



M4

A White Paper

An In-Depth Look At The Community M4 Midrange Driver

by Pat Brown





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C O M M U N I T Y P R O F E S S I O N A L L O U D S P E A K E R S

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M4™ A Device for the Critical Midrange



The purpose of this investigation is to describe a device that has been designed specifically for the frequency band that extends from about 200 Hz to 2000 Hz, which is commonly referred to as the "midrange." It shall be put forth that this decade is critical to sound reproduction since it contains the majority of audio information for speech and music. Support for this assumption will begin with an overview of the basic information from which it is drawn.

The human hearing system is sensitive to modulations, or ripples, in the atmospheric pressure that surrounds us. This pressure can be "rippled" anywhere from less than once per second to many thousands of times per second. The typical human ear/brain system is sensitive to frequencies between 20 and 20,000 cycles per second, or Hertz (Hz). The purpose of a sound reinforcement system is to reproduce this spectrum with sufficient acoustic power to produce a desired level at some listener distance. The human ear, as marvelous a device as it is, does not cover the entire spectrum with equal sensitivity. In fact, the human ear/brain system is optimized for reception of the middle portion of the entire passband, most likely due to our dependence upon speech in our day-to-day lives. This middle portion, which we will call the midrange, contains the most critical information for the human receiver. With this in mind, it is interesting that this critical band has long been overlooked in the design and implementation of sound reinforcement components and systems. One purpose of this paper is to put forth the idea that the midrange is the most critical part of the spectrum, and that superior reinforcement systems can be constructed if developed around the midrange decade.

The first step of this investigation is to determine the requirements for an optimum transducer for coverage of the middle decade of the audible spectrum. Such a transducer would serve as a complement to the already existing variety of excellent transducers for the other two decades of the audible spectrum. While there are many approaches to component design, perhaps the best course is to allow such a

device to define itself based upon the physics of the sound that we wish to reproduce. This approach will discourage any temptation to take an existing transducer and stretch its parameters to include the middle decade, a common practice in the audio industry. As with the other two decades in the audible spectrum, the requirements are clearly defined by the physics of the sound that we wish to reproduce.

The essential requirements for optimum midrange reproduction shall include, but not be limited to:

- High Efficiency**
- Low Distortion**
- High Power Handling**
- Durability**

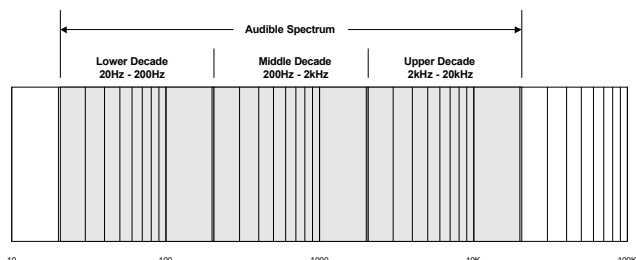
Let us first describe the desired passband of such a device based on the essential parameters of wavelength and distortion.

The response of all transducers is frequency dependent. This simply means that they do not and cannot behave the same at all frequencies, due to the physical characteristics of sound waves. In everyday life, the physical size of something nearly always determines how we handle it. The same is true for sound waves. Since sound propagates at a velocity that is not frequency dependent, the physical size of a sound wave is directly related to its duration and velocity. This can be understood by considering that a single cycle of a 1000Hz tone lasts for about 1ms. Since this wave is traveling at approximately 1130 feet per second (344 meters/sec), distance (or length) is equal to the product of time and velocity, the length of the wave will be:

$$\lambda = \frac{1130}{f} = 1.13 \text{ ft} \quad \text{or} \quad \frac{344}{f} = 0.344 \text{ m}$$

