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LIVE SOUND INTERNATIONAL

INSIDE

THE SAME DESTINATION

Regardless of a system's size and scale, the goals are remarkably similar

Tech Topic

COHERENCE & REVERBERATION

Real-world observations on separating the signal from the noise.

by Merlijn van Veen

Coherence is a common feature found on many analyzers that enables us to distinguish signal from noise. It will indicate whether you're measuring a loudspeaker or, for example, a moving light.

Coherence is subject to change. One of the aspects involved, among others, that we'll explore in-depth is the relationship between the direct sound of a loudspeaker in a room and the room's reverberation. This is an attempt at putting real-world observations into context. Many of the results have been obtained experimentally, and some concepts that don't directly further our understanding have been deliberately omitted.

Loosely defined, coherence is a function that indicates contamination of the measurement data. It's proportional to the ratio of signal to the sum of signal plus noise. Here's the relationship:

$$Coherence \propto \frac{S}{S+N}$$

In other words, coherence is an indicator for the signal-to-noise ratio (SNR) spectra in **Figure 1**, and by extension, speech intelligibility and related. In practice, it suffices to think of coherence as a data quality indicator.

High-coherence data is reliable and actionable, informing us how to move forward with tuning a sound system. However, all of this is based on the assumption there's actual signal left over...

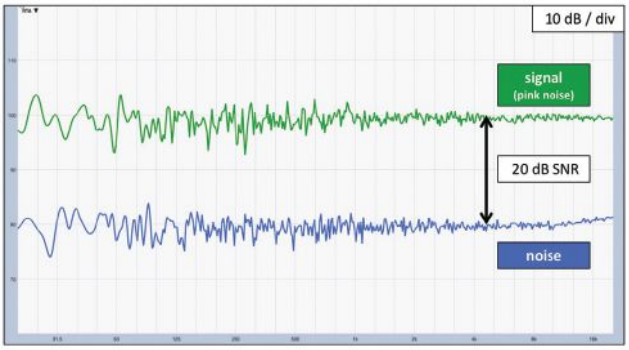

Figure 1: Signal-to-noise ratio (SNR) spectrum view.

DESTRUCTIVE INTERFERENCE

Destructive interference will destroy signal, which in turn will be replaced by whatever is left over. Typically, residual noises like HVAC, moving lights, generators, audience enthusiasm and the like. The spectrum of a comb filter is shown in **Figure 2**. Comb filters manifest themselves as an alternating pattern of peaks and nulls in the frequency response. They're caused by adding multiple copies of the same signal (produced by other speakers or reflections), arriving at different times, together.

Comb filters are inevitable whenever there's physical displacement between multiple sources reproducing the same signal and/or surroundings made of specular, reflective boundaries. These phenomena are particularly noticeable when walking the room while listening to pink noise and are typically described as phasing, flanging or chorusing. Sound familiar?

Comb filters appear to have audible pitch. The frequencies of the peaks of a comb filter constitute a harmonic series and the apparent pitch is equal to the frequency of the first peak (fundamental). Which pitch you perceive is uniquely defined by your listening position with respect to the sound system and surrounding boundaries, regardless of the program material itself, and is therefore a moving target that typically goes up and down in frequency as you walk the room, because you're dealing with a purely spatial problem.

Whenever you hear phenomena like these, realize that equalization is no longer a viable option unless the apparent pitch remains constant over space (which I yet have to encounter). Just note that every time the direct sound has been cancelled for whatever reason, signal has been replaced by noise (at the nulls or cancels) and both SNR and, inherently, coherence decrease.

It should be emphasized that all of this

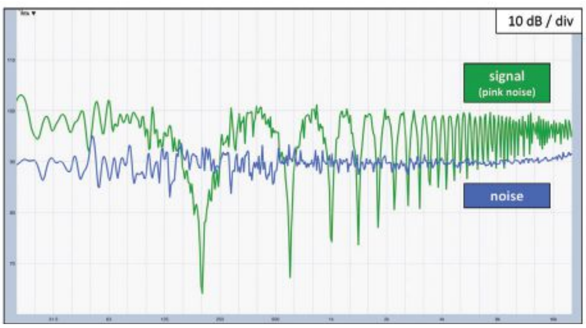

Figure 2: SNR and destructive interference.

happens typically after the sound left the loudspeaker(s), and signal processing typically has limited to no merit other than properly time aligning multiple sources, which doesn't resolve room interaction!

RIPPLE

The difference between a comb filter's maxima and minima, expressed in dB, is known as ripple **(Figure 3)**, and it's arguably the most important metric when designing sound systems.

Ripple is a function of the relative level offset between two or more copies of the same signal **(Figure 4)**. An in-depth explanation of ripple is beyond the scope of this article. (For more information, please consult the chapter entitled "Summation" in all three editions of Bob McCarthy's "Sound Systems: Design And Optimization." In the meantime, I encourage you to work with the phase calculator that I've set up on my website [merlijnvanveen.nl] to gain insight into the balancing act between relative level and time offsets.)

Figure 5 shows several transfer functions of the same comb filter with varying amounts of ripple while competing with different amounts of background noise. Notice how coherence (red trace) is greatly affected by both ripple as well as background noise.

In general, less ripple (less degraded, more robust signal) results in overall improved coherence. Interference between multiple copies of the same signal is minimized when relative level offset comes to the rescue. Simultaneously, lower background noise levels translate into more SNR, which also improves coherence.

So, how does ripple typically evolve over distance indoors?

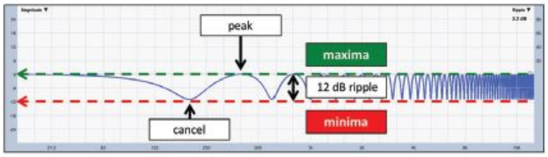

Figure 3: Ripple, the difference between maxima and minima expressed in dB.

