Motional Control of Loudspeakers

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Abstract—**Loudspeakers are imperfect electromechanical transducers that are normally sold to end-consumers without any form of compensation. Any non-linearities will therefore negatively affect sound quality. Reducing non-linearities requires elaborate mechanical constructions and tight tolerances, forcing the best-sounding systems to be very expensive. This paper details the use of Simulink to contrast the effectiveness of electronic feedforward and feedback techniques, discusses further mathematical techniques to improve compensation techniques, and introduces an intuitive methodology for loudspeaker enclosure and power amplifier design that the average audio enthusiast could follow. It is shown that a closed-loop controller is more effective at improving the bass response of a loudspeaker compared to a well-known feedforward controller topology.**

Index terms— **Feed-forward Control, Linkwitz Transform, Loudspeaker, LTSPICE, Motional Feedback, Simulink, State Estimation, State-Variable Feedback, Thiele-Small Parameters.**

I. INTRODUCTION

Generally, loudspeakers should be as small as possible, respond as linearly and across as much of the frequency spectrum as possible, to distort minimally, and to consume and emit power efficiently [1]. Practically, this is impossible to achieve without multiple-driver systems equipped with appropriate power and filter electronics, all of which drives the cost of such a system ever higher. Most end-consumers must therefore accept lower-quality sound reproduction.

The worst-offending type of loudspeaker is the subwoofer; since it must move larger volumes of air to reproduce bass sounds, their construction is larger, and the margin for error in operation increases. In the sub-70 Hz "sub-bass" range, performance is considered unreliable [2]. Improving the sound quality of a subwoofer can therefore be considered the biggest improvement to any multi-driver sound system.

One method to improve sub-bass sound quality is to use a feedforward controller. A filter can be introduced into the signal chain to manipulate line level audio to increase the magnitude of bas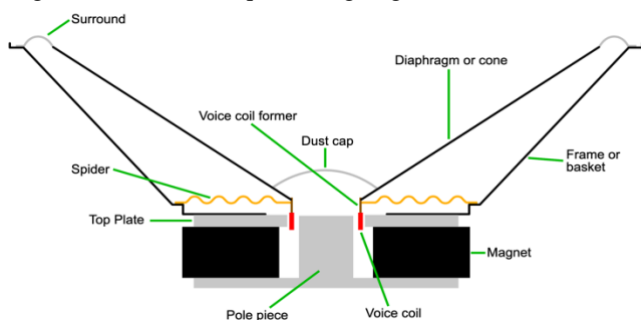s frequencies going into the subwoofer, which strengthens the sub-bass response of the subwoofer. Another method is to use an electronic controller as part of a feedback loop. Some parameters can be measured from the subwoofer as it operates and controlled against the input signal to try to make the subwoofer's performance track the input signal as closely as possible.

Using electronic controllers is much cheaper for manufacturers to design and build, as controllers can be simulated easily and quickly compared to iterating through mechanical designs, builds, and testing. Similar design techniques for the controllers can be used for other, similar electromechanical transducers.

The aim of this project is to determine which electronic controller is best able to linearize and boost the frequency response of a cheap subwoofer system, to prove that good sound quality can be achieved cheaply.

II. ECONOMIC, LEGAL, SOCIAL, ETHICAL AND ENVIRONMENTAL CONTEXT

The pursuit of high-quality audio reproduction is a niche interest, but the predicted growth of the loudspeaker market globally [3] will bring with it an increase in consumer understanding of the equipment being purchased. Consumers are ever less willing to settle with poor sound quality, especially with the proliferation of online reviewers giving prominence to better quality gear, particular that which sports some novel technology that helps it to stand out from competitors. The challenge for any audio company to stay relevant in an expanding market is thus becoming more difficult. The ideas in this, and similar, reports can help both consumers and companies; by introducing as-yet uncommercialized technology into a saturated market, at a fraction of the price of very expensive uncompensated speakers, consumers and reviewers can get access to better-quality loudspeakers for less money, and companies can stay relevant and continue making a profit. The possibility of improving loudspeaker sound quality whilst taking into account the changing nature of the loudspeaker's physical properties could reduce waste electrical and electronic equipment (WEEE) by dissuading consumers from carelessly disposing of perfectly functional systems, and by preventing overdesign from manufacturers.

The same technology can be used with similar electromechanical system topologies to improve the quality

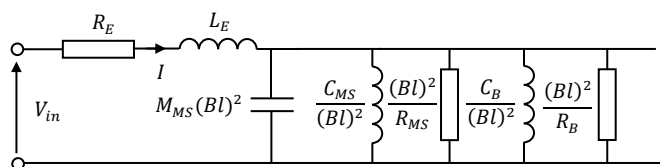Fig. 1 Loudspeaker anatomy, from the Wikimedia Commons.

Fig. 2 Loudspeaker electrical and equivalent mechanical circuit

of their performance, with potentially niche problems and markets receiving invigoration through such novel solutions.

