APPLICATIONS OF DYNAMIC DIFFUSE SIGNAL PROCESSING IN SOUND REINFORCEMENT AND REPRODUCTION

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1   INTRODUCTION

In many sound reinforcement and sound reproduction settings, multiple loudspeakers are used to deliver adequate sound pressure levels (SPL) across audience/listening areas. Depending on the application, these acoustic signals may be coherent and derived from a common source. This leads to position-dependent constructive and deconstructive interference (comb filtering), where the received frequency response is a function of a given location's position relative to the loudspeaker array. This presents problems when attempting to deliver a consistent audience experience and one that is true to the source material. Further to this, surface reflections may combine constructively or destructively, leading to an uneven spatial distribution of sound energy. The variance in frequency response across a measurement grid has been quantified in prior work by the metric of spatial variance (SV), measured in decibels[2], which is described in more detail in section 3.1.1.

Perceptually transparent decorrelation of discrete coherent sources is a potential solution to large SV resulting from coherent source interference. This is because decorrelated sources sum by the powers of the contributing waveforms, and the summation is not dependent on relative phase[3]. This work investigates the use of a time-varying decorrelation algorithm (dynamic diffuse signal processing) for the reduction of SV and other comb filtering effects in several sound reinforcement and reproduction scenarios. The performance of dynamic diffuse signal processing (dynamic DiSP) is evaluated alongside non-time varying decorrelation (static DiSP) in several environments. The following section will provide a brief overview of existing audio signal decorrelation methods, and justification for the choice of decorrelation method employed in this work.

1.1   Audio signal decorrelation overview

There currently exists a large number of methods for signal decorrelation. Many of these have been developed for use with stereophonic echo cancellation systems, where signal decorrelation is required for correct acoustic path identification for echo cancelling adaptive filters[4-9]. Methods include amplitude modulation, additional low level white noise, interleaved comb filtering, time varying all-pass filtering, and additional non-linear distortion.

These methods have generally been assessed in relation to limited bandwidth, voice conferencing applications. Amplitude modulation would likely be too destructive for full spectrum, high quality audio applications. Adequate decorrelation by the addition of random noise is unlikely to be achieved unless the noise is at an audible level relative to the signal[5]. Interleaved comb filtering methods rely on signals acoustically summing perfectly at the listening position to avoid perceptual coloration. Time varying all-pass filters have been found to degrade audio quality without careful constraint of the time varying parameter[6-7]. Methods regarding the introduction of non-linear distortions including quantization noise were assessed subjectively in a MUSHRA style test and found to degrade audio quality[10].

To be suitable for high quality, wideband sound reproduction applications, the decorrelation method must be perceptually transparent, introduce minimal system latency and be scalable to an arbitrary number of discrete sources.

A library of decorrelation filters can be constructed from a frequency domain specification with unity magnitude response across all frequencies, with the phase response defined by a random number sequence[11]. Filter coefficients are obtained via an inverse Fourier transform (IFFT) and applied via convolution with the input signal. Decorrelation via a library of pre-generated decorrelation filters offers the option of scalability to any number of sources as well as low-latency implementation, as all that is required is a convolution operation for each discrete source in the system. However, this methodology is only valid if the decorrelation filter generation algorithm provides filters that are adequately decorrelated from each other and are perceptually transparent. Kendall[11] and Boueri[12] discuss potential issues regarding perceptual transparency with the aforementioned method. High variations in the phase response lead to magnitude deviations from unity between frequency bins. This in turn leads to signal coloration post-IFFT.

A further method for the generation of a library of decorrelation filters is presented by Hawksford[13], referred to as diffuse signal processing (DiSP). *Temporally diffuse impulses* (TDIs) are generated by the summation of cosine waves of increasing frequency up to Nyquist. Each cosine wave is subject to a frequency dependent exponential decay, with the decay constant increasing with frequency. The phase of each cosine is dictated by a random number sequence. To ensure each cosine frequency component carries equal energy, each wave is normalized by its RMS value. An all-pass response is achieved with minimum phase equalization, which preserves the random excess phase response of the impulse.

The application of DiSP for the reduction of low-frequency SV was previously carried out by the authors[12.] A variation on the frequency dependent exponential decay method is described, where decay constant is definable by frequency bin. As the choice of decay constant for each cosine wave is directly linked to the resulting level of decorrelation and perceptual impact in that frequency range (i.e. the lower the decay constant for a frequency component, the greater the level of decorrelation between TDIs in that bin, but the greater the perceptual impact), this allows for flexibility regarding trading level of decorrelation with perceptual impact throughout the spectrum. To ascertain subjectively valid values for decay constant by octave band, a subjective test was carried out by the authors[13]. Using this method of TDI generation results in a library of decorrelation filters that are perceptually transparent and adequately decorrelated from each other.

