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A New Method for Transient Distortion Detection

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ABSTRACT

Transient distortion, or 'loose particle' measurement, is an important loudspeaker production line quality control metric that identifies and facilitates troubleshooting of manufacturing issues.

This paper introduces a new enhanced loose particle measurement technique that discriminates more accurately and reliably than current methods. This new method introduces 'prominence' after envelope detection, a new metric for audio measurements, that effectively isolates transient distortion in the presence of periodic distortion. This technique also offers the unique ability to listen to the isolated transient distortion waveform which makes it easier to set limits based on audibility and has widespread applications.

1 Introduction

Transient distortion, or 'loose particle' measurement, is a valuable quality control metric because it identifies non-periodic distortion, for example, rattling parts, separately from periodic distortion such as rubbing or buzzing parts. This facilitates troubleshooting of manufacturing issues.

This paper introduces a new transient distortion measurement technique that is more accurate and reliable than current methods. In addition to improved performance, this new algorithm also aids understanding of the correlation between measurement results and audibility, since it is possible to isolate and listen to just the transient distortion artifacts.

Although this analysis method was developed for measuring loose particles in loudspeaker drivers, it is also valuable for measuring rattling parts such as buttons, fasteners, and loose wires on various audio devices, and measuring impulsive distortion or Buzz, Squeak and Rattle (BSR) in automotive audio applications [1].

2 What is Transient Distortion? Why does it matter?

Transient distortion is caused by random clicking, popping, and other noises in the time domain. In a speaker or headphone driver, this might be caused by foreign particles such as glue or magnet fragments trapped in the gap behind the diaphragm or dust cap.

In a device such as a smart speaker, transient distortion might come from a loose volume control button on the device that rattles when sound is played.

In an automotive application, it could be characterized as buzz, squeak and rattle from loose wires, screws or fasteners in a car door that the loudspeaker is mounted in.

In all cases, the sound is undesirable, so devices that exhibit such faults should be identified and rejected.

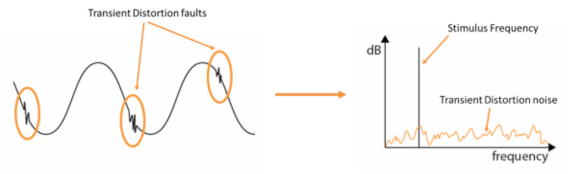

Figure 1. Loose particles visible in the waveform on the left (circled) show up as broadband noise in a spectrum.

In the recorded time waveform, transient distortion faults appear as impulsive noises added on the stimulus wave. These impulses are not related to the frequency of the stimulus, but rather to the vibration caused by the displacement amplitude of the diaphragm. The transient distortion is more frequent and significant when the speaker is driven near or below its resonant frequency, where the displacement of the diaphragm is the greatest.

Although the sound - a random clicking, buzzing or popping noise - can sometimes sound similar to higher order harmonic distortion (Rub & Buzz), such defects are not clearly reflected in the frequency spectrum of the waveform. Figure 1 shows a waveform with transient distortion, and the corresponding frequency spectrum. The vertical black line represents the stimulus frequency and the orange broadband noise spectrum indicates the transient distortion. Transient distortion is best identified at the *time* the transients occur, unlike Rub & Buzz distortion which is best identified by the *frequency* at which it occurs [3].

3 Prior Measurement Methods

The concept of using time-frequency analysis to measure transient distortion was introduced in 2003 with the Loose Particle algorithm [3].

This early model was based on time envelope analysis, a method widely used for detecting faulty bearings and gears in the machine industry. This represented the first use of this technique for audio measurements; a refined version and several adaptations for this new application were introduced in 2004 [4]. The algorithm uses a sine sweep stimulus and applies a tracking high pass filter and RMS envelope analysis to the response waveform.

Within this envelope, pulses are detected and counted. This was the first method to separate transient distortion from periodic distortion and is widely used on production lines, primarily for driver manufacturing. However, faults can be hard to discern under certain conditions, and it can be complex to configure and set limits with several variables that must be balanced to optimize results for each product and test environment.

