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Kick-drum signal acquisition, isolation and reinforcement optimization in live sound

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ABSTRACT

A critical requirement for popular music in live-sound applications is the achievement of a robust kick-drum sound presented to the audience and the drummer while simultaneously achieving a workable degree of acoustic isolation for other on-stage musicians. Routinely a transparent wall is placed in parallel to the kick-drum heads to attenuate sound from the drummer’s monitor loudspeakers, although this can cause sound quality impairment from comb-filter interference. Practical optimization techniques are explored, embracing microphone selection and placement (including multiple microphones in combination), isolation-wall location, drum-monitor electronic delay and echo cancellation. A system analysis is presented augmented by real-world measurements and relevant simulations using a bespoke Finite-Difference Time-Domain (FDTD) algorithm.

1. INTRODUCTION

The importance of a robust kick-drum sound for popular music sound reinforcement cannot be understated. The kick-drum emphasizes the beat of a song while providing great aural and physical impact. The significance of this impact necessitates the drum to be accurately captured by the kick-drum microphone(s), with minimal interference from nearby system components.

Many concert venues produce strong late-arriving reflections which can cause confusion at the drummer position while attempting to maintain a beat. These reflections can stem from the ceiling, rear wall or balconies in indoor venues and from nearby buildings in outdoor venues. The arrival time of these reflections can be multiple hundreds of milliseconds after the initial sound.

To alleviate the confusion at the drummer position, a small loudspeaker system is routinely placed directly behind or to the side of the drummer. This system is termed a drum-fill. The drum-fill can consist of any combination of monitor system components including in-ear monitors, monitor wedges and subwoofers. This system provides the drummer with direct monitoring capabilities of any instrument on stage and overpowers the late-arriving reflections, making the drummer's job much easier.

Unfortunately, the presence of this monitoring system can adversely affect the signal received at the kick-drum microphone(s), especially when a drum subwoofer is utilized. The signal propagating from the subwoofer contains both electronic delay (introduced from the mixing console and/or processing units) as well as acoustical propagation delay from the subwoofer to the microphone. The delay between the arrival of the direct and subwoofer signals results in strong comb-filtering [1], thus diminishing the accuracy of the reinforced kick-drum signal.

The problem is complicated by the requirement of an isolation shield directly in front of the drums in order to insulate the drums from the rest of the stage, thereby reducing stage sound pressure levels. The reflections from the shield will aggravate the already-troublesome comb-filtering at the microphone.

It is important that both the drummer and microphone position responses are addressed, as the response at the drummer location can vary with that at the microphone. A severely comb-filtered signal at the drummer may cause the monitor engineer controlling the drum-fill to boost the amplitude at key frequency bands, potentially degrading the signal at the microphone.

It is therefore important to consider both the microphone and drummer positions when optimizing kick-drum reinforcement. This paper will demonstrate both frequency-independent and physical configuration optimization techniques achievable with existing industry-standard hardware as well as proposing frequency-dependent optimization concepts, including echo-cancellation.

The system optimization sections will be preceded by a discussion on the history of kick-drum reinforcement in live sound and followed by the authors' recommendations for future work on the subject.

2. KICK-DRUM REINFORCEMENT HISTORY

The first appearance in the public eye of a kick-drum microphone in the context of a large-scale concert may be the film "Monterey Pop." The microphone can be seen in photos of Mitch Mitchell's drum kit during Jimi Hendrix's famous "burning guitar" sequence. This took place in June, 1967.

This may have been specifically for recording purposes, rather than sound reinforcement, as Wally Heider's mobile recording service had been hired to document the show for film and later record release. The dynamic vocal microphones are Shure SM56s with foam windscreens, while the kick-drum microphone appears to be the similar Shure SM58, released in 1966 for the studio market [2]. The SM58 is essentially the same microphone as the SM56, but with a built in pop filter and no shock mount. Placement was approximately 45 cm from the front head on center.

Photos from April, 1968 of the Who at the Fillmore East in New York City show a microphone placed between Keith Moon's twin kick-drums, although this may have been for the snare drum. The unintended consequence of amplifying the double kick-drums did not stick with the Who's sound technician, however, as later photos show no kick-drum microphone.

At the Woodstock Music and Arts Fair in August, 1969 Santana drummer Mike Shrieve's kick-drum can be seen with a Shure Unisphere microphone (also known as a 565D-LC). This was a dynamic microphone, already available as the ubiquitous model SM58, repackaged in polished silver finish with a chrome plated ball windscreen for the live rock market.

It can be assumed that a large number of 565D-LC style microphones were available at Woodstock, as they can be seen used on a wide range of instruments including: vocals, drums, percussion, bass guitar and organ. The kick-drum microphone is about 20 cm off the front head at a 45° downwards angle. This position may have been derived when one of the stage hands took a vocal microphone on a boom stand, which typically would have been pointing up at a singer's mouth, and turned the boom over to point in the general direction of the kick-drum.

By 1975, concert sound reinforcement had advanced to include a specialty microphone for the kick-drum. British bands traveled with their own dynamic microphone, which included a large square windscreen and an oversized diaphragm, supported internally on springs. This was the AKG D-12 and was marketed in the United States under the Norelco brand name. Strangely, it was shown in advertisements as a female vocal microphone.

The AKG D-12 produced a very deep sound that was associated with British recordings of the time period. The microphone was positioned close to the front head and rim of the kick-drum. In the era of disco music, with its quarter note kick-drum patterns, more emphasis began to be placed on the kick-drum over the previous centerpiece, the snare drum.

