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.

# Introduction Github Repo

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 bass 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.

# Economic, Legal, Social, Ethical and Environmental Context

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# Theory

## 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 *RE* and inductance *LE*. 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 *MMs*, inductance *CMs*, and resistance *RMs* respectively [3]. 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:

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 *RB*, 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 *CB* (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 [4] shows the final equivalent circuit used for the project.

## Loudspeaker System Diagram and Classification

A full system block diagram for a loudspeaker is shown by Fig. 2 [5] [6]. 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 [4] 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.

## Thiele-Small Parameters

The Loudspeaker driver manufacturers do not directly quote values for *MM*, *CM*, *RM* etc. because it is difficult to measure these values directly. Instead, a set of values known as Thiele-Small parameters (TSPs), described in [7]. 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 *fs*. For this project, *fs* 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 *XMAX*, the maximum possible excursion of the cone. If the cone travels further than *XMAX*, 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.

## 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 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 [8], with resonant frequency *fs*­. This high-pass action can be described in the complex-frequency domain as two pairs of poles and zeroes.

The Linkwitz Transform [9], 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 [10]. 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 *XMAX*, and this must be investigated and, if possible, corrected for.

## Feedback Controllers

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

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 [11] 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 [6]. 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.

# 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 [12].

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.

## 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 [13]. It also features two voice-coils, which introduces the possibility for novel simultaneous driving/sensing setups.

## 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:

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 *XMAX*. The relationship between *Ipk* and the volume of the box *VBOX* may be derived:

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 *Ipk* calculated to determine *VBOX* 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:

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 *XMAX* 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 [14] was chosen as the candidate amplifier to operate in bridged mode, for a maximum of 80 W power delivery.

## Enclosure Construction

Although the work in [15] 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 *VBOX*, given the material’s thickness. An Autodesk Fusion 360 model for the box (Fig. 5) 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 thick 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.

## 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 [13]. 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.

## Linkwitz Transform Tuning

Design tools are already available on the Linkwitz Transform’s website [10] 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.

## Simulations

Simulink [16] was used to construct a suite of graphs that give insights into the predicted performance of the loudspeaker in its various configurations. LTspice [17] 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 [12] for more details.

The two models that were constructed in Simulink are available on this project’s Github page [18]. 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.

# Results

1. Impedance analyses show that fs is off by 20 Hz – only has a small impact on TSPs.
2. Analyses also show that voice coils are in phase, therefore if linked, driving-sensing setup could occur without phase-correction circuits. Evidence to suggest however that they aren’t linked as shown by impedance measurements from IN-1 to OUT-2 (appendix)
3. Mounting driver in enclosure raises fs as predicted.
4. LT increases bass response
5. Voltage control is the most ideal but will probably break the subwoofer
6. LT second-most ideal, could have massive coil current/excursion but ultra-low frequencies can be filtered out to avoid this
7. Current-controlled give you a LOT of bass by making the 30W amp perform to its limits (Ipk for 30Wrms amp is 10.1A). Bridged amp necessary for safe operation.

REQUIRED GRAPHS:

* a, b: Impedance analysis csv imported into MATLAB
* c, d: output comparison plots
* d: excursion comparisons
* e:

## Impedance Analyses

Impedance analyses show that the real value for the PLPW6D’s *fs* 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, but by rounding up the box size, this was A close up of a map

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Description automatically generatedalready taken into account. The voice coils have slightly different phase responses, 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 connection between probes.

## Effects of Mounting Driver in Enclosure

C

## Comparing Compensators

# Discussion

A close up of a map

Description automatically generatedA close up of a map

Description automatically generatedThe results of the impedance analyses show that the design for the box is too small for the original poweramp requirement, as the magnet cannot actually generate enough force to overcome the decreased compliance of the box. However, this is easily rectified by simply increasing the power delivered to the speaker, which is acceptable since the original power requirement was so modest.

Measuring across separate voice coils lead to a massive impedance measurement, implying the existence of a dielectric, thereby confirming that the voice coils were independent from one another. The perceived error in impedances and phase responses between the two for the selected subwoofer is small enough such that no correction circuitry is required, so the driving-sensing setup discussed earlier can be undertaken with confidence.

The results of the SPICE simulations prove that the Linkwitz Transform not only increases the magnitude of response at sub-bass frequencies, but also improves the group delay of the subwoofer system in the range of operation. This means that adding the transform would enable the subwoofer to produce more bass, more linearly, and with less of a lag from when an input voltage is applied to it. The latter analysis shows that this subwoofer system would be appropriate to A close up of text on a white background

Description automatically generatedtransplant into a full sound system instead of existing as a stand-alone research item.

The current Linkwitz Transform circuit is not optimised well for a real-world subwoofer. Under 20 Hz, massive current and cone excursion requirements render accurate sound reproduction difficult, with risk to the physical integrity of the system high.

A close up of a map

Description automatically generatedThe Simulink simulation may be considered a ‘worst-case’ scenario for the system’s operation – at low frequencies the cone must move the furthest to move the required volume of air to reproduce the driving frequency. Nevertheless, it is clear that the cone excursion does not exceed 4mm from equilibrium in either direction of its travel once the system is stable. This Simulink model could be used further into the project to test the validity of closed-loop circuits.

Frequencies lower than 20Hz were not considered worth simulating given that most recordings do not contain sound data below 20Hz. The action of one closed-loop controller will be to limit the cone’s excursion to *XMAX* to avoid damage to the system and distortion to reproduced sound.

# Evaluation

TBC

# Future Work

1. Create loudspeaker equivalent circuit from datasheet values whilst waiting for subwoofers to arrive.
2. Conduct impedance analyses on subwoofers and edit equivalent circuit to reflect real-world values.
3. Derive box dimensions using equivalent circuit theory.
4. Build box and mount subwoofer.
5. Measure frequency response of system in anechoic chamber and take new impedance measurements.
6. Use measured frequency response to design and tune Linkwitz Transform circuit.
7. Measure new frequency response of subwoofer with Linkwitz Transform.

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# Appendix

## MATLAB Enclosure Volume Design Script [19]

%%%%%% USER EDITS GO HERE %%%%%%

%%% SPECIFICATION BLOCK %%%

% Add loudspeaker Thiele-Smalll 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\_int)

top\_bottom\_length = int\_depth

top\_bottom\_width = mounting\_face\_width\_ext

## MATLAB Enclosure Volume Design Script

+*VS* = 15 V, –*VS* = –15 V [14]

*C1* = 197 nF, *C2* = 69 nF, *C3* = 62 nF

*R1* = 15.4 kΩ, *R2* = 13.3 kΩ, *R3* = 442 kΩ

## A close up of a map Description automatically generatedMATLAB Enclosure Volume Design Script