In this work, the method of TDI generation by variable decay constant will be employed, utilizing the decay constants from prior work[14]. As this method of signal decorrelation requires updating filter coefficients over the course of milliseconds, a pre-generated library of filters is necessary. The previously-discussed method allows for an arbitrary number of perceptually transparent decorrelation filters to be generated.

1.2   Rationale for Time Varying Decorrelation

In the investigation into the effectiveness of DiSP for the reduction of low frequency SV it was found that whilst SV was effectively reduced in most cases using an anechoic model, performance was significantly reduced using an image source model[14]. This is because whilst the discrete sources in the image source model were decorrelated, surface reflections were highly correlated to direct sounds. It is proposed here that a time varying decorrelation method will increase DiSP effectiveness in reflective acoustic spaces, provided the update rate of TDI coefficients is shorter than the arrival time of the first problematic early reflection.

This was investigated further by Moore[15]. A dynamic variant of DiSP is described, which makes use of a large library of pre-generated TDIs. Each frame of each discrete source audio stream is convolved with a new TDI drawn from the library. An overlapping output window is used to minimize perceptual effects. It was found that dynamic DiSP provided superior results in reducing low frequency SV across a wide audience in both an anechoic and image source model.

It is expected that for applications where surface reflections are a factor, time variance is required, although further real-world measurements are necessary to validate this.

2 DYNAMIC DIFFUSE SIGNAL PROCESSING APPLICATIONS

Decorrelation algorithms have many potential uses in audio reproduction. Decorrelation can be used in mixes to increase spaciousness[16], up-mix mono sources to pseudo-stereo or multichannel mixes[17], increase externalization in headphone listening[18] and reduce comb filtering effects that cause timbral colouration or strong low frequency SV across audience areas[19]. This work examines the effectiveness of dynamic DiSP when applied to a number of such applications, inspecting both modelled and real-world results.

2.1 Low-frequency spatial variance reduction across wide audience areas

A real-world space was utilized, with dimensions 10.6 m x 11.6 m x 9.2 m. Two subwoofers were positioned on the floor at coordinates (2.9 m, 4.7 m) and (7.7 m, 4.7 m). The two speakers were therefore 4.8 m apart. A measurement grid of 20 points was positioned in the audience area. The total grid covered 4.8 m x 3.6 m with a point to point spacing of 1.2 m.

Three configurations were investigated: the unprocessed system (where both subwoofer signals were identical), the system with static DiSP applied (where the subwoofer signals were decorrelated by convolving each frame of each loudspeaker's audio stream with the same TDI pair), and the system with dynamic DiSP applied (where the subwoofer signals were decorrelated by convolving each audio input frame with a different TDI pair). The methodology of dynamic DiSP is described further in section 3.1.2.

A musical signal was used (Tom Sawyer by Rush) to excite the space, as a standard impulse response measurement is impossible with time-varying signal processing such as dynamic DiSP. The transfer function was obtained by analyzing each measurement based on the input signal.

2.1.1 Calculation of spatial variance

The spatial variance (SV) in magnitude response across an audience area can be quantified in decibels with Equation 2.1:

$$SV = \frac{1}{N-1}\sum_{i=1}^{N}\left|A_i - \overline{A_i}\right|^2$$
Eq. 2.1

where N represents the number of measurement positions, A_i represents the sound pressure level (SPL, dB) of a frequency bin at measurement position i, and $\overline{A_i}$ represents the mean SPL (dB) of that frequency bin over all measurement positions. SV is analyzed here over the frequency range 30 – 200 Hz (the lower limit of 30 Hz was necessary due to signal to noise ratio constraints, and the upper limit of 200 Hz due to the subwoofer crossover frequency) and is given as a mean value for all frequencies in this range.

2.1.2 TDI library generation

Appropriate selection of a TDI update rate for dynamic DiSP is crucial, and is informed by the size of the acoustic space and source/measurement locations. TDI update rate can be defined by adjusting the audio stream frame size. In this work an audio sample rate of 44100 Hz is used in all tests. The appropriate TDI update rate for a given acoustic space is approximately determined by the first reflection arrival time to a central measurement position which is one half wavelength delayed from the direct sound (at the highest analysis frequency).

For the previously-described space, a required TDI update rate of 55.4 ms was calculated, indicating that an audio frame size of 2443 samples was needed. Within each output audio frame, a 1/3 overlap factor was used to reduce perceptual effects from the constantly changing TDIs,

meaning that 3 different TDIs were part of the generation of each audio output frame, therefore the actual TDI update rate was 1/3 of that which can be seen as an absolute maximum for dynamic DiSP effectiveness. To further reduce audible effects, the three TDIs per output frame were obtained via interpolation from one set of TDI coefficients to the next.

