can generally achieve the best performance from random sensor placement on the array, which offers the highest microphone field density and placement for the highest dynamic range measurements. [3]

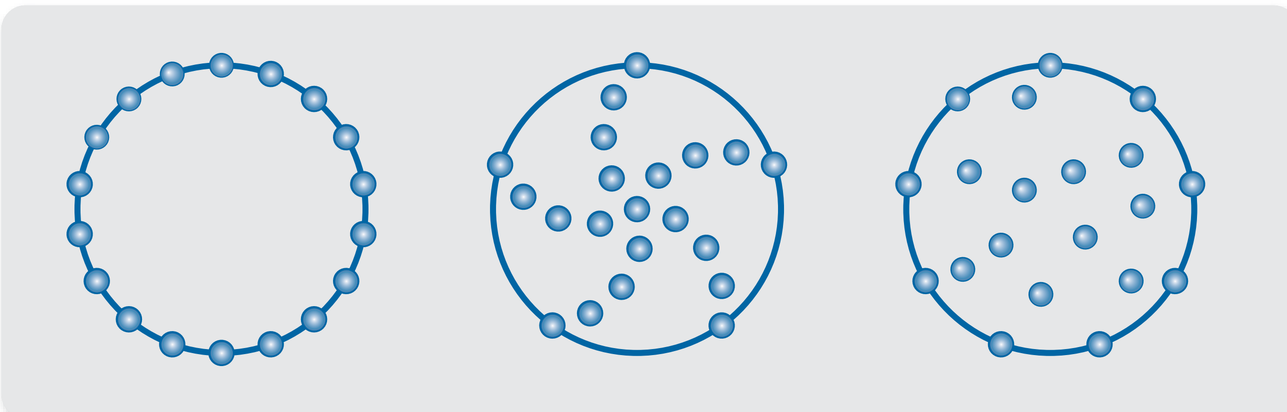

Figure 4. The three main configurations for microphone arrays used in acoustic beamforming from left to right are ring, spiral, and random.

As mentioned earlier, the spatial resolution of the system is directly related to the distance from the sound source. Essentially, the larger the array, the better spatial resolution you can achieve for your measurement system.

Traditional calibration techniques used with microphones are often unwieldy when working with microphone arrays because of the large number of microphones that need to be calibrated. It can be costly from both a time and resource perspective to individually calibrate all 30+ microphones in an array. When determining sound source location, however, absolute accuracy of the microphones is not as important as detecting changes in sound levels. You can take advantage of this to calibrate the entire array at once. By generating a tone at a known location in the field, you can create weighting factors for each microphone that result in correct location identification. Studies have shown that this method can provide system accuracy of ±0.5 dB when using microphones off by more than ±4 dB. [4]

Another downside of working with the large number of microphones used in an array is the cost of sensors themselves. Measurement-grade microphones can get expensive quickly, and you may be tempted to reduce the number of microphones in your array (and thus the end results of your tests) to save cost. However, recent developments in microphone technology have given rise to low-cost microphones, such as a microelectromechanical system (MEMS), that offer fair quality at low cost. NASA engineers took it upon themselves to research the viability of these microphone types for beamforming applications and determined that they offer sufficient quality for noise source detection when used in arrays. [5]

Wiring Considerations for Microphone Arrays

As explained earlier, higher dynamic range results in a better quality measurement. One of the easiest ways to increase the measured dynamic range of your system is to block out electronic interference in the signal. Most often, this interference happens in the cable from the sensor to the data acquisition device, which can act as a large antenna that picks up interference noise. To prevent this, most microphone sensors connect to coaxial cabling such as BNC or SMB that offers shielding of the signal wire to prevent noise pickup. Often microphone suppliers have the appropriate cabling to connect directly between the sensors and the data acquisition system (such as SMB microphone to BNC data acquisition connectors.