III. THEORY

A. Loudspeaker Equivalent Circuit Diagram

A loudspeaker may be modelled as two circuits which interact through a magnetic field. From a design perspective, this is a very powerful tool – a full electrical simulation for a subwoofer in a box can also be combined with compensation circuit simulations, which saves time and makes the design process much easier. Fig. 1 shows the anatomy of a typical loudspeaker; the electrical circuit is the voice coil resistance R_E and inductance L_E. The mechanical circuit represents the mass of the cone and air, the spring property of the spider, or suspension, and the total mechanical damping effects as an equivalent capacitance M_{Ms}, inductance C_{Ms}, and resistance R_{Ms} respectively [4]. These two circuits are linked by a transformer that represents the back-emf/force constant, Bl, which represents the constant of proportionality between force on the cone and current through the coil:

$$F = Bl\, I \tag{1}$$

In accordance with this introduced force-current proportionality, and with the intuition that, since all the mechanical parts of the loudspeaker are attached, they must share the same velocity, the simple circuit can be improved. Using standard techniques to refer values on the secondary of a transformer to its primary, the equivalent mechanical circuit parameters can be placed in parallel with the electrical circuit parameters, and the coupling effect is removed.

An enclosure represents an additional mechanical resistance R_B, since the loudspeaker is now affixed to a non-moving mounting face. The compression of air behind the speaker that the box stiffens the cone suspension, thereby decreasing compliance, which is represented as an additional equivalent parallel inductance C_B (adding inductors together in parallel decreases the total inductance). This represents the final additions to the equivalent electromechanical circuit that are relevant from a design perspective. Fig. 2 [5] shows the final equivalent circuit used for the project.

B. Loudspeaker System Diagram and Classification

A full system block diagram for a loudspeaker is shown by Fig. 3 [6] [7]. The electrical system forms a low-pass filter, whose cut-off frequency will lie above the frequency of operation. The mechanical system forms a band-pass filter – below its resonant frequency, for a subwoofer, the cone will be moving slowly but pushing a large volume of air, which requires large forces, and therefore large currents.
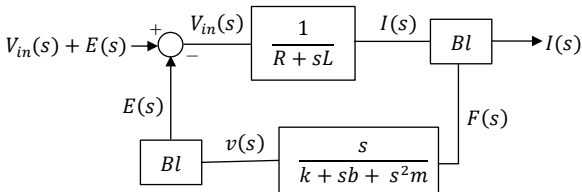
Above resonance, less air is being moved but the cone moves faster; as the cone's acceleration increases, the force and therefore the current required increases. To achieve this project's aim, it is more important to satisfy below-resonance requirements as opposed to those above resonance. Of note in [5] is that, assuming that the force on the cone is non-linear, it can be shown that the effects of a non-linear cone suspension are stronger than at higher frequencies, highlighting further the benefits of focusing compensation techniques on subwoofers as an overall quality-boost to a multi-speaker sound setup.

C. Thiele-Small Parameters

The Loudspeaker driver manufacturers do not directly quote values for M_M, C_M, R_M etc. because it is difficult to measure these values directly. Instead, a set of values known as Thiele-Small parameters (TSPs), described in [8]. These values are easier for loudspeaker driver manufacturers to measure and can be converted into an equivalent circuit by the user if necessary. They also give a viewer a more intuitive view of the driver's performance – whilst the equivalent circuit parameters describe individually each aspect of the mechanical parts of the driver, the correlation between n their variation and a change in the driver's performance is not obvious. With TSPs, the equivalent circuit for a loudspeaker can be derived, and a box's compliance and mechanical resistance can be added into it. It can be the case that an actual driver's TSPs will vary from the nominal ones given on a data sheet; it is therefore advisable to measure all TSPs.

The parallel RLC network seen in Fig. 2 has a resonant frequency equal to the subwoofer's mechanical resonant frequency TSP f_s. For this project, f_s should be kept as low as possible, so that the subwoofer uses less power to produce lower frequency sounds.

Another important TSP that serves as a limit to a subwoofer's performance is X_{MAX}, the maximum possible excursion of the cone. If the cone travels further than X_{MAX}, non-linearities in the cone suspension will become much more impactful against the quality of sound, and in extreme cases, the cone could become blown out from the rest of the driver.

D. Feedforward Controller

The effect of a feedforward controller is to manipulate the input signal into the subwoofer in such a way that the bass frequencies are more prominently represented, and thereby more prominent in the sound produced. Referring to the system block diagram of Fig. 3, there are two impedances at play. The impedance of the voice coil has a first-order low-pass action, which does not need to be corrected for as the 3-dB point of the series resistor-inductor
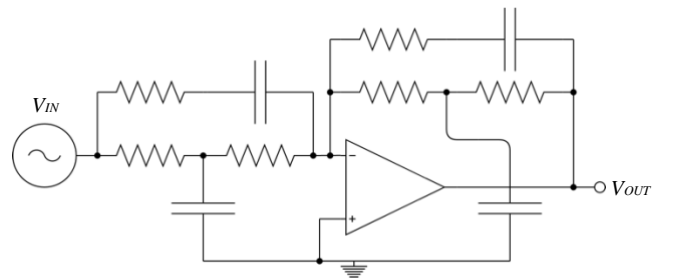

Fig. 3 Loudspeaker system diagram

Fig. 4 Linkwitz Transform [10]

circuit will fall way above the critical operating range of the subwoofer. The impedance of the mechanical circuit has a second-order high-pass action [9], with resonant frequency f_s. This high-pass action can be described in the complex-frequency domain as two pairs of poles and zeroes.