As disco music faded in popularity and sound reinforcement systems became more sophisticated with multi-band crossovers, separate subwoofers and larger power amplifiers, the kick-drum's importance continued to grow at a disproportionate rate to everything else on the concert stage. When punk music became popular, a second microphone was added to the inside of the drum for a stronger transient response known as "attack".

More sophisticated (and expensive) microphones originally used in studios began to appear on stage, including the Electro Voice RE20, which was a mainstay of radio announcers for its "Variable-D" (no proximity effect) design, the Sennheiser 421, which was a popular saxophone microphone in Europe and a staple of German television on just about every instrument, and the newly created Crown PZM, a style of condenser boundary microphone that was mounted to a flat plate and could be placed directly inside a kick-drum.

By the late 1980's even the most expensive microphones such as the Neumann U-87, which was usually reserved for Frank Sinatra or Barbara Streisand's studio vocal microphone, had found its way into Charlie Watts' (Rolling Stones) kick-drum.

In the 21st century, new microphones combine the two distinct technologies, condenser and dynamic, in one unit such as the Audio Technica ATM250 DE which allows both elements to be located as close as physically possible, picking up the source at virtually the same place and time.

3. CONVENTIONAL SYSTEM OPTIMIZATION

Current industry-standard hardware (digital mixing consoles, signal processing units, etc.) allows for straightforward signal manipulation which can be used to enhance the response at the kick-drum microphone and drummer position. Physical configuration adjustments provide further control of the overall system response including: kick-drum internal damping, front drum head choice and microphone/ isolation shield placement.

A common kick-drum/microphone/drum subwoofer setup is used in this section to illustrate the advantages/disadvantages of various approaches using real-world measurements. This initial setup excludes the isolation shield from the system.

All measurements utilize a 51 cm diameter (46 cm depth) kick-drum and a dual-18" Gand Concert Sound GSL subwoofer situated directly behind the drummer. A Shure Beta 52A microphone (B52A) was located in front of the drum with a Shure Beta 91 microphone (B91) placed inside the drum. An Audix TM1 measurement microphone (TM1) was placed at the drummers head to record the drummer position response. All signals were routed through a Yamaha LS-9 16-channel digital mixing console and digitally recorded with a sample rate of 48 kHz (16 bit). The measurement configuration is shown in Figure 3.1.

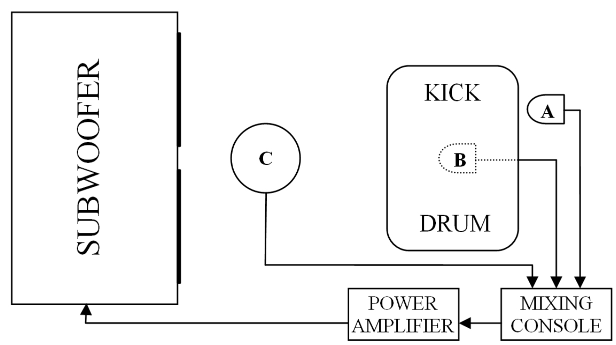

Figure 3.1 Measurement setup for kick-drum and drum subwoofer system (A = B52A, B = B91, C = TM1)

All measurements/simulations in this section were judged against a measured "dry" kick-drum signal (Figure 3.2), using the kick-drum with a pillow placed inside for damping purposes and a front drum head with an offset hole. The B52A microphone was placed 0.07 m in front of said hole, while the B91 microphone was located 0.25 m from the front head, inside the drum. The TM1 microphone was located just above the drummer's head; a horizontal distance of 1.05 m from the front head. The subwoofer and isolation shield were not included in the system for the "dry" measurements.

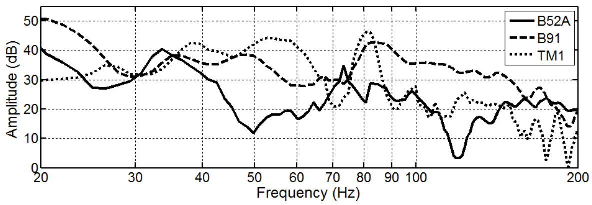

Figure 3.2 Reference kick-drum frequency responses (no subwoofer/isolation shield)

Microphone/subwoofer/isolation shield configurations will be analyzed based on their deviation from the reference responses. The closer the microphone response is to its reference signal, the better. The drummer response is also ideally expected to match its reference response, although at a higher amplitude due to the reinforcement requirements. As this particular project concentrates on the effects of the drum-fill subwoofer on the reinforcement system, spectral analysis will be performed up to 200 Hz. The reasoning behind exclusive focus on the subwoofer band will be detailed in Section 3.2.

3.1. Physical drum configuration

The first important decision when configuring a kick-drum is whether the front head will contain a circular hole. The addition of the hole is analogous to adding a port to a subwoofer, where the hole/port will affect the frequency response of the enclosure. The measured effects of adding a hole to the front head are illustrated in Figure 3.3.

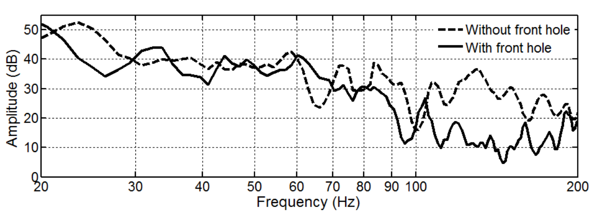

Figure 3.3 B52A measured frequency responses with (solid line) and without (dotted line) a front head hole (no subwoofer/isolation shield/internal damping)

The addition of the hole to the front head causes an observable reduction in magnitude response from 100 – 200 Hz. This can be attributed to the fact that the front head no longer exhibits uniform higher-order resonances due to the discontinuity on its surface, leading to less coupling between the two heads [3].