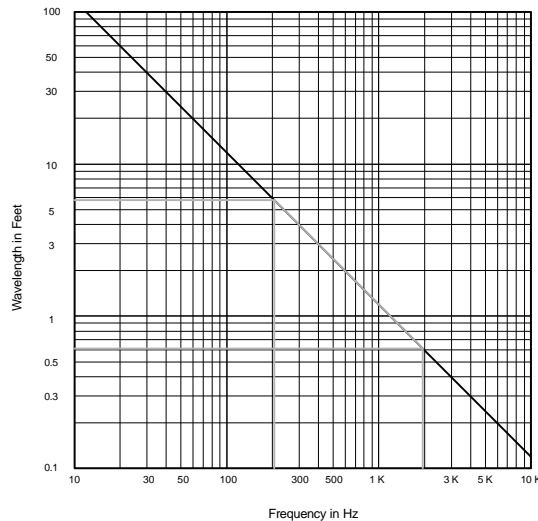
where:

f is the frequency in Hertz



A 100Hz tone has a period of about 10ms, and since it travels at the same speed as the 1000Hz wave (but lasts longer) its physical length becomes about 11 feet (3.44 m). The common term used to describe the physical length of a wave in free space is "wavelength." Since wavelength is inversely proportional to frequency, as the frequency goes lower, wavelengths get longer. And, as wavelengths get longer, the physical attributes of the transducers that reproduce them must be altered accordingly.

Since we are defining the characteristics of an optimal midrange device, the passband of such a device should be considered in terms of wavelength. While such a description will clearly define the physical dimensions of a pattern control horn, it will also serve to define the size and other physical properties of the piston that shall ultimately drive such a horn. Our optimum midrange device must be capable of generating large amounts of acoustic power from 250Hz to 2500Hz, and the waveguide must provide directional control for the corresponding wavelengths ranging from about 6 inches to 6 feet.



Nomograph showing the relationship between wavelength and frequency.

Speech vs. music

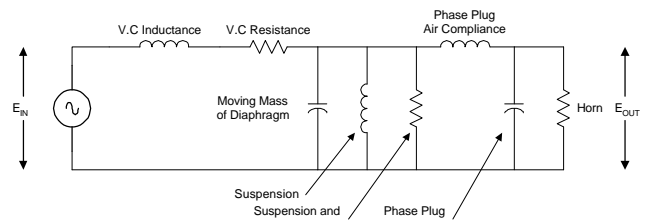
It may be demonstrated that the requirements for speech systems are unique, and are not necessarily met by systems that are favored for music reproduction. Music is an inherently non-linear event, meaning that there is often no exact criteria by which to evaluate it, and that colorations that were not present in the original are often tolerated or even desirous in reproduction. We may conclude that music systems are best evaluated by listeners and not test instruments.

Research has shown that speech is not so subjective, and that many of the artifacts that are considered acceptable or even desirable in music systems cannot be tolerated in speech systems. The reverb and delay added to a singer's voice suddenly becomes unacceptable when the person stops singing and begins speaking. Unfortunately, many loudspeaker systems have this same "Jekyll and Hyde" personality.

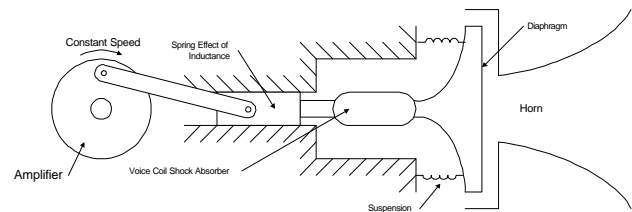
Design of a Midrange Driver

The circuit in the following figure depicts a simplified electrical equivalent circuit of a compression driver, and the analysis of it yields a series of useful equations for driver parameters. These are set forth in several texts, dating back

to Olson's work in the 1930's and 40's. Viewing transducers as ladder networks of resistive and reactive components leads to an understanding of the inherent signal delay properties of such systems. It is the energy storage characteristics of the devices reactive elements that cause phase (time) anomalies. The clever transducer designer understands that these are present and accounts for them in the design.



A more revealing depiction would be a mechanical model, as shown below:



Notice that the only thing that stands between the amplifier and the diaphragm is the mechanical resistance R_{EM} , which simplifies to:

$$R_{EM} = \frac{B^2 V_{VC}}{p}$$

where:

V_{VC} is the volume of the voice coil

p is the resistivity of the voice coil wire

B is the flux density

This simple relationship tells us that the *higher the field intensity and the larger the volume of the voice coil, the greater the efficiency* (due to lower losses in R_{EM}).

The M4 maximizes both of these important parameters, making the M4 the most efficient loudspeaker on the market to date. Let us look deeper into the anatomy of the M4 and examine why it stands alone as the optimum driver for the critical midrange.