The Linkwitz Transform [10], shown in Fig. 4, is a high-pass filter that eliminates the driver's poles with a pair of zeroes, and reintroduces a new set of poles at a new lower frequency.

The advantage of this particular filter topology as opposed to more rudimentary op-amp circuits is that the filter additionally improves the time response of the loudspeaker, such that it responds faster and takes less time to settle to an input [11]. The disadvantage is that it extends bass performance at the expense of the subwoofer's cone excursion; there is no way for this filter to determine if the subwoofer's cone is going beyond X_{MAX}, and this must be investigated and, if possible, corrected for.

E. Feedback Controllers

Fig. 2 shows that the moving mass of the loudspeaker is represented by a capacitor; using the well-known capacitor current equation:

$$I = C\frac{dV}{dt} \qquad (2)$$

in conjunction with equation (1), it can be shown that the voltage across this capacitor is proportional to the velocity of the mass of the cone, known as the moving mass. This result shows that any controller should be attempting to force the velocity of the moving mass to remain in phase with the input voltage. There are a number of different ways to obtain a measurement for the velocity of the moving mass, which should be contrasted against each other. This result is also useful for simulation purposes.

Referring again to Fig. 3, it can be seen that a loudspeaker has multiple states. State variable feedback [12] is a method by which a system can be controlled by taking into account all of its available states, with the result of moving the closed-loop poles of the system. This is highly advantageous as it allows one to specify a desired performance for the loudspeaker and implement a controller that forces it to track this defined performance. An additional technique, state-estimation, can be combined with this to build a robust and extensive system model of a loudspeaker that can directly emulate simulations of the equivalent circuit by estimating the values of otherwise unmeasurable states [7]. An example of an unmeasurable state for the loudspeaker is the voltage across the capacitor seen in Fig. 2; if this were estimated, then the velocity of the cone would be directly known without transducers or detectors. The confidence in this velocity value would be better than if it were derived from a measurement of cone acceleration or displacement. Disadvantageous is the difficulty for an average end-user to follow this methodology, as compared to simpler closed-loop control circuits.

IV. METHODOLOGY

The general methodology of the project will now be described. A more complete description of all the steps taken, and initial mistakes made, may be found in the Interim Report for this project [13].

The project's innovative steps of designing and building open- and closed-loop compensators require a subwoofer unit to work with – therefore, the first part of this project revolves around constructing a complete subwoofer unit from scratch. This methodology is advantageous as it can be easily replicated by someone that only has access to a driver's datasheet.

The COVID-19 pandemic forced the project to an unexpected end around the end of March; some steps in this methodology therefore never happened. This shall be indicated on the steps in question, which are still described here to give the reader a full understanding of the project's aims and goals.

A. Driver Selection

The aim of this project is to fulfil the fundamental aims of loudspeakers introduced in section I. The chosen driver, the Pyle PLPW6D, reflects these aims, with its small form factor and low cost [14]. It also features two voice-coils, which introduces the possibility for novel simultaneous driving/sensing setups.

B. Enclosure and Power Amplifier Design

The given TSPs for the PLPW6D can be used to calculate the volume of an enclosure. As discussed in section III, an enclosure stiffens the driver's cone suspension, which means that adding any form of enclosure raises the resonant frequency of the system. It is thus easier to calculate a box size that allows the subwoofer to operate around a new, chosen higher resonant frequency for the system. The peak current through a loudspeaker's coil at a frequency f may be expressed as a combination of equation (1) and well-known classical mechanical relationships:

$$I_{pk} = \frac{M_{MS}(2\pi f)^2 \cdot X_{MAX}}{Bl} \qquad (3)$$

This peak current is an important variable; a power amplifier suitable for the system's operation around the new chosen resonant frequency can be chosen in conjunction with a suitable box size that allows the subwoofer to produce sound without breaking the limit of X_{MAX}. The relationship between I_{pk} and the volume of the box V_{BOX} may be derived:

$$V_{BOX} = \frac{VAS \cdot X_{MAX}}{C_{MS}I_{pk}Bl - X_{MAX}} \qquad (4)$$

where VAS is a common TSP provided by loudspeaker manufacturers on datasheets.

At low frequencies, the dominating limiting factor to the loudspeaker's performance is its compliance. Acceleration at these low frequencies is low; therefore, the force required to reproduce the input signal is at its highest. The current required to satisfy this force requirement will therefore be at its largest. The value of I_{pk} calculated to determine V_{BOX} is adopted as this maximum value, in order to balance power requirements below and at resonance. The power rating of the power amplifier P may be determined using:

