Lost & Found Orchestra – Sonic Screwdriver

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BSc Report

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Abstract

This thesis showcases the development process of a tool for achieving a sustained, non-percussive sound that captures the sonic essence of ceramic objects, a so-called ‘sonic screwdriver’, meant for implementation in iterations of Daniël Maalman’s Lost & Found Orchestra sound art installations. The tool is meant to use a different way of sound excitation than Maalman’s conventional method of simply tapping objects with cores of solenoids. This is done by means of creating an audio feedback loop on the surface of the objects, which allows the objects speak in their own voice by using its resonant frequencies in an audio feedback loop. The audio feedback loop is composed of a contact microphone, a surface transducer and an audio amplifier. The system achieves accurate pitch control of the audio feedback at the resonant frequencies of an object by means of a control signal being input into the audio feedback loop via a second surface transducer. The developed solution can be used as a powerful tool in the creation of many types of sound art.

Acknowledgements

First of all, I would like to thank Daniël Maalman for giving me the opportunity to work on this project, inspiring me with his way of working and being available to answer questions day and night. It has been a truly interesting and rewarding challenge. Furthermore, I would like to thank Edwin Dertien for helping me stay on track in times of confusion and frustration and his continuous support and guidance, as well as Erik Faber's. Erik Faber, Eddy de Weerd and Douwe de Bruijn all deserve my gratitude for helping me out with the experiments with the HP 4194A Impedance/Gain-Phase Analyzer. Finally, a word of thanks is in place for Koka Nikoladze and Diemo Schwarz, who both helped me out via email despite both of their busy schedules and not being affiliated to the University of Twente.

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1 Introduction

Sound art is a relatively young art discipline, with the origin of the definition going back only to between the 1910's and the 1930's. Camilo Salazar, manager of Music Technology program at Morley College London describes it as "an art form that focuses on sound as a fundamental element". He adds that it is an everchanging definition as with time, more and more genres are already categorized as sound art [1]. It is often combined with elements of visual art and/or experimental music.

Because sound is at the center of the latter too, it is easy to confuse sound art with music. In general, it can be said that music can be a form of sound art, but not all sound art is considered music [2]. Oftentimes, music is more time- and melody-focused than sound art, though this is not always the case; there is some overlap between the two and many sound artists are also musicians, causing musical aspects or elements to often be present in sound art.

One such sound artist who is a musician as well is Daniël Maalman. Maalman is an AKI Academy of Fine Arts Enschede graduate (2010) who is specialized in building temporary sound art installations based on spaces and sites. In his works, among which Lost & Found Orchestra (2015), he aims to uncover hidden sounds in materials and objects. As Maalman is a musician too, he tends to use these hidden sounds for musical purposes, by using them in compositions for example [3]. One typical example of how he does this is his series of Lost & Found Orchestra installations, in which the objects and the space they are placed in act as the instruments. The objects or space produce sounds by being tapped on by different actuators such as solenoids or vibration motors, for example. These actuators are controlled in real-time from a central MIDI trigger board, creating sounds with the objects. A mix of this 'orchestra' is created by feeding contact microphones (or: piezo discs) that are placed on the surface of objects into a mixer, allowing post-processing of the sounds. [4] Because Lost & Found Orchestra is meant as an installation that the spectator can walk around in as it is being played, all the actuators have an LED attached to them that lights up when activated, so that the spectator can see more easily where each sound that they hear is coming from exactly. Here too, sound art is combined with visual aspects.

Both Lost & Found Orchestra and his artist residencies at ekwc (European Ceramic Work Centre) and STEIM (Studio for Electro-Instrumental Music) serve as the research phase for a next project called Foundscape. At ekwc, Maalman researched the effects that the type and shape of ceramics would have on their sonic properties. At STEIM, he researched the ways in which ceramics can be excited to produce sounds, for example with which actuators. One of the aims of Foundscape is for Maalman to develop his own electronic toolkit for use in sound art projects. This is where the Sonic Screwdriver comes in. The project description was as follows [5]:

> *"LFO is a project by sound-artist Daniel Maalman who is currently doing residency project at EFKW and later this year at STEIM. For this assignment a universal multi-channel midi trigger board needs to be developed which can use different types of actuator to tap, stroke, rattle and otherwise make objects release inner hidden sounds. In the same system a pickup (microphone, piezo, etc.) will be interfaced allowing audiosonic feedback. Basically, you are designing a swiss army knife of sound - or 'sonic screwdriver' if you like."*

What remains to be innovated is different types of actuation from simple tapping, stroking or rattling. What the different actuators have in common, is that they all mostly rely on the initial impact for the sound that is eventually produced, with the only color that is given to the sound is caused by the reverberation within the object or effects added in post-processing. The produced sound decays quite quickly and the amount of sounds that can be produced with the actuators is rather limited.

As all art is a form of personal expression, it is quite necessary to have a very flexible toolkit, so that the artist has optimal creative freedom and does not have to worry about not being able to properly express themselves because of limitations imposed by the tools used. In case of a painter for example, this could mean having a large collection of different paint brushes. In the case of sound artist Daniël Maalman, it means a more flexible method of creating sounds using ordinary objects as musical instruments than simply hitting an object repeatedly. That is why the research question is as follows:

How can a sustained, non-percussive sound be produced from a ceramic object, in which only the characteristics of this specific object determine the tone?

Finding an answer to the research question will be handled as follows: first, an analysis of the state of the art is made based in both scientific literature and already existing projects and products. After that, the creative techniques that will be used in finding the solution are explained. This is followed by an ideation phase, in which many ideas for possible solutions are brainstormed and the scope of the project is defined at first, after which these ideas are narrowed down to only the best ones to allow for a better focus on a solution. These best ideas will be specified further and stakeholders will be analyzed, after which the realization phase starts, in which the appropriate components for the task are selected and used for building different prototypes, of which the final one should be able to reliably produce a sound out of a ceramic object. These prototypes will be subjected to a series of tests and experiments that are developed based on practical application of the concepts that lay at the foundation of the research done for this project. These tests and experiments will be carefully documented and evaluated, after which a conclusion will be drawn about what is the best way to keep the ceramic objects in sustain and recommendations for further research will be made.

2 State of the Art on sustaining sounds in ceramic materials

2.1 Introduction

In this chapter, two things will become clear. The first one is the theory of the workings behind sustaining a sound that is produced in an object, the second one is the practical applications of this theory that is presently already in use in products and projects. This will be done by means of a scientific literature analysis first, followed by a state-of-the-art analysis.

2.2 Background research[1]

2.2.1 Introduction

There are many ways to sustain a sound. One type of sound that has a long decay time by definition is audio feedback, which can theoretically have an infinite sustain and is also very employable and usable in a musical context, as shown by The Beatles and Jimi Hendrix, to name a few. This literature research aims to give the reader insight in the different ways in which audio feedback can be controlled and used musically in sound art installations in order to allow for more sonic flexibility in sound art pieces. Towards this purpose, a definition of audio feedback will be given first. After that, an overview of the means in which audio feedback can be controlled will be given and finally, a conclusion will be drawn about what the best way to control audio feedback is.