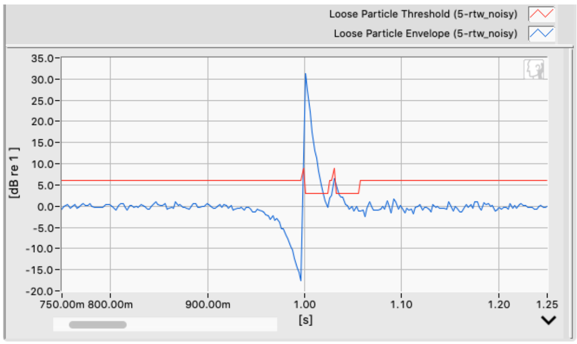

Figure 2. The impulsive background noise counted twice due to ringing.

In the example shown in Figure 2 with impulsive background noises, it can be tricky to ignore the impulsive background noise and only count the transient loose particles coming from the loudspeaker. The impulsive background noise is getting counted twice by the loose particle threshold limit (in red) due to ringing. Tweaking the parameters of the algorithm (Figure 3) can avoid this but it is very difficult to make the same parameters work for all cases.

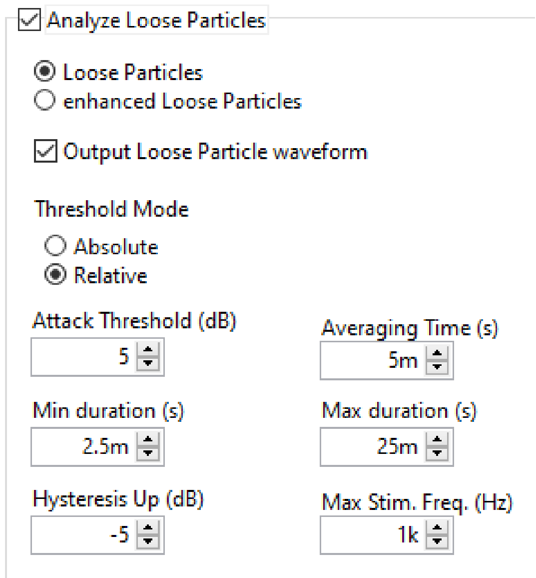

Figure 3. Loose particle threshold parameters for old algorithm.

Another approach to Loose Particle detection is Crest Factor analysis. This method uses a tracking 2nd order high pass filter at ten times the stimulus frequency and examines the peak to RMS ratio of the waveform as shown in Figure 4.

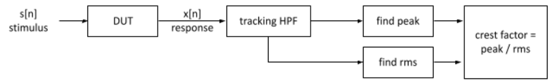

Figure 4. Crest Factor analysis flow diagram.

Sine waves have a low crest factor of 3 dB but transients typically have crest factors greater than 10 dB, making them easy to highlight. Harmonic distortion artifacts also have an elevated crest factor, and this method identifies both types of distortion together.

While the single distortion metric that this method produces can be useful for production line pass/fail testing, it does not reveal the precise failure mode.

This is important as transient and periodic distortion are caused by very different mechanisms. In a speaker driver, periodic distortion, often known as Rub and Buzz, is caused by a rubbing component, such as an improperly centered voice coil [4].

Transient distortion is caused by loose particles becoming trapped under the voice coil and rattling randomly with the excitation of the voice coil. Accurate failure mode analysis enables rapid troubleshooting of the specific problem causing the defect, reducing downtime and increasing yield.

Crest Factor analysis, like many distortion measurements, is prone to interference from background noise which typically has a crest factor greater than 10 dB. This is particularly challenging in a noisy factory where intermittent loud noises, such as a stamping machine, air compressor, or something being dropped, can trigger a false positive. It also requires significant care and effort to correctly establish limit curves for optimum performance.

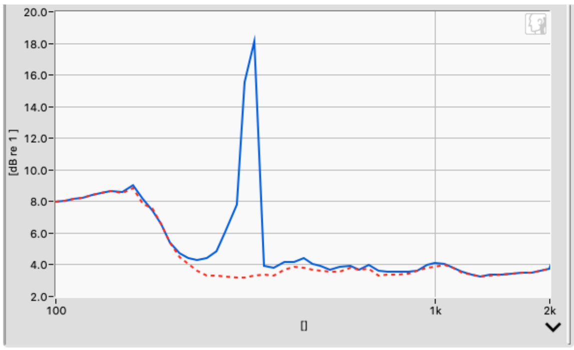

Figure 5. Loudspeaker Crest Factor (red curve) in a quiet environment and an impulsive noisy environment (blue curve).

