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Wide-area psychoacoustic correction for problematic room modes using non-linear bass synthesis

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ABSTRACT

Small room acoustics are characterized by a limited number of dominant low-frequency room modes which result in wide spatio-pressure variations that traditional room-correction systems find elusive to correct over a broad listening area. A psychoacoustic-based methodology is proposed whereby signal components coincident only with problematic modes are filtered and substituted by virtual-bass components to forge an illusion of the suppressed frequencies. A scalable and hierarchical approach is studied using the Chameleon Subwoofer Array (CSA) and subjective evaluation confirms a uniform large-area performance. Bass synthesis exploits parallel nonlinear and phase vocoder generators with outputs blended as a function of transient and steady-state signal content.

1. INTRODUCTION

Room modes in small- to medium-sized closed acoustical spaces often cause wide variations in low-frequency response across a listening area. This acoustical spatial variance will result in largely different impressions of a room and/or sound system, commonly with adjacent listeners experiencing antithetical acoustical conditions.

A surplus of research exists concerning room-mode correction/suppression, including passive, active and hybrid systems. Many well-accepted correction systems perform effectively in decreasing the consequences of room modes, but often fall short concerning spatial variation minimization and can require highly-complex signal processing. Depending on the system configuration, there are often room modes that are nearly impossible to fully correct, resulting in an incomplete solution to the problem at hand.

As room modes are a physical phenomenon, it is proposed that a psychoacoustical method could strengthen a physically-based correction technique or possibly even operate as a standalone correction procedure. This would ease the physical requirements of the system and allow for problematic room modes to be addressed largely within the psychoacoustical domain.

A system is presented in this paper whereby the "principle of the missing fundamental" (or virtual-bass synthesis) is utilized to create the impression of the presence of certain narrow frequency bands while in actuality these dominant bands are removed from the audio signal. This procedure effectively eliminates the physical reinforcement of the most problematic room modes with the aim of significantly reducing spatial variation across a listening area while maintaining consistent signal fidelity.

Common small-room, low-frequency correction procedures will be briefly discussed, highlighting the pros and cons of each system followed by an analysis of the standard procedures used to implement the principle of the missing fundamental by means of a novel hybrid system. The virtual-bass procedure will then be described in the context of room-mode suppression, including subjective evaluation results. Finally, the virtual-bass correction procedure will be analyzed in the context of the Chameleon Subwoofer Array (CSA) low-frequency wide-area correction system [1, 2] with the purpose of dealing with problematic room modes to ease system requirements.

2. LOW-FREQUENCY ROOM ACOUSTICS

Room acoustics are influenced largely by room modes below the Schroeder frequency, as defined in Equation 2.1 [3].

$$f_s = 2000\sqrt{\frac{RT_{60}}{V}} \qquad (2.1)$$

where the Schroeder frequency, f_s (Hz), is defined by the reverb time, RT_{60}, and the room volume, V (m^3). This limit for the low-frequency range of a closed space operates on the principle that above the Schroeder frequency room modes become sufficiently spatially and spectrally dense so as not be subjectively distinct, largely due to masking within the human hearing mechanism.

Room modes are a consequence of standing waves between one or more set of parallel reflecting surfaces and arise at frequencies with integer multiples of their half-wavelengths fitting perfectly within the standing wave pattern (Equation 2.2). Listeners experience largely different steady-state and transient responses at these frequencies, depending on their location relative to the complex standing wave pattern (Figures 2.1 & 2.2).

$$f_m = \frac{c}{2}\sqrt{\left(\frac{\eta_x}{x}\right)^2 + \left(\frac{\eta_y}{y}\right)^2 + \left(\frac{\eta_z}{z}\right)^2} \qquad (2.2)$$

where the room-mode frequencies, f_m, are calculated for η_x, η_y and η_z from zero upwards with x, y and z representing the dimensions (in meters) of a rectangular space and c, the speed of sound in air (m/s) [4].

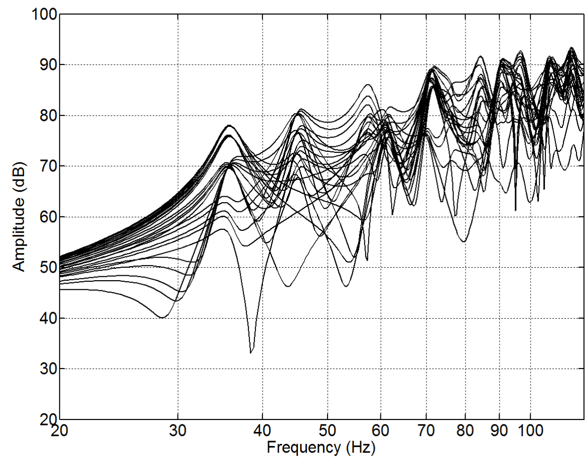

Figure 2.1 Simulated frequency responses of 25 listening locations in a 5 m x 4 m x 3 m room with a single subwoofer in the room corner