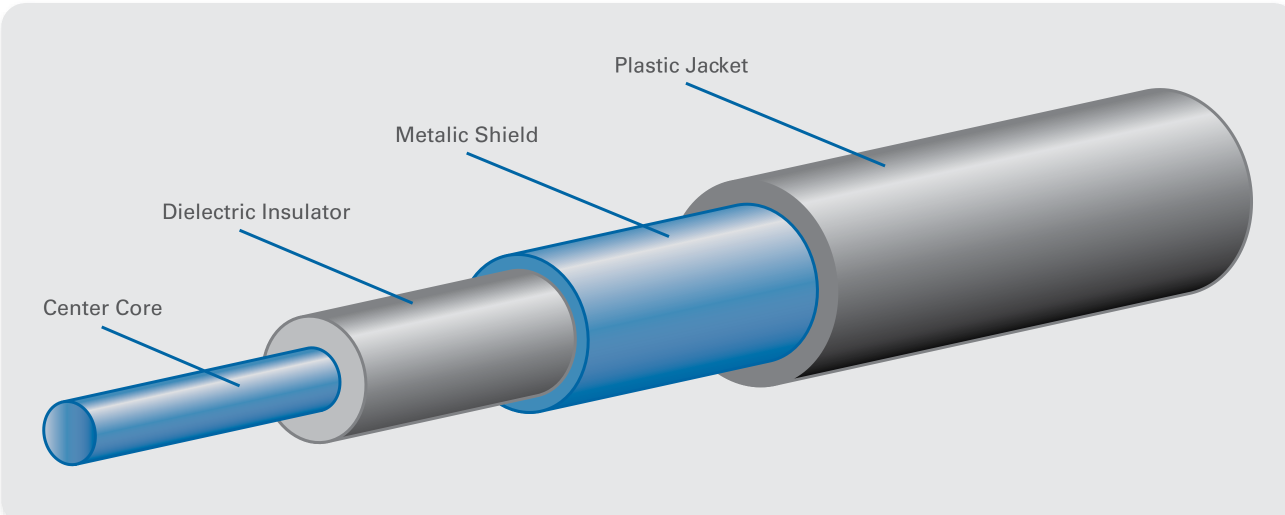

Figure 5. Coaxial cables shield the signal wire to prevent the electrical interference of the signal from environmental sources.

Cable Length

Cable length can be a concern for your application if you are running sensor cable farther than 300 ft. Two important signal characteristics can be affected by long cable runs: maximum signal frequency and signal amplitude. This is due to inherent impedance (or resistance) and capacitance in the cable caused by the gauge and material that makes up the internal signal wire of the coaxial cable. Generally speaking, lower gauge (thicker) wires have both lower impedance and higher capacitance.

Capacitance in a cable is the same as adding a capacitor in series with your sensor and acts as a lowpass filter to attenuate higher frequencies. Cable capacitance is specified in pico-Farad per linear foot, with higher capacitance resulting in a lower frequency cutoff for the lowpass filter. The maximum frequency that can be transmitted through a cable is as follows. [6]

$$fmax = \frac{10^9 \times (IEPE\ current\ (mA) - 1)}{2\pi \times capacitance\ per\ meter\ (pF) \times cable\ length\ (meters) \times max\ signal\ voltage\ (V)}$$

From this equation, you may note that you can increase your maximum frequency without changing any cable characteristics by simply increasing the excitation current to the sensors. While this is true, increased excitation current may not be supported by the sensors and may cause battery-powered current exciters to fail sooner because you are using more energy. Higher currents also lead to more internal heating of the microphone sensor, which may cause it to operate at a temperature outside its specifications—at best degrading measurement performance and at worst destroying the sensor.