This conclusion can be supported by examining the B91 microphone measurement from inside the drum, where changes in resonant behavior of the front head should be much less pronounced as the internal microphone will largely be influenced by the internal modal behavior of the drum and the direct sound (Figure 3.4).

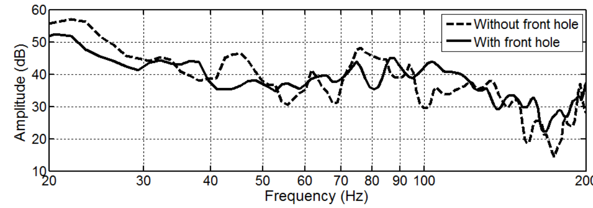

Figure 3.4 B91 measured frequency responses with (solid line) and without (dotted line) a front head hole (no subwoofer/isolation shield/internal damping)

This comparison confirms that the reduction from 100 – 200 Hz is largely due to changes in the resonant behavior of the front head, as this effect is minimally noticed at the internal microphone location.

For pop/rock music applications, excessive ringing at the outside microphone location can be detrimental to the overall kick-drum sound as it deemphasizes the ideally sharp drum attack. In jazz music, however, the boost in the low-mid band may be desirable since the kick-drum is used more for musicality than for pure impact.

In addition to the hole on the front kick-drum head, internal damping is often added to further reduce any ringing to sharpen the impact of the kick-drum. This is commonly achieved by placing a pillow or thick blanket inside the drum (Figure 3.5).

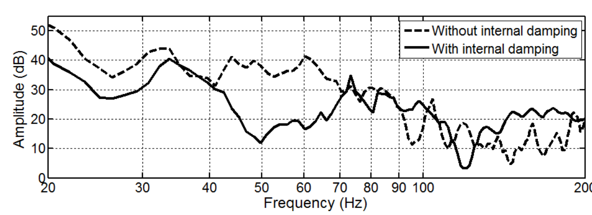

Figure 3.5 B52A measured frequency responses with (solid line) and without (dotted line) internal damping (no subwoofer/isolation shield/internal damping)

The frequency response comparison between the damped and undamped drum shows a large reduction in amplitude between 40 – 70 Hz. While this may seem undesirable, the advantage of utilizing damping becomes clear by examining the change in transient response (Figure 3.6).

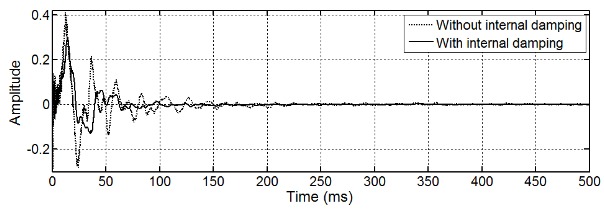

Figure 3.6 B52A measured responses with (solid line) and without (dotted line) internal kick-drum damping (no subwoofer/isolation shield)

The damped drum measurement shows a much shorter kick-drum attack, with minimal ringing. This variety of kick-drum response is considered desirable among pop/rock music live sound engineers since it delivers a strong atonal impact.

3.2. Kick-drum reinforcement

Sound reinforcement at the drummer position can be achieved with a drum-fill. Based the authors' collective experience, most of today's touring drummers utilize in-ear monitor systems (IEM); only utilizing a subwoofer in the drum-fill to achieve the physical impact of the kick-drum. IEM systems provide reinforcement over the remaining bandwidth. This trend provides justification for only addressing the low-frequency band (20 – 200 Hz) for this project.

Reinforcing the kick-drum signal in such close proximity to the drum itself poses a number of issues. Firstly, this opens the possibility of system instability if the positive feedback level between the subwoofer and microphone meets or exceeds unity, causing runaway levels at one or more frequencies. Secondly, the signal emanating from the subwoofer will arrive at the kick-drum microphone(s) as a delayed and frequency shaped version of the direct signal, resulting in comb-filtering.

While runaway feedback can be controlled with an equalizer and/or noise gate, comb-filtering effects can be much more difficult to control. Even if the system can be arranged so that the microphone receives a signal closely resembling the dry kick-drum signal, the drummer is located a distance from that microphone and likely will not experience the same benefits as the microphone location. This is akin to issues with single-point equalization where optimizing a single listening location often worsens other listening locations [4].

A series of measurements were taken utilizing electronic time delay applied to the drum subwoofer signal, as well as simple polarity reversals, in an attempt to limit comb-filtering. Measurements were taken using the configuration in Figure 3.1, with the B52A microphone feeding the subwoofer. Frequency responses were calculated from the measurements and the mean absolute error (MAE) in response from the reference response in Figure 3.2 was determined for both the drummer and microphone positions. The microphone and drummer MAE values were then averaged to give a single metric representing system performance. Lower MAE deviations indicate high correlation between microphone, drummer and reference responses (Figure 3.7).

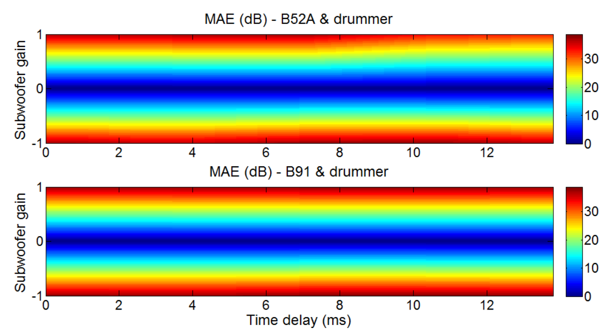

Figure 3.7 Mean absolute error (MAE) measurements between the drummer and microphone references for variable subwoofer gain and electronic time delay

The subwoofer delay versus gain MAE measurements indicate that although the delay may alter the frequency at which comb-filtering occurs, no amount of delay will cause a significant reduction in MAE of the frequency response at the microphone and drummer. The results in Figure 3.7 indicate that only turning off the subwoofer will achieve a flat input-output response, which defeats the purpose of having a reinforcement system in the first place.