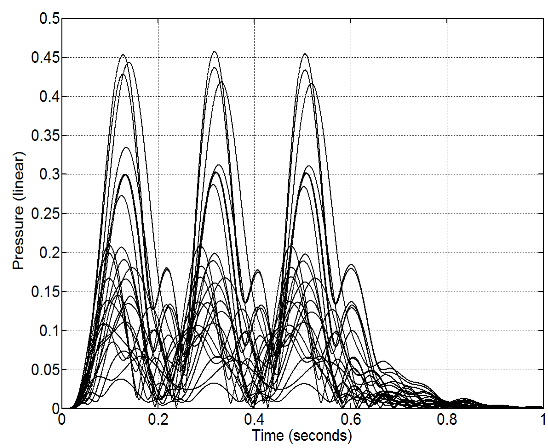

Figure 2.2 Simulated 80 Hz tone burst measurements at 25 listening locations in a 5 m x 4 m x 3 m room with a single subwoofer in the room corner

The largely varying frequency response over a listening area, shown in Figure 2.1, demonstrates how greatly the low-frequency steady-state acoustical response differs between closely-spaced listening locations. At some locations certain frequencies may be overpowering while at other locations the same frequencies can be virtually non-existent. The related transient response also experiences similar spatial variation among listeners, often causing difficulty in perceiving detailed time-domain nuances (i.e. following the bass line) within a signal, illustrated for example by the 80 Hz tone burst [5] shown in Figure 2.2.

3. COMMON ROOM-MODE CORRECTION PROCEDURES

Low-frequency room-mode correction can be addressed with a number of well-known passive and active procedures. Each of these techniques approaches the modal problem from a different perspective, resulting in varying advantages and disadvantages between the methods. These positive and negative aspects of each approach must be considered when selecting a correction procedure that meets system requirements.

3.1. Passive correction – surface absorption

Increasing the surface absorption in a room is a simple technique to mask modal problems within a space. Absorption is generally increased by adding soft, porous materials such as foam to the walls of a room. The decreased reflections from the walls limit the buildup of standing waves and cause the modes to exhibit a wider Q, resulting in less noticeable room resonances due to an increased modal overlap (Figure 3.1).

While Figure 3.1 highlights a noticeable decrease in sharp room modes as absorption is increased tenfold, spatial variance only decreases by a marginal amount of 5.0%. Even though the acoustic space will exhibit significantly fewer low-frequency resonances with added wall absorption, a strong variance will still exist between listeners.

Simple foam-based absorbers are regularly used to reduce high-frequency reflections, but can be difficult to implement for lower frequencies since their size is dictated by the wavelengths in the target frequency range. This would require unreasonably large absorbers for low frequencies; hence this correction method may not be suitable for low-frequency spatial variance reduction purposes.

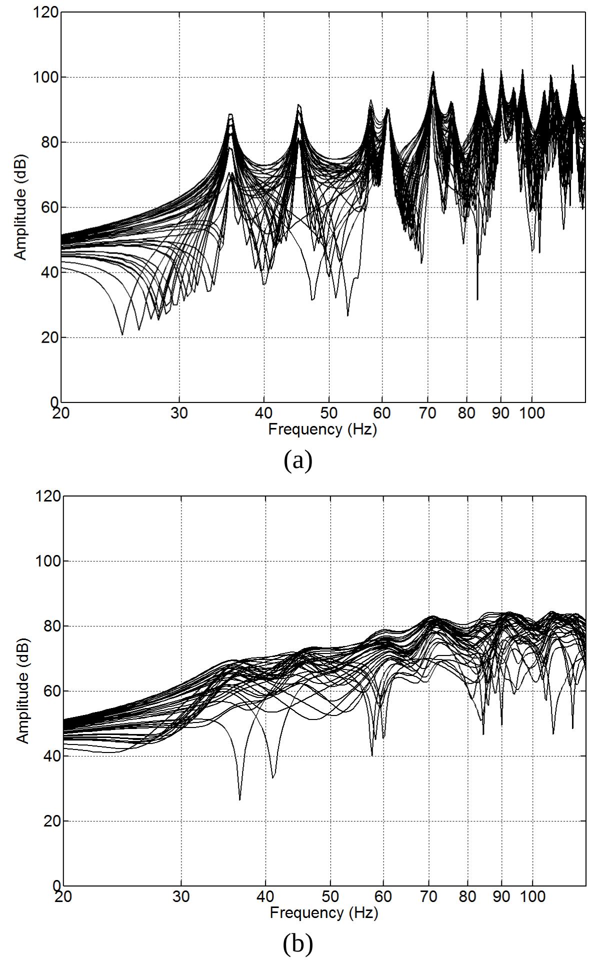

Figure 3.1 Simulated frequency responses over a listening area with (a) 2% wall absorption and (b) 20% wall absorption

3.2. Passive correction – source placement

In situations where additional absorption is not practical and there are minimal signal processing options within the system, intelligent source placement can provide a significant reduction in spatial variance.