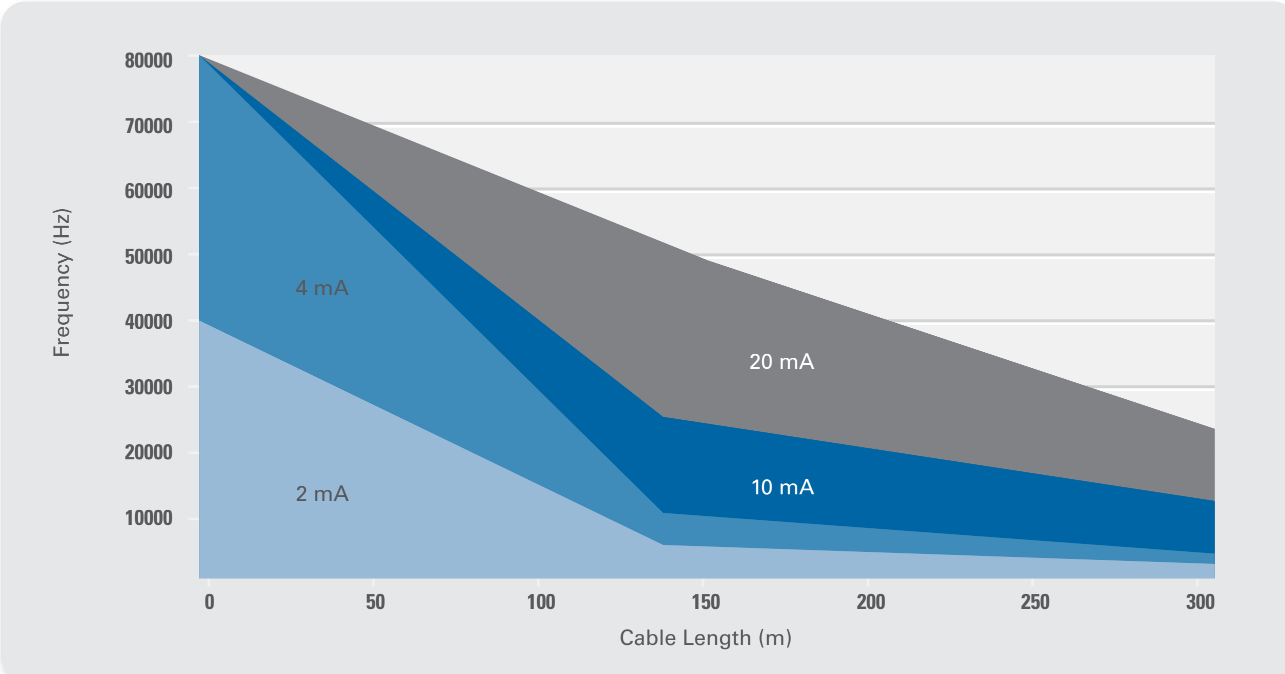

Figure 6. Acceptable Frequencies and Cable Lengths at Various Current Excitation Levels [7]

Signal attenuation results from the impedance in a cable, which is a function of both the physical parameters of the cable and the frequency of the signal passing through it. Most cable signal attenuations are specified in decibels per linear foot (or meter) for typical frequencies. Because attenuation is dependent on frequency, you can use cables specifically manufactured for certain frequency ranges. So if you are forced to run long wires and signal attenuation is a concern, you may want to seek out cables designed for your specific frequency ranges of interest. [8]

Signal Conditioning and Data Acquisition for Microphone Arrays

Microphone arrays are complex sensors. To function properly, they require a large amount of signal conditioning hardware and adept data acquisition hardware including excitation, anti-alias filtering, simultaneous sampling, and high streaming rates. Additionally, you can select optional features like IEEE 1451.4 (TEDS) to make life a lot easier.

Triggering

The system should first have some form of start trigger to tell the data acquisition system when to begin acquiring data. Because sound waveforms can take up such a large amount of storage space, you should record only the sound from the microphone sensors only when the vehicle is in range of the microphone array. To accomplish this, you can set a trigger at the start of the test track to signal when the vehicle is about to enter the microphone array's line of sight.

Often a trigger is made out of an IR light source, and a reflector is placed on opposite sides of the track. When the vehicle moves down the track, it passes between the source and reflector, interrupting the signal and firing the start trigger in the data acquisition system.

A camera for recording the image in the microphone's sound field can also be triggered off this same signal, ensuring that the video and sound location mapping are properly synchronized. For more information about displaying sound location results, see that section of this document.

Sensor Excitation

As far as sensor conditioning requirements, first consider the excitation for the microphones in the array. Traditional measurement-grade microphones use a constant-current excitation to power the internal electronics that detect sound waves. This type of excitation is generally known as integrated electronic piezoelectric (IEPE) excitation. This excitation can be provided by either external signal conditioning hardware or the data acquisition hardware itself. The advantage of integrating it into the data acquisition hardware is the reduction in system complexity and cost. By integrating as many of the system pieces as possible, the points of failure are reduced and system reliability is increased.