A more detailed analysis can be performed using the absolute error (AE) between the reference and drummer/microphone responses, performed at a fixed subwoofer gain (Figure 3.8).

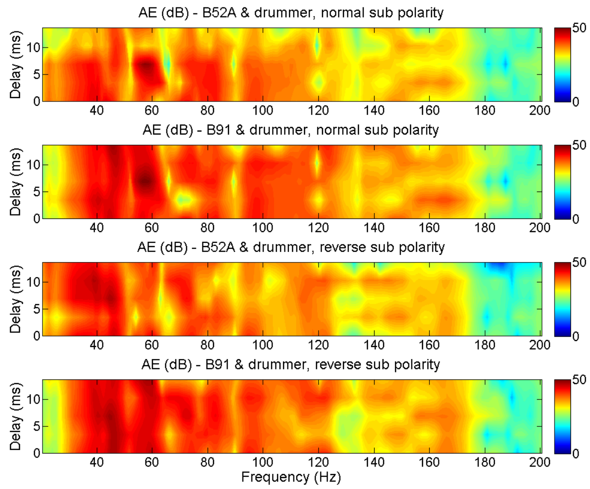

Figure 3.8 Absolute error (AE) measurements between the drummer and microphone references for variable electronic time delay versus frequency

The absolute error plots validate the claim that although the problematic frequencies shift with varying time delay, the mean absolute error is largely unchanged with electronic time delay and/or polarity reversal.

Interestingly, when the subwoofer was set to normal polarity with 6.90 ms of electronic delay, the drummer position exhibited a subjectively pleasing response. The top plot in Figure 3.8 indicates a strong deviation from flat at 60 Hz for this configuration. This likely manifested itself as a strong boost in signal amplitude at 60 Hz. While this may be acceptable for the drummer, the front-of-house (FOH) engineer would likely receive a signal strongly colored by the drum monitor system.

3.3. Drum kit isolation

The electronic delay and polarity tests did not consider any isolation requirements. Isolation of sound from the drum kit is commonly required as it is desirable to keep stage sound pressure levels to a minimum, providing a much safer acoustical environment for musicians and stage personnel while also limiting the output requirements of the stage monitor system.

The primary means of drum isolation employs a clear shield placed directly in front of the drum kit. While the isolation shield effectively limits the drum sound contamination over the rest of the stage, it provides another early refection into the kick-drum microphone and drummer position causing additional comb-filtering.

The MAE analysis performed in Section 3.2 can be repeated using measurements taken with the isolation shield at different distances from the front kick-drum head (Figure 3.9). Only the B52A microphone measurements were used for this analysis, where similar behavior can be assumed for the B91 microphone.

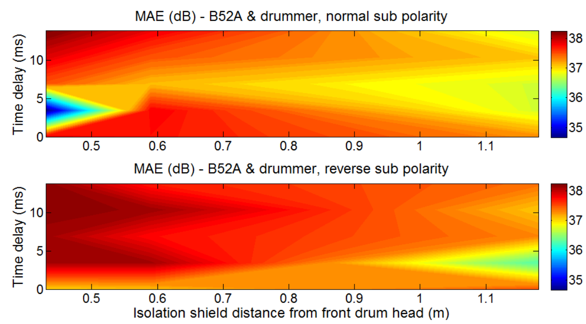

Figure 3.9 Mean absolute error (MAE) measurements between the drummer and microphones for variable electronic time delay and isolation shield position

The time delay versus isolation shield location shows improved performance for certain combinations, although these improvements only reduce the MAE by a few decibels. While these simple methods of system adjustment can give subjectively pleasing results at the drummer location, they cannot predictably guarantee an uncolored response at the microphone location.

Systems limited to simple configuration adjustments ultimately have to make a compromise: either adjust the system according to the drummer's preferences or the FOH engineer's preferences. It can be argued that the system should be (and often is) configured for the drummer and the FOH engineer can equalize the signal appropriately, but this is likely to smear the transient response due to phase shift introduced by multiple bands of parametric equalization.

3.4. Practical considerations

In the constantly changing live sound acoustical environment, techniques and settings used one day may not always yield the same results the next day. Stage and room acoustics can vary from venue to venue, requiring different equalization to achieve similar responses at the drummer position. These drum-fill equalization changes may negatively affect the sound at FOH as well, due to interference at the microphone.

Drum riser and stage construction will also have an effect on the way the drum-fill will interact with the kick-drum microphones. It is often beneficial to keep the drum-fill off the drum riser to minimize mechanical coupling, although stage size can prevent this option, leaving few options for speaker placement.

Drum risers also restrict the placement of the isolation shield, commonly providing less than 0.5 m of space in front of the drums for the shield. Also, at many outdoor events an isolation shield cannot be used due to strong winds. Being aware of all available options and their positive and negative effects on the system responses allows for the best choice to be made.

4. COMPREHENSIVE OPTIMIZATION

Manual configuration processes, detailed in the previous section, are imprecise in nature and can be time consuming; an issue that cannot be overlooked considering the tight production schedules in live sound. It is therefore beneficial to explore automatic, environmentally-adaptive system optimization which can reliably and predictably deliver the desired response at both the microphone and drummer.