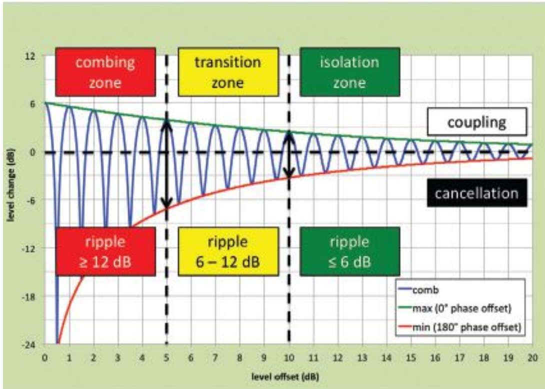

Figure 4: Ripple as a function of level offset.

CRITICAL DISTANCE

Ripple goes hand in hand with the direct-to-reverberant ratio (D/R). For frequencies with wavelengths much smaller than the dimensions of a given room, we can resort to statistics. This criterion needs to be met for all acoustics equations that follow from here on. Under such circumstances, the direct sound drops with 6 dB per doubling pf distance (inverse-square-law), whereas reverberation tends to maintain its level regardless of distance **(Figure 6)**.

In the direct field, the direct sound dominates over the reverberation with positive D/R values. In the reverberant field, it's the other way around with negative D/R values. The distance where direct and reverberant see eye to eye at the same level, with a D/R value of zero, is called critical distance. It's where the scale tips.

Ultimately, listeners experience and measure the combined SPL of both direct plus reverberation which implies that in the reverberant field, beyond critical distance, the inverse-square law is typically no longer observed or experienced.

As long as the direct sound is dominating, there will be little or no ripple (6 dB or less) and high coherence because any reflections that could possibly cause destructive interference are soft by comparison. We're effectively isolated from the room and obtain near-anechoic data. Conversely, if the reverberant sound dominates, the reflections are so strong by comparison that they wreak havoc on the direct sound causing ripple in excess of 12 dB and low coherence.

My favorite tool, for the sole purpose of explaining the underlying mechanism involved, is the Hopkins-Stryker equation. With this equation we can estimate

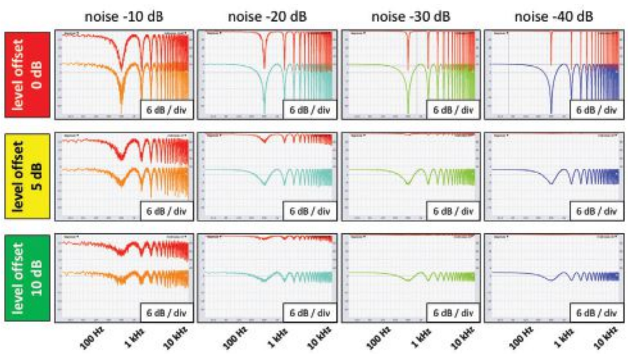

Figure 5: Coherence, ripple and background noise.

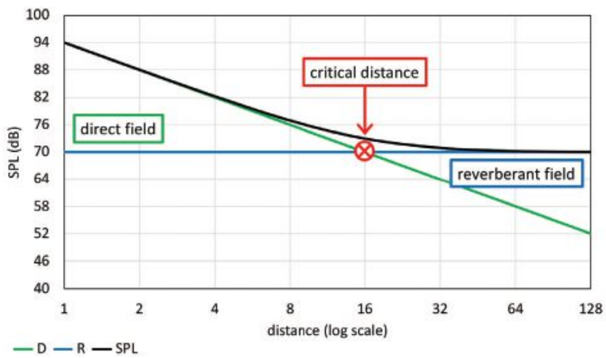

Figure 6: Critical distance.

L_p = Sound Pressure Level
L_W = Sound Power Level
Q = directivity factor
D_x = distance
S = surface area
$\bar{a}$ = absorption coefficient
K = constant

$$L_p = L_W + 10log\left(\underbrace{\frac{Q}{4\pi D_x^2}}_{\text{direct}} + \underbrace{\frac{4}{S\bar{a}}}_{\text{reverb}}\right) + K$$

Figure 7: The Hopkins-Stryker equation.

(at best) critical distance. The accuracy of this equation gets us into the "ballpark," bringing us into the right order of magnitude; however, it should be treated with scrutiny. In its simplest form, without the additional modifiers Ma and Me, here's the equation:

$$L_p = L_W + 10\log\left(\frac{Q}{4\pi D_x^2} + \frac{4}{S\bar{a}}\right) + K$$

Figure 7 provides a detailed explanation of all the variables. What makes this equation interesting is the part between parentheses. The first and second fraction determine how direct and reverberant levels evolve over distance respectively. Independent of the sound power level (SWL) or simply put, volume or loudness of the source.

Notice that only the first fraction contains a D_x^2 in the denominator. That's the $1/r^2$ dependency or inverse-square law. Reverberation relies solely on the venue's total surface area and the average absorption coefficient of that combined area.

If we set D_x to d_c, as in critical distance, and make direct and reverberation equally loud, the condition at critical distance, we obtain this equation:

$$\frac{Q}{4\pi d_c^2} = \frac{4}{S\bar{a}}$$

If we solve this equation for d_c, it leads to a new equation:

$$d_c = 0,141\sqrt{QS\bar{a}}$$

This indicates that in practice, critical distance depends primarily on Q and $\bar{a}$ because surface area is a given unless you intend to bring a wrecking ball. We'll take this further in my next installment, looking at issues such as directivity factor, the absorption coefficient and more. LSI

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