Analyzing the same measurement as in the previous graph with the impulsive background noise (Figure 2), using Crest Factor, shows the difference between a quiet recording and one with the impulsive background noise in Figure 5. Unfortunately, there is no way to discern if the increase in Crest Factor is due to loose particles or background noise.

	Crest Factor Analysis	**Existing Loose Particles Algorithm**
How it works	Uses a tracking high pass filter and examines the peak to RMS ratio or crest factor of the waveform. Sine waves have a low crest factor and transient distortion has a high crest factor	Uses a sine wave stimulus and applies a tracking high pass filter and envelope analysis to the response. Within the envelope, transient events are detected and counted.
Pros	Single Distortion Metric - Harmonic distortion also has a high crest factor so both types of distortion are identified together	Separates transient distortion from periodic distortion; this helps identify the cause of the fault, accelerating troubleshooting
Cons	• No insight into failure mode as harmonic and transient distortion are lumped together •Susceptible to background noise as noise also has a high crest factor • Requires significant care and effort to correctly establish limit curves for optimum performance	• Complex to configure and set limits • Can be hard to discern faults under certain conditions

Table 1. Summary Comparison between Crest Factor and old Loose Particle Algorithms.

4 The New Algorithm, Comparison and Results

A new algorithm, enhanced Loose Particle (eLP), was developed to improve upon the accuracy of existing methods, while making it easier to use and set limits. It uses a tracking notch filter to suppress the stimulus and allow artifacts to be clearly detected. Instead of filtering and counting pulses, additional analysis is applied to measure the prominence, or relative impulsiveness, of the detected artifacts.

Prominence is an established mathematical concept used in other branches of science, most commonly topography. In a topographic context, it describes the elevation of a summit relative to its surrounding terrain. This differs from its absolute elevation, which measures the height of the summit above sea level.

Our research determined that the audibility of impulsiveness of artifacts depends on how that event compares to adjacent audio events – thus, the audibility of artifacts is viewed through the lens of their adjacent events as opposed to just their absolute level. This fits well with the prominence paradigm.

The new algorithm, therefore, calculates the prominence of each loose particle event to indicate the importance of an audio event relative to adjacent audio events. This is, we believe, the first application of prominence to audio measurements.

Figure 6 and Figure 7 show the same underlying example curve with three peaks (A, B, and C). Figure 6 shows the absolute elevation of each of the peaks from some fixed level. Figure 7, on the other hand, shows the prominence of each peak – this is obtained by comparing each peak relative to its highest adjacent valley.

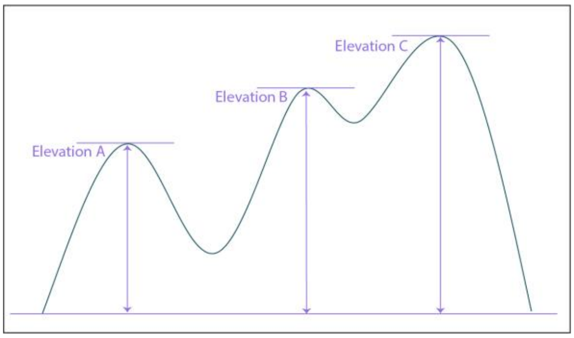

Figure 6. Absolute Elevation of several peaks.

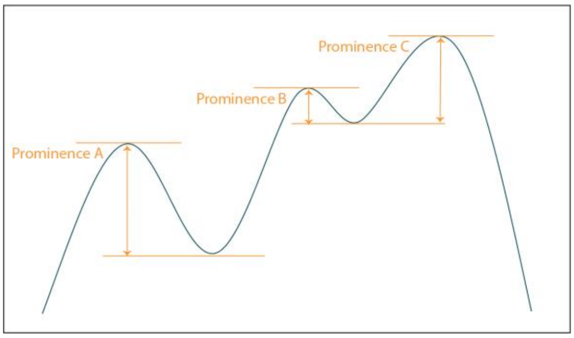

Figure 7. Prominence is elevation relative to its surroundings.

