LOW DISTORTION WIDE BANDWIDTH MIDRANGE LOUDSPEAKERS

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1 INTRODUCTION

Poor vocal intelligibility is a common cause for complaint in live music, and a critical safety concern in public address systems. The dominant format for sound reinforcement loudspeakers consists of direct radiating cone drivers, crossed over to compression drivers between 1-2kHz. While this provides a benefit in size reduction, we present measurements showing this format results in significant distortion in the critical range for speech intelligibility and music reproduction (1-5kHz) as well as limiting useful output at high levels due to the large difference in transducer efficiency (typically 10-15dB). Many modern systems attempt to compensate for this with extensive equalization, we discuss technical reasons why this may not be a satisfactory solution in real applications.

Finally, we present measurements of an alternative approach; a wide bandwidth, low distortion midrange device, requiring no corrective equalization. Total distortion levels are typically 20dB lower in the same bandwidth. Directivity measurements are also presented and discussed, as well as some psychoacoustic reasons for why this approach may be preferable.

2 BACKGROUND

2.1 PROPERTIES OF ACOUSTIC RADIATORS

The frequency range of audible sound is astonishingly wide, stretching over ten octaves from approximately 20Hz up to 20kHz (for young people). This bandwidth presents numerous challenges to sound reproduction, mostly as a consequence of the correspondingly huge range of wavelengths (17m to 1.7cm). With reference to this, one fundamental problem in loudspeaker design is that the size of an acoustic radiator determines its directivity. A useful model for understanding the radiation from a loudspeaker diaphragm is the radiation from a plane piston in an infinite baffle:

$$\tilde{p}(\mathrm{r},\theta)=-\mathrm{jka}^{2}\rho_{0}c\tilde{u}\frac{e^{\mathrm{jkr}}}{2r}\left[\frac{2J_{1}(\mathrm{ka}\sin\theta)}{\mathrm{ka}\sin\theta}\right]$$

Equation 1: Radiation from a baffled circular piston[1]

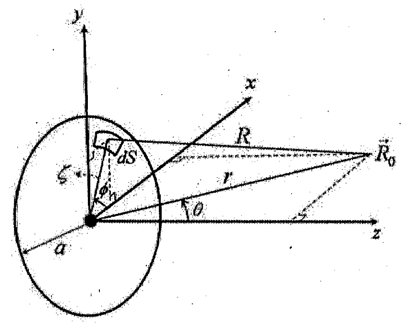

Figure 1: Geometry of equation 1

Equation 1 is an expression for the radiation from a baffled circular piston. The variation of pressure with angle (θ) at a distance *r* is determined by the directivity function in square brackets, where *J1* is the Bessel function of the first order, *k* is wavenumber (2πf/c), and a is the radius of the piston.

This provides a simplified model for the directional behaviour of loudspeakers, although ignores the effects of cone geometry and breakup.

The directional behaviour of a plane piston can be simplified with reference to *ka* in three general regions;

ka << 1 (low frequencies)	ka ≈ 1 (mid frequencies)	ka >> 1 (high frequencies)
Wavelength is large relative to the radiator. Dispersion is omnidirectional, approximating the behaviour of a point source.	Wavelength is similar to the dimensions of the radiator. Dispersion becomes unidirectional.	Wavelength is small relative to the radiator. Dispersion becomes increasingly narrow, additional side lobes begin to appear above ka ≈ 5

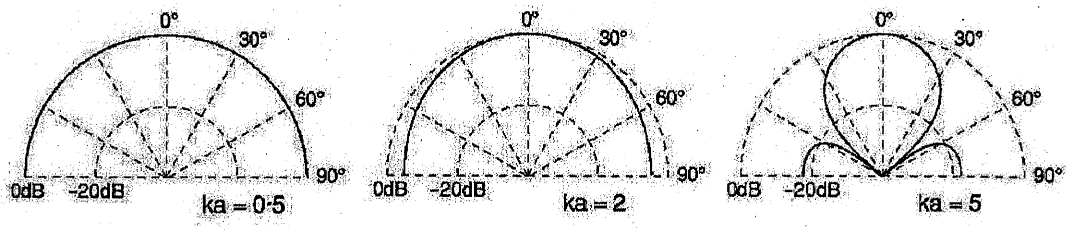

Figure 2: Plots of equation 1 at ka = 0.5, 2 and 5

Another important parameter in Equation 1 is u – the velocity. For a purely theoretical massless piston, this is constant, but for a real piston with mass and a constant driving force, velocity decreases with frequency according to Newtonian mechanics (or in the impedance analogy, the equivalence of mass to inductance). In an electrodynamic loudspeaker this is compensated for by the rising acoustic impedance with frequency to produce a region of flat response, often referred to as the piston band or mass controlled region.