Often the goal is to achieve maximum low-frequency output without high levels of amplification. This has been achieved by keeping subwoofer to room-mode coupling in mind [6]. When a subwoofer is placed at an anti-node of a room mode, coupling will be maximized. When the subwoofer is placed at a node, coupling will be minimized (theoretically zero) due to placement at the standing wave's zero crossing [4]. Placing the subwoofer in the corner has the added benefit of the Waterhouse effect, where each nearby boundary contributes a 6 dB boost to the sound pressure level, giving an 18 dB boost for a corner location [7].

While this simple expedient projects greater low-frequency energy into the room, it does not provide an equal response at all listening points. This is due to the strong dependence on source to listening location coupling. To provide more uniform coverage at low-frequencies when using a single subwoofer, it should be placed close to as many nodes as possible. Although the center of a room generally contains the most frequency nodes common to a single point, it is normally an impractical location for a subwoofer. In addition, center placement does not benefit from the Waterhouse effect, resulting in lower system output.

Due to the drawbacks of central subwoofer placement, a compromise can be made by placing the single subwoofer ground level at a wall midpoint (Figure 3.2). This placement provides a 16% spatial variance reduction compared to corner placement (while central placement results in a 38% reduction). However, as with passive absorption techniques, single subwoofer placement cannot provide sufficient spatial variance reduction to guarantee equal listening experiences for all listeners.

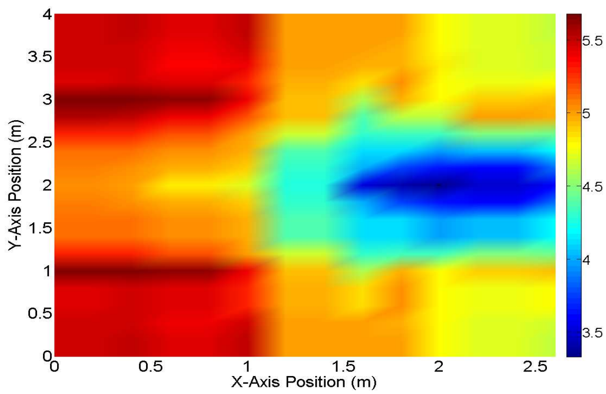

Figure 3.2 Simulated spatial variance values for various single subwoofer placements over the first half of a 5 m x 4 m x 3 m room

Research has also been conducted using multiple omnidirectional subwoofers to minimize spatial variance, concluding that four subwoofers located at wall midpoints is the most practical solution giving significant spatial variance reduction (Figure 3.3) [4, 8].

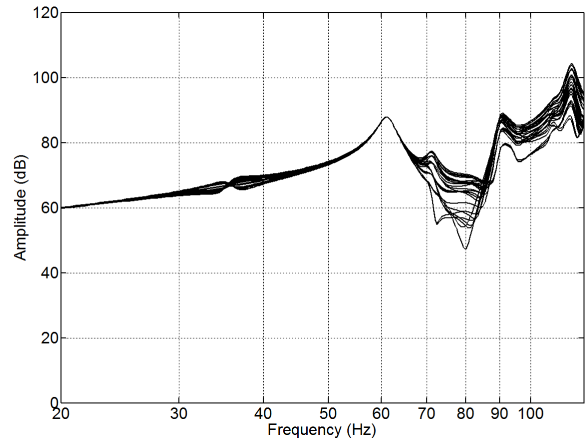

Figure 3.3 Simulated frequency responses over a listening area with a four subwoofer system consisting of one omnidirectional unit at each wall midpoint

This configuration provides a more uniform listening experience throughout a space, but with lower efficiency due to the destructive interference used to limit the buildup of standing waves and also the nodal placement of the sources (low source to room coupling).

3.3. Active correction – parametric equalization

A simply implemented room-mode suppression technique involves parametric equalization. This method usually targets three to five of the most problematic room modes by applying notch filters centered at these frequencies to limit their reinforcement within the sound system. This strategy limits the buildup of standing waves at these frequencies and can help to reduce spatial variance.

While this low-frequency correction method does not address all modal problems within a space, it can be easily used as a quick fix for the worst acoustical problems in a room. Problematic modes can either be identified and handled automatically with room measurements or addressed manually by ear.