In this example, even though Peak C has the highest elevation, it does not have the largest prominence value. Prominence A, associated with peak A, has the largest prominence value – even though it has the lowest elevation of the peaks shown. Consequently, Prominence A is considered more significant from an audibility perspective.

In both the methods shown in the previous section (Loose Particle algorithm and Crest Factor algorithm), the loudspeaker's fundamental or linear response shape impacts the level of the individual transients when sine sweeping through the response of the loudspeaker, thus making it difficult to use a fixed flat threshold for limits when counting the number of transients.

A simple horizontal straight-line user-defined prominence threshold determines at what level the peak will be counted (Figure 8), and a user-defined count of events over the time window determines the pass/fail threshold.

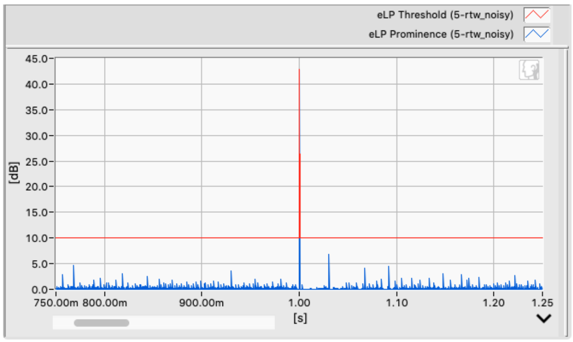

Figure 8. Prominence clearly identifies transients.

Analyzing the same measurement again with the impulsive background noise (Figure 2), using prominence, makes it much easier to discern the impulsive background noise from the other transients with the greater dynamic range.

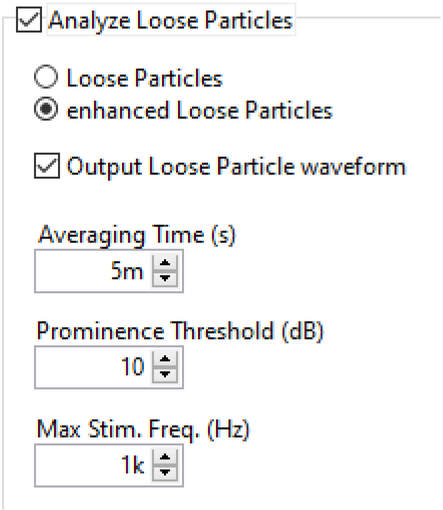

Figure 9. New and simpler loose particle parameters (compared to Figure 3).

A factory background noise event typically only occurs once or twice during a measurement, whereas many loose particle transients will occur during the same timeframe. The event count is also user-determined and is set according to the background noise in the measurement environment. These are the only parameters (Figure 9) that the user needs to define, so limit setting is simple, well correlated to audibility, and can be configured to give reliable results even when background noise is present. Figure 2

It is also easy to relate objective results obtained by this method to subjective analysis as the user can listen to the recorded time waveform with the fundamental removed, hearing only the loose particles or transients (Figure 10). This enables the prominence threshold to more easily be set based on defect audibility.

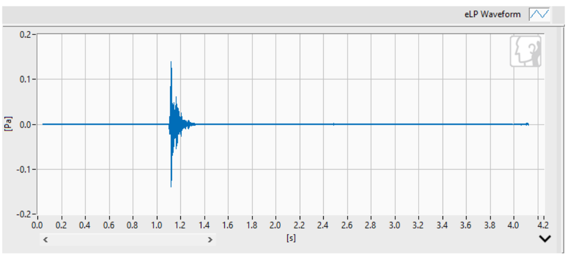

Figure 10. Loose Particle waveform with Fundamental removed.

This algorithm offers three significant advantages over other methods. First, it differentiates random transient distortion from periodic harmonic distortion for rapid and efficient troubleshooting of production line defects. Second, since it relies on a cumulative event count rather than a single event triggering a fail, false positives due to background noise are minimized. Finally, since the measured metric is the number of loose particles greater than a certain threshold in a given timeframe, limit setting is simple as it is not frequency dependent.