The TDI library was generated using a uniform probability density function for random phase generation, with phase constrained to ±0.94π[13]. Decay constants are listed in Table 2.1[15].

Frequency band (Hz)	Median decay time audible threshold (ms)	Frequency band (Hz)	Median decay time audible threshold (ms)
<63	179.8	250-500	19.7
63-94	104.8	500-1000	15.7
94-125	78.8	1000-2000	12.7
125-187.5	36.8	2000-4000	8.2
187.5-250	27.6	>4000	3.7

Table 2.1 Decay constant values for TDI generation[15]

Decay constants were set at the center of each frequency band, with intermediate decay constants obtained by linear interpolation[15]. As decorrelation is not required above 4 kHz for any of the tested applications, it is desirable to set the decay time of these frequency components to an arbitrarily short time (<5ms) so that they only contribute to the initial impulse of the TDI and not the noise tail, eliminating any potential audible artifacts for frequencies above 4 kHz.

This also has the effect of scaling the amplitude of the noise tail in relation to the initial impulse, which can be useful in TDI generation. The rapid exponential decay of these high-frequency components gives them a low RMS value over the full length of the TDI. When they are normalized by their standard deviation (i.e. standard deviation set to 1), their overall amplitude is increased. The summation of these cosines from 4 kHz to Nyquist leads to a large increase in the amplitude of the initial impulse over the first few samples of the TDI. Once the initial impulse amplitude is normalized to 1, the amplitude of the noise tail is greatly reduced. This is illustrated in Figure 2.1.
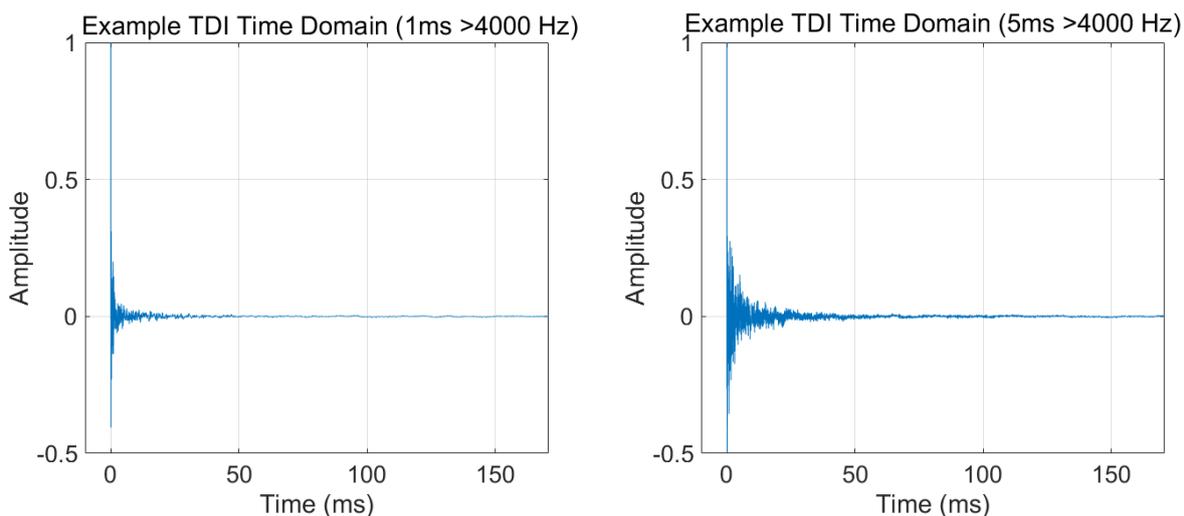

Figure 2.1 Example TDIs. Each TDI was generated with the same decay constants (Table 3.1) except the left plot shows TDI generated with all cosine waves above 4000 Hz set to 1 ms decay, and the right plot shows the result of setting these cosine waves to 5 ms decay. The increased amplitude of the noise tail in the right plot equates to greater decorrelation when multiple TDIs are used, but also greater audibility of the filter.

Two example TDIs are shown where all TDI generation parameters are identical except the choice of decay constant for components above 4 kHz. The left plot shows the resulting TDI with a 1 ms selection, and the right plot shows the TDI with a 5 ms selection. The two TDI libraries will perform differently, with the 5 ms selection providing greater decorrelation but with increased perceptibility.

In this testing, two TDI libraries were generated, identical apart from the selection of decay constant above 4 kHz (1 ms and 5 ms, respectively). Further work will include subjective testing to establish a perceptually-valid limit for this selection.

Each TDI library consisted of 100 TDIs. Dynamic DiSP was achieved by drawing a different TDI for each source per audio frame. Each output frame consisted of the overlapping of 3 successive input frames. For static DiSP, a random pair of TDIs was chosen from the library and was convolved with each input frame to generate the two decorrelated source signals.