The limiting factor for low frequency output is decreasing radiation impedance. Below ka=1 radiation impedance drops rapidly, eventually becoming purely reactive and no power is transferred to the air. Understanding these behaviours, and the contradictory physical requirements of low and high frequency transducers, clearly it is impossible to construct a single acoustic transducer that is capable of reproducing the entire audible range (at least not at the sound pressure levels required for public address). Engineers realised this after the development of the first loudspeakers, and as soon as the 1930s the 2-way loudspeaker had appeared. This combined the work of Jensen & Pridham (horn loaded compression-type drivers) for high frequencies and Rice & Kellogg (direct radiating cone drivers) for low frequencies, with a passive crossover filter network at around 300-500Hz. The first commercially successful two way loudspeaker was the Altec Shearer Horn in 1935, which won an Academy award for technical achievement in 1936.

The combination of a small radiator for high frequencies and a large radiator for low frequencies also enables more consistent directivity across the audio spectrum. Constant directivity with frequency is a critical requirement for public address systems in order to achieve consistent sound pressure level (SPL) and frequency response throughout the intended coverage area.

The improvements in bandwidth and directivity have resulted in multi way loudspeakers becoming the dominant format in all areas requiring high quality sound reproduction.

2.2 DISTORTION

However, the primary motivating factor for the development of the first 2-way systems was probably not improved directivity, but to achieve a wider frequency range, and subsequently to reduce the severe distortion produced by early compression drivers. Several studies, including Voishvillo (2007) have found that due to the psychoacoustic effect of *masking*, distortion at low frequencies is less objectionable to listeners than midrange or high frequency harmonic distortion. This shifts the focus to compression drivers, which usually operate from 1000Hz and above.

There are many sources of distortion in electrodynamic loudspeakers, and almost all of them are functions of diaphragm displacement (Table 1).

It appears that in order to design truly high fidelity loudspeakers, diaphragm displacement must be made as small as possible, while still retaining the required sound pressure levels.

Non-linearity	Function of:	Frequency range	Mechanism
Compliance (C_{MS})	Displacement (x)	Below Fs	Non-linear restoring force
Force Factor (Bl)	Displacement (x) Current (i) Velocity (u)	Low frequencies (highest displacement)	Non-linear f=Bli Non-linear damping e=Bli
Coil Inductance (L_E)	Displacement (x) Current (i)	HF modulation by LF High frequencies	Variation of inductance with coil position Non-linear permeability of steel
Coil Resistance (R_E)	Current (i)	All frequencies	Resistance increases with temperature
Young's Modulus	Strain (ε)	Modal frequencies of cone	Stress as a non-linear function of strain
Modulation (IMD)	Displacement (x)	HF modulation by LF	Variable time shift in propagated sound producing modulation distortion

Table 1: Overview of non-linearities in electrodynamic loudspeakers (adapted from Beranek, Acoustics (2012))

The inspiration for this paper was Klipsch's papers "Modulation in distortion in loudspeakers" Parts 1-3 (1969. 1970 and 1974). Klipsch's measurements provide an early glimpse at the potential sound quality benefits of horn loudspeakers, and the importance of modulation distortion in determining subjective sound quality.

Boer et al. (1998) found that intermodulation distortion was the most annoying, and harmonic distortion the least annoying. It would make sense that modulation distortion performance is critical, as a multi-tone excitation is more similar to a real music signal than a single frequency excitation (as in the case of a harmonic distortion measurement). Modulation products can be relatively high order compared to harmonic distortion products in loudspeakers.

Various studies have found that even order (asymmetric) harmonic distortion is much less objectionable to listeners than odd order (symmetric) harmonic distortion. Fortunately, most harmonic distortion in loudspeakers is dominated by the second harmonic.

Tan, Moore and Zacharov (2003) investigated subjective ratings of hard and soft clipping on speech and music signals, finding strong correlations. Relevant to this work, they also recorded the outputs of real transducers, which were rated by participants, and found a moderately strong negative correlation between listener preference and the distortion measure (DS) devised for the study.