What is Acoustic Power?

The concept of acoustic power is not readily considered by most people, due in part to the industry's preoccupation with sound pressure. To put it simply, sound pressure is the result of sound power. Sound power is the *cause*, sound pressure is the *effect*. The goal in driver design is to maximize the amount of acoustic power available from the device, given the constraints and trade-offs between the available parameters. Once the acoustic power has been realized, we then turn our focus to pattern control devices (horns) that will channel this power to a smaller unit area, increasing the sound *intensity* for that area, increasing the sound *pressure* (at a given listening position) beyond what it would have been if no pattern control was used.

In many conventional cone-type transducers, it can take as much as 100 electrical watts to produce one acoustical watt, yielding an efficiency of about 1%. At the other extreme, a perfectly efficient transducer (that unfortunately does not and cannot exist) would produce one acoustic watt with the application of one electrical watt. The M4 comes closer to this ideal than any other driver available today, producing nearly one-half watt of acoustic energy from a single watt of electrical energy.

One acoustic watt is equivalent to 107.5 dB SPL at four feet from an omnidirectional source. A device's directivity can be described with a term called "Q" which is an indication of its ability to confine the applied energy to a smaller unit area. Q, therefore, is a parameter of the horn, not the driver. If we take one acoustic watt and couple it to a horn with a Q of 2 (hemispherical), the intensity and therefore the pressure will increase by a factor of 2 to 1 in the area covered by the device. This is quite useful, since it allows more energy on the audience and less on the walls and ceiling. It is common practice to convert the Q rating into decibels by the formula:

$$DI = 10 \log Q$$

where DI is the directivity index

By considering Q in terms of the Directivity Index, the increased sound pressure provided by increasing Q beyond unity ($Q = 1$) is readily apparent. Let's look at the numbers and get a feel for what a large source of acoustic power and a high Q horn can do for a system design (see figure below). When you consider that a 3dB difference in efficiency represents a two-to-one power ratio, the benefits of the M4 become very apparent. Also consider that the M4 is more efficient in its passband than some conventional mid/hi frequency drivers are in their passband. This means that it can require several high-frequency drivers for each M4 in the system, for flat power response.

Power Capacity

With 200 watts of electrical power applied, the M4's power output at 250Hz is 100 acoustic watts. This output increases as frequency increases, due to lower excursion requirements. While electrical power handling capability is a much more common specification, it tells nothing about how much sound comes out when that power is applied to the device. It only indicates how much amplifier power can be applied to the device without damage.

The electrical power capacity of a loudspeaker is determined primarily by two factors: its excursion limits and its thermal limits (caused by voice-coil heating). Since both vary with frequency, the actual power capacity of a loudspeaker is said to be frequency dependent, therefore being a complex function of frequency.

The power capacity of the M4, by the AES standard, is 200 watts. This 200 watt rating primarily describes M4's thermal limits. At low frequencies, below 300Hz, M4's power capacity will be primarily excursion limited. For voice or musical program material, the 200 watt rating describes the M4's power capacity accurately.

How much sound can you get from one electrical watt?

SPL at 4 feet

Source of Acoustic Power	% Efficiency	Acoustic power from 1 electrical watt	With omni horn Q=1	With 90x40 horn Q=10	With 60x40 horn Q=16	With 40x20 horn Q=50
Cone Loudspeaker	3%	0.03	93dB SPL	103dB SPL	105dB SPL	110dB SPL
2" Mid/Hi Driver	35%	0.35	103dB SPL	113dB SPL	115dB SPL	120dB SPL
M4	43%	0.43	105dB SPL	115dB SPL	117dB SPL	122dB SPL

Achieving Efficiency

Efficiency is a measure of how much acoustic power results from an application of electrical power.

In the early days of audio, transducer efficiency was a prime consideration, mainly because the largest amplifiers were able to produce only a few watts of power. High efficiency meant that most of this power could be converted into sound. With the advent of solid state electronics, amplifier power became quite economical. These days, amplifiers in the thousands of watts are readily available and (at least compared to the old days) inexpensive. So why would we continue to be concerned about efficiency?