2.1.3 Results

SV across the previously-described measurement grid was calculated for the frequency range 30-200 Hz for the unprocessed (correlated) signals, static DiSP processed signals and dynamic DiSP processed signals. The results are presented as a percentage change in SV from the unprocessed signal over several analysis window lengths to observe temporal DiSP performance. Unprocessed SV was consistently in the range of 15-20 dB for all analysis windows.

Figure 2.2 shows the performance of static and dynamic DiSP over analysis windows ranging from 50 ms to 10 s. Auditory temporal integration time has been suggested in previous work to be in the region of 170-200 ms[20-21] and therefore an analysis window of 170 ms is included. Dynamic DiSP has superior performance to static DiSP even when utilizing TDIs with the same generation parameters, indicating that time varying decorrelation is desirable in reflective acoustic environments. It is shown that static DiSP with a 4 kHz decay time of 1 ms is not effective, although the dynamic variant successfully reduced SV over a long enough analysis window.
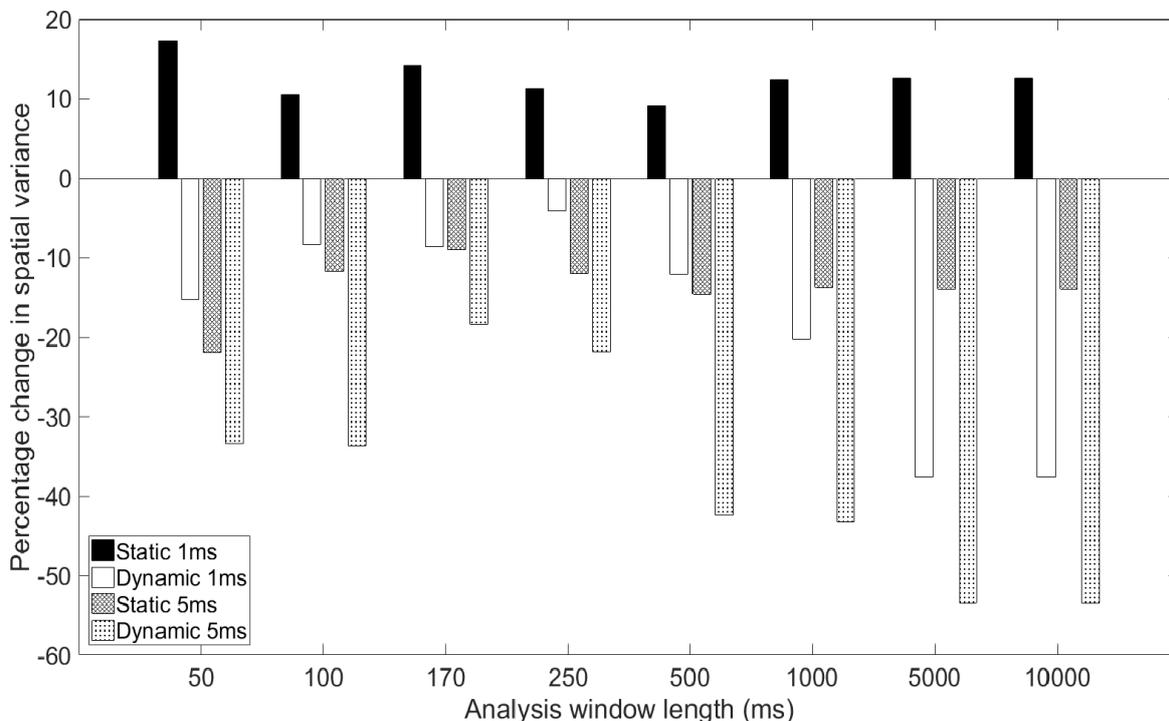

Figure 2.2 Percentage change in SV for static and dynamic DiSP using both 1 ms and 5 ms TDI libraries for analysis windows ranging from 50 ms to 10 s

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2.2 Low-frequency control in small rooms

Small- and medium-sized rooms are typically subject to high spatial variance due to room-modes[3]. It is possible that dynamic DiSP may alleviate this issue due to the lack of correlation between surface reflections.

For this test, a room of size 3.9 m x 4.3 m x 1.9 m was inspected. The walls consisted of painted brick, the floor of concrete and the ceiling of plaster on lath. Two full-range loudspeakers were placed 0.95 m off the floor at coordinates (1.4 m, 0.5 m) and (2.9 m, 0.5 m). The system was inspected following the same procedure as detailed in section 2.1, except the measurement grid spanned 2.0 m x 2.2 m. A required TDI update time was calculated as 10.7 ms. Unprocessed SV was measured to be between 10-17 dB for all analysis windows.