The drawback to this technique is that it eliminates information in the targeted frequency bands; therefore listeners may be missing key elements of the audio signal in exchange for modal suppression.

3.4. Active correction – single point equalization

Another common correction technique similar to automatic parametric equalization is single-point equalization. This operates by taking a measurement at the primary listening point in a room, generally with a maximum length sequence (MLS) test signal. Once the frequency response is calculated from the measurement, an inverse filter will be created based on the magnitude response within the target frequency band and applied to the system output.

Single-point equalization can be applied over the entire audible frequency range (20 Hz – 20 kHz) and typically performs well at the target listening point. Unfortunately, this is not the case for the non-target points, where source-to-listener coupling is different from the target location. This causes low correlation between the correction benefits at the target point and non-target points. Spatial variance will usually not exhibit a noticeable change using single-point equalization (Figure 3.4); therefore this technique is only effective for scenarios where there is only one listening location and listener using the system at a time.

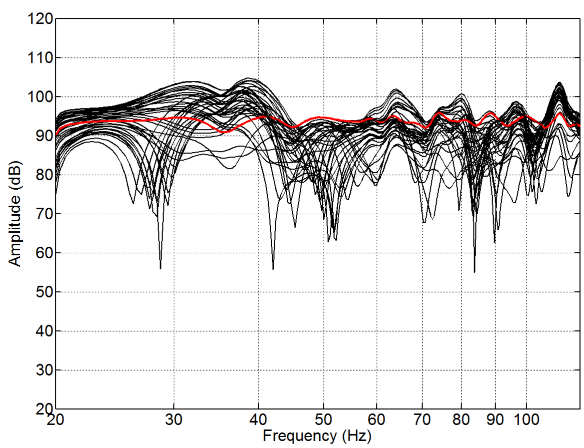

Figure 3.4 Simulated frequency responses over a listening area after single point equalization (red line = target equalization point)

3.5. Active correction – other techniques

There exist a number of additional low-frequency correction techniques that have been addressed in previously published literature. Many of these techniques involve multiple-point equalization techniques whereby measurements are taken throughout the listening area and are grouped based on similarity and/or weighted based on location importance to give significant spatial variance reduction [9 – 13].

Some of these methods employ fixed equalization (one-time measurements) while others utilize adaptive systems where measurements are continuously taken as the system is operational, leading to a problem in having measurement microphones throughout the listening area at all times. This problem has been eliminated with the system presented in [13] since measurements are taken in close proximity to the subwoofer drive unit.

An additional room correction method that has been the topic of investigations is active absorption [14]. Active absorption combines the principles of passive correction methods and single/multiple point equalization methods. This method generally operates with a set of one or two primary sources placed at one end of a room. In addition, an array of secondary sources is placed at the opposite end of the room, with each drive unit usually containing a measurement microphone to monitor the signals received from the primary sources. These secondary sources will then reproduce their measurements with reverse polarity in an attempt to suppress wall reflections, giving a traveling wave in the room as if it were anechoic.

Active absorption can require a large number of secondary drive units to effectively suppress reflections within the room, making them difficult and costly to implement. When properly calibrated, though, these systems can create a virtual anechoic environment where all points (at a sufficient distance from the secondary units) will experience the same response both in the time and frequency domains.

4. VIRTUAL-BASS SYNTHESES

This section describes a low-complexity room-mode correction process that builds upon the concept of parametric equalization by incorporating a psychoacoustic-motivated procedure known as *virtual bass*. The process is compatible with a wide range of sound reinforcement systems. The core problem with conventional parametric equalization is that in order to reduce spatial variance it suppresses the frequencies that maximally excite room modes, thus filtering potentially important audible information. The enhancement presented in this paper seeks to compensate for this loss of information by substituting a subjectively equivalent signal based upon virtual bass synthesis.

Virtual bass synthesis operates on the doctrine of the missing fundamental. The missing fundamental, or the residue pitch, is a result of the complex pitch-extraction mechanism within the inner ear and the brain. When presented with a spectrally-complex sound, the pitch extraction mechanism attempts to make sense of the received signal by relating various spectral components to one another [15]. When such spectral components are equally spaced, this will result in a perceived pitch corresponding to the greatest common factor of the frequency values (in Hz) that falls within the audible range of 20 Hz – 20 kHz. For instance, if the source contains spectral components at 200, 300, 400 and 500 Hz the overall perception will correspond to a harmonically-rich tone at 100 Hz.

This effect can operate using only two higher harmonic components (e.g. second and third) of the fundamental. Adding additional harmonics will increase the sharpness of the signal timbre (sound quality) as the average frequency of the components increases [15].

When applying the missing fundamental for low-frequency applications, it is important to keep the average frequency of all spectral components to a minimum so that the perceived pitch is as close in timbre to the fundamental as possible. Minimizing the amount of harmonic components introduced will also preserve the fidelity of the source signal since these virtual-bass components are a form of distortion which should ideally be kept to a minimum.