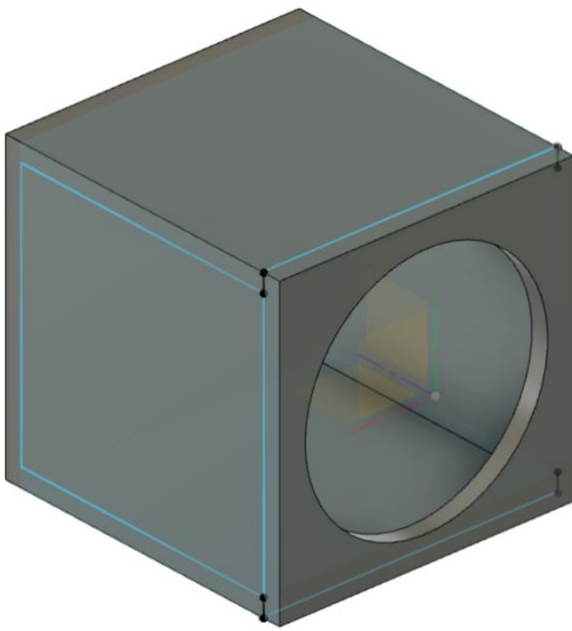$$P = \left(\frac{I_{pk}}{\sqrt{2}}\right)^2 Z_{NOM} \qquad (5)$$

Fig. 5 Autodesk Fusion 360 enclosure render

At frequencies above resonance, power requirements increase dramatically as seen by the squared frequency term in equation (3), therefore it is unrealistic to assume that the subwoofer can excurse to X_{MAX} too far beyond its resonant frequency, but this is excusable as the focus of this project is on the sub-bass frequencies as discussed in the introduction; a good crossover filter network will ensure that midrange speakers supplement subwoofers in this region, typically 100-200 Hz.

Although power amplifiers will always feature some non-linear behaviour, the magnitude of error is much smaller compared to that which is being investigated in this project. Therefore, it is acceptable to purchase a pre-built power amplifier.

A MATLAB script was used to perform all the described calculations, this is given in the Appendix. The volume of the box for the Pyle PLPW6D was calculated and rounded up to 6 litres, with a power amplifier capable of delivering 30 W into one of its voice coils.

For the purposes of this project, the Texas Instruments LM1876 [15] was chosen as the candidate amplifier to operate in bridged mode, for a maximum of 80 W power delivery.

C. Enclosure Construction

Although the work in [16] shows that a cubic box is not necessarily the best shape for delivering the most neutral frequency response, it is certainly the most convenient shape to work with. Due to the large pressure variations that a subwoofer will generate during operation, its enclosure should be as strong as possible without compromising on weight. It must also exhibit absolutely no air leaks, as these create high-order dynamics which drastically alter the frequency response of the system (a ported enclosure was not considered for this project).

To achieve these aims, 12 mm MDF was chosen as the candidate material for construction. Using the MATLAB script mentioned above and given in the Appendix, the dimensions for each panel of material is automatically calculated to give the desired internal volume equal to V_{BOX}, given the material's thickness. An Autodesk Fusion 360 model for the box (Fig. 5) [17] was created to assist during manufacture.

The 12 mm MDF panels were cut to size using a laser cutter and were joined together initially with wood glue, then with 3mm diameter screws. Banana ports were embedded securely into the back of the enclosure and were soldered onto the speaker's spade connectors. Before the enclosure was completely sealed shut, bathroom silicone was applied to every joint and left to cure, to guarantee an air-tight interior.

D. Impedance Measurements

The Bode 100 Vector Network Analyser was used to take accurate impedance measurements of each voice coil of the unmounted PLPW6D, which were used to verify the TSPs given on the datasheet [14]. Impedance analyses of the mounted subwoofer were not possible due to the effect of the on-going COVID-19 pandemic but would've formed the data to tune the Linkwitz Transform with.

E. Linkwitz Transform Tuning

Design tools are already available on the Linkwitz Transform's website [11] that make trivial the process of tuning its circuit elements in accordance with standard resistor and capacitor values. These tools all function in a similar way: by taking the original resonant frequency and Q-factor of the subwoofer, zeroes are introduced to cancel these existing poles and by specifying new values for each, new poles are introduced to modify the response. The full filter designed for the PLPW6D is given in the Appendix.

F. Simulations

Simulink [18] was used to construct a suite of graphs that give insights into the predicted performance of the loudspeaker in its various configurations. LTspice [19] was initially used for the earlier stages of the project to verify the background theory and as a check to verify the Simulink results. See the Interim Report [13] for more details.

The three simulations that were constructed in Simulink are available on this project's GitHub page [20]. Overall, six circuits exist: one to model the subwoofer's uncompensated response, one to model the Linkwitz Transform-compensated subwoofer, three to compare different models of closed-loop compensated subwoofers, and one to determine the responses of the systems. The first five models were designed to be easy to use and understand, with clear indications of where readings are being taken from, and which states are being used as feedback. These models are most appropriately used to understand the effect of different compensation techniques on the base system on a granular level. Data collection for these models was manual in nature and no overarching script can be used to generate useable data.

