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TECH NOTE

Motional Feedback Loudspeaker Essentials

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In 2017, Grimm Audio introduced the DMF ("Digital Motional Feedback") technique for its subwoofers, that uses active feedback for more accurate bass reproduction. Motional Feedback as Philips introduced in the seventies was brought back to life using modern digital electronics. An essential element in these systems is an acceleration sensor that is mounted on the cone, which measures the radiated sound. Sometimes alternative types of sensors are introduced that sense the diaphragm position or velocity, with potentially lower cost as advantage. It can be shown that these alternatives offer limited or no use at all to improve sound reproduction quality. In an earlier paper series, I dove deep into the theory of motional feedback, aiming at technically skilled people. Because unfounded claims about alternative sensors often surface however, the need for a more comprehensive blog on this subject for a wider audience was felt.

Measure for feedback

In an active system, **feedback** is the term for measuring the result of an action and then comparing this with the intended result to correct mistakes caused in the action. These corrections indicate that feedback is aimed at reducing errors. For audio this means reduction of distortion but also reduction of dynamic effects like resonances in loudspeakers.

Feedback is all around us, not only in electronic systems with amplifiers, but also in ourselves. Imagine someone is pushing you. You feel and see (measure!) a displacement and since you don't want to fall down, your brain will activate your muscles to exert a force opposite to the pushing force in order to reduce the displacement. This means that a feedback controlled system consists of the following elements:

- A driving mechanism that performs a demanded action, like muscles or a motor.
- A measurement device that monitors the resulting action, like eyes, nerves, a camera or a sensor.
- A control device that compares the measured action to the demanded action and calculates the appropriate correcting action to reduce the difference between the two, like your brains or a computer.

It is obvious that all three elements are required and that they are specific for a certain action, depending on its nature. Furthermore, it is important to be aware that the corrective action is a **reaction** to the deviation, which means it always needs a bit of time. This time delay is the reason why the reduction is never perfect and research in feedback control is continuously targeted to cope with this limitation.

Let's now take the loudspeaker as example to explain the importance of not making mistakes in the measurement. What we want is a faithfull reproduction of music that is previously recorded. This means that we want to measure the sound and compare it with the signal of the recording. The control computer, for instance a DSP, can then send a correcting signal to the power amplifier / driver combination such that the sound is less distorted and non-resonant.

One can safely conclude that in a feedback controlled system the measurement determines the best attainable final result. An error in the measurement will give the wrong information to the computer and it will make the amplifier/driver combination act to compensate this wrong measurement. As a consequence the sound will contain the error in the measurement. A clear example of such a measurement error is electronic noise. A noisy sensor will result in a noisy loudspeaker and this is a key factor in selecting a sensor.

How to make sound

Let's start with an observation. Any audio enthusiast will recognise that low frequencies require more diaphragm motion to produce a certain sound level than high frequencies. It is however less known what the cause of this effect is.

Well, it has everything to do with the ability of the diaphragm to compress air to a certain pressure level. Let's first define some terms with help of a simple picture. In Figure 1 the voice coil generates the motion by exerting a force on the diaphragm and the dust cover, which drive the air.

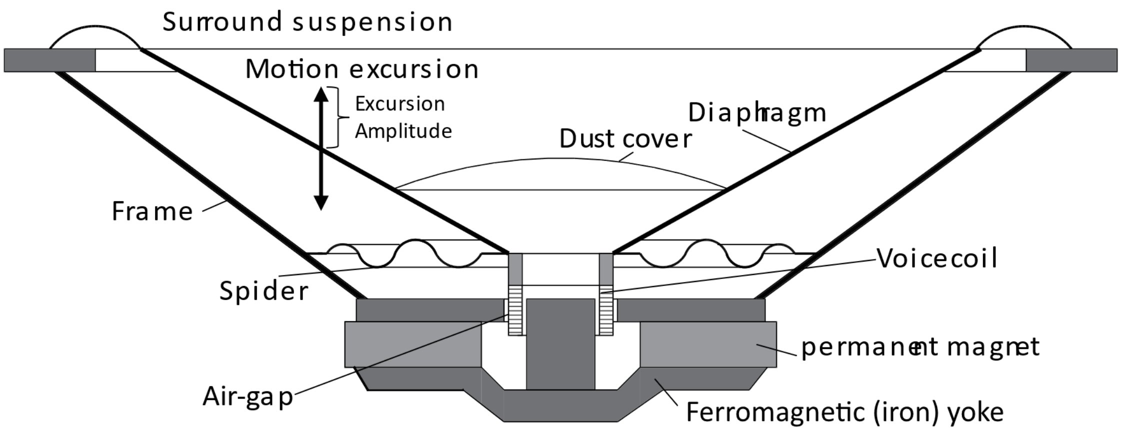

Figure 1: Cross section of a loudspeaker driver.