There are two primary approaches to implementing the virtual-bass effect which both offer unique advantages and disadvantages. These two techniques will be presented in the following sections.

4.1. Nonlinear device virtual bass

A nonlinear device (NLD) is the most common harmonic generator implemented within virtual-bass systems for a number of reasons. First, the NLD is memoryless, allowing for real-time applications. NLDs generally operate using a polynomial approximation of a chosen function. The calculated coefficients are then applied to the input signal as defined in (4.1).

$$y = \sum_{i=0}^{N} h_i x^i \tag{4.1}$$

where, h is a vector containing the N polynomial coefficients with x and y representing the signal input and output, respectively [16].

The NLD virtual-bass technique operates in the time domain, applying the effect over all spectral components of the signal. However, this process normally introduces intermodulation distortion to the signal if there are two closely-spaced spectral components in the input signal. While it has been argued that these components cause minimal auditory artifacts due to psychoacoustical masking at the Basilar membrane in the inner ear [16], intermodulation distortion is an unwanted peripheral to the NLD virtual-bass system, which must be handled with care.

Early virtual-bass research utilized a full-wave rectifier (FWR) for the NLD [17]. The FWR is simple to implement, but suffers from the fact that it generates only even-order harmonics. A FWR applied to a 100 Hz pure tone would result in harmonic distortion introduced at 200, 400, 600 Hz and so on. Following the principle of the missing fundamental, this harmonic series should result in a perceived pitch of 200 Hz rather than 100 Hz. The perceived pitch is a full octave higher than the target pitch perception which results in an inaccurate virtual-bass effect.

This problem has led to a significant body of research to develop the ideal NLD for virtual-bass applications. A wide range of NLDs are presented in [16], where they are each objectively and subjectively evaluated to best judge performance. The second exponential-type NLD in [16] was rated highly in both objective and subjective tests and was therefore chosen as the NLD for this work. The input-output relationship is shown in Fig. 4.1.

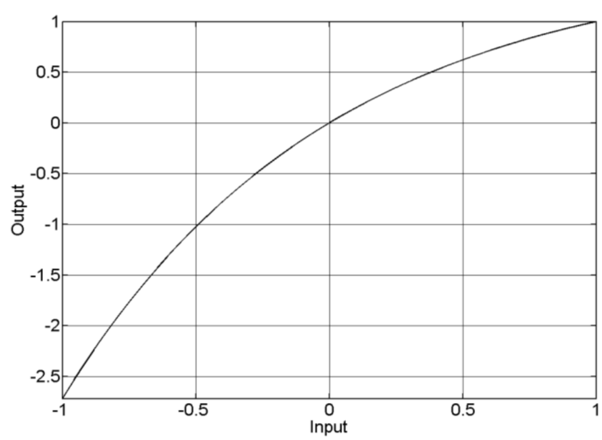

Figure 4.1 Input-output relationship for exponential NLD virtual bass

NLD virtual-bass systems are implemented with a series of filters to give approximate control of the spectral components of the effect. The input signal is first processed by a low-pass filter (LPF) with a cutoff frequency set to the upper limit of the required low-frequency extension. This low-passed signal is then processed by the NLD, generating the harmonic components.

Next, the NLD output is sent through a bandpass filter (BPF) to remove the fundamental spectral components and to roughly shape the harmonic components. If only a low-frequency boost is required (as opposed to a bandwidth extension), the BPF can be replaced by a LPF. After the BPF, gain is applied to the signal and then combined with a delayed version of the original signal. The overall NLD virtual-bass process is shown in Fig. 4.2.

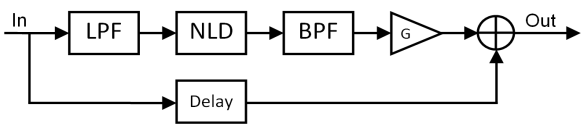

Figure 4.2 NLD virtual-bass procedure

A widely-utilized commercial NLD-based virtual-bass system is called *MaxxBass* [18]. In addition to the system architecture in Fig. 4.2, *MaxxBass* uses equal-loudness processing to provide a virtual-bass effect subjectively equal in level to the unprocessed signal.

4.2. Phase-vocoder virtual bass

An alternative to the NLD virtual-bass approach has emerged in recent years utilizing a phase vocoder (PV) as the harmonic generator [19]. The PV virtual-bass approach provides superior harmonic control, allowing for selective harmonic inclusion in the effect. Since this approach operates in the frequency domain, the intermodulation distortion can be effectively avoided, unlike with NLDs.