Lower cost MEMS microphones also require excitation to power them, but it is typically a simple voltage supply rather than constant current IEPE. As an example, the ADMP521 from Analog Devices requires a voltage supply of between 1.2 V and 3.3 V to function. In the case of MEMS microphones, a simple power supply provides the voltage to the sensors, rather than relying on the data acquisition hardware or external signal conditioning. This difference between excitation philosophies is due simply to the low cost and easy availability of voltage power supplies as opposed to IEPE constant-current power supplies.

Anti-Alias Filtering

Anti-alias filtering removes any alias frequency components from the analog signal by lowpass filtering it before it is digitized by the data acquisition hardware. Without an anti-alias filter, frequencies higher than the Nyquist frequency of the data acquisition hardware register as lower frequencies and could skew the results of the beamforming.

Similar to sensor excitation, this filtering was traditionally performed with external signal conditioning hardware. Today, however, most microphone data acquisition hardware, such as the NI cDAQ-9178 and NI 9234 C Series module, incorporates filters, reducing system cost and complexity.

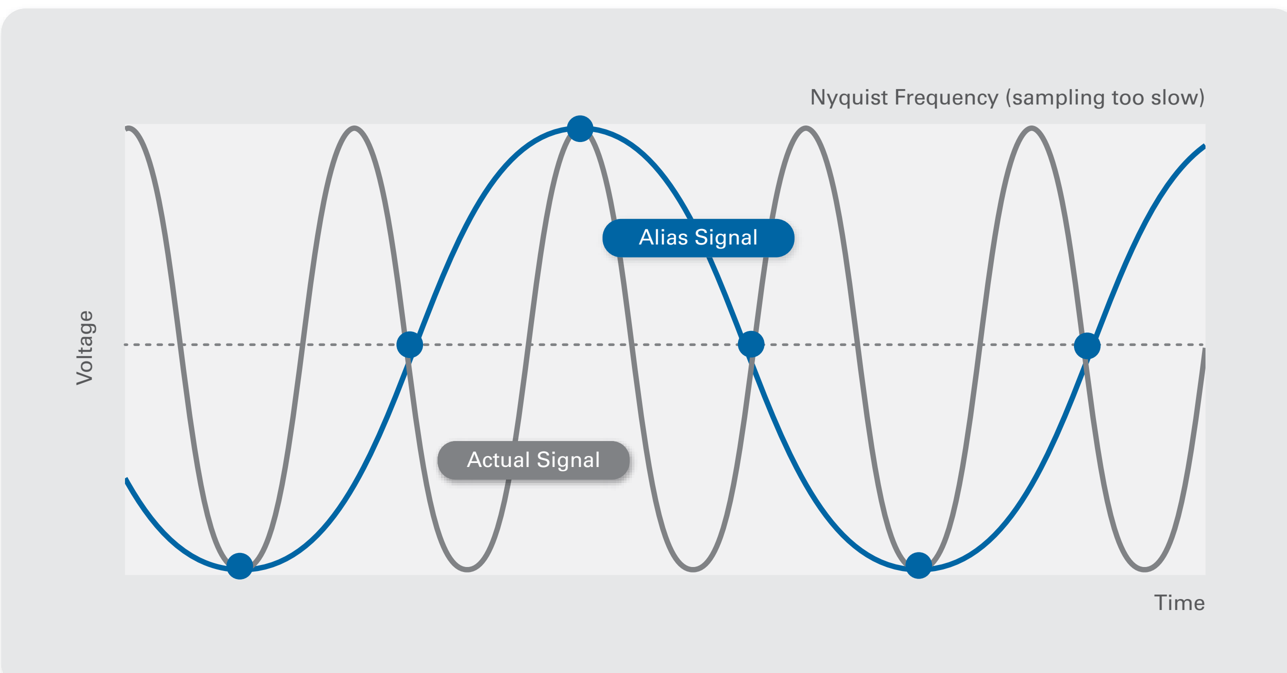

Figure 7. Without an anti-alias filter, frequencies higher than the Nyquist frequency of the data acquisition hardware register as a lower frequency and skew the results of the beamforming.