In contrast, Geddes (2003) found that non-linear harmonic distortion was *not* a significant predictor of listener preference.

2.3 COMPRESSION DRIVERS

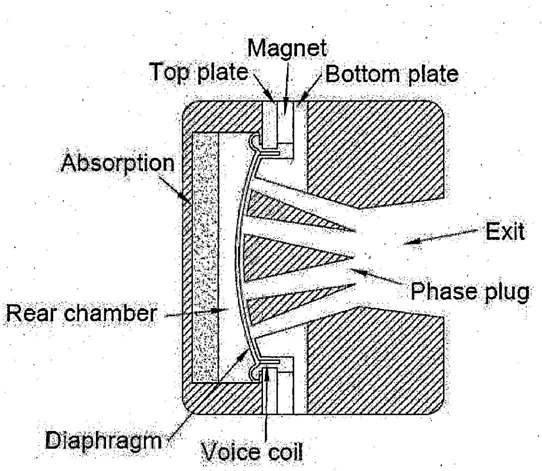

Figure 3: Compression driver schematic diagram

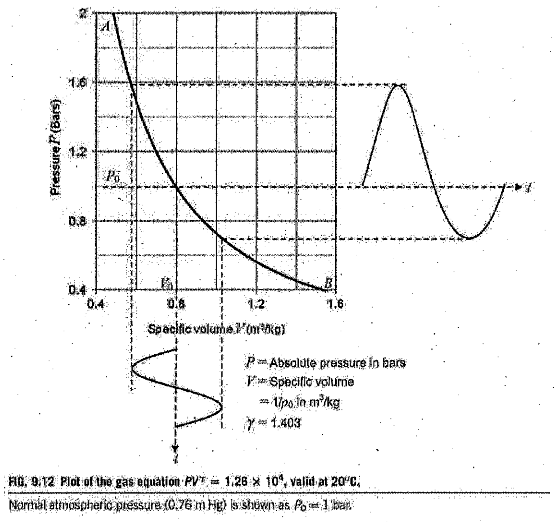

Figure 4: Plot of the gas equation at 20C (From Beranek, Acoustics: Sound Fields and Transducers)

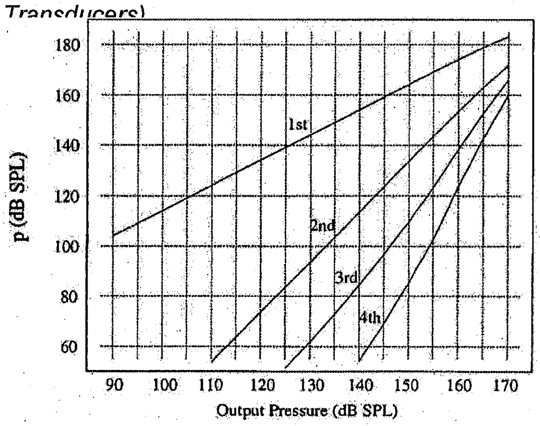

Figure 5: Predicted harmonic level vs. SPL at horn mouth at 1kHz (Holland and Morfey 1996)

Compression drivers are a sub-type of moving coil / electrodynamic loudspeaker, consisting of a small (2-4" diameter) dome shaped diaphragm constructed from a lightweight, stiff material such as Aluminium or Titanium, radiating through radial or concentric slots, providing compression loading. The outlet (exit) of the driver is connected to a horn flare. The very high air load on the diaphragm from compression loading, and impedance matching of the horn results in extremely high efficiencies, on the order of 103-113dB at 1W/1m.

However, compression drivers are far from ideal transducers. Due to the small size of the diaphragm and reliance on compression and a horn to load it, the acoustic impedance rapidly drops with frequency, and the diaphragm becomes "unloaded", resulting in a large increase in diaphragm displacement, and consequently distortion.

Voishvillo claims that the strongest source of intermodulation distortion is the modulation of the compression chamber volume with displacement.

In addition to displacement-related forms of distortion, another source of distortion is the non-linearity of air at high sound pressure levels. At extreme sound pressures, the change in air density begins to affect the propagation speed of the wave, distorting it and producing harmonics (Figure 4). Sound pressure levels in the compression chamber can exceed 160dB which is reaching the non-linear region for audible frequencies. Holland and Morfey (1996) proposed a model for nonlinear sound propagation in horns (Figure 5).