The answer lies in what a transducer does with the electrical power that it receives. Each watt drawn by the transducer is converted into one of two types of energy: sound or heat. While sound is what we are after, it is inevitable that heat will also be produced, and heat is a main culprit for many transducer failures in high-level systems.

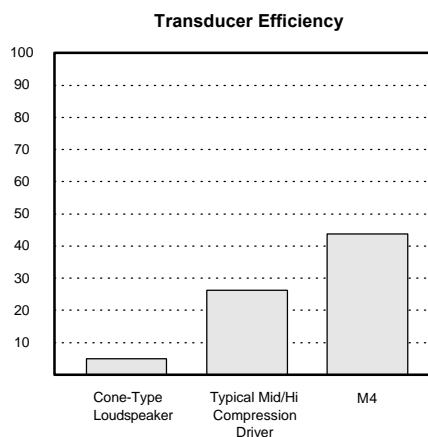
While a frequency response specification is useful, it may be misleading at high frequencies due to high-frequency beaming, a phenomenon that occurs as the wavelength being reproduced by a driver becomes small when compared to the diameter of the driver. Even though the on-axis response is flat, the pattern has narrowed and there is less off-axis coverage. Power response is best measured on a constant-directivity horn or plane wave tube, which gives a better representation of these two parameters. Simply stated, a highly efficient transducer produces more sound and less heat.

Efficiency in a transducer is proportional to the amount of magnetic flux that can be concentrated into a magnetic field gap. This magnetic field gap should be made as narrow as possible, yet not so narrow as to allow the voice coil to rub as it moves through the gap.

The M4 achieves a highly concentrated magnetic field by using a unique focused geometry and an added flux stabilizing ring. This combination helps keep the magnetic field from varying under high power inputs.

This voice coil and magnetic assembly, when combined with other design features, result in an amazing 43% efficiency rating for the M4. This means that 43% of the incoming electrical energy is actually converted into acoustical energy, and only 57% is converted into heat. The following chart shows the typical efficiency ratings for various types of transducers. The high efficiency of the M4 means that it will be less prone to heat related problems, such as power compression, and therefore much more reliable than other devices.

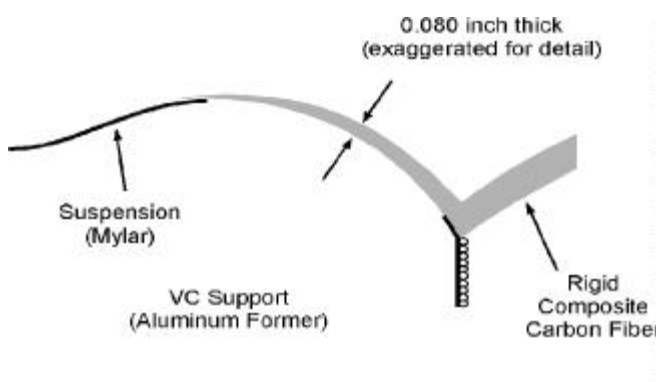
Power compression results from the tendency of voice coil impedance to rise as temperature increases. This causes less power to be drawn from the amplifier, and hence a reduction in acoustic output. The power compression in many devices can be several dB. Due to the M4's efficiency, virtually no power compression occurs.



Composite Diaphragm

A large diaphragm is essential for a 4-inch midrange driver capable of producing 100 acoustic watts. The diaphragm of an M4 is a full 6.65" (16.89 cm) in diameter and has an effective piston area of about 40 in² (258 cm²). This is due in part to the use of the outer suspension as part of the piston. The phase plug throat area of 8 square inches with this 40 square inch (258 cm²) diaphragm results in a compression ratio of about 5:1. The M4's maximum diaphragm excursion is +/- 0.1 inches (2.54 mm), rivaling that of some woofers.

All loudspeaker diaphragms begin to break-up at some high frequency. This is the point where the diaphragm is no longer acting like a piston. The design goal of the M4 was to extend high-frequency response to beyond 2000Hz. Originally this was accomplished by the development of a sandwich diaphragm with 2.0 mil aluminum "skins" and a rigid, low-density foam core. To further reduce breakup, the outer edge of the diaphragm was rolled, greatly increasing its stiffness.



The M4 uses a carbon fiber diaphragm to raise the frequency of the first breakup mode out of the device's passband.