Simultaneous Measurements and Channel Synchronization

Ensuring that you can successfully synchronize and simultaneously sample high-channel counts of microphones is one of the most important requirements of your data acquisition system. The most common calculation method used for beamforming is to accurately measure the time delay between the different microphone sensors in the array. To accomplish this, all of the sensors must be sampled at the same time, or the skew in the sampling throws off the results of the calculations. To this end, the data acquisition hardware must offer simultaneous, synchronized sampling of all the microphones in the array.

When there are just a few microphones, you can easily accomplish this because the required number of data acquisition channels can be contained in a signal data acquisition chassis or even module. But large arrays typically require more channels than can fit in a single chassis or device, so the ability to synchronize between devices becomes crucial.

Data Streaming Rates and Data Transfer Bus Selection

When dealing with large amounts of sensors, you must consider data streaming rates, that is, the amount of data that the connection between your data acquisition modules and your computer can handle. This is especially true with microphone arrays for which data acquisition samples are larger in size to accommodate high dynamic range, and samples are taken more often so that higher sound frequencies can be sampled.

To calculate how much data bandwidth you need for your application based on the maximum detectable audio frequency you want to detect, you can use the following equation. Note that data acquisition systems almost always acquire data in either 2 bytes per sample or 4 bytes per sample. So though your data acquisition card may offer 18- or 24-bit resolution, the bits per sample as far as data transfer and storage are concerned is still 32 bits (or 4 bytes).

$$data\ rate\ (Mbps) = 2 \times maximum\ frequency\ (Hz) \times bits\ per\ sample\ (bits)$$

You can select from several communication buses to transfer this data. Each offers its own advantages and disadvantages that you need to weigh for your particular application. The most common bus types for high-bandwidth data acquisition like the kind required for beamforming are USB, Ethernet, and PCI or PCI Express.

USB is the most common bus for consumer use and has made its way into the data acquisition market. It offers the advantage of plug-and-play ease of use and commercial off-the shelf costs coupled with fast bandwidth (256 Mbps). USB cables, however, tend to be shorter and have a maximum length of 5 m. USB is a good all-around communication bus if you can place your data acquisition system in the same room as the storage and signal processing computer.

Ethernet data acquisition systems use the same technology as Internet connections to provide data transfer. Though the consumer market is slowly starting to support Gigabit Ethernet, most data acquisition systems are still limited to 100 Mbps transfer lines. Ethernet's main advantage, though, is its ability to run cables long distances. So if your data acquisition needs to be separated from the storage and signal processing computer by some distance, than Ethernet may be a good bus choice.

PCI and, more recently, PCI Express take advantage of the same technology used in computers to transfer data internally, such as between the processor and the video card. PCI technology can be used for data acquisition either with a data acquisition board that plugs directly into a PCI or PCI Express computer slot or by using a PXI system, which extends the PCI bus outside the computer, usually with a MXI connection. The advantage of using a PCI-type communication bus is raw data transfer speed, which is up to 16 GB per second of data. This does come at a cost, however, because PCI and PXI systems tend to be more expensive than either USB or Ethernet. High-channel-count systems are a better fit for this bus technology because the high bandwidth is truly needed to accommodate large amounts of data transfer.

	Advantages	Disadvantages	Use Case
USB	■ Medium bandwidth ■ Lower cost ■ Easy connectivity	■ Short cable length	Data acquisition can be close to the computer; medium-channel counts
Ethernet	■ Long cable length ■ Lower cost ■ Easy connectivity	■ Low bandwidth	Data acquisition needs to be far from the computer
PCI Express	■ High bandwidth	■ More expensive ■ Short cable length	Very high-channel count (above 100 channels)

Table 1. Comparison of the Most Common Communication Buses for High-Bandwidth Applications

Transducer Electronic Data Sheets (TEDS)

TEDS can significantly help with the setup and configuration of your microphone array systems. TEDS store information about the sensor on a tiny piece of EEPROM memory inside the sensor. When connected to a TEDS-enabled data acquisition system, the sensor can send information identifying itself and provide calibration and sensitivity information to enable the autoconfiguration of large amounts of channels.

Signal Processing and Analysis for Acoustic Beamforming

So far you have focused on simply getting the acquired sounds from the microphone array to the computer for signal processing. Now examine the actual signal processing technique to determine the noise source.