For the following it is important to remember that sound is perceived via an alternating pressure variation around the average pressure of the surrounding air. The motion excursion (or just shortly "excursion") of a loudspeaker diaphragm is defined as an alternating displacement with a certain maximum value, called excursion amplitude.

It is not difficult to imagine that slow movements of the diaphragm create little pressure. When moving with a certain excursion amplitude at low frequencies the loudspeaker diaphragm moves relatively slow, which means that only little air pressure is created, while the same excursion amplitude of the diaphragm at higher frequencies requires faster movement, causing more air pressure.

So far, so good. We can conclude that:

1. It is the velocity (speed) of the diaphragm that causes the sound pressure.
2. A higher frequency with the same motion amplitude implies more velocity.
3. 1+2 ➔ A higher frequency with the same motion amplitude gives more air pressure.

This all means that we need less motion or a smaller surface of the diaphragm for higher frequencies.

Let's look at it in more detail now. How do the motion amplitude, frequency, diaphragm surface and sound pressure exactly relate to each other? The radiated sound pressure is 1:1 proportional to the motion amplitude and to the surface of the diaphragm as long as the diaphragm is small in respect to the wavelength of the sound. At higher frequencies bundling occurs but in this simplified explanation we focus on the lowest frequencies up to ~500Hz (= ~60cm wavelength). This all seems quite logical.

Less logical is the relation between sound pressure and frequency. Due to one of the properties of air, called *impedance*, the coupling of the moving diaphragm to the air is frequency dependent. Combined with above relation between velocity and frequency (= multiplied), the frequency-dependent impedance of the air **causes a squared relation between sound pressure and frequency**. In other words, a factor 2 higher frequency causes a factor 2 x 2 = 4 more sound pressure!

So, if a loudspeaker diaphragm would move with equal amplitude for all frequencies one would hear a very shrill sound with no bass. This means that for natural sounding music it is necessary that the motion amplitude response of the loudspeaker diaphragm **decreases** with a factor equal to the **frequency squared** at increasing frequencies. Fortunately, nature helps us to automatically generate this frequency dependent behaviour of the motion amplitude...

As illustrated in Figure 1, the diaphragm of a loudspeaker driver is excited by a linear motor, consisting of a voice coil in a permanent magnetic field. This motor creates a force to the diaphragm as function of the current from the power amplifier. When neglecting parasitic effects of the self-inductance, which is allowed for low frequencies, the current level and correspondingly the force are proportional to the signal voltage from the power amplifier. Physics laws (Newton) dictate that an object to which a force is exerted will be accelerated proportional to this force divided by its mass. And most importantly the related mathematics dictate that the acceleration level of an alternating moving diaphragm at constant motion amplitude will increase with increasing frequency to the frequency squared!

To fully understand this, we need a small amount of mathematics and a graph.
First of all, a musical signal can be thought of as a collection of numerous sinusoidal alternating signals. Figure 2 shows an example of a sinusoidal alternating excursion, with its velocity and acceleration. It also explains the mathematical term *differentiation* in a moving system as the change of the value over time.

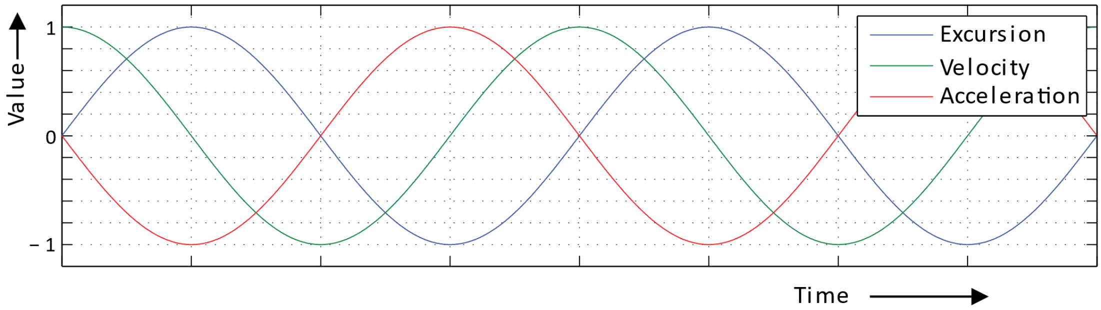

Figure 2: Sinusoidal alternating excursion with its derived (differentiated) velocity and acceleration.