Let's take a more in-depth look at the algorithm, outlined in Figure 11.

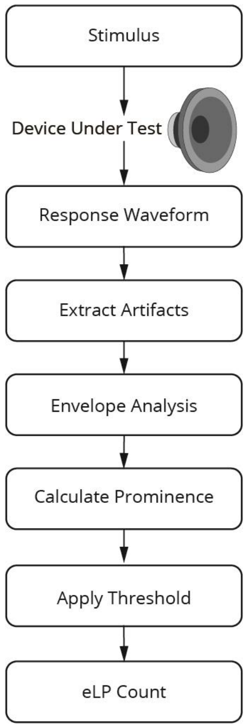

Figure 11. Schematic showing new enhanced Loose Particle (eLP) algorithm.

The measurement begins with a stepped sine stimulus, $s[n]$, which has a smooth transition from one frequency step to the next. Figure 12 shows the signal processing associated with extracting artifacts and the envelope analysis at each frequency step.

Figure 12. New algorithm flow diagram.

After playing the stimulus $s[n]$ through the device under test (DUT), the response waveform $x[n]$ is acquired and processed using a notch filter centered the at the stimulus frequency f_c Hz:

$$y_1[n] = x[n] - 2cos(\omega_c)x[n-1] + x[n-2] + 2acos(\omega_c)y_1[n-1] - a^2 y_1[n-2] \quad (1)$$

where,

a controls the filter bandwidth,
$\omega_c = 2\pi f_c/F_s$,
and F_s= sampling rate in Hz

The output of the notch filter, $y_1[n]$, called the Loose Particle waveform suppresses the stimulus while highlighting the artifacts in the response waveform. Figure 13 shows the raw recorded acoustic waveform and Figure 14 shows the extracted artifacts in the Loose Particle waveform. This Loose Particle waveform can be played through a reference loudspeaker or headphone to listen to the recorded artifacts and enable correlation to the measured results. With the stimulus removed, the transient artifacts and other distortions e.g., rub & buzz, are clearly audible – this makes it easier to set limits.

Next, the energy envelope, $y_2[n]$, of the Loose Particle waveform, $y_1[n]$, is computed by using a first-order low pass filter (LPF) with a time constant of τ:

$$y_2[n] = \left(1 - e^{-1/\tau F_s}\right)(y_1[n])^2 + e^{-1/\tau F_s}(y_2[n-1]) \quad (2)$$

Averaging time (2τ) is a user-defined parameter – smaller values cause the energy envelope to be more reactive to changes in the underlying waveform, while larger values cause the energy envelope to be relatively smoother. Varying this parameter allows the user to adjust the sensitivity of the measurement to impulsiveness.

Figure 15 shows a time domain measurement of the energy in the recorded waveform, plotted against time. Transient defects are revealed as a random burst of energy. The Prominence of each peak is calculated, and a Prominence Threshold, the level above which an event will be counted, established. This is shown in Figure 16; the horizontal red line indicates the Prominence Threshold, and events above the threshold are highlighted in red. In our evaluation, we identified a threshold of around 10 dB as a good starting point, although this may vary depending on various factors such as exactly what is being measured (for example, loose particles in a driver versus a rattling button) and the acceptable tolerance for the product. Finally, the number of events over the measurement duration is counted. This results in a numerical output upon which a pass/fail limit can be set.

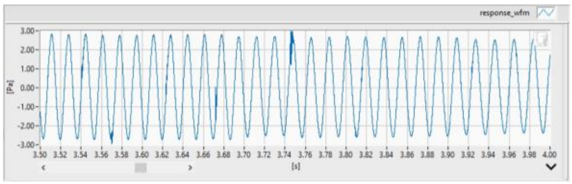

Figure 13. Raw recorded acoustic waveform with loose particle

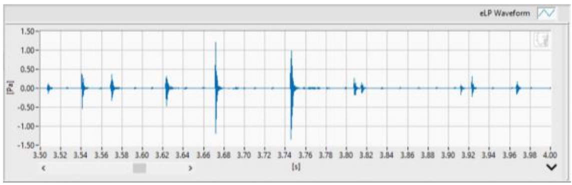

Figure 14. Extracted artifacts in the Loose Particle waveform .