Noise source detection using beamforming is accomplished through a "delay and sum" method. As you have seen, an array of microphones is used to measure the sound from the unit under test. All microphones are simultaneously sampled, and their recorded data is then shifted to compensate for time delay due to differences in the microphone distance from some origin point. The time-shifted recorded sounds are then summed to determine the direction of highest sound pressure.

Because beamforming operates in the far field, sound waves can be treated as planar, meaning that if the sound source were centered in front of the microphone array, it would reach all microphones at the same time. If the sound source were offset to one side, it would hit the microphones closest to the source first, followed by the ones farther away with a time delay proportional to the distance the microphones are apart [9]. Beamforming takes advantage of this delay in sound wave propagation between multiple microphones to calculate exactly where the sound originated from in front of the microphone array.

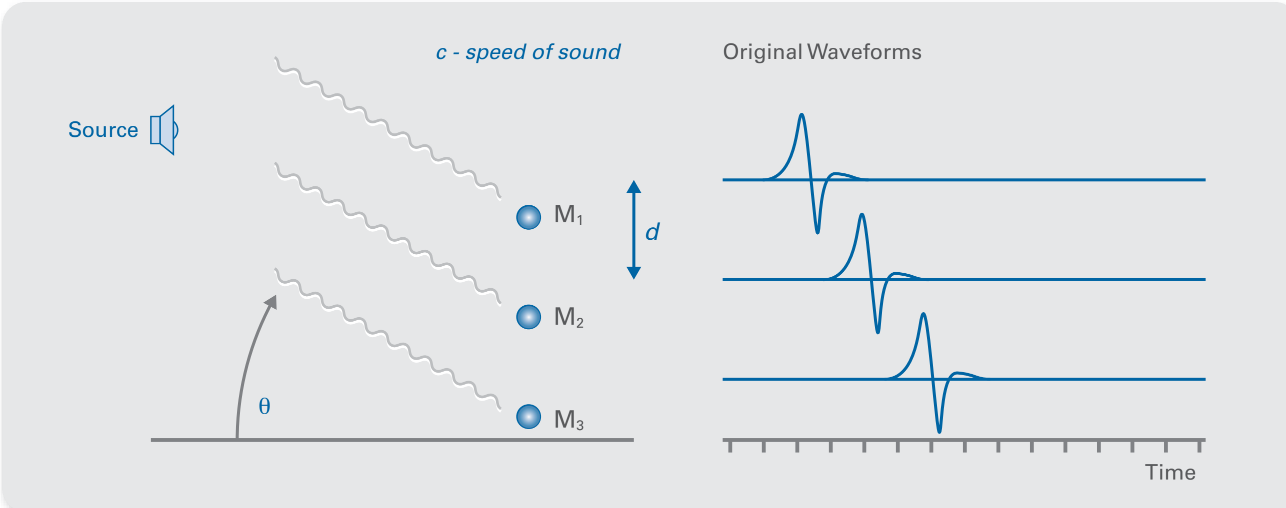

Figure 8. If the noise source is offset from the microphone array, the sound waves reach the closest microphones first, causing a measurable time delay between each microphone based on the distance between them.

The process of using the delay and sum beamforming technique is as follows for a single line microphone array. Once you understand the derivation, you can extrapolate this to 2D geometry.

1. Sound waves are individually multiplied by a weighting factor corresponding to each microphone. This compensates for calibration and known noise in the test field.
2. An initial sound wave incident angle is chosen and the time delay for each microphone based on that angle is calculated using the following equation:

$$t_n = \frac{(n-1)d \cos \theta}{c}$$

3. The time delay for each microphone is applied to the sound waveform recorded for that microphone, shifting the start time of the recorded sound for that microphone by the time delay.
4. The waveform from each microphone is summed to give a total power level.
5. The process is repeated for all possible incident angles, giving an intensity map across the microphone array's field of sound.
6. The location of the highest intensity on the map corresponds to the sound source location.