4.1. System configuration

In order to meet all system specifications at both the microphone and drummer position, it is necessary to provide a minimum of two degrees of freedom towards correction. The first degree of freedom required is frequency-dependent subwoofer equalization. The equalization ideally achieves the desired response at the drummer location. This response can be anything within reason, but for this investigation a targeted flat input-output response will be assumed.

An additional degree of freedom must be implemented to make the kick-drum microphone insensitive to both the signal from the subwoofer and the reflections off the isolation shield. This can be achieved via a straightforward application of echo-cancellation [5] where an equalization filter is applied to the microphone output to suppress all received sound except the direct kick-drum signal.

The system topology (Figure 4.1) includes electronic and acoustical delay times as well as numerous gain and transfer function variables, all used to calculate the theoretical input-output relationships for the microphone and drummer positions.

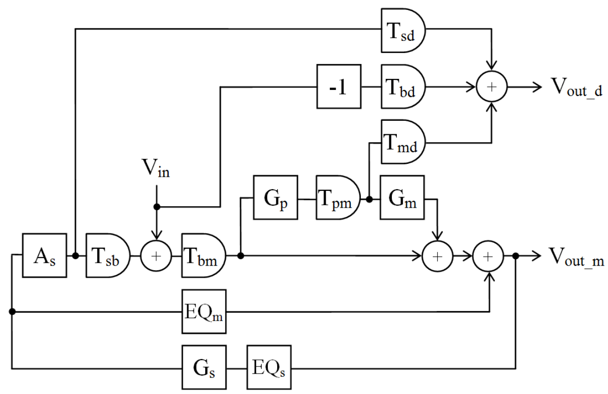

Figure 4.1 Comprehensive optimization system configuration for kick-drum/subwoofer/isolation shield setup utilizing subwoofer EQ and echo-cancellation

The signal flow in Figure 4.1 begins at the beater head of the kick-drum (V_{in}). The first path of the direct sound flows to the microphone with a corresponding beater to microphone propagation delay, T_{bm}. The direct sound is also reflected off the isolation shield (if present). This is modeled by applying the reflection factor of the shield, G_p, along with the round trip microphone to shield propagation delay, T_{pm}; the "p" stands for poly(methyl methacrylate) or PMMA, the most common material used for isolation shields. A microphone gain, G_m, is applied to the reflection to approximate the 180° polar response of the microphone. This reflection is then added to the direct sound path of the microphone.

The microphone output signal, V_{out_m}, is fed through two processing stages. The first, EQ_s, filters the signal according to the frequency response requirements at the drummer position. Next, signal gain is applied with G_s according to the power amplifier settings. The transfer function of the subwoofer is approximated by A_s and summed with the direct input signal including the subwoofer to beater propagation delay (T_{sb}).

Echo-cancellation is applied to the microphone output with a feedforward path containing the microphone equalization filter, EQ_m. This signal is taken from the processed subwoofer signal after G_s has been applied.

The drummer location input-output relationship, V_{out_d}, is determined by a sum of three signals. First, the inverted direct signal from the beater is taken with a beater to drummer propagation delay, T_{bd}. The kick-drum beater strikes the drum head, forcing it away from the drummer, creating a negative pressure wave thus necessitating the inverted direct signal at the drummer.

The second signal considered at the drummer position is the subwoofer signal with the appropriate subwoofer to drummer propagation delay (T_{sd}). Lastly, the reflection from the isolation shield is included, taken from the microphone location with the microphone to drummer propagation delay (T_{md}).

Processing delay, which is specified at a maximum of 2 ms for the LS9 console [6], was not included in this model, but is be essential for practical implementations of this technique.

All calculations for this system were carried out assuming a recursive nature to the reflections and subsequent signals received at the microphone and transmitted through the drum subwoofer.

4.2. Microphone response echo-cancellation

The first system unknown, EQ_m, corresponds to the echo-cancelation at the microphone. The microphone output can be found by analyzing the system configuration (Equation 4.1) and simplifying/solving to give the microphone position transfer function (Equation 4.2).

For full echo-cancellation, the denominator of Equation 4.2 must reduce to the form of the numerator, with the exception of the beater to microphone propagation delay term (Equation 4.3). Solving for EQ_m yields the first optimization equation (Equation 4.4). The solution for EQ_m correctly indicates that the properties of EQ_m are dependent on EQ_s, the subwoofer equalization function, which must be addressed for the echo-cancellation procedure to be accurate.

$$V_{out_m} = \left(1 + G_p G_m e^{-j\omega T_{pm}}\right)\left(V_{in} + V_{out_m} EQ_s G_s A_s e^{-jwT_{sb}}\right)e^{-jwT_{bm}} + V_{out_m} EQ_s EQ_m G_s \tag{4.1}$$

$$\frac{V_{out_m}}{V_{in}} = \frac{\left(1 + G_p G_m e^{-j\omega T_{pm}}\right)e^{-j\omega T_{bm}}}{1 - EQ_s G_s\left(A_s e^{-j\omega(T_{sb}+T_{bm})} + G_p G_m A_s e^{-j\omega\left(T_{sb}+T_{pm}+T_{bm}\right)} + EQ_m\right)} \tag{4.2}$$

$$-EQ_s G_s\left(A_s e^{-j\omega(T_{sb}+T_{bm})} + G_p G_m A_s e^{-j\omega\left(T_{sb}+T_{pm}+T_{bm}\right)} + EQ_m\right) = G_p G_m e^{-j\omega T_{pm}} \tag{4.3}$$

$$EQ_m = -e^{-j\omega T_{pm}}\left(\frac{G_p G_m}{EQ_s G_s} + G_p G_m e^{-j\omega(T_{sb}+T_{bm})} + A_s e^{-j\omega\left(T_{sb}+T_{bm}-T_{pm}\right)}\right) \tag{4.4}$$

4.3. Drummer position equalization

As the primary function of the subwoofer is to serve the needs of the drummer, the input-output relationship at the drummer position must be calculated (Equation 4.5) and used to solve for the transfer function (Equation 4.6). This relationship can be used to find the solution for the subwoofer equalization coefficients, EQ_s.