PVs operate by splitting an input signal into short time-domain windows (generally between 50 – 250 ms). The PV takes the fast Fourier transform (FFT) of each time window, applies the required processing while maintaining phase coherence and then generates the output signal either by sum-of-sinusoids or inverse Fourier transforms where each window is overlap-added to minimize amplitude-modulation effects. This present work utilizes the sum-of-sinusoids method.

A disadvantage to the PV arises due to the trade-off between time and frequency resolution. Virtual-bass systems require adequate frequency resolution to allow for accurate harmonic generation in addition to avoiding intermodulation distortion. Frequency resolution can be determined by (4.2).

$$f_{res} = {}^{1}/_{t_w} \tag{4.2}$$

where, f_{res} is the frequency resolution (Hz) and t_w is the window length (s). For example, a 125 ms window gives 8 Hz resolution while a 500 ms window gives 2 Hz. This issue leads to smeared transient performance which is clearly evident when applied to audio signals such as drum beats.

Previous solutions to this problem have involved reinitializing the phase within the algorithm when a transient is encountered [20] and also removing any transients from the input signal and then reinserting them, unprocessed, at the PV output [21]. The phase re-initialization solution can prove difficult as it relies on precise transient detection; otherwise, phase re-initialization will occur in excess causing poor phase coherency for the steady-state signal components. The transient removal method has had low ranking in subjective tests since transient signal components are not addressed within the effect [21].

Even through the PV cannot handle transients perfectly it does perform well on pitched signal components. Unlike the NLD system, PV virtual bass does not require a LPF on the input stage, as the algorithm can selectively apply the effect to frequency bins. Within the PV the selected frequencies are pitch shifted to the desired harmonic frequencies and amplitude adjusted to match any equal-loudness requirements; therefore no BPF or HPF is necessary on the output stage.

Since the PV virtual-bass system is more computationally demanding, it is necessary to down-sample the input signal for real-time applications. This requires a LPF before the down-sampling process to avoid any spectral aliasing. Once the signal has been processed, it can be up-sampled to the original sampling rate and recombined with the delayed original signal. The overall PV virtual-bass process is shown in Fig. 4.3.

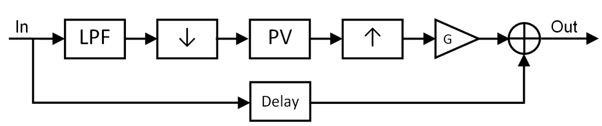

Figure 4.3 Phase vocoder virtual-bass procedure

While PVs are commonly used for audio effects such as pitch shifting and time stretching [22], there are no known commercial applications for PV virtual bass.

4.3. Hybrid virtual bass

A virtual-bass system that exploits the respective strengths of the NLD and PV systems but circumvents their weaknesses should provide a bass synthesis less sensitive to changes in input signal content. When the input signal has a high transient content, the system favors the NLD output and conversely, when the signal is more pitched the PV effect is utilized.

This hybrid approach utilizes a transient content detector (TCD) which analyzes successive time domain windows of the input signal and appropriately weights the respective virtual-bass algorithms that are running in parallel (Fig. 4.4).

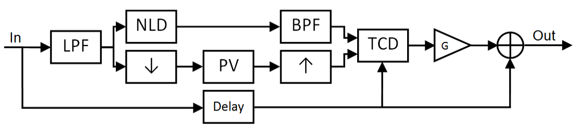

Figure 4.4 Hybrid virtual-bass procedure

The implementation of the hybrid virtual-bass procedure is discussed in detail in [23] including results from subjective evaluations which rated the new procedure alongside NLD and phase vocoder approaches over a wide range of musical genres. The hybrid approach showed less sensitivity to input content and was therefore chosen as the virtual-bass procedure for the work presented in the remainder of this paper.

5. VIRTUAL-BASS ROOM-MODE CORRECTION

Virtual bass can be used as a supplemental component within the parametric equalization structure to help suppress the most problematic room modes but without losing crucial audio information, as alluded to in the previous section. When used as a standalone application, virtual bass can often produce an artificial sounding effect which can detract from a natural listening experience. These applications are often targeted towards bandwidth extension of restricted loudspeakers where there are few alternatives to achieve strong low-frequency perception.

However, if the virtual-bass effect was limited to a narrow-band application many of the artificial artifacts may be masked by the surrounding frequencies of physically reproduced energy. The narrow band(s) removed from the signal through parametric equalization could then be reinforced psychoacoustically with the narrow-band virtual-bass procedure to maintain any information present within these frequency bins. This room-mode correction procedure is highlighted in Figure 5.1.