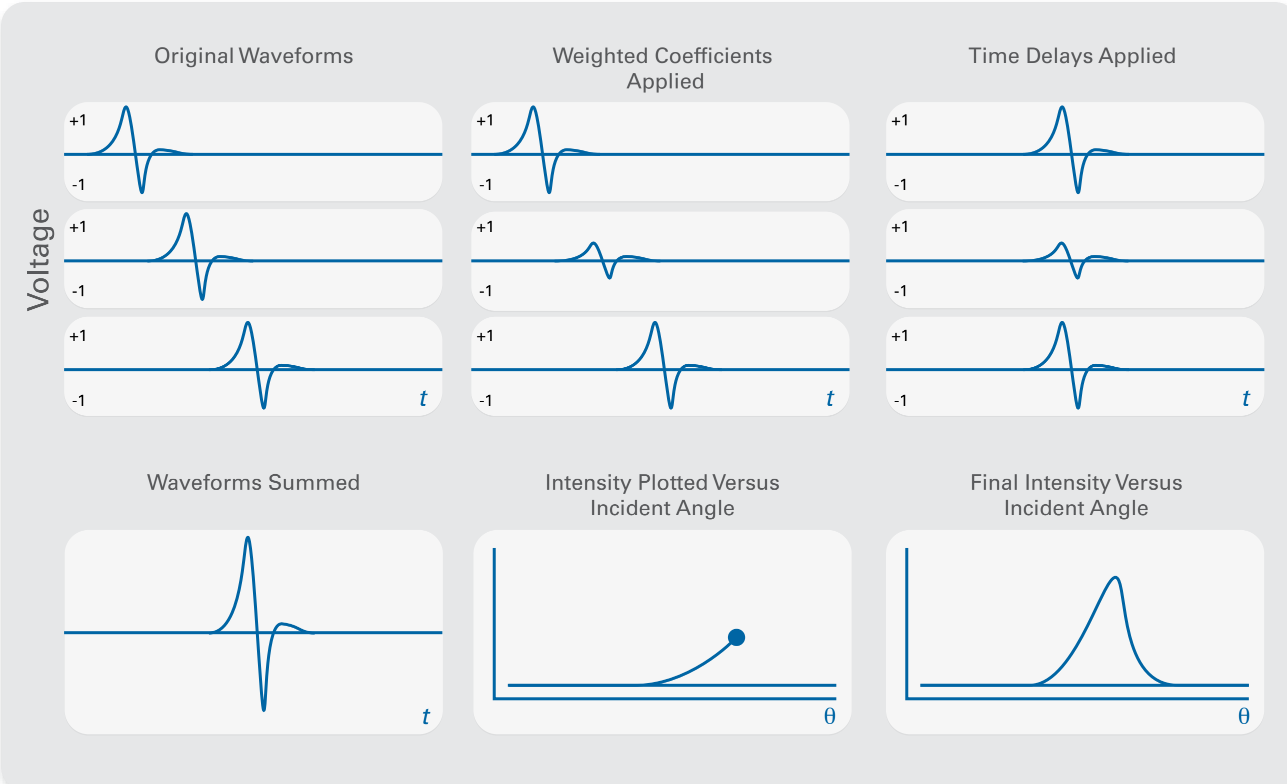

Figure 9. The delay and sum method is one of the simplest and most common algorithms used for acoustic beamforming.

Calibration and Weighting Filters

Before you apply time delays to the acquired sound waveforms, you multiply them by weighting constants. Each microphone has its own weighting constant associated with it that helps compensate for local noise in the test field.

You also can use these weighting filters to calibrate the microphone array. By placing a noise source at a known location in the test field, you can adjust the weighting constants of the microphones until the intensity map output from the analysis matches the known location.

Displaying Noise Source Results

To display the results, you ideally project the location of the detected noise source onto an image of the passing vehicle. You can implement this by combining live camera footage with an intensity map overlaid to display the noise location as shown in Figure 10.

Some microphone arrays come with a camera already placed at the center of them. However, if you are building your own array or yours did not come with a camera, you can place a simple USB or Ethernet webcam in the center of the microphone array facing the test track. By triggering the camera recording start to the sound waveform recording start from the microphone, you can properly align the calculated values of the sound location to a location in the camera frame.

Figure 10. By laying the intensity map from the delay and sum beamforming method over a camera image taken at the same time you recorded the sound, you can visualize the noise source better.