Assuming the microphone position has had echo-cancellation applied (V_{out_m} is a delayed version of V_{in}) the drummer input-output relationship can be simplified as shown in Equation 4.7. The second term in Equation 4.7 can be used to solve for EQ_s, noting that V_{out_d} must match a delayed and inverted input signal plus the required gain, G_d (Equation 4.8). This gives a direct solution for the subwoofer equalization (Equation 4.9).

The set of equations developed in this section can be used to predict the performance of the system and to generate filters for practical system performance testing. The following sections will compare the mathematically predicted behavior to simulation results using filters generated by applying the proposed equations.

$$V_{out\_d} = V_{in}\left(G_p e^{-j\omega(T_{pm}+T_{md}+T_{bm})} - e^{-j\omega T_{bd}}\right) + V_{out\_m} EQ_s G_s A_s \left(e^{-j\omega T_{sd}} + G_p e^{-j\omega(T_{sb}+T_{pm}+T_{md}+T_{bm})}\right) \quad (4.5)$$

$$\frac{V_{out\_d}}{V_{in}} = \frac{V_{out\_m}}{V_{in}} EQ_s G_s A_s \left(e^{-j\omega T_{sd}} + G_p e^{-j\omega(T_{sb}+T_{pm}+T_{md}+T_{bm})}\right) + G_p e^{-j\omega(T_{pm}+T_{md}+T_{bm})} - e^{-j\omega T_{bd}} \quad (4.6)$$

$$V_{out\_d} = V_{in}\left[-e^{-j\omega T_{bd}} + e^{-j\omega T_{bm}}\left(G_p e^{-j\omega(T_{pm}+T_{md})} + EQ_s G_s A_s \left(e^{-j\omega T_{sd}} + G_p e^{-j\omega(T_{sb}+T_{pm}+T_{md}+T_{bm})}\right)\right)\right] \quad (4.7)$$

$$-G_d e^{-j\omega T_{bd}} = e^{-j\omega T_{bm}}\left(G_p e^{-j\omega(T_{pm}+T_{md})} + EQ_s G_s A_s \left(e^{-j\omega T_{sd}} + G_p e^{-j\omega(T_{sb}+T_{pm}+T_{md}+T_{bm})}\right)\right) \quad (4.8)$$

$$EQ_s = \frac{-1}{G_s A_s}\left(\frac{G_d e^{-j\omega(T_{bd}+T_{bm})} + G_p e^{-j\omega(T_{pm}+T_{md})}}{e^{-j\omega T_{sd}} + G_p e^{-j\omega(T_{sb}+T_{pm}+T_{md}+T_{bm})}}\right) \quad (4.9)$$

4.4. Theoretical performance

The above equations can be validated mathematically by generating filter coefficients using Equations 4.4 and 4.9 and then substituting into Equations 4.2 and 4.6 to calculate the theoretical transfer functions. If the filter coefficients have been calculated correctly, the corresponding responses should be perfectly flat with an additional gain present at the drummer position.

The modeled configuration matches that used for the measurements with the microphone, subwoofer, drummer and isolation shield placed 0.07 m, 1.4 m, 1.05 m and 0.59 m from the front kick-drum head, respectively. The 180° polar response, G_m, was set to 0.178, according to the B52A's specifications [7]. The subwoofer was set to deliver 6 dB of reinforcement and the isolation shield reflection coefficient was set at -0.5, taking sound dispersion into account.

The responses were first calculated with the subwoofer equalization, EQ_s, set to unity and the microphone equalization, EQ_m, turned off (set to zero) (Figure 4.2). Responses were plotted up to 500 Hz to best illustrate the comb filtering problems, where frequencies above 160 Hz are outside the subwoofer operating range, but nonetheless suffer from reflections off the isolation shield.

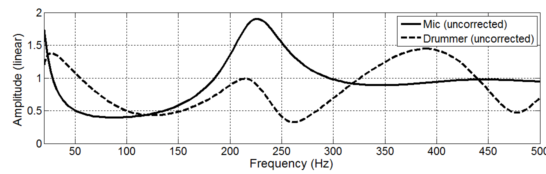

Figure 4.2 Theoretical uncorrected frequency responses at the microphone (solid line) and drummer (dotted line)

The uncorrected frequency responses support the problem at hand, where the microphone and drummer responses differ greatly. The microphone position response shows constructive interference around 200 – 250 Hz which is in agreement with the 1.47 m subwoofer to microphone distance, corresponding to one full wavelength at 233 Hz. Destructive interference seen in the microphone response occurs between 40 – 170 Hz, where one half-wavelength of 117 Hz fits into the subwoofer to microphone spacing. Similar analysis can be performed on the drummer position, noting that additional comb filtering can be seen due to a greater distance between the shield and drummer.

The benefits of the subwoofer equalization and microphone echo-cancellation can now be demonstrated by calculating optimal values for EQ_s and EQ_m (Figure 4.3). The target drummer level, G_d, was set for 6 dB sound pressure level (SPL) of reinforcement. The calculated complex filter coefficients giving the results in Figure 4.3 can be inspected to ensure they are realistic for practical implementations (Figure 4.4).