With a little reasoning one can imagine that velocity creates a displacement and that acceleration is needed to reach a certain velocity. In other words, the higher the velocity the faster the excursion will change and the higher the acceleration, the faster the velocity will change. You can see this illustrated in Figure 2. At the start of the graph the position (blue) is increasing in the positive direction from zero, with a steep slope. At the same time the velocity (green) is at a maximum value, and it hardly changes at this maximum value. Therefore the acceleration (red) has a zero value at this moment in time.

With mathematics this relation is better shown, because it includes the influence of frequency. See the table below. To calculate velocity from excursion, or acceleration from velocity, one needs to apply *differentiation*. So when the diaphragm moves with a sinusoidal alternating excursion x at an excursion amplitude A, its velocity v and acceleration a are calculated by means of differentiation as function of the frequency f [Hz]. You will recognize the resulting sin and cos functions in Figure 2.

In case the acceleration is known and you need to calculate the velocity and excursion, the opposite of differentiation is needed, which is called *integration*. This is shown in the second column of the table.

Differentiation	**Integration**
$x = A\sin\omega t$	$a = A\sin\omega t$
$v = A\omega\cos\omega t$	$v = -\dfrac{A}{\omega}\cos\omega t$
$a = -A\omega^2\sin\omega t$	$x = -\dfrac{A}{\omega^2}\sin\omega t$

With the angular frequency $\omega = 2\pi f$ in [rad/s]

It goes beyond the purpose of this tech note to show how these relations are derived as the principle is well known and can be looked up elsewhere (for instance on Wikipedia). More importantly it clearly leads to the following conclusion: **when driving a moving diaphragm with a constant acceleration level at all frequencies, its excursion amplitude will <u>decrease</u> with the frequency squared!** Which is exactly what we need for a frequency independent sound radiating loudspeaker driver.

The optimal loudspeaker sensor for feedback

What does this tell us about what to measure for feedback? At first it seems most straightforward to measure the sound with a microphone. Indeed this can be done, however... a microphone will measure all sounds and not only the sound from the loudspeaker, like people talking, noises of cats and dogs, but also reflected sound via walls. Especially the latter are a problem as they come later and cause the feedback controller to correct something that is no longer there, a typical cause for instability. One may think that mounting the microphone inside the enclosure would solve this. Unfortunately however, environmental sound can enter the enclosure through the membrane and cabinet walls. Also, the sound pressure inside the enclosure is different from the sound pressure at the outside.

So we should find a measurement method that is sufficiently accurate in respect to the radiated sound, without the problems of the microphone. Many experiments have shown that for low frequencies the movement of the diaphragm fulfils this demand. It is heavy enough to not detect other sound sources, and the sensor can invisibly be mounted on the loudspeaker driver. Now the next question is which part of the diaphragm motion should be measured: its position, velocity or acceleration?

Above we have seen that a frequency independent acceleration level automatically creates a frequency independent sound level. This means that **measurement of the acceleration creates a flat response when used for feedback on a loudspeaker driver!**

Then why do some people use velocity feedback or even position feedback? Well, that depends on what you want to achieve. Following from the notion that feedback causes a correction of the measured value, the following can be concluded:

- Position feedback gives a correcting force to unwanted displacement
  ➔ it acts like a stiff spring.
- Velocity feedback gives a correcting force to unwanted velocity
  ➔ it acts like a damper.
- Acceleration feedback gives a correcting force to unwanted acceleration
  ➔ it acts like a mass.

Now we also know that a driver mounted in an enclosure shows resonances of which the lowest is determined by the stiffness (k) of the support and the moving mass (m) of the diaphragm plus voice coil. Below this lowest resonance frequency the radiated sound will decrease because the motion amplitude becomes frequency independent. The peak (Q) of the resonance is determined by a damping action with coefficient (c) in the following way:

$$f \propto \sqrt{\frac{k}{m}} \qquad Q \propto \frac{1}{c}$$

Combining these equations with the insight from the bullets above, we see that position feedback increases the stiffness k, so the resonance frequency will shift to higher frequencies (see Figure 3). This reduces the low frequency response, which is not what we would like.

Velocity feedback increases the damping c, and thus reduces the peak of the resonance. This can be useful with loudspeakers that don't have much damping on their own, but it does not further improve the performance.