Figure 2.3 shows the results of the small room test. It is shown that dynamic DiSP outperforms static DiSP in the reduction of low-frequency SV, but that it takes an initial 250 ms or so for the full effect to be realised. This is due to the TDI length of around 170 ms. For dynamic DiSP to work, there must be sufficient time for several TDIs to be applied. Static DiSP with 1 ms decay outperformed static DiSP with 5 ms decay over most analysis windows, which highlights another issue with static DiSP. Even when TDI generation parameters should lead to combinations of TDIs providing greater levels of decorrelation, it is still possible for individual TDI combinations to provide weak decorrelation due to similarities in the phase response. This is especially true where only 2 sources are used, providing only 2 degrees of freedom. Dynamic DiSP circumvents this by utilizing a library of TDIs and constantly updating TDI combinations over time.
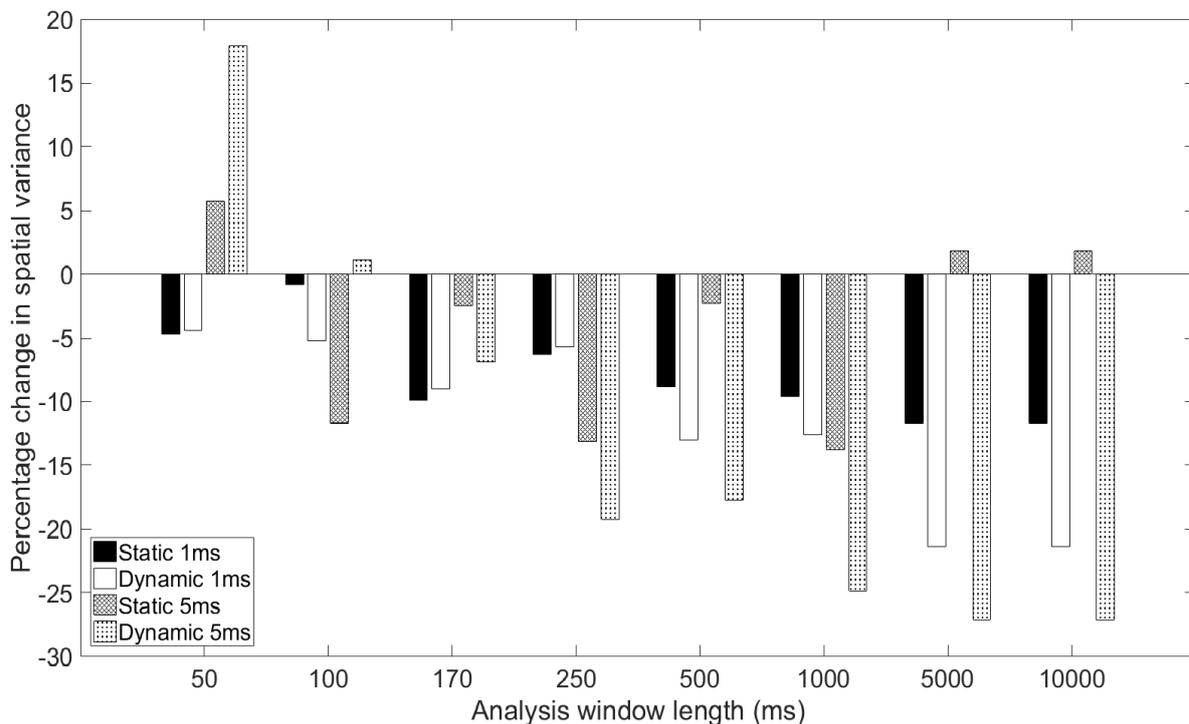

Figure 2.3 Percentage change in SV for static and dynamic DiSP using both 1ms and 5ms TDI libraries for analysis windows ranging from 50ms to 10s in small room test

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2.3 Low-frequency control in small rooms with a single source

With dynamic DiSP, it is interesting to consider the case of a single source and whether the low-frequency characteristics of a small room can be made to resemble that of a more diffuse space.