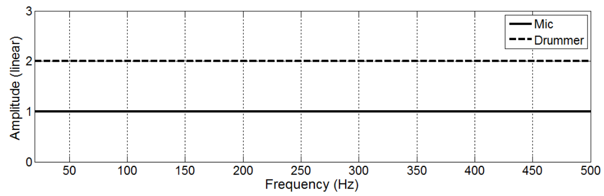

Figure 4.3 Theoretical corrected frequency responses at the microphone (solid line) and drummer (dotted line)

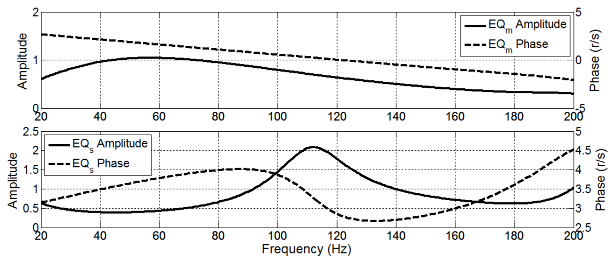

Figure 4.4 Amplitude (solid line) and phase (dotted line) values of the calculated microphone (top) and subwoofer (bottom) filter coefficients for $G_d = 1$

The coefficient values are reasonable for adequate handling by professional sound reinforcement systems and remain so for any required drummer position level, G_d (Figure 4.5).

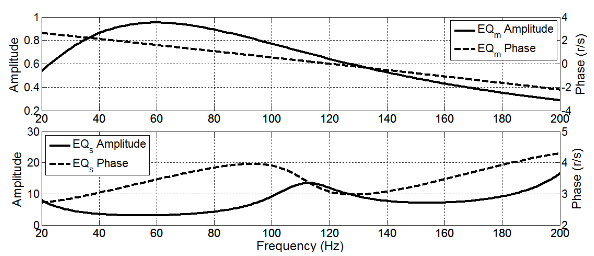

Figure 4.5 Amplitude (solid line) and phase (dotted line) values of the calculated microphone (top) and subwoofer (bottom) filter coefficients for $G_d = 9$

The input signal can now be filtered to give the resulting subwoofer signal, using Equation 4.10. A Dirac delta function was utilized for the input signal for illustrative purposes (Figure 4.6). The generated subwoofer signal shows components which effectively suppress reflections from the isolation shield.

$$\frac{V_{sub}}{V_{in}} = \frac{V_{out\_m}}{V_{in}} EQ_s G_s A_s \qquad (4.10)$$

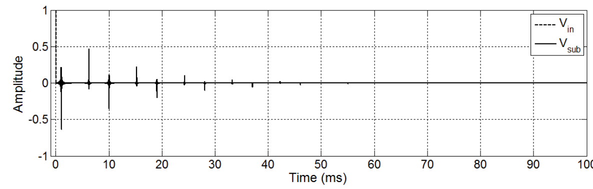

Figure 4.6 Generated subwoofer signal (solid line) due to a Dirac delta function input (dotted line at t = 0 ms)

System stability can be examined by calculating the required equalization filters (Equations 4.4 & 4.9) and then altering the microphone position before calculating the theoretical responses. In this case the microphone position was altered by ± 10 cm (Figure 4.7).

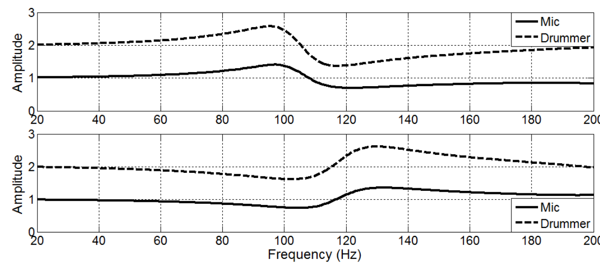

Figure 4.7 Corrected microphone and drummer frequency responses for a microphone position offset of 10 cm (top) and -10 cm (bottom)

The correction errors due to microphone offset are minimal, especially in the subwoofer band (20 – 120 Hz). It is not uncommon for the kick-drum microphone to move slightly during a performance, but usually no more than a few centimeters. Additionally, it is useful to examine the correction area for the drummer. Drummer position changes of 0.5 m towards/away from the kick-drum were tested (Figure 4.8).

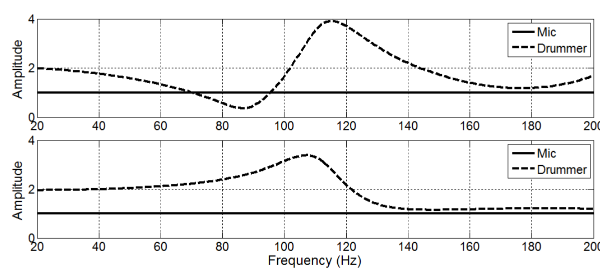

Figure 4.8 Corrected microphone and drummer frequency responses for a drummer position offset of 50 cm (top) and -50 cm (bottom)

As the drummer moves away from the kick-drum (Figure 4.8, top plot) the response drops in the subwoofer band. Alternatively, a boost is seen in the same band as the drummer moves closer to the drum (Figure 4.8, bottom plot). While the frequency response certainly is position dependent, it changes in a natural manner, whereby moving away from the drum causes quieter levels and vice versa. The microphone response is unaffected by drummer movement, as expected.