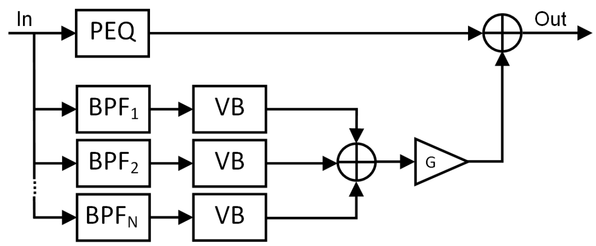

Figure 5.1 Virtual-bass room correction procedure

In Figure 5.1 the unprocessed signal is sent via the parametric equalization (PEQ) routine with notch filters centered at the most problematic modes within the room (generally three to five modes). At the same time, the unprocessed signal is sent in parallel through N bandpass filters (BPF_x) centered at each target frequency and is then run through the virtual-bass procedure (VB), as detailed in Section 4 of this paper. All virtual-bass outputs are then summed with appropriate gain (G) applied to the resulting signal. The final virtual-bass signal is then recombined with the parametric equalization output to give the fully processed signal to be sent through the remainder of the signal chain.

5.1. Subjective evaluation procedure

Since the virtual-bass effect occurs within the human hearing mechanism and the brain it is necessary to subjectively evaluate the proposed virtual-bass correction procedure. Since the parametric equalization routine will remove various narrow bands from the physically reproduced signal, the resulting signal is expected to have slightly reduced low-frequency impact, but with the aspiration of maintaining high fidelity with minimal obvious artifacts due to the harmonic distortion from the virtual-bass effect.

Tests were carried out in the Audio Research Laboratory listening room at the University of Essex with room dimensions of 6.0 m x 5.8 m x 2.8 m. The listening room sound system consisted of two sealed box subwoofers placed on the ground at wall midpoints to the left and right of the listening positions along with two left and right main stereo loudspeakers. Two adjacent listening locations were chosen where the right location received strong low-frequency energy due to the close proximity of many mode antinodes while the left location received little energy due to nodal placement.

The four strongest room modes (41, 58, 67 and 84 Hz) were chosen as targets based on FDTD simulations using proprietary software [24, 25] and confirmed with previous room measurements.

Ten high-fidelity musical recordings were chosen for the tests, each from a distinct musical genre as detailed in Table 5.1.

Genre	*Artist*	*Song*
Classical	Frank Zappa	*Dog Breath Variations*
Jazz	The Bad Plus	*Big Eater*
Blues	Bernard Allison	*Mean Town Blues*
Rock	Jeff Beck	*There's No Other Me*
Pop	Robert Randolph	*Diane*
Vocals	The Blind Boys of Alabama	*These Bones Gwine Rise Again*
Reggae	Bob Marley	*Get Up Stand Up*
Country	The Drive-By Truckers	*Bob*
Folk	Alison Breitman	*Tenafly*
Hard Rock	Audioslave	*The Worm*

Table 5.1 Musical selections by genre

Subjects were presented first with the unprocessed musical sample and instructed to move between the two seats both to judge overall sound quality and low-frequency variance between the two locations. Sound quality was rated on a one hundred point scale with one hundred being the best possible score. Low-frequency variance was also rated on a one hundred point scale with one-hundred representing the significant variance and zero representing no noticeable variance.

The test subjects were first presented with a list of the musical selections and asked to choose three based on their musical preferences. Next, a sample track was played, allowing subjects to become accustomed to the test procedure. Each unprocessed/processed clip pair was then played until the subject had assigned ratings. The entire test generally required fifteen to twenty minutes to complete. The signal processing procedure was not revealed to the listeners to avoid any possible biasing.

5.2. Subjective evaluation results

The test subjects were composed of fifteen males and six females ranging in age from twenty-three to sixty-three years old. Each subject completed the test during independent sessions. The subjective evaluations results are presented in Figure 5.2.

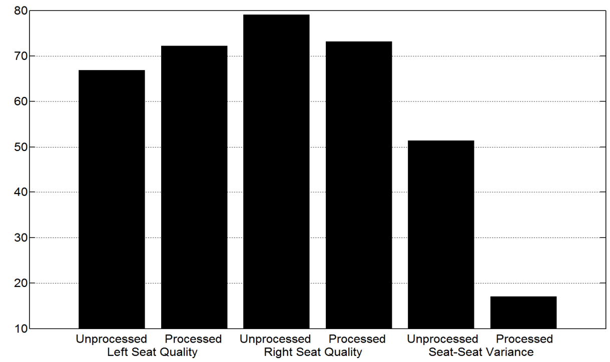

Figure 5.2 Virtual-bass low-frequency room mode correction subjective evaluation results

The subjective evaluation results indicate that the right seat generally received quality ratings in the "good" range which can be largely attributed to strong low-frequency presence. The left seat, on the other hand, received quality ratings in the "fair" range with subjects generally commenting that they sensed the left seat lacked certain musical information. These differences in quality ratings are reflected in the unprocessed seat-to-seat variance ratings in the "moderate" range.