For this test, the experiment detailed in section 2.2 was repeated with one source positioned 0.95 m from the floor at coordinates (2 m, 0.5 m). The results are shown in Figure 2.4.
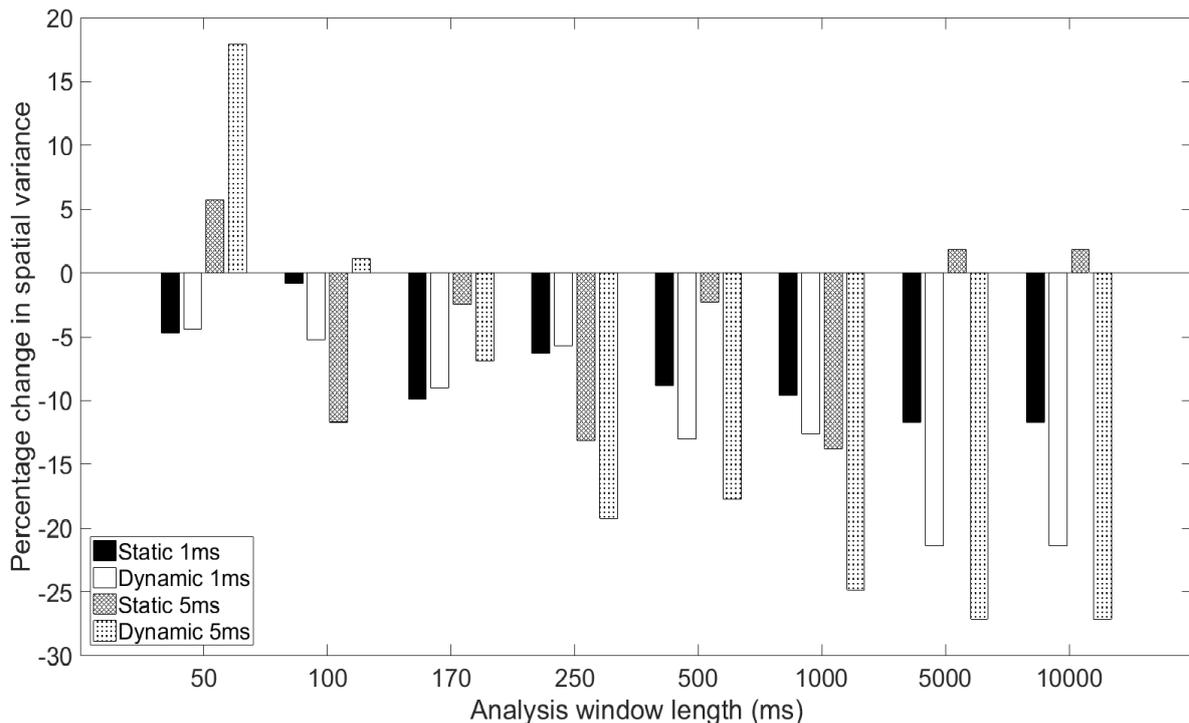

Figure 2.4 Percentage change in SV for static and dynamic DiSP using both 1ms and 5ms TDI libraries for analysis windows ranging from 50ms to 10s with a single source.

Dynamic DiSP is shown here to be reasonably effective at reducing SV with even a single source, generally being shown to increase in effectiveness over longer analysis windows due to the application of multiple TDIs. Of interest is the performance of static DiSP with 1 ms decay. Static DiSP should provide little impact on SV with only one source as it is expected that diffuse behaviour will not be achieved with a single TDI. This may indicate that there is some level of acoustic decorrelation applied when sound reflects off room surfaces. The utilisation of DiSP with a single source requires further investigation and is the focus of future research.

2.4 Crossover Network Design

Loudspeaker crossover networks are subject to timbral colouration caused by comb filtering around the crossover frequency. Decorrelating transducers in a single loudspeaker unit is possible provided unique TDIs are used for each drive-unit. Similarly, a set of line arrays may be decorrelated from a subwoofer system in a live sound reinforcement setting.

In this test, an image source simulation of a loudspeaker with a single tweeter and woofer spaced at 10 cm was used, and a crossover frequency of 2000 Hz. The simulated space was 4 m x 4 m x4 m and typical living room absorption coefficients at 2000 Hz were used (walls set to 0.05 for plaster board on battens, ceiling set to 0.05 for plaster on lath and floor set to 0.75 for 9 mm pile carpet). A fifty-point measurement grid was used, on axis and in the vertical plane.