Funktion One founders Tony Andrews and John Newsham recognised the problem of distortion in compression drivers in the early 1970s, and began developing methods of extending the midrange bandwidth upwards. This uniquely solves the problem of distortion in compression drivers simply by restricting their output to above 4-6kHz, drastically reducing displacement.

3 MEASUREMENTS

3.1 HARMONIC DISTORTION

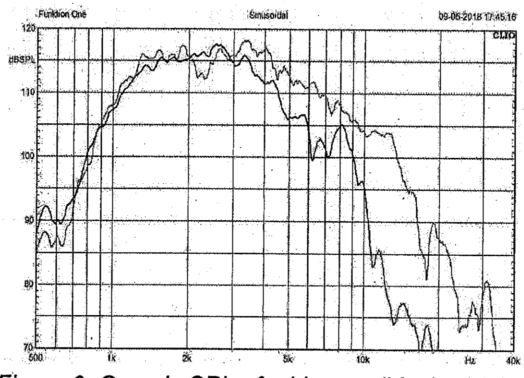

Figure 6: On axis SPL of midrange (blue) and CD (orange) 1200Hz 4th order Linkwitz-Riley HPF

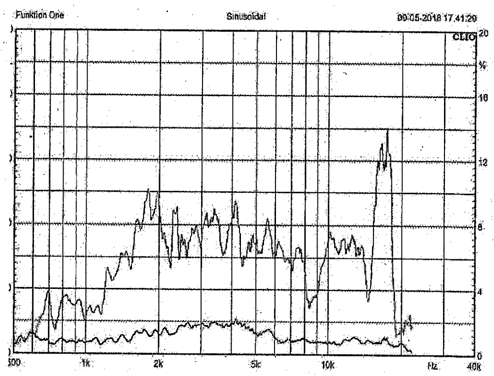

Figure 7: Comparison of % THD in midrange (blue) and compression driver (orange)

Distortion measurements were performed using the Clio 12 system from Audiomatica. The microphone was a calibrated B&K 4006, powered by a custom preamplifier. The measurements were performed in a low reflection room.

Figure 7 shows total harmonic distortion (THD) at 10% power (normalised to fundamental). The two devices have the same RMS power rating and sensitivity, and were measured with the same $20V_{RMS}$ stimulus, high pass filtered at 1200Hz, providing a valid comparison in the mid frequency range. An important note is that the midrange can be used from *200Hz*, but rolls off from 4kHz, making comparison above this frequency invalid. Maximum THD is 2% (-34dB) in the midrange and 10% (-20dB) in the compression driver. At 2kHz, the midrange THD is 1.4% (-37dB), while the compression driver is at 10% (-20dB).

3.2 MODULATION DISTORTION

Modulation distortion was measured with a test stimulus of two tones at 1kHz and 4kHz at 106dB at 2m. Note the difference in level of modulation products, as well as the presence of higher order products in Figure 9. The tones at 3 and 5kHz are are 20dB and 15dB lower in the midrange device. Further contributing to the difference in performance, the modulation products in the midrange above 4kHz would be out of band in practical use, in contrast to the compression driver.