Finally, acceleration feedback decreases the resonance frequency due to an effective increase of mass m. (*Note that this is not real mechanical mass, which would reduce the efficiency. It is only virtual mass that changes the dynamic behaviour*). As a result, the flat response above the resonance frequency is now extended to lower frequencies, which is the preferred behaviour.

In summary, this means that position feedback is useless for loudspeakers, while velocity feedback is only useful around the resonance frequency. Acceleration feedback, however, is useful over a wider frequency band, thereby also reducing harmonic distortion at higher frequencies.

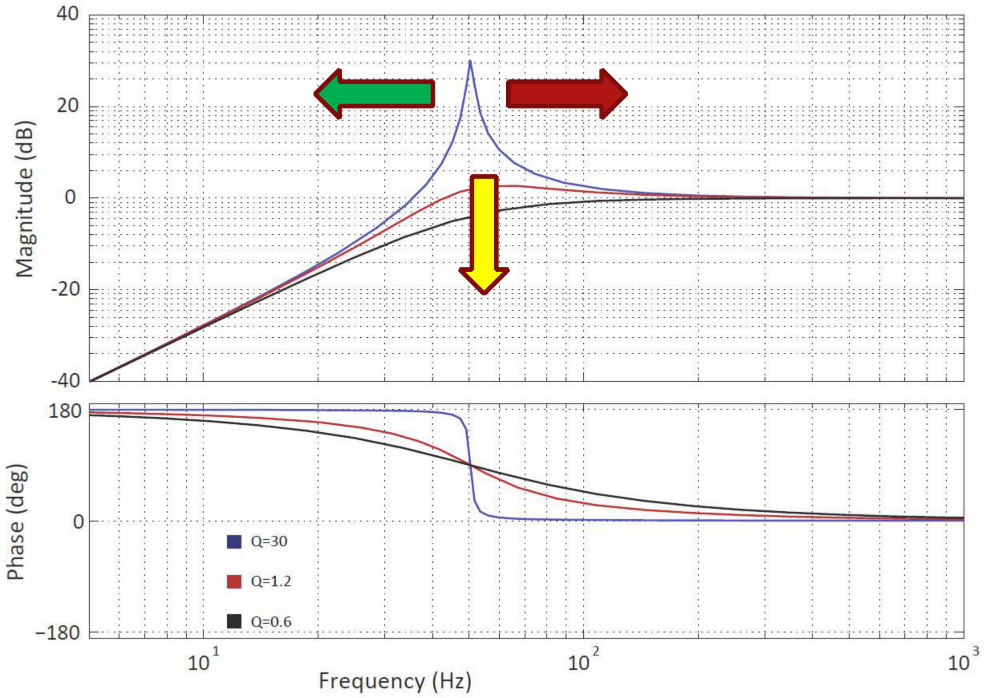

Figure 3: Effect of different kinds of feedback on the resonance behaviour of a loudspeaker. Position feedback (red arrow) increases the resonance frequency, which would limit the low frequency response. Velocity feedback (yellow arrow) gives a lower Q of the resonance because of the created damping. Acceleration feedback (green arrow) lowers the resonance frequency, which increases the flat response to low frequencies.

Suitable sensors and final remarks

It has been shown that acceleration sensing is the best way to correct errors in loudspeaker drivers by means of active feedback control. Unfortunately this is also the main reason why Motional Feedback of loudspeakers has never become mainstream. It requires the integration of an acceleration sensor with vulnerable wiring in the driver, making it more expensive and prone to damage.

As cheap alternative it is possible to derive a velocity signal from the voltage and current through the motor, which is sometimes done for velocity feedback. And in theory one can derive the acceleration from this velocity signal by means of electronic differentiation. Unfortunately electronic differentiation increases the noise at higher frequencies (remember the $A\omega$ term for the amplitude with differentiation) and as was explained in the beginning, this noise will be transferred to the driver…

A final remark for all those who think they invented the best sensor and come up with a position sensor, for instance based on a reflected laser beam or a bending element. Even when such sensor would be perfect for position sensing, to get an acceleration signal it requires twice differentiation (with $A\omega^2$) which increases the noise even more.

Unfortunately, modern cheap MEMS accelerometers with integrated AD conversion (as used in smart phones) are not yet suitable because of the limited dynamic range, noise and bandwidth. At least 16 bit (96dB SNR) performance is required with a bandwidth of over 3 kHz. For that reason analogue piezo-electric sensors are still the best choice. Which is why they are used in our DMF subwoofers.