The final model was constructed from a scripting standpoint – clicking one button runs all necessary simulations and generates three graphs. This was done for convenience towards latter stages of the project to show the overall performance of each system against another.
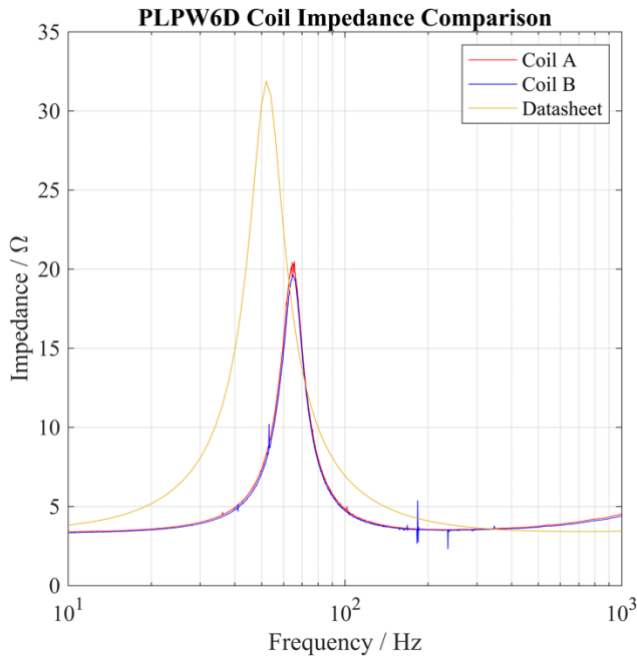
Fig. 6 Pyle PLPW6D voice coil magnitude response comparison

V. RESULTS

A. Impedance Analyses

Impedance analyses (Fig. 6) show that the real value for the PLPW6D's f_s deviated about 20 Hz from that quoted in the datasheet. In practice, this has a minimal impact on the TSPs and can be compensated for simply by increasing the power amplifier's rating – this was already taken into account by rounding up the box size as discussed earlier. The discrepancy in peak impedance values can be attributed to the nature of the voice coil inductance not being constant across every subwoofer. The voice coils have slightly different phase responses (Fig. 7), which would justify designing phase-correction circuitry if they were electromagnetically linked. However, when measuring across the input of one and the output of another, the collected data is meaningless and represents a lack of
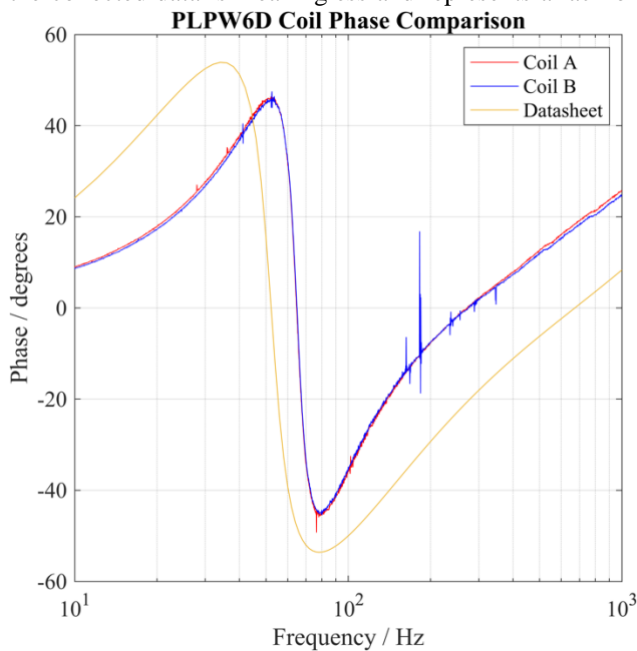

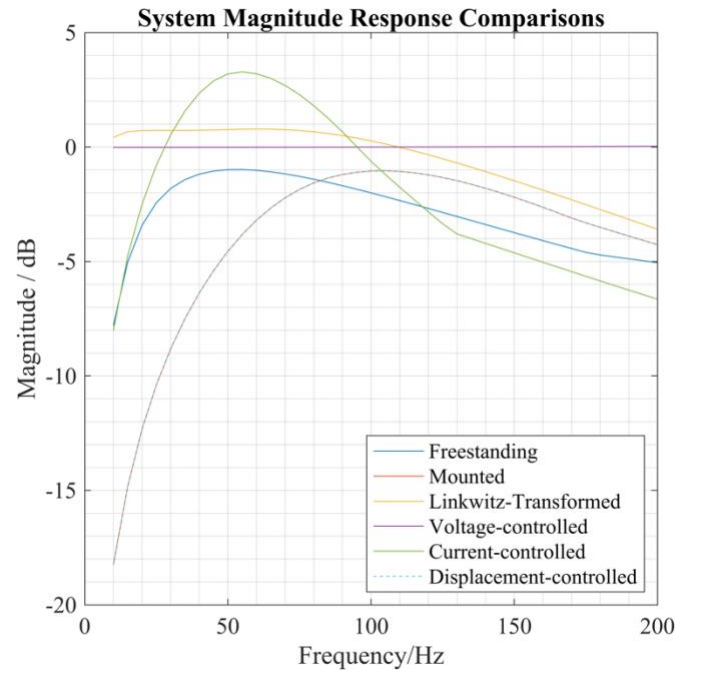Fig. 7 Pyle PLPW6D voice coil phase comparison response

Fig. 8 Pyle PLPW6D simulation magnitude response comparisons

connection between probes. This data is given in the Appendix.