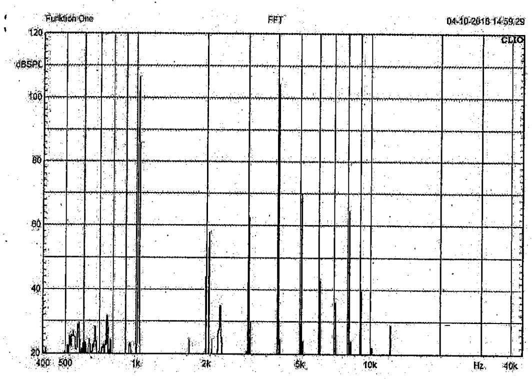

Figure 8: Modulation distortion of 1k and 4k tones in midrange

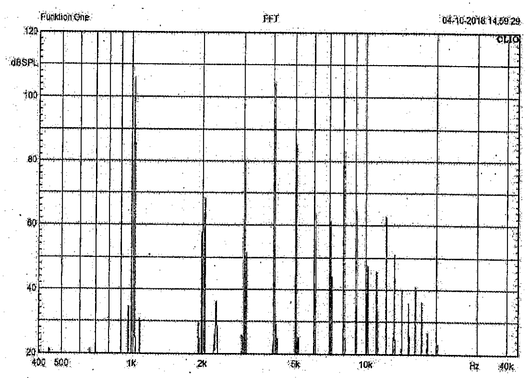

Figure 9: Modulation distortion of 1k and 4k tones in compression driver

3.4 DIRECTIVITY

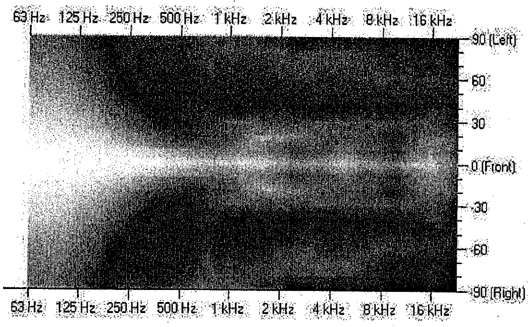

Figure 10: Horizontal directivity of 2x Evo 6E cluster (4kHz crossover)

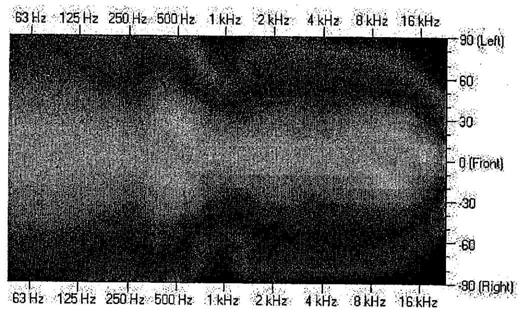

Figure 11: Horizontal directivity of a 3-way line array element (~1.5kHz xover)

The principal disadvantage to raising the crossover point is the increasing disparity in directivity between the two sources. This can be addressed with waveguide and crossover design. Figure 11 shows a direct radiating line array type loudspeaker with a midrange crossed over at approximately 1.5kHz. The narrowing of the midrange devices is apparent, centred around 1kHz. The beamwidth returns to a nominal 130° above 2kHz. In contrast, Figure 10 shows the horizontal directivity of a 2-way horn loaded loudspeaker with a 4kHz crossover point. The controlled directivity and summation of the waveguides results in constant 100° directivity throughout most of the operating range.

3.5 MAXIMUM SPL AND EQ

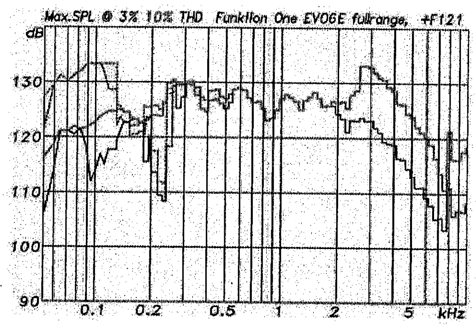

Figure 11: Max SPL of Funktion One Evo 6E and F121

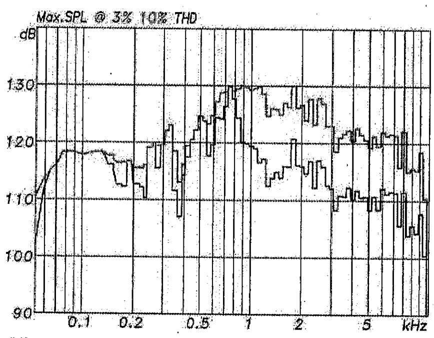

Figure 12: Max SPL of equalised 2-way loudspeaker design

Figure 11 shows the max SPL of an Evo 6E at 3% and 10% THD. This design uses a 10" midrange from 200Hz to 4kHz, and a 1.4" compression driver above 4kHz. No equalisation is required. The frequency response is balanced throughout the midrange, and similar to the small signal response. The 5% and 10% curves overlap due to the system limiters. Figure 12 shows the maximum SPL of a 2-way loudspeaker design with a direct radiating LF driver and 1.4" compression driver, using extensive equalisation to produce a flat response. At 3% and 10% THD, the frequency response is far from flat and there is a clear >10dB difference in maximum SPL between the compression driver and LF driver. Arrays of multiple loudspeakers will balance the LF output, but decreasing summation and increasing air absorption will further reduce the high frequency output above 10kHz.