Figure 2.5 shows a sound pressure level map (dB) of the 50-point measurement grid without DiSP (top), and with dynamic DiSP (bottom) at the crossover frequency of 2000 Hz for an analysis window of 1 second.
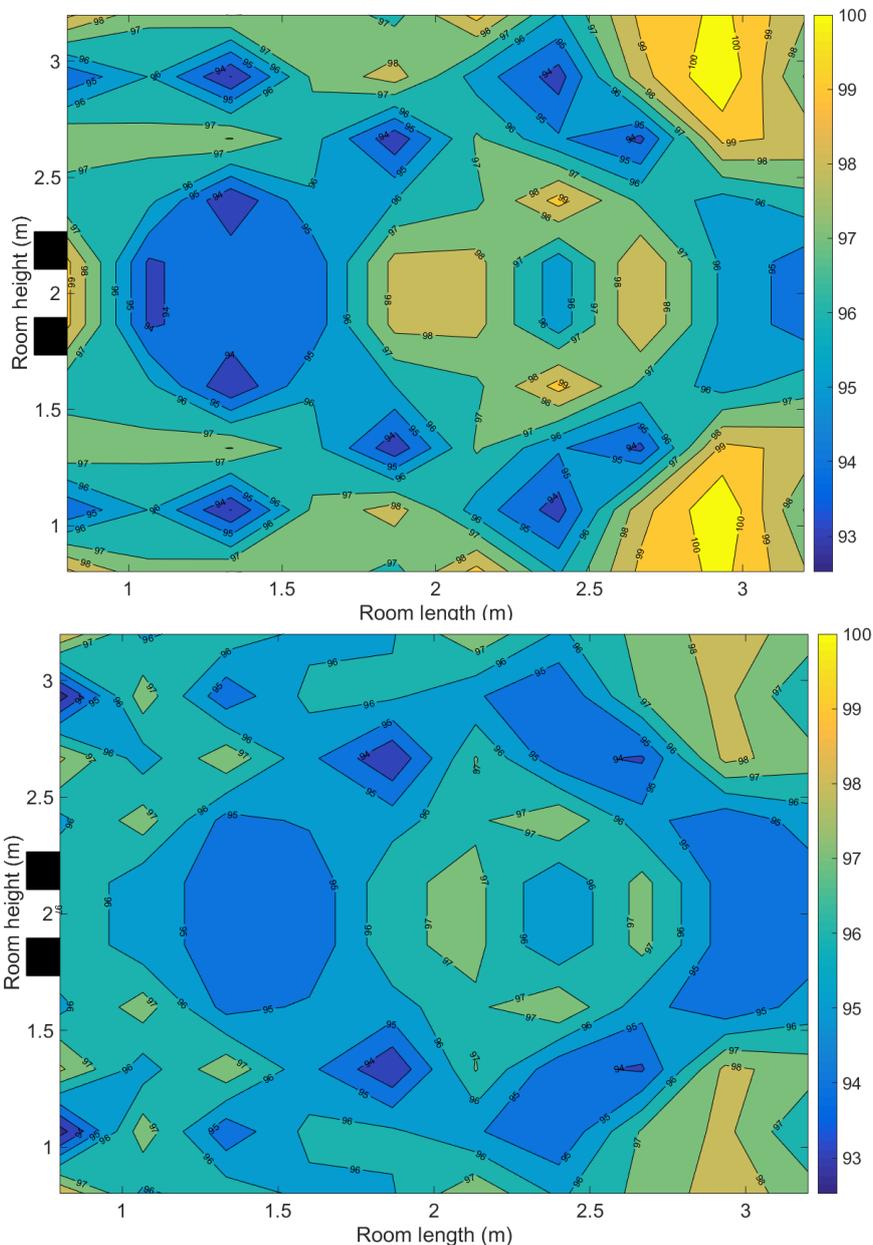

Figure 2.5 Sound pressure level map (dB) of the 50-point measurement grid without DiSP (left), and with dynamic DiSP (right). Black boxes indicate drive-unit locations.

Dynamic DiSP is shown to reduce the SV at 2000 Hz across the measurement grid, with an overall reduction of 43.5%. Areas of minimal and maximal pressure are still present, but are reduced both spatially and in terms of magnitude difference from the SPL mean.

This indicates the effect of comb filtering has been reduced and should result in a reduction of timbral colouration of source material. Further real-world tests are needed to assess the subjective impact of this.

2.5   On-stage monitoring systems

To ensure performers are given an accurate and audible reference signal, typically either one or more loudspeakers are employed per performer on stage. These may take the form of monitor wedges and/or side fills. Whilst the individual mixes for each speaker may differ, the elements within the mixes originate from the same sources, and are therefore correlated. This leads to a situation where dispersion patterns of many speakers may overlap in a relatively small area, causing comb filtering and timbral shifts as performers move around the stage.
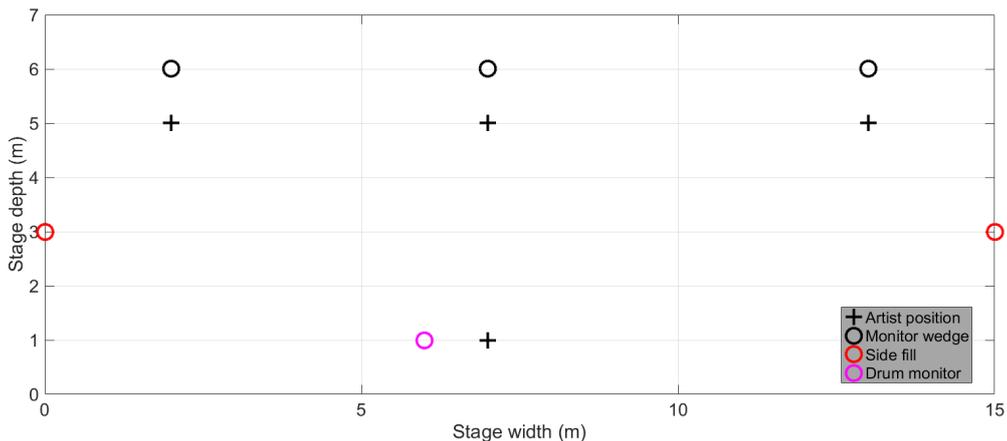