B. Comparing Compensators

Figs. 8, 9, and 10 show the simulated magnitude responses, phase responses, and group delays of the various Pyle PLPW6D subwoofer system configurations. Figs. 11 and 12 show measurements of coil current and cone excursion as each system configuration is driven by a chirp signal, ranging from 10 Hz to 200 Hz and lasting for 1 second, with an amplitude equivalent to the peak input voltage from the power amplifier. This was chosen to simulate a 'worst-case' scenario for the subwoofer's operation. Phase response data must be interpreted whilst taking into consideration the way that the ear perceives sound – by detecting differences in pressure generated by
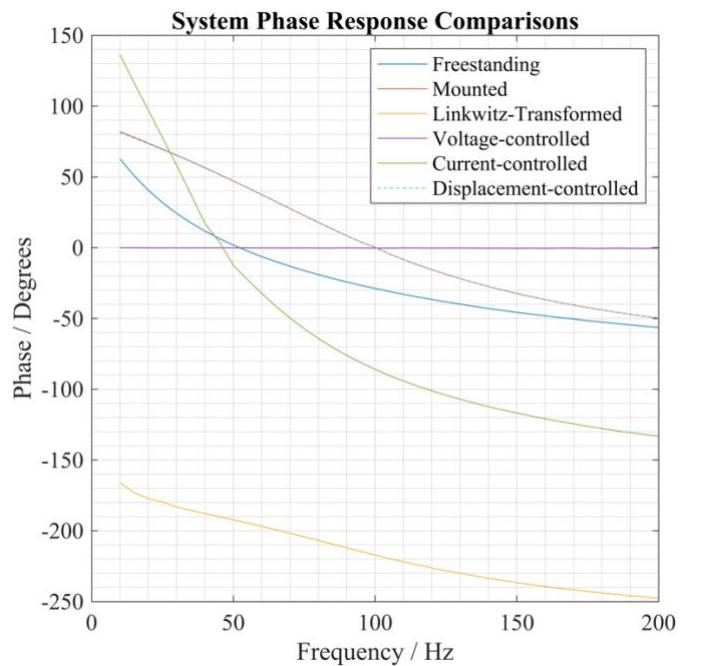

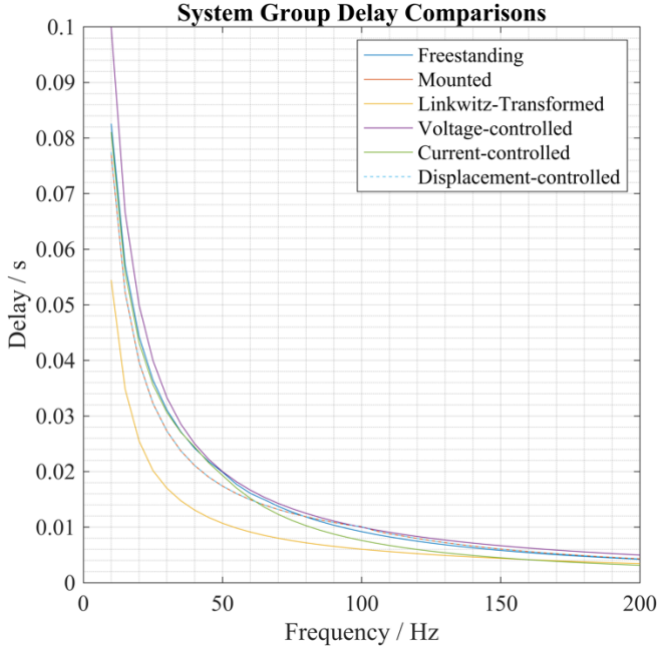Fig. 9 Pyle PLPW6D simulation phase response comparisons

Fig. 10 Pyle PLPW6D simulation group delay comparisons

sinusoidal movements of the subwoofer's cone. This means that a phase response of $\pm180°$ is still perceived as the same change in air pressure by the ear.

The sudden changes in gradient seen in two graphs of Fig. 8 is the result of only 39 data points being saved. This was done to save time when generating the data.

The voltage-controlled system exhibits a perfectly linear magnitude and phase response. It also exhibits much larger coil current and cone excursions.

The displacement controller has no discernible impact upon the mounted system's performance.

The current controller increases the low-frequency bass response of the subwoofer the most, but not linearly. It exhibits the second-best group delay from around 60 Hz to 160 Hz, and the best group-delay thereafter. However, X_{MAX} is exceeded by 6 mm towards low frequencies.
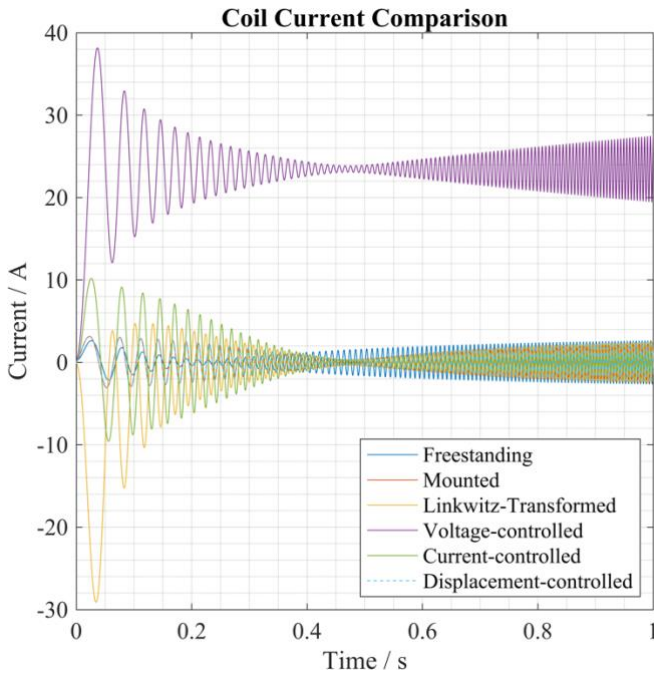

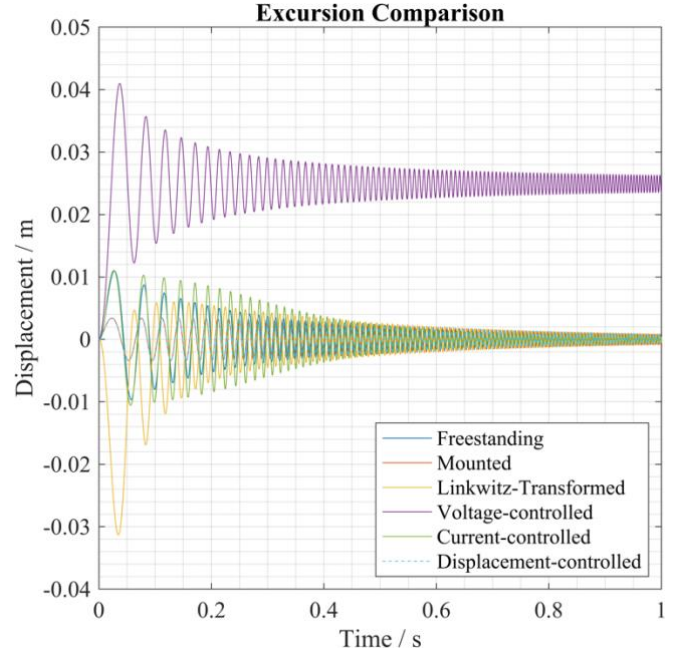Fig. 11 Pyle PLPW6D simulation coil current comparisons

Fig. 12 Pyle PLPW6D simulation cone excursion comparisons

The Linkwitz Transform exhibits a flat magnitude response to around 75 Hz before falling linearly as frequencies increase. At all frequencies below 160 Hz, the transform exhibits the best group-delay. At the very extreme low-end of 10 Hz, the current requirement and resulting cone excursion are massive, before falling below those of the current controller at around 40 Hz.

VI. DISCUSSION

The displacement controller does not satisfy the aims of the project because it does nothing to enhance the performance of the subwoofer. This is because the measurement of cone displacement is always much smaller in comparison to the input signal, so the compensated signal into the speaker will only be negligibly different. More testing needs to be conducted to determine whether the controller can make the subwoofer perform exactly as it should even if non-linearities are introduced.

The voltage controller completely fulfils the aims of the project. This is because the output voltage of the system is directly proportional to the input voltage. However, the physical integrity of the system would be at risk if it were implemented. Such high sinusoidal coil currents would increase the DC resistance, and potentially lower the inductance, of the voice coil due to a sharp increase in temperature. Either scenario would result in the compensator performing sub-optimally, as the system states against which it was tuned would have changed. With enough heat, the voice coils could even melt – a serious possibility given the thinness of the wires used in the PLPW6D. The high excursions would cause the non-linear cone suspension to dominate the subwoofer's operation, which the voltage controller cannot compensate for, and unwanted oscillations may unexpectedly occur. In any case, the cone also runs the risk of moving so far out of the enclosure and driver housing that it simply becomes detached and blows out from the subwoofer. Extracting a measurement for this voltage is not convenient – a voltage bridge setup around the cone is necessary, which would

confine the system to laboratories due to impracticality. A larger subwoofer which is built to withstand much larger excursions and coil currents would be a better candidate for testing a voltage-controlled setup.

The current controller partially satisfies the project's aims by enhancing the subwoofer's bass response, but without introducing linearity. The peak current requirement of this configuration, around 10 A, would drive a 30 W_{RMS} power amplifier to the limits of its operation (the maximum peak output current I_{pk} is 10.1 A). This controller is best used with a subwoofer which needs to produce as much bass as possible, irrespective of the magnitude of bass per frequency band, such as in a cinema, or as part of a live concert rig.

The Linkwitz Transform mostly satisfies the project's aims by providing boosted, somewhat linear bass. It also has the unique advantage of exhibiting the best group delay for most of the critical bass frequencies. The current requirement and cone excursion at around 10 Hz may be dismissed with some confidence, because not much recorded sound contains data for sub-20 Hz frequencies. Since the circuit is designed to work with Line level voltage inputs, the Linkwitz Transform is best paired with a subwoofer that is used as part of a studio or home speaker setup.

VII. CONCLUSION

The total cost of all parts, including those bought but not used, came to just below £75, which is a fraction of the price for high-end uncompensated sound systems. Ultimately though, the effects of COVID-19 on the project prevented any real-world data from being used or generated. This means that all results must be taken with the stipulation that they are from simulations which do not accurately reflect the nature of real physical systems. This project therefore does not satisfy its specification [13] but at least serves as guidance for future work. The results are also specific to subwoofers of similar build to the PLPW6D – similar control techniques applied to much larger concert or cinema subwoofers would need to be investigated further. The results are satisfactory to make changes to a home audio or similar hobbyist setup.

VIII. FUTURE WORK

Some guidelines for work that may be conducted based upon the findings of this project are given:

- Tune all compensators against real-life TSPs of the enclosed subwoofer measured using an impedance analyser.
- Implement and test all compensators firstly using analogue electronics, then using a field-programmable analogue array [21] that can be reprogrammed to compensate for changes to the subwoofer's TSPs over time.
- Investigate the potential use of state-observers, disturbance-observers, and estimators to enhance the ability of closed-loop systems to react to unwanted disturbances from the environment and predict changing TSPs over time without human intervention.

- Extend similar ideas and principles to similar electromechanical transducers, such as shaker tables or biological agitators.

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APPENDIX

A. MATLAB Enclosure Design Script [22]

```
%%%%% USER EDITS GO HERE %%%%%
%%% SPECIFICATION BLOCK %%%
% Add loudspeaker Thiele-Small parameters here
Cms_spec = 0.00063
Bl_spec = 6.16
VAS_spec = 15.38
fs = 52.2
Xmax = 0.004
Mms = 0.0148
Z_nom = 4
speaker_face_diameter = 0.165 % Also known as Sd

% Select MDF thickness and define mounting face
dimensions
mdf_thickness = 0.012
mounting_face_length_ext = 0.200
mounting_face_width_ext = 0.200

% Choose new resonant frequency
f_above = 100

%%%%% USER EDITS END HERE %%%%%
a_above = (2*pi*f_above)^2 * Xmax
F_above = Mms * a_above
Ip_above = F_above / Bl_spec
poweramp_above = (Ip_above/sqrt(2))^2 * Z_nom

%%% Ideal box size for Xmax
Ip = sqrt(poweramp_above / Z_nom)*sqrt(2)
F_req = Ip * Bl_spec
Cms_box = Xmax / F_req
VAS_box = (Cms_box / Cms_spec)*VAS_spec
V_box = (VAS_box*VAS_spec)/(VAS_spec-VAS_box)

%%% BELOW RESONANCE POWER REQUIREMENTS %%%
VAS_box = (VAS_spec * V_box)/(VAS_spec + V_box)
Cms_box = (VAS_box/VAS_spec) * Cms_spec
fb = fs*sqrt(VAS_spec / VAS_box)
F_req = Xmax / Cms_box
Ip = F_req / Bl_spec
P_diss = (3.4 * (Ip)^2)/2
four_ohm_poweramp_below = (((Ip)^2)/2) * 4
eight_ohm_poweramp_below = (((Ip)^2)/2) * 8

%%% DIMENSIONS %%%
int_vol = V_box / 1000
box_depth_int = int_vol /
((mounting_face_length_ext-(2*mdf_thickness)) *
(mounting_face_length_ext-(2*mdf_thickness)))

box_depth_ext = (2*mdf_thickness) +
box_depth_int
int_vol = V_box / 1000

mdf_thickness = 0.012
speaker_face_diameter = 0.165
mounting_face_length_ext = 0.200
mounting_face_width_ext = 0.200
mounting_face_length_int =
mounting_face_length_ext-(2*mdf_thickness)

mounting_face_width_int =
mounting_face_width_ext-(2*mdf_thickness)

int_depth = int_vol /
(mounting_face_length_int*mounting_face_width_in
t)
top_bottom_length = int_depth

top_bottom_width = mounting_face_width_ext
```
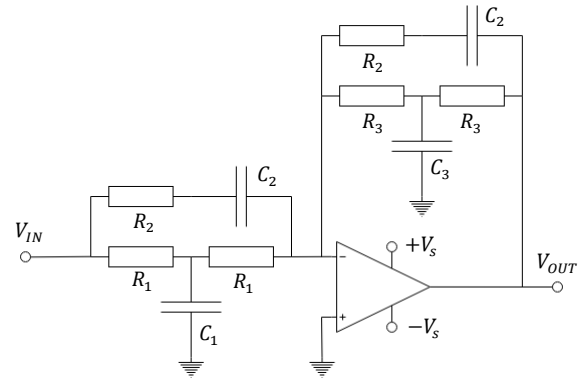
B. MATLAB Enclosure Volume Design Script

Fig. 14 Linkwitz Transform designed for enclosed PLPW6D

$+V_S = 15$ V, $-V_S = -15$ V [15]
$C_1 = 197$ nF, $C_2 = 69$ nF, $C_3 = 62$ nF
$R_1 = 15.4$ kΩ, $R_2 = 13.3$ kΩ, $R_3 = 442$ kΩ

C. MATLAB Enclosure Volume Design Script