Data Storage

As you saw earlier, beamforming and other microphone array applications can generate significant amounts of data. This is a concern not only for data streaming and transfer rates but also for data storage. Typically sampled waveforms need to be saved to disk during the tests, either for later analysis or for archival and later reference, because designs of this vehicle under test change. Saving data sets this large can be difficult if not impossible for standard hard drives. You can choose from two solutions to this problem: newer solid-state drives (SSDs) or redundant arrays of inexpensive disks (RAIDs).

SSDs offer fast write speeds (up to 435 MB/s in third-party benchmarking tests [10]), but because they rely on newer technology, they tend to be more expensive and provide limited storage capacity. Though the write speeds are akin to the requirements of microphone arrays, the storage sizes are generally not large enough to accommodate the data sets, but they may be a viable solution depending on your application.

RAIDs are the typical solution to overcoming the limitations of traditional hard drives. They work by placing many traditional off-the-shelf drives in parallel and then treating them as one hard drive through a software driver. The more drives you add in parallel, the larger the storage space and the higher the write speeds you can achieve. In addition to these two advantages, you can choose from a variety of RAID configurations such as mirroring to create redundancy in your data and protect you in case of drive failure. [11]

The disk storage configuration you choose ultimately depends on the channel count, sample rate, and sample length of the data you acquire. While slower, lower-channel-count systems may be able to hobble along with a standard laptop drive, higher-channel-count systems rely on RAID configurations.

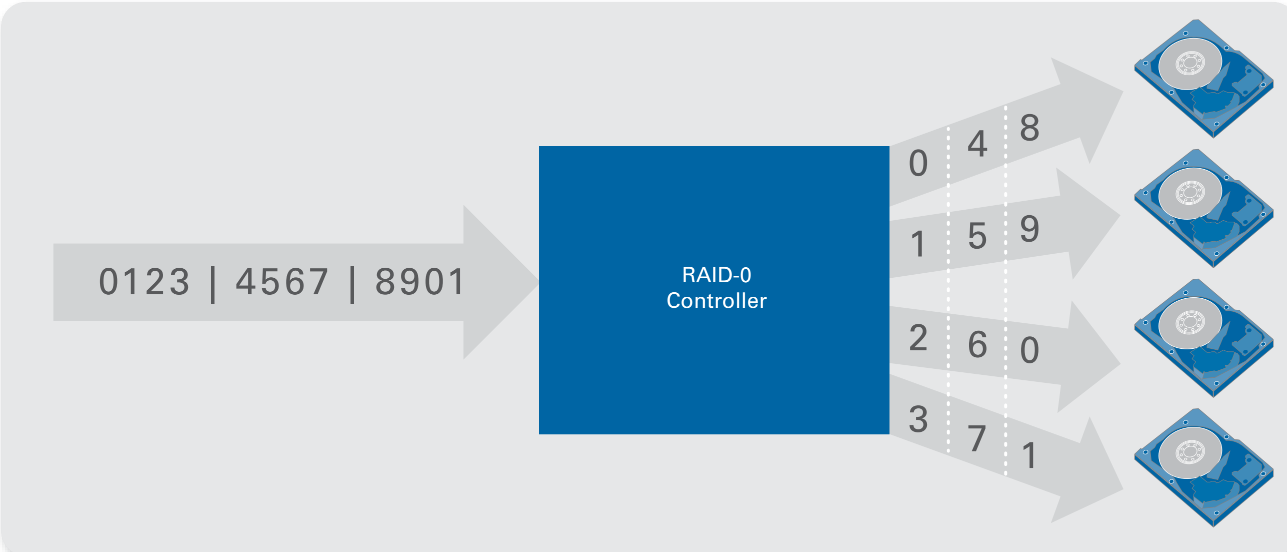

Figure 11. RAIDs place multiple traditional, inexpensive drives in parallel to increase write/read speeds and storage capacity.

Additional Resources

- Case Study: Acoustic Beamforming in Vehicle Pass-By Noise Tests – Read how UFSC used NI CompactDAQ and LabVIEW to develop a portable and affordable acoustic beamformerf for noise source identification in pass-by noise measurements.
- National Instruments Acoustic Beamforming System – View pricing for the NI Acoustic Beamforming System with NI CompactDAQ and LabVIEW and configure it for your application.

Notes

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