4.5. Simulated performance

Although the proposed system generates theoretically acceptable results, it is essential to test the system in a practical acoustical environment. In this case, a virtual acoustic environment was utilized. The Finite-Difference Time-Domain (FDTD) simulation toolbox [8, 9] was chosen due to its flexibility of simulation parameters, since both the subwoofer and microphone signals require different equalization.

The subwoofer and microphone correction signals were generated using the theoretical model and the dry B52A measurement (Figure 4.6). The signals are based on 6 dB SPL of reinforcement required at the drummer position. No isolation shield was simulated in this case.

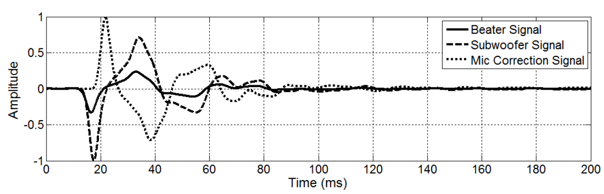

Figure 4.6 Generated subwoofer (dashed line) and microphone correction (dotted line) signals using the dry B52A kick-drum measurement (solid line)

The subwoofer signal shows just over double the amplitude of the dry kick-drum signal while the microphone correction signal is a time-delayed inverse of the subwoofer signal in order to completely suppress that signal as it arrives at the microphone.

These signals were fed into the FDTD simulation toolbox with an identical system setup as the theoretical model. The simulated signal at the microphone required post-processing to apply the echo-cancellation using the generated correction signal, while the drummer position simulation could be analyzed as is. Frequency responses were generated and compared to the dry kick-drum frequency response (Figure 4.7).

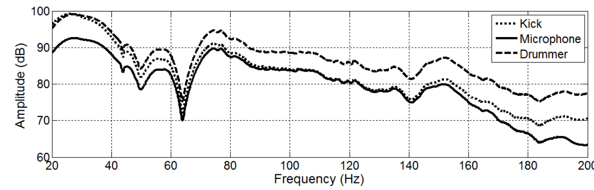

Figure 4.7 Simulated frequency responses for the drummer position (dashed line) and microphone (solid line) as compared to the dry B52A kick-drum measured frequency response (dotted line)

The results are as expected from 70 – 150 Hz, where the microphone response is nearly identical to that of the dry kick-drum signal and the drummer position response is similar, with 6 dB of gain. The errors outside this frequency range can be attributed to the nonlinear frequency response of the modeled subwoofer. At very low frequencies the subwoofer produces less output, therefore the echo-cancellation will over-correct, thus attenuating the response.

Errors in the high-frequency band (150 – 200 Hz) are the consequence of the internal crossover of the FDTD toolbox. To avoid aliasing, the toolbox splits .wav files into low- and high-frequency bands. This reduces the simulation response as frequencies approach the crossover point. The effect of this is similar to the subwoofer's high-pass characteristics, where the microphone signal is attenuated due to over-correction and carries over through the subwoofer to the drummer position. This issue is illuminated by determining the simulated transfer functions of both the microphone and drummer position (Figure 4.8).

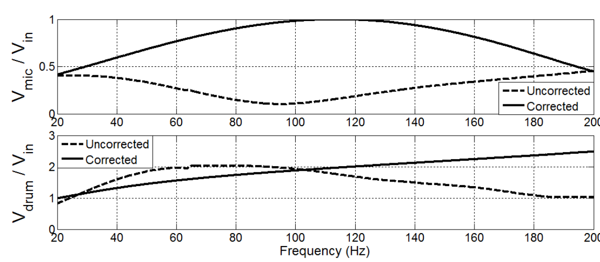

Figure 4.8 Simulated uncorrected (dotted line) and corrected (solid line) transfer functions for the microphone and drummer position

The transfer function calculations indicate that the microphone position benefits from the correction procedure as the destructive comb-filtering in the uncorrected system is eliminated in the corrected system. The uncorrected drummer position transfer function also highlights problematic interference patterns which largely disappear after correction.

The simulation results indicate that the theoretical model cannot be precisely transferred to a realistic situation. This leads the authors to conclude that a system operating using measurements must be implemented in future versions of this method in order to ensure its practicality.

5. CONCLUSIONS & FUTURE WORK

An overview has been presented of currently-available and conceptual methods of kick-drum reinforcement optimization, highlighting positive and negative aspects of each technique.

Optimization utilizing current industry-standard hardware comes down to a simple trial-and-error approach, where the optimal solution depends on the physical system configuration, including the acoustical properties of the drum, itself. The subwoofer signal can be adjusted in increments corresponding to the quarter-wavelength of the dominant kick-drum frequency for quick system tuning. It is also useful to designate one microphone for FOH and one for monitors, to reduce the impact of the monitoring system on the FOH system.

A conceptual optimization method has been proposed, whereby echo-cancellation is utilized to eliminate any effect the subwoofer has on the kick-drum microphone. This idea, although proven theoretically, has been problematic when implemented within a virtual space. This is largely due to inconsistencies between the theoretical and simulation models, which will likely be amplified in real-world scenarios.

The issues of correction by purely theoretical means leads to a number of possibilities concerning future work in this area. First, system performance may be greatly improved and stabilized if actual measurements are used for calibration, whereby system propagation delays and gains can be directly measured, followed by filter generation using the provided equations.

Alternatively, an adaptable system may be beneficial, where the correction routine is constantly updated by monitoring the received signals at the microphone(s). This system ideally would be less sensitive to temperature/humidity changes and to movement of the kick-drum microphone.

If one of these improved systems were to be successfully implemented, the authors can envision an expanded system that provides echo-cancellation for all drum-kit microphones, thus eliminating the emphasis on noise gates to control the signals passed to the mixing console from each microphone.

6. ACKNOWLEDGEMENTS

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