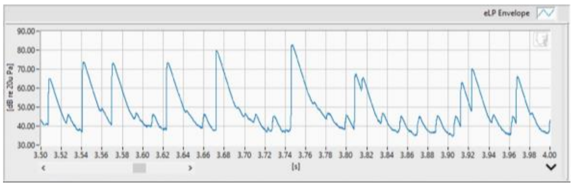

Figure 15. Time Envelope of loose particle waveform.

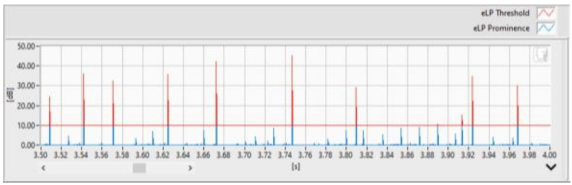

Figure 16. Prominence with Threshold for loose particle waveform.

5 Results

Let's compare the results obtained with this algorithm for a good speaker and one with known 'loose particle' transient distortion defects.

Figure 17 shows both the response waveform and the enhanced Loose Particle metric (transient distortion) for a good speaker with no known defects. Figure 18 shows the same data for a speaker with a significant transient defect. The defect is clearly visible on the response waveform, and the correlation between the magnitude of the distortion and the magnitude of the eLP prominence (red spikes) is clear.

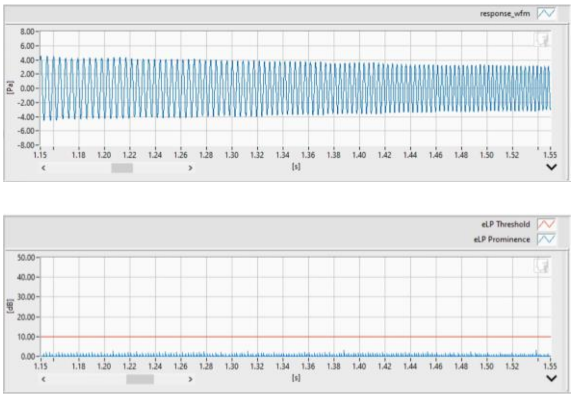

Figure 17. Response waveform (top) and corresponding transient distortion (bottom) for a good loudspeaker.

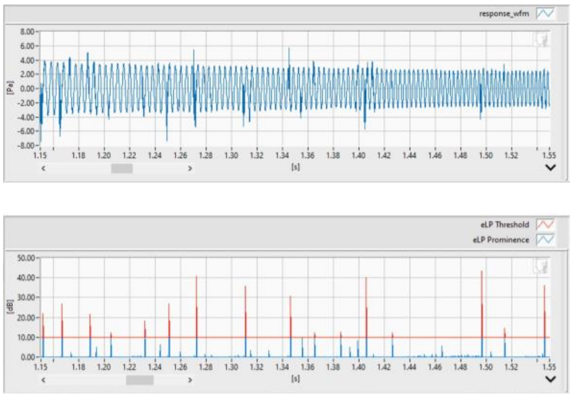

Figure 18. Response waveform (top) and corresponding transient distortion (bottom) for a loudspeaker with transient distortion.

6 Conclusions

This new 'enhanced Loose Particle' algorithm, significantly improves the accuracy and reliability of transient distortion detection. It is more accurate than earlier algorithms, and minimizes the risk of false positives due to background noise that is inherent with other measurement techniques.

Furthermore, it distinguishes loose particle defects from periodic distortion such as Rub & Buzz, facilitating troubleshooting both on the production line and in R&D applications.

The ability to play back just the loose particle recording and compare it to the visual representation facilitates understanding of the correlation between measurement results and audibility, and aids limit setting. Limit setting is simple as the user simply has to define the Prominence Threshold and the event count, and default values work well in the majority of cases.

In addition to speaker and headphone driver testing, this method has applications including Buzz, Squeak and Rattle (BSR) measurements in cars, and rattling components in audio devices.

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