Figure 3.1 Example monitor speaker/performer location set up used for simulation

In this test, an example stage set up was simulated. Six loudspeakers were positioned as in Figure 3.1. To simulate an outdoor setting, only the floor (stage) of the model was made reflective, and all other surfaces were set to anechoic. The frequency response measurements were generated in the same way as the other tests, over the course of 1 s. Dynamic DiSP was used up to 4 kHz, again by setting decay times above this at 1 ms, and results are shown from 0-4000 Hz.

Figure 3.2 shows the magnitude responses of 50 measurement positions across the simulated stage area, with 1/10th octave smoothing applied to both plots.
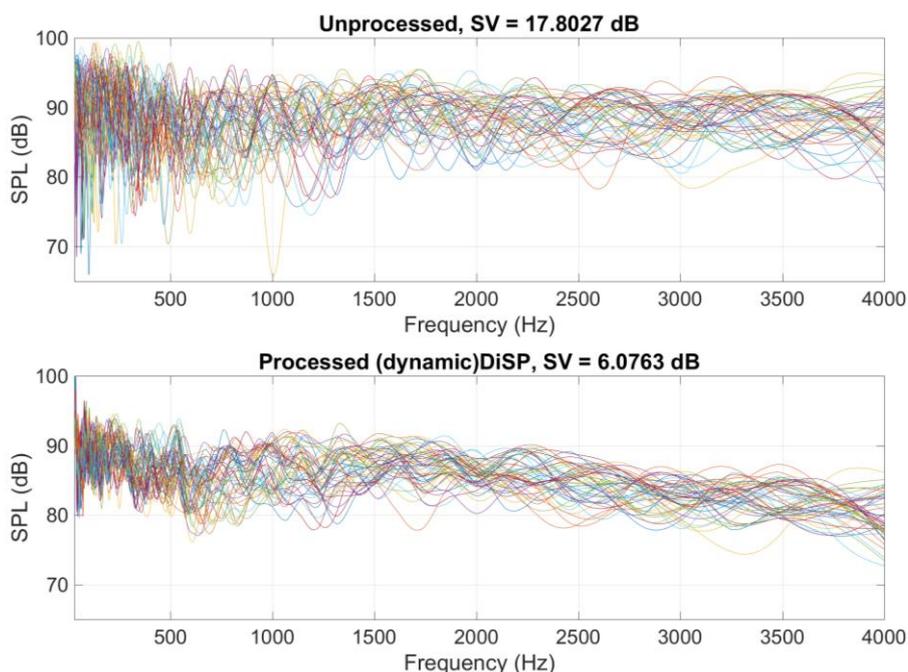
Figure 3.2 Magnitude responses of 50 measurement positions across the stage area for dynamic DiSP (lower plot) and the unprocessed system (upper plot)

The lower plot in Figure 3.2 shows that dynamic DiSP is successful in smoothing many of the amplitude nulls seen in the upper plot (where DiSP is not active), whilst bringing more uniformity to the responses as indicated by the reduction in spatial variance from 17.8 dB to 6.1 dB. Perceptually, this equates to a more uniform response when moving around the stage, and more accuracy regarding the source material.

3   CONCLUSION

This work has investigated the application of dynamic and static DiSP to several typical audio reinforcement and reproduction scenarios. It has been shown that the dynamic variant of DiSP performs better than static DiSP in the reduction of low-frequency SV when the analysis window is long enough to include the overlapping of several TDIs. This indicates that in the case of dynamic DiSP, SV caused by coherent surface reflections is being tackled as well as SV that is a result of coherent source interference, implying that dynamic DiSP may be a useful tool in the treatment of modal room behaviour. If that is the case, dynamic DiSP is potentially a more attractive solution than existing methods of modal treatment, which include the use of large volume absorption[22], multiple speaker placement[1] or multi-point equalisation techniques.

Other potential applications have been investigated via image source simulation. It has been shown that dynamic DiSP may be useful in reducing comb filtering around crossover frequencies, and can be used for broadband decorrelation of multiple speakers resulting in the reduction of amplitude nulls and peaks caused by comb filtering. Further investigations and real-world measurements are necessary to validate these preliminary results, and this will be the focus of future research.

Further analysis and optimisation is required, especially regarding perceptual effects of the processing. It is noted that small rooms may require TDI update rates of >10 ms. The perceptual effects of dynamic DiSP are more noticeable with short window lengths, so it is possible further modifications to the dynamic DiSP methodology are necessary.

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