2.2.2 Audio feedback

To find a way to expand the musical palette of a sound artist by means of audio feedback, a clear understanding of what exactly feedback is, is necessary. As also cited in Kollias [6] and Ferguson et al. [7], Morris [8] discerns three different types of feedback:

1. Electronic feedback, the most basic type. It occurs when the output of an electronic circuit is directly connected to the input of a circuit as a signal, making the signal enter the system again, but with some small delay. Electronic feedback makes for a highly predictable signal.

2. Digital feedback. This type uses data within one software program or between multiple programs as an input and output, rather than physical signals.

3. Electroacoustic feedback, the most relevant type for this paper. This is also the type musicians simply refer to as feedback or audio feedback. It occurs when the output of a circuit is connected to an actuator such as a simple speaker, instead of being connected directly to the input of the same circuit. The signal that the speaker produces is picked up by a sensor such as a microphone, which is connected to the input of an audio amplifier that outputs the received signal to the same speaker again. In electroacoustic feedback, the delay from the output back into the input is determined by the distance between the speaker and the microphone among others. This process produces the so-called Larsen effect, which results in a sustained high-pitched sound.

[1] This section is largely part of the literature review project for the course 'Academic Writing'. It has been submitted to drs. W.B. Tollenaar and graded on 14/05/2019.

Audio feedback is usually considered undesirable, as it often tends to be distracting and unpleasant to the ear. One such example is when it occurs as a loud squeal in a PA system when somebody is trying to publicly address a crowd[9]. In this research though, it is actually desired because producing audio feedback is a requirement for the solution.

Morris [8] also adds that electroacoustic feedback can make for much more complex and less controllable results than electronic feedback, because the environment that is between the speaker and microphone are placed in can heavily color the original signal before it is picked up by the microphone. One example of this comes from Lucier [10] in the form of a sound art piece that was created by iteratively recording himself reciting a text. Lucier made one recording of himself, which he then played back over speakers to record it again. He would then play back the next recording and record it, et cetera. After a couple of minutes, the recited words started becoming less and less intelligible, because the resonances that were created by characteristics of the environment had taken over and eventually became all that can be heard. To conclude, there are three different types of feedback, of which electroacoustic feedback is the one that is used in music the most.

2.2.3 Methods of Feedback Control

The lack of control over the outcomes of electroacoustic feedback makes it hard to use predictably, which is an undesirable trait when approaching it from a musical perspective because musical instruments traditionally have a highly predictable relationship between control input and sound output [7]. There are different ways of controlling electroacoustic feedback for use in musical context, each with varying degrees of success. Di Scipio [11] presented his attempt at controlling feedback, which he achieves by means of digital signal processing (DSP) on a computer. The DSP algorithm dynamically adjusts itself based on the input signal. The algorithm controls the gain of the feedback, which keeps the system in equilibrium volume-wise. He claims that this way, he uses sound as both content for listeners and as a control signal for the feedback gain of the audio feedback [12][13] Similarly, Ferguson et al. [7] controlled the gain of a bandpass filter using a DSP algorithm to steer the pitch of the audio feedback of a guitar and an amplifier. However, this did not guarantee the specified pitch to be produced, as the strings may not have a natural resonance at this pitch. The approach of choosing sonic features to make the audio feedback conform to was also proposed by Kim et al. [14] by means of evolutionary control of second-order structures.

Kollias shows a similar view to Di Scipio regarding letting sounds control themselves by using them as a control signal for DSP, as well as the focus point for the listener. However, he argues that letting the sound only control the feedback gain grants almost no control of the overall sonic shape of the feedback[15]. [6], [9], [15], [16] and [17] propose a system of self-organizing sounds, a term that Kollias invented himself. He explains this term as "'intelligent' sound/music systems characterized among others by autonomy, distributed/decentralized feedback processes and of environmental awareness"[2, pp. 1]. He invented this term to prevent any confusion that might be caused by the overuse of the word 'feedback' in the many different fields of science [6]. While Di Scipio's system is technically self-organizing according to Kollias' criteria, its implementation is quite elementary and lets many opportunities for controlling different aspects of the sound go to waste [15]. However, Di Scipio, Morris [18] and Hayes & Stein [19] and argue that this lack of control over the feedback simply puts the traditional role of composer of music into the hands of sound itself, sidelining the actual composer as merely the engineer of the dynamical feedback control system [12].

To prove that more accurate control of audio feedback rather than just feedback gain is in fact possible, Kim et al. [17] propose a more flexible DSP system that uses multiple signal-inferred characteristics of the acoustic environment to further control the sound of the audio feedback. The DSP algorithm is updated in real-time based on the audio that is connected to the input of the system, after which it takes this signal, modifies it, and outputs it. In this system, the room reverberation can control the tempo-scale characteristic, the ambient noise level can control the amplitude and the resonances of the acoustic environment determine the timbre (energy distribution over frequency spectrum) of the audio feedback. Just as Ferguson et al. [7], Kim et al. do warn that despite a sophisticated DSP algorithm, audio feedback is still hard to control, as a very slight increase of gain or decrease of distance between speaker and microphone can already violate the Barkhausen stability criterion. This is the criterion that is required for oscillation and thus producing audio feedback [20]. Feedback can still not be predicted accurately short-term due to nonlinearities, though long-term tendencies can be predicted.

Additionally, Morris [8] proposes the option of iterative feedback to give the artist more control over the output, by which he means a feedback process that is not necessarily implemented in real-time, but can also be implemented afterwards in editing. An example of iterative feedback is the aforementioned *I Am Sitting In A Room* [10]. To conclude, the ways in which audio feedback can be controlled are various types of gain-adjusting DSP algorithms, DSP algorithms that model characteristics of the acoustic environment, and iterative feedback.

2.2.4 Conclusion

In conclusion, though there are in fact ways to control audio feedback predictably, there are relatively few, each with varying degrees of success. It seems that most options involve the use of computers, be it for dynamically adjusting DSP algorithms or iterative feedback. Audio feedback remains an unpredictable phenomenon due to the enormous impact that even the smallest of changes to the acoustic environment can have on the produced sound, making accurately controlling it for musical purposes quite the challenge. While iterative feedback grants the artist the most precise control over the final produced sound, the process takes quite long, making it unsuitable for live performances. DSP algorithms are the best option for real-time usage. These algorithms can range from simple general gain adjustment for automatic amplitude control, to more intricate band pass gain adjustment for pitch tendency control, to complex acoustic environment characteristic modeling, with which tempo-scale characteristics, amplitude and timbre of the audio feedback can be controlled fairly precisely.

However, not every single adjustment option is completely necessary for a sound artist and having too many options may harm the user-friendliness of a system. For the Lost & Found Orchestra, the only set-in-stone requirement is real-time operation. That is why a band pass gain adjustment DSP algorithm would likely be the best fit for the Lost & Found Orchestra: it is a fairly flexible and precise control method, while not being too intricate to use.

For further research, the next step would be to look into options for a DSP system that allows control of as many feedback controlling parameters as possible, both automatically and by hand. Automatic parameter control could then be achieved by means of preset selection by the user, after which the resulting feedback sound will simply behave a certain way at the push of a button. For manual parameter control, a computer system would be used in guiding the user in the process of parameter tweaking in real-time. This would be done by means of an artificial intelligence algorithm based focused on not violating the Barkhausen stability criterion, which can very easily happen in control of audio feedback. This could possibly be the most flexible and user-friendly audio feedback control system.

2.3 State-of-the-art review

2.3.1 Introduction

Currently, a multi-channel MIDI trigger board is implemented [21]. Currently, a Korg nanoKONTROL Studio sends MIDI data to a computer that converts the data to RS-485 and sends it to a custom-made 24V LED dimmer board (Figure 2.1). Different types of actuators can be connected to the output of the dimmer board, which can then all be controlled individually using MIDI signals. As mentioned before, solenoids are usually used for this purpose, but they solely rely on impact for the production of sound. In this chapter, examples of technologies currently in use are analyzed. They already conform to certain aspects that the final product of this graduation project should conform to as well in their current form. While none of them are exactly the solution to the research question, as that would make this research pointless, and they all have their own limitations that make them unsuitable solutions, these technologies will be discussed in this chapter to draw inspiration from.

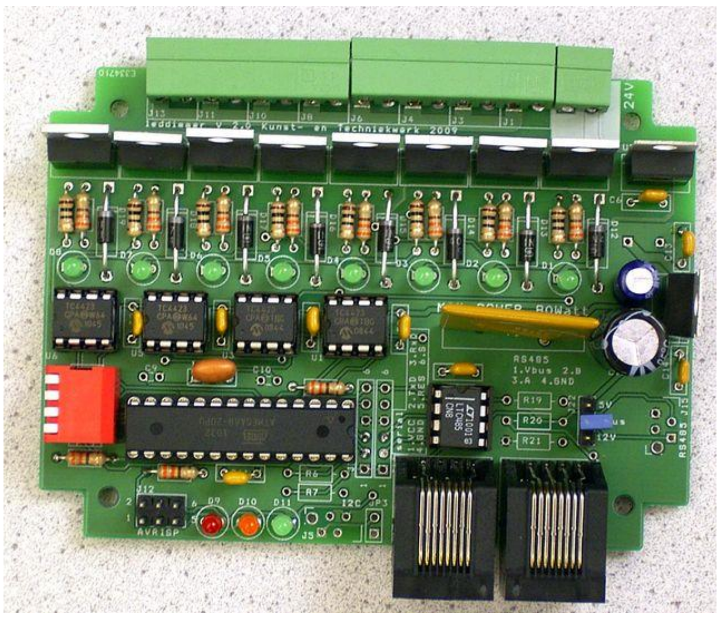


Figure 2.1: Edwin Dertien's 24V LED dimmer board. [2]

2.3.2 EBow

The EBow (Figure 2.2), short for 'electronic bow' or 'energy bow', is a device for use on electric guitar. It was invented in 1969 by Greg Heet and is currently produced by Heet Sound Products in Los Angeles. What it does, is make a guitar with metal strings produce sounds on one string without the use of a plectrum. The sound that is produced with it is often compared to the sound that is achieved when running an actual bow (such as the one used on a violin or a cello) across the strings of the guitar, hence the name.

[2] Source: http://wiki.edwindertien.nl/lib/exe/detail.php?id=projects%3Aleddimmer%3A24vdimmer&media=installations:leddimmer:dimboard.jpg

Figure 2.2: An EBow placed on an electric guitar.[3]

The way it works is as follows: as can be seen in Figure 2.2, the device has both an input coil and an output coil that are placed at a fixed distance from each other, at which they do not directly interfere with each other. Meanwhile, a guitar string is held perfectly centered along the same axis as the coils, also at a constant distance from the coils because of the string guides underneath the device (Figure 2.4) which hold on to the two neighboring strings, also silencing them. The two coils are connected to a LM386 low voltage audio power amplifier, which is an amplifier integrated circuit that requires very few external components (Figure 2.5). It runs on a standard 9V battery. Because of the magnetic field that is created by the circuit, the guitar string will start to vibrate and because of the feedback loop, this vibration will provide an input for the circuit, which will make the output coil produce a new magnetic field and keep the string vibrating for as long as the user wants or until the battery runs out.

Moving the device along the strings will allow the user to discover a place where they like the produced feedback sound best and find other sweet spots. For electric guitars, this is usually near one of the magnetic pickups, for acoustic guitars, it depends. The constant distances between the three main components responsible in the production of sounds, namely the input coil, output coil and the string, makes this audio feedback loop one that is very predictable and controllable. This is something that is rarely seen in audio feedback.

[3] Source: https://fleetsound.com/products/e-bow-ebow-plus-electronic-bow-for-guitar

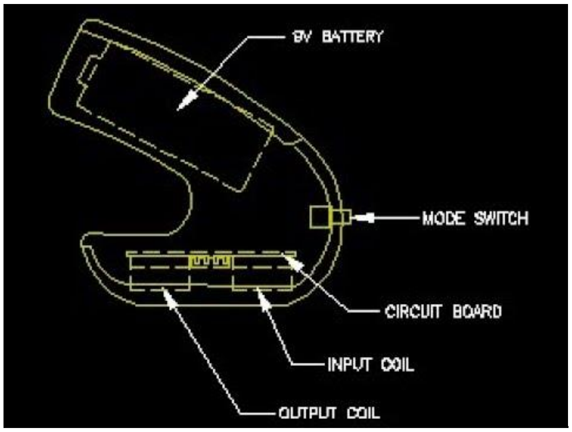


Figure 2.3: a high-level overview of an EBow.[4]

Figure 2.4: underside of an EBow.[5]

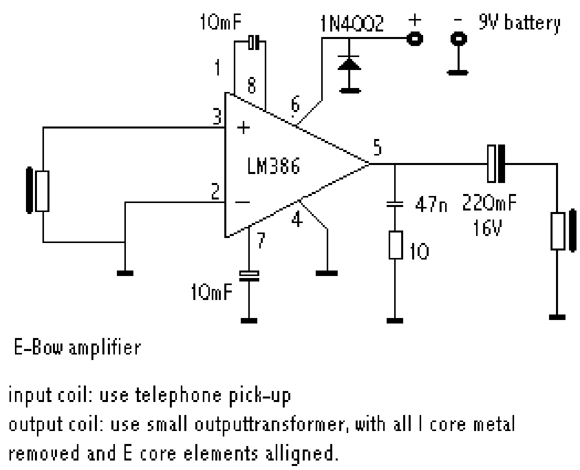


Figure 2.5: DIY-schematic of an EBow.[6]

The EBow allows for a lot of sonic flexibility for guitarists: the user guide describes it as "virtually an instrument in itself". It also provides plenty of examples for different playing techniques and even mentions ways to use it to make an electric guitar sound like a flute, cello, horn or harmonica [22]. This flexibility is very relevant to the current research. The small form factor is a big plus too and the ergonomics make for an intuitive user experience. It would seem that the EBow checks all the boxes of the project requirements. However, the tonal flexibility when used on guitar is also the device's main downside: due to its design, it only works on metal-stringed instruments. This makes it useless for this project in its current state, as the material that needs to produce a sound is ceramic. It could be that adding metal to the ceramic would allow it to be manipulated with the EBow, but that requires further research. What could potentially be useful in

[4] Source: https://www.youtube.com/watch?v=98zGR3QSJ74
[5] Source: https://buzzmusic.co/products/ebow
[6] Source: https://www.diystompboxes.com/smfforum/index.php?topic=80620.0

finding a solution is to build the EBow circuit on a breadboard and experiment with different types of sensors and actuators connected to the input and output of the circuit.

2.3.3 Koka's Magnetic Bow

Koka Nikoladze is a well-known Georgian sound artist. Just as Maalman, Nikoladze is a musician too. His sound art therefore incorporates musical elements as well. For example, he makes his own musical instruments and even small boxes with all kinds of objects inside and on top that play a small composition when turned on. He calls these boxes Beat Machines. One of the musical instruments that he made is called Koka's Propeller Trinity, which uses a propeller for generating many types of different sounds. Included in the trinity is Koka's Rotary Magnetic Bow [23], which, as the name implies, uses permanent magnets mounted on a rotating disk (Figure 2.6). According to Nikoladze, when the magnets spin in front of any musical instrument, it causes a magnetic oscillation, which can be controlled by changing the rotation speed, which is controlled with a fader-type potentiometer. By changing the magnetic oscillation, different frequencies can be targeted and resonate [24]. For example, when placed in front of a plugged-in electric guitar as showcased in his demonstration video, a droning sound is created that can go on for as long as the artist wants it to. The bow seems to have to be mounted at a fixed distance from the instrument, which does make for very stable audio feedback.

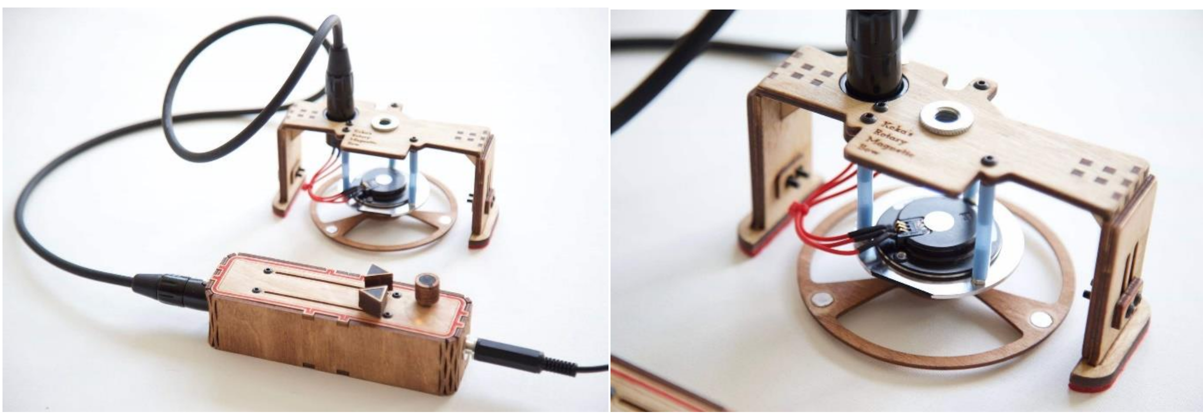


Figure 2.6: Left: Koka's Rotary Magnetic Bow full setup, right: close-up shot of the rotary bow.[7]

The form factor of the device is quite small, and it features a convenient mounting hole for attaching it to a microphone stand. Though the stable audio feedback is a desirable feature, the pitch control of the device leaves a bit to be desired as the pitch is almost entirely determined by adjustment of a potentiometer. Though intuitive, this is quite an imprecise method. Of course, by replacing the regular potentiometer with a digital one, it can be adjusted precisely, but the sound that is produced at a different setting will differ heavily based on what instrument the device is used on. It is hard to determine if the device is easy to integrate in other types of setups, as the only currently known instance of Koka's Rotary Magnetic Bow being used is in the demonstration video on an amplified electric guitar. Even though it does not conform to all requirements, this device is worth building as a test setup to get familiar with the way it works, for the purpose of sonification of ceramics.

2.3.4 The Sound of Ceramics

The Sound of Ceramics was an exhibition by Lu Wang and Polly Apfelbaum at Brown University, Rhode Island, which showcased the sonic properties of ceramics in different geometric shapes as they were

[7] Source: https://www.facebook.com/pg/nikoladzeofficial/photos/?ref=page_internal

suspended from the ceiling with a wire (Figure 2.7) [25]. The pieces of ceramic material were used as a melodic percussive instrument during a performance, which was accompanied by prerecorded sounds produced by the same setup. Unfortunately, no footage of this exhibition of any sort could be found, so the way in which the shape of the ceramics influenced the sounds that they produced remains unknown. However, it would have been a good example of the sonic properties of ceramic materials.

Figure 2.7: The Sound of Ceramics.[8]

While ceramics are being used to create different types of sounds which are all easily and intuitively controllable, the sounds produced in this setup are all impact-based and decay quite quickly, just as in the current state of the Lost & Found Orchestra. Aside from this, the system is all but small and unobtrusive, and would be hard to integrate in a Lost & Found Orchestra in this form. However, this is not really a shame, as there is no added benefit in adding more objects in the Lost & Found Orchestra that produce sound in exactly the same way that the current objects already do.

2.3.5 Ceramic Sound Art Instruments

In 2016 and 2017, sound artist Fedde ten Berge created a few interactive ceramic sound art installations in collaboration with Frank van Os from Barst! Keramiek. The shapes of the installations invite viewers to touch them, after which the installation responds with amplified electronic signals applied to its surfaces, generating sounds while also showcasing the electronic properties and vibrating qualities of the used ceramics. The experience that is produced is different for every viewer. Three examples of these ceramic installations (Figure 2.8):

- The Egg. This installation starts to crackle when the viewer moves their hands closer to it. As they move their hands away, a chord is produced using phase aligned formant synthesis. When the viewer puts their hand in the water contained in the egg, the ceramic material will produce the loudest sound [26]. All this interaction is made possible by means of Bela (Figure 2.9), an embedded computer platform that allows for digital interfacing of sensors in a low-level manner (not unlike Arduino), while simultaneously being capable of ultra-low latency on-board audio processing, which makes for interesting real-time interactive applications [27].

[8] Source: http://www.pollyapfelbaum.com/2016/the-sound-of-ceramics/

- The Trunk. Close proximity capacitive sensing is used for producing sounds in both the tree trunk and the ceramic water trays of the installation. The water triggers samples of water splashing when touched. The trunk is equipped with three piezo disks at the bottom, so that sounds produced by tapping on it can be extended into what seems to be infinity using software [28]. This extended sound is close to that what needs to be achieved on ceramics for this research.

- The Shroom. Another Bela-based artwork that uses sensors to sense different types of vibrations, such as electromagnetic or acoustic, but also material vibrations and responds to people approaching it with different sounds. The ceramic dome has steel dust baked inside of it and as the inside is covered with aluminum foil, this makes for a capacitive proximity sensor. Also inside the bowl are electret microphones for picking up acoustic vibrations and piezo disks for picking up vibrations caused by knocking [29]. What type of actuators are used is currently unknown, but it could be worth getting into contact with the artist for this too, as this installation can create sustained ceramic sounds as well.

Figure 2.8: From left to right: The Egg, The Trunk, and The Shroom by Fedde ten Berge.[9]

[9] Source: https://blog.bela.io/2017/12/12/fedde-ten-berge-instruments-bela/

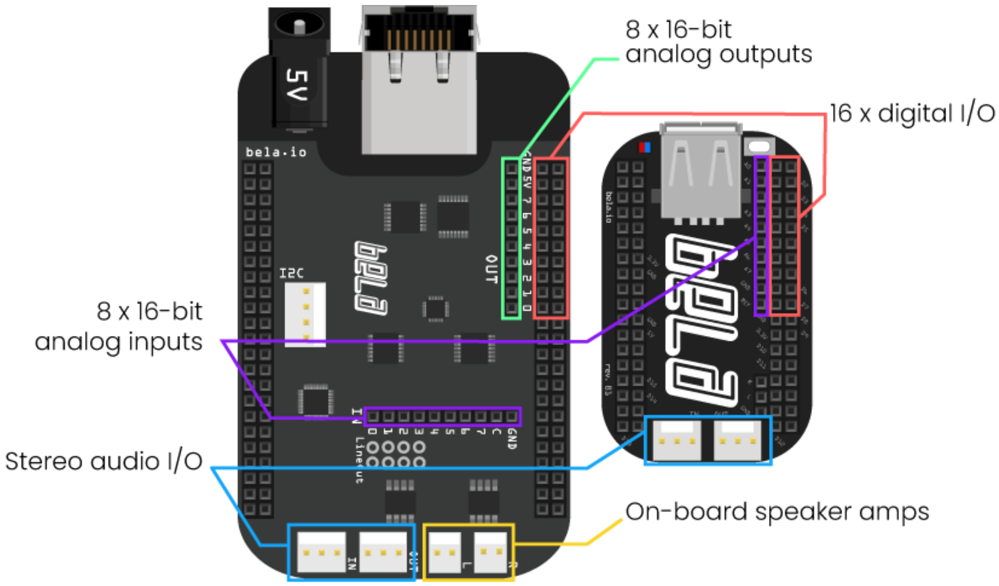


Figure 2.9: a graphic overview of Bela.[10]

2.3.6 MIDI Sprout

A device developed by Joe Patitucci from zero-waste record label Data Garden, who try to combine electronic music and the environment to show a relationship between the two. The MIDI Sprout uses changes in conductivity on the surface of plant leaves combined with a complex algorithm to create music (Figure 2.10). This device has also sparked the advent of Plants FM, an online radio station that streams music generated by plants and the MIDI Sprout all day long. The usage of changes in conductivity of a surface for generating notes is an interesting concept for the Lost & Found Orchestra too. Through the aforementioned The Shroom, it became clear that the conductivity of ceramic clay can be changed by mixing it with something like steel dust before baking. If a way of changing the conductivity of already baked ceramics can be found, maybe such as bending it etc., this could be relevant for the Lost & Found Orchestra as well in some way, though not in the context of capturing the actual acoustic characteristics of a ceramic object, which is purely an acoustical context.

[10] Source: https://bela.io/about

Figure 2.10: MIDI Sprout interfaced with a MacBook Pro via MIDI over USB and electrodes placed on a plant as input.[11]

2.3.7 Conclusion

What can be gathered from the state of the art is that there is a plethora of examples of products and projects that can make for sonification of a wide range of objects, but none of them are directly capable of solving the research question as-is. Elements from the working principle of the EBow can definitely be used to get an indefinite sustain, aspects of other projects such as Ten Berge's involving conductivity less so. The remainder of this research will be dedicated to finding ways to sonify objects by implementing knowledge from the state of the art, such as constructing an EBow circuit and connecting different types of sensors and actuators to the input and output. At the very least, this analysis has given an insight in which techniques might not work as well, such as capacitive sensing.

[11] Source: https://www.midisprout.com/

3 Methods & Techniques

3.1 Introduction

This chapter is devoted to the methods and techniques that will be put into practice to research audio feedback in ceramic objects. This will be done by means of the design process that is adhered to in most projects in the Creative Technology BSc. program that was designed by Mader and Eggink [30], which can be found in Figure 3.1. The methods will be named and explained, and the used research techniques that belong to these main methods will be listed in under the method they belong to.

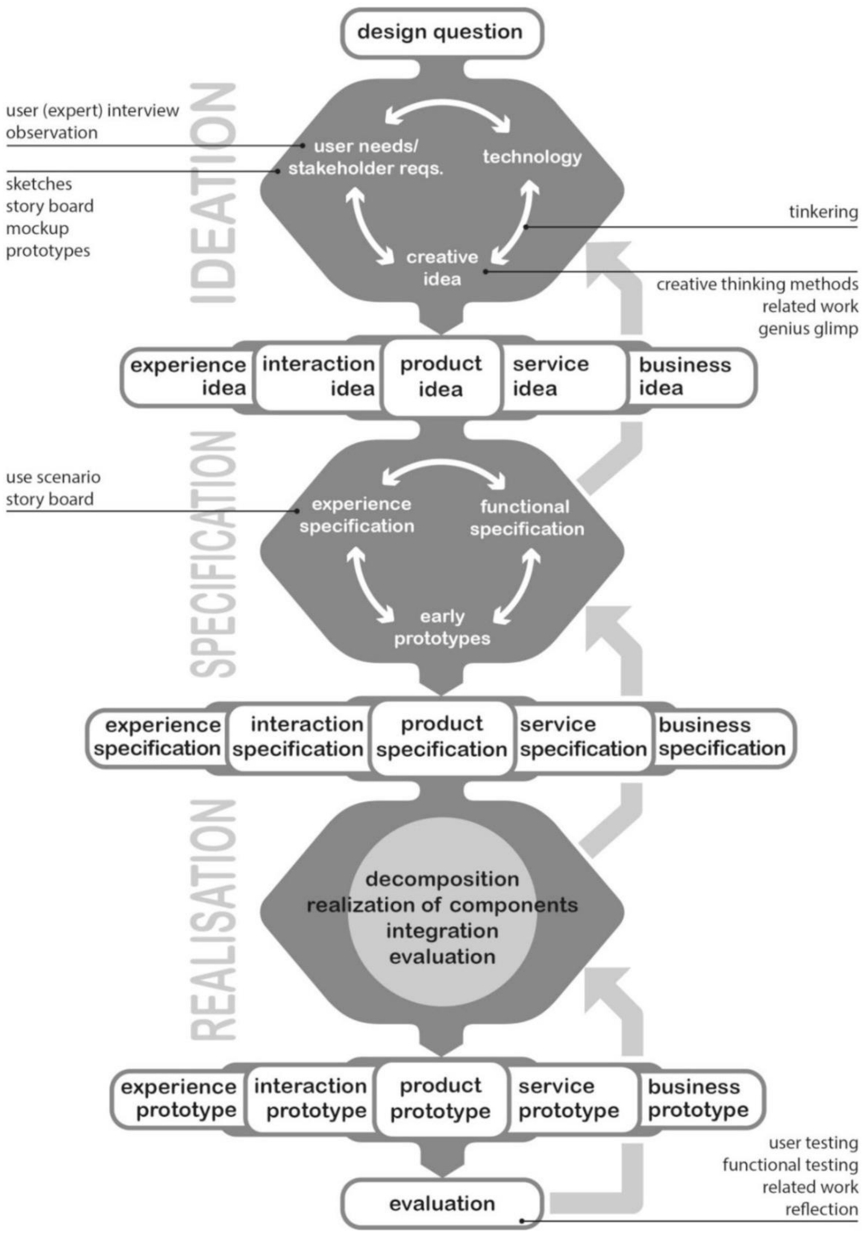


Figure 3.1: The design process of Creative Technology [30].

3.2 Ideation

When looking at Figure 3.1, there is one design question that leads this project into the ideation phase, which is the research question of how to sonify ceramic objects in a non-percussive manner. In this specific project, the techniques used are user/expert interview (part of the co-design process that is required for this project, as it is made specifically for one sound artist) and sketches. Ideation is the phase of divergence, meaning that the territory of the design project, its boundaries and the project goals are defined by means of research. This research can be done via many means, such as surveys, focus groups literature study or lab experiments. Most of the research that needs to be done, is done in this phase. The goal is to make clear to the designer what exactly needs to be accomplished with the design challenge to make the design achieve its objectives. Eventually, a combination of a product idea and an interaction idea is what came out of this phase, which required some iteration when in the specification phase, it turned out that some of the preliminary requirements were either not feasible or not defined clearly enough.

3.2.1 Scientific literature research

Though some scientific literature analysis has been done already for chapter 2.1, the analysis of relevant literature is an iterative process that should occur in every chapter of any scientific publication, such as this graduation project. It helps the researcher stay focused and prevents a tunnel vision mentality, which could prevent the researcher from finding an answer to their research questions. New sources will be searched for as soon as a lack of scientific evidence for a statement occurs. In the process of analyzing this literature, interesting new findings will also be incorporated into this report, which may lead to new research questions.

3.2.2 Co-design

Co-design sessions need to take place, at which both the commissioner and student should be present. This session should serve to brainstorm and to discuss the view that both parties have on the project and if either of those views is realistic. A final design concept should be a project that still has the core functionality of what the commissioner aims to achieve with the project, but it should also be feasible to make for the developer, which is in this case the student. Feasible is meant in both the sense of completing the project in the given timeframe of 10 weeks, as well as it being realistically possible to make. The developer and the commissioner will work side by side and bounce ideas off each other as soon as they occur in order to gain new insights and work further from these insights.

3.3 Specification

After the design territory and scope have been made clear in the ideation phase, they need to be reduced as much as possible. This might sound counterintuitive as some effort has already been put into this in the divergence stage, but the thinking area needs to be narrowed down in order to set a practical focus for the project. Transformation is all about prioritizing objectives of the design and create a sketch of the real-world situation that is as realistic as possible so that it can be decided which things not to lose track of, but also which things to specifically not pay attention to. After defining the problem, identifying the variables and setting the objectives, the focus of the specification phase should be on convergence. This means uncertainties have to be removed from the design one by one, until one final design remains [31]. This part of the process could theoretically be taken over by a computer the most easily, as it is a process that requires rigidity in decision making. Humans may be more likely to compromise on aspects such as safety of their design in favor of the look, because they have more stake in that part of the design. A computer simply applies a set of rules that it strictly adheres to so that no compromises will be made and the best product possible will be brought onto the market. If a certain value of a variable turns out not to be tolerable, the process will revert to the transformation stage and all variables and goals will have to be reconsidered. In this chapter, the main categories of components that should be used will be discussed and different for each of these categories will be compared. This will result in an updated version of the list of preliminary requirements, called the system requirements. In this project, these requirements were the product specifications and the interaction specifications. Co-design is used for defining these requirements as well, as Maalman's preferences are still very relevant here. Some light prototyping is done here as well to check for feasibility of using certain components and set specific goals for the prototype, but nothing concretely focused on answering the research question is built yet. This will have to wait until the realization phase

3.4 Realization

Based mostly on the requirements, thinking needs to be done about building prototypes that can be tested, so that eventually, a solution can be developed. It will turn out soon enough if the requirements that were set in the previous chapters are specific and concise enough, as if they turn out to be not, a prototype cannot be built properly. This is because it is impossible to leave too many variables to the imagination, as the prototype will likely not achieve the goal that it is set out to achieve.

3.4.1 Building prototypes

The insights gained during the scientific research will be used to select the working principles of an eventual end product, which will be thoroughly tested in isolation by means of various prototypes. Based on these principles, the appropriate components will be evaluated and selected for building the prototypes. Building prototypes is an iterative process, so if part of one of the prototypes turns out not to function as expected, it will be tinkered with to see the influences that that can have [30] and looked at scientific theory as to why something does not work the way that is expected.

3.4.2 Laboratory experiments

The usage of and experimenting with laboratory equipment to analyze characteristics of ceramics is an obvious component of the design of a prototype for this project. Delivering a prototype is part of this project after all. Experiments will be based on interesting facts that are discovered in the research of scientific literature, questions that arise in the design process, or on other experiments. The experiments will all be designed carefully and meticulously, so that as little unnecessary time as possible is spent in a laboratory environment, making time management easier. This time that could otherwise have been wasted is instead used to write clear and concise documentation or all aspects of the research.

4 Ideation

4.1 Introduction

The ideation phase mainly focuses on designing with the stakeholder in mind, which is why they need to be interviewed. Some brainstorming will be done based on this, after which preliminary requirements can be set.

4.2 Stakeholders

The main target audience of the to-be developed technology is sound artists, making them a large stakeholder in this project. To get an accurate picture of exactly what they are looking for in this technology, it is necessary to consult them in person. It is easy to make a technology very flexible and let the end user configure every parameter, but not all end users are equal, and some may have less of an understanding of the influence of certain parameters and may not be able to properly configure the system. This means that putting some constraints to the flexibility is necessary. To gain these insights, their previous work needs to be analyzed and interviews need to be scheduled with them. Interviews and co-design with Daniël Maalman himself and an interview with Jaap Mutter (an artist who has worked on numerous projects with Maalman) contributed the most information to the list of preliminary requirements that will be mentioned later.

The designer of the system is the second large stakeholder in this situation. The designer or developer is also the person responsible for integrating measures to prevent abuse into the system. This is mostly because the designer does not want to be held accountable for anything that might go wrong while the product is being used by the end user, such as damage to either the user or their possessions. This is an incentive to make the product as safe to use as possible. One way of doing this is by significantly reducing the flexibility of the technology, so that the end cannot possible navigate themselves into any sort of dangerous usage scenario. The hard part is though, that the technology still needs to be flexible enough for the end user, as it has a purpose to fulfill as a creative tool.

Consumers of the art created with the use of this technology are the third stakeholder. Though not as important as the previous two, the final result of the usage of the technology should still be pleasing to the people who enjoy sound art. If it is not, they sort of become the victim of the developed technology and they will lose interest in the work of the artist, which in turn reduces the income of the artist. Polling whether they like the produced results or not is not the concern of the developer, but rather the concern of the sound artists, as after all, they are the ones responsible for making their own living. This stakeholder will not be taken into account for further development, as it is the responsibility of the artist to cater to their preferences.

Usually, personas would be made based on the characteristics of the aforementioned stakeholders. However, for this specific project, this would not lead to a gain of any valuable new insights, as the main user of this technology is one specific person: Daniël Maalman. It would not make sense to model him as a persona, as required user insights can simply be inquired by asking Maalman. A persona-analysis is not extremely useful as the technology is developed with mostly one application in mind, which is usage in Maalman's sound art installations. This is why no persona-analysis will be carried out. However, other sound artists will be taken into consideration during development, as other artists may get inspired after experiencing Maalman's work. Afterwards, the project requirements will be summed up.

4.3 Brainstorming

Brainstorming for possible solutions to the research question was done in a few different ways. In order to set the boundaries for what was possible to get sound out of a ceramic object, Edwin Dertien recommended making a morphological overview of ways to destroy a ceramic dinner plate. This morphological overview can be found in Figure 4.1. While being a lighthearted exercise, this diagram provoked thought about how all of these different methods of destroying a ceramic object would sound if they were to be executed in real life, most of which would not produce a long, sustained sound. Now that the boundaries were set for breaking a ceramic object, the working space for keeping a ceramic object intact in the process of producing a sound from it was defined.

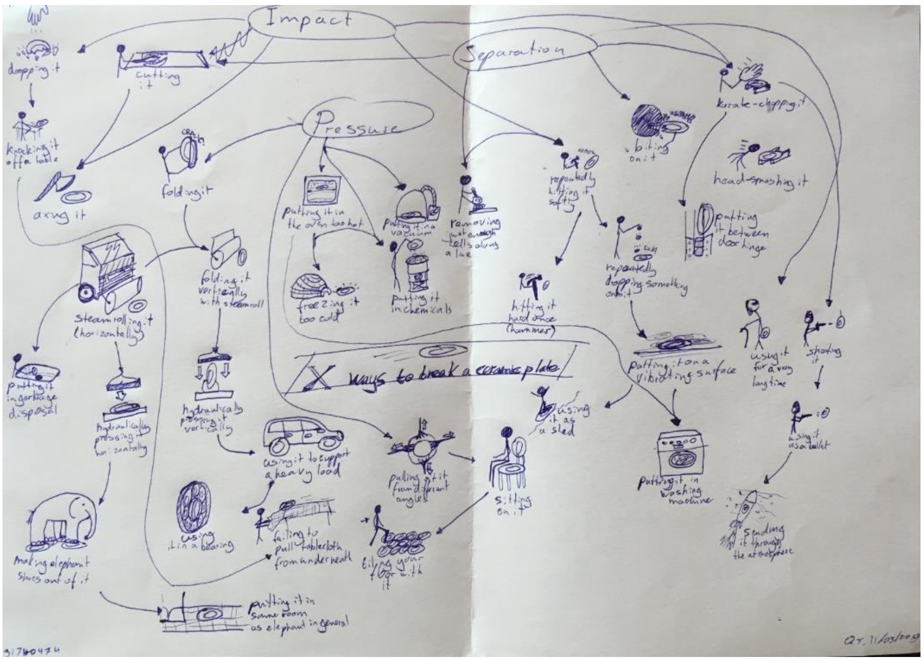


Figure 4.1: X ways to break a ceramic plate.

4.4 Interviews

As is the case with co-design, carrying out an interview with the person by whom the design is going to be used. is extremely important, as they play a large role in determining the preliminary requirements and maybe even part of the system requirements. The interview with Maalman was not done in a traditional single Q&A session style, but rather by means of planning in parts of different days with him and playing around with his equipment while asking things that were unclear and Maalman telling things he thinks are important for the project. As such, this interview cannot possibly be documented as a traditional interview. That is why the outcome of the interview is mostly summed up in the list of preliminary requirements. As Maalman is very experienced in the field of sound art, his opinions should be valued highly.

Through Maalman, an interview session with all-round artist Jaap Mutter was also arranged, in which he gave his opinions on object sonification. He mentioned he had once tried to get a relay to switch at its resonant frequency, which made it produce a sound. An option he mentioned was using an analog distance sensor placed above the object and then connecting the distance sensor to an audio amplifier. It would have to be an analog sensor, as a digital one might cause aliasing or over-quantization in the to-be amplified signal, as it only has a certain bitrate and bit depth. An analog sensor simply produces a continuous output voltage, that would change smoothly depending on the distance of the plate. This would then have to be combined with a similar type of actuator to keep the object vibrating to obtain a sustained sound.

4.5 Preliminary requirements

The requirements below will be prioritized based on the MoSCoW principle as defined by Clegg and Barker [32]. MoSCoW is an acronym that stands for Must, Should, Could, Won't, with the o's added for pronunciation convenience. The musts are the main factors that determine whether the project can be counted as a success. If even one of these is not met, the project should be considered a failure. The should are the factors next in line of importance, but the project will not fail it they are not met. The coulds are aspects that could improve overall satisfaction with the result but are not considered critical and may be added if time and resources allow for it. The won'ts are aspects that have the lowest priority to be added and are therefore not included in the scope of the project. They will likely not be considered for a following iteration either. Requirements proposed by Maalman are in **bold**.

Musts:

- **Plug & play for existing Lost & Found Orchestras**
- **Use the characteristics of the object itself in the produced sound**
- Accurate pitch control
- **No damage to the object**

Shoulds:

- Affordable
- **Expandable**
- **Easy to move around and remove**

Coulds:

- Intuitive user interface
- Portability

Won'ts:

- Control over the inherent resonant frequencies of a ceramic object.

4.6 Building plans

The current plan is to build a system in which an audio feedback loop will occur, using a ceramic object as a medium for the sound to propagate itself through. This idea is based on the literature research that has been done, the fact that the microphones to accomplish this with are already present in the current system and the fact that Jaap Mutter also mentioned it as an option. It will be worked out more in-depth in the specification chapter

5 Specification

5.1 Introduction

5.2 Object sonification

The ways in which people have tried to sonify objects are plentiful. However, there are three elements that they all seem to have in common:

1. A source of excitation
2. A means of picking up
3. A means of amplification

Coincidentally, these are also the three main required components for an audio feedback loop, as mentioned in 2.3.2 and 2.3.3.

Starting with the most obvious part of object sonification, a ceramic object needs to be present to obtain a sound in the first place. One ceramic artifact that was made during Maalman's artist-in-residence was made available by him for testing purposes (Figure 5.1), though he noted that lots of time and money had gone into their production. This meant that if one of the ceramic artifacts were to break, it would be very expensive to replace. Based on a recommendation of Maalman, a dinner plate will be used in the setup as a substitute for the ceramic artifacts that Maalman plans on using in the future installations. The specific dinner plate that is used in testing is a Grocier Innovation porcelain 256mm dinner plate, as can be seen in Figure 5.2. Using a dinner plate for testing was the advice of the commissioner and this specific type of plate was at hand in multitude. The plate will need to be suspended in some way, for which a special shock mount needs to be designed. This one can be found in the realization chapter.

Figure 5.1: Ceramic artifact during Daniël Maalman's artist-in-residence program at ekwc.

Figure 5.2: Grocier Innovation 256mm porcelain dinner plate.[12]

Maalman mentioned the following: there is a difference between ceramic material and the enamel coating. Both have an impact on the sound that is produced in the end. In 8TET1, he only used the ceramic material porcelain. Porcelain is made by adding powder to water with some other elements, which results in castable clay. The clay is then poured into a cast, which is then baked. The result is a finished ceramic object. Then, a mixture for enamel (a glass-metal composite [33]) is made, which is then slathered onto the baked ceramic object and subsequently baked at a slightly lower temperature to obtain a glossy finish. The enamels used all contain lead. The orange and yellow enamels seemed to pull the

[12] Source: https://www.svz.nl/nl/bord-265mm-innovation

material more and therefore break the ceramics more often than the blue and green enamels, which Maalman suspects has an effect on the sounds the ceramic can produce.

5.3 Excitation

5.3.1 Introduction

The source of excitation is the part that makes the surface of an object vibrate, which makes it produce sound. In Maalman's Foundscape project: 8TET1, [34] he sonifies homemade ceramic plates by placing solenoid modules with springs attached above the plates. Using Dertien's RS485-based 24V LED dimmer board [35], the solenoids are controlled via MIDI signals provided by Ableton Live. Based on the characteristics of the ceramic plates (types of additives used, such as lithium), the impact power of the solenoid, the tip used on the solenoid and the location where the ceramic plate is hit, a different sound will be produced. For example, most of the ceramic plates as seen in Figure 5.3 are hit with a small, rubber tip, but one of them is hit with a Dremel sanding tip, which produces a sound with many more higher harmonics. However, this is again a sound produced by means of impact.

Figure 5.3: 8TET1 by Daniël Maalman[13]

5.3.2 Surface transduction

When looking at a loudspeaker, the part that produces sound by moving up and down is the diaphragm of the speaker, which often has the shape of a cone, after which it is aptly named as can be seen in Figure 5.4. The cone should have a large surface so that a sufficient amount of air can be displaced.

[13] Source: https://deskgram.net/p/2052307506612555614_25690963

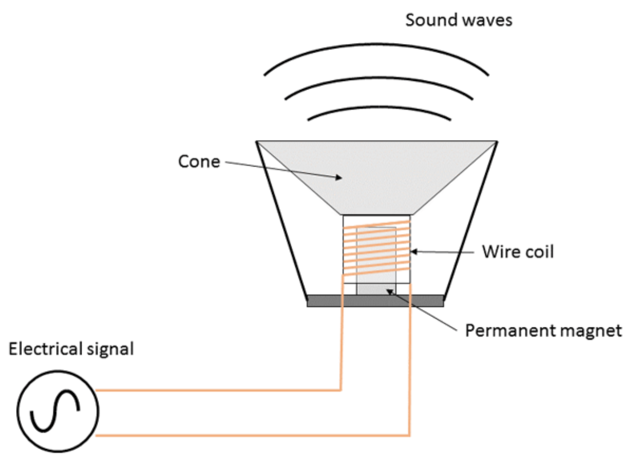


Figure 5.4: Simplified anatomy of a standard loudspeaker.[14]

In the case of an object vibrating and therefore producing sounds, the object itself can be thought of as the diaphragm of a speaker. Instead of simply attaching a speaker to the object, there is a way to waste as little signal as possible. Removing the diaphragm results in a speaker that can be put onto any surface, which will turn that surface into a diaphragm. These types of speakers exist, and they are called surface transducers. One example of a surface transducer in use is the project called "IKEA plate reverb", YouTuber LeoMakes actually uses the Adafruit Medium Surface transducer. In his initial setup, he bought two metal IKEA 'BROR' shelves and frames and modified them so that they would be about coffee table height. He put four plastic knobs in the corners of the surface of the first shelf as spacers to rest the surface of the second shelf upon. This is to minimize contact area so that the second plate can optimally resonate when a surface transducer is put on top (Figure 5.5), where it stays in place by means of its grippy, rubber bottom.

Figure 5.5: Bottom plate with spacers on the left, top plate with surface transducer and contact microphones on the right.[15]

The surface transducer is connected to one output channel of a Velleman MK190 2x 5W mp3-player amplifier DIY kit, which is a simple TDA1517-based amplifier (low-wattage stereo B-class power amplifier integrated circuit). The output of a digital audio workstation or *DAW* is fed into the amplifier

[14] Source: https://naturallysciencesite.wordpress.com/2017/06/17/how-do-headphones-work/
[15] Source: https://www.youtube.com/watch?v=CZWAntOnrx4

using a digital-to-analog converter or *DAC*. The contact microphones are two 7BB-12-9 12mm piezo elements, which are attached with double-sided tape. Their output is fed into an analog-to-digital converter or *ADC*, which can record the signal that is filtered by the plate using a DAW. The DAC and ADC are combined into one convenient device known as an audio interface. At the end of the video, Leo remarks that the piezo discs do pick up quite a bit of ground noise, which can be notch-filtered out, and that their output impedances are not matched with the input impedance of the ADC, which could be fixed using an amplifier circuit.

However, in a follow-up video, he managed to fix most of his problems: he was able to make the plate resonate better by cutting off the bent edges, completely sanding off the powder coating and suspending the plate from four corners using springs. The impedance matching problem he thought he had turned out to not be a problem, as no loading effects occurred in the ADC because of the high 1MΩ input impedance built in to the audio interface. The ground noise that occurred was fixed by simply running a wire from the resonating plate to the same grounding point the audio interface was using.

In this project, the shelf surface acts as a filter for the sound that is sent in via the surface transducer because of its inherent resonant frequencies. As discussed before, a common way of picking up sounds on a surface is by means of piezo sensors used as contact microphones. They send the vibrations they detect to the ADC as a voltage signal, which is received as a sound. This sound can then be recorded in the DAW.

For making a ceramic plate stay in resonance, a principle similar to the aforementioned plate reverb is needed. The main difference is that the output of the piezo sensors would need to be amplified and sent back to the surface transducer, so that a feedback loop can arise. Looking at available surface transducers, Adafruit is a well-known manufacturer of them. They produce two basic types:

- Adafruit Medium Surface Transducer (Figure 5.6)
    - Effective bandwidth 100 - 15000Hz
    - Listening frequency 50 – 2000Hz
    - Input voltage 3.7V
    - Resonant frequency 100Hz
    - 3W RMS, 5W peak

- Adafruit Large Surface Transducer (Figure 5.7)
    - Effective bandwidth 100 - 15000Hz
    - Listening frequency 50 – 2000Hz
    - Input voltage 6.3V
    - Resonant frequency 50Hz
    - 5W RMS, 10W peak

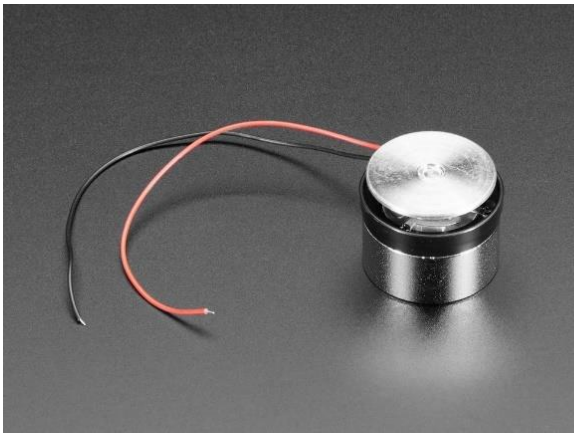


Figure 5.6: Adafruit Medium Surface Transducer.[16]