After the virtual-bass processing, through, the subjective ratings show a noticeable shift. The right seat, while rated "good" unprocessed, has decreased by approximately five points to the lower bound of the "good" range. The left seat received "fair" quality ratings unprocessed, but has increased into the "good" range. The left and right seats' processed ratings are within two percent of each other which is strongly reflected in the processed seat-to-seat variance ratings in the upper portion of the "not noticeable" range.

The subjective evaluations have shown that virtual-bass room correction can provide a reasonable amount of spatial variance reduction between seats. The compromise is that seats with naturally superb responses tend to experience slight decreases in fidelity in order to increase the fidelity of naturally poor sounding seats. The virtual bass ensures that all musical information present in the unprocessed signals is perceptually maintained in the processed signals.

6. CHAMELEON SUBWOOFER ARRAY APPLICATIONS

The impetus behind the development of this virtual-bass room correction procedure is the chameleon subwoofer array (CSA) room-correction system, first proposed in [1] and elaborated on in [2]. CSA room correction operates using subwoofers with multiple drive units, allowing for frequency-dependent polar patterns giving an increased number of degrees of freedom for system control. Measurements are taken at key target points within the listening area and complex correction filter coefficients are calculated for each drive unit based on a set of target frequency responses (Figure 6.1).

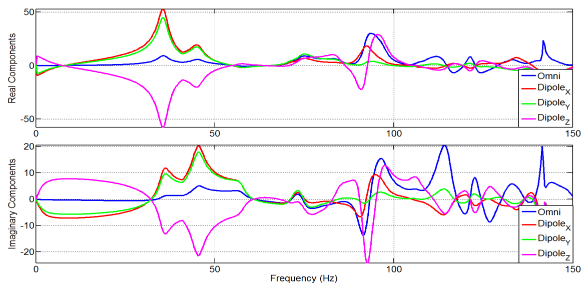

Figure 6.1 Calculated CSA coefficients for 4-unit array

CSA systems generally exhibit ideal behavior when the multi-driver subwoofers are placed based on passive placement configurations that give low spatial variance. A four subwoofer CSA system, for example, with units at wall midpoints can achieve spatial variance reduction of over 90% (Figure 6.2).

The drawback to CSA correction is that often a certain configuration will have difficulties correcting for one or two narrow frequency bands, as seen in Figure 6.1 (40, 85 and 140 Hz). The higher filter coefficients will limit the overall efficiency of the corrected system.

This efficiency problem is naturally addressed with the virtual-bass correction procedure, whereby the few frequencies that are difficult/impossible to correct for can be entirely eliminated within the correction system algorithm and replaced by virtual-bass components. This will cause significantly lower filter coefficients, resulting in a more efficient correction system.

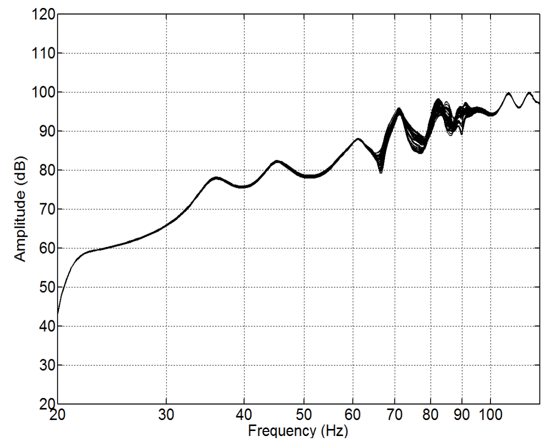

Figure 6.2 Simulated frequency responses over a listening area with a subwoofer system consisting of a CSA unit at each wall midpoint
(uncorrected responses shown in Figure 2.1)

7. CONCLUSIONS

A wide-area low-frequency room-mode correction method based on a combination of parametric equalization and the virtual-bass effect has been presented as a simple solution for reducing the negative effects of room modes (primarily high spatial variance) while maintaining all input signal information and reasonable fidelity.

Correction method complexity is reduced by not relying on a purely physical correction system, but using a hybrid physical/psychoacoustical system which can be applied to nearly any sound system. This technique was developed primarily out of the need to address the few narrow frequency bands that prove difficult to correct for using a chameleon subwoofer array (CSA).

The virtual-bass mode correction algorithm was subjectively tested by comparing overall sound quality between unprocessed and processed signals as well as seat-to-seat variance between the two versions. Results clearly show a sharp drop in spatial variance due to the virtual-bass procedure with a decreased difference between sound quality ratings at each seat.

While this correction procedure was developed as an add-on to a specific low-frequency correction method, it can easily be implemented in any system in a similar manner to parametric equalization.

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