3.6 INTELLIGIBILITY IN A REVERBERANT SPACE WITH BACKGROUND NOISE

Because intelligibility is strongly related to the ratio of direct to reflected sound, loudspeaker directivity is usually a more critical parameter than distortion in determining STI. For this reason a midrange device with a 90° waveguide was compared to the 90° compression driver from the previous measurements. The 10" device is included for comparison as this has been recently used in stadium PA systems due to its ability to meet very high STI and SPL specifications. The loudspeakers were installed in a reverberant space, with an additional loudspeaker generating a pink spectrum background noise at 68dBA. The RT_{30} was ~1.2 seconds. All loudspeakers were tested in the same position measuring 96dBA at 1m and ~80dBA at the receiver. STI was calculated in Clio, including noise and masking effects. The results are in Table 2:

Loudspeaker	250	500	1k	2k	4k	8k	STI (Female)	STI (Male)	STIPA (IR)
Compression driver on 90° waveguide	N/A	0.432 (I)	0.652 (C)	0.655 (C)	0.711 (B)	0.751 (A)	0.60 (D)	0.62 (D)	0.60 (D)
5" midrange on 90° waveguide	N/A	0.443 (H)	0.681 (B)	0.701 (B)	0.755 (A)	0.826 (A+)	0.66 (C)	0.63 (D)	0.63 (D)
10" midrange on 55° waveguide	0.456 (H)	0.536 (F)	0.639 (D)	0.690 (B)	0.792 (A+)	0.753 (A+)	0.68 (B)	0.65 (C)	0.65 (C)

Table 2: Speech transmission index results

4 DISCUSSION

A possible criticism of this work is that the results are obvious – as discussed in the introduction, basic electroacoustics show that a larger radiator will have a smaller displacement and reduced distortion compared to a smaller radiator. But the industry has embraced the use of compression driver midrange so universally that a closer investigation is warranted. Additionally, obvious results are no less scientifically valid than surprising results!

One of the most interesting aspects of these results is the magnitude of reduction (~20dB) in harmonic and modulation distortion in the 2-5kHz range. This frequency range is the most sensitive in human hearing, and it is reasonable to suggest that reduction of distortion in this range may even be perceptually more significant than an equivalent reduction in distortion in the low or high frequency ranges.

The assumed caveat of reducing directional consistency is shown to be avoidable with careful acoustic design. Reproducing the speech band with a single transducer has clear benefits when examining the shape of the maximum SPL curves.

The Speech Transmission Index results are compelling, but the improvement in STI in the midrange devices is most likely due to their increased directivity, rather than reduced distortion. Nevertheless, this demonstrates another advantage of the approach. Analysis of Clarity (C_{50}) (measured without background noise) shows a similar incremental improvement, with the 90° compression driver at 9.11 followed by the 90° 5" waveguide at 9.23, and finally the narrow directivity 10" waveguide reaching 9.87.

5 CONCLUSIONS

We have demonstrated that the single greatest advantage to adding a dedicated midrange driver to a system is the vast reduction in distortion; extending the frequency response of a radiator upwards in frequency is preferable to extending the response downwards, reducing displacement and distortion rather than increasing it. Maintaining consistent directivity is possible with advanced waveguide design. We have also demonstrated that equalisation is not required to design high performance sound reinforcement loudspeakers, and has significant disadvantages when the system is required to produce very high acoustic outputs. Additionally, resonances cannot be fixed with equalisation – time cannot be fixed with amplitude.

Reproducing as much of the voice band (100Hz – 5kHz) as possible with a single transducer makes sense from a perceptual perspective – real human speech is heard every day, and human hearing is highly adapted for maximum sensitivity to speech both physiologically in the ear, and neurologically (see Cocktail Party effect etc.). Purely considering the acoustic characteristics, it is a reasonable hypothesis that attempting to reproduce this range from two or more sources which are disparate in both size (directivity) and sensitivity, and joining them with electronic filters is unlikely to ever sound as "natural" or "convincing" as a real human voice; rigorous perceptual testing of this hypothesis would be a useful direction for further work.

Although the current industry paradigm is to focus solely on consistency of SPL across area, we hope that continual improvements in understanding of sound quality will filter through to the wider industry, and bring awareness that consistency of sound level is just the very first step in providing truly immersive listener experiences.

6 FURTHER WORK

Perceptual testing of this approach compared to standard methods would certainly produce interesting results. Generally, it would be useful to investigate listener preferences and perception thresholds for linear and non-linear distortion in loudspeakers. A significant difficulty in this is the necessity of exposing participants to high enough sound pressure levels, for the results to be realistically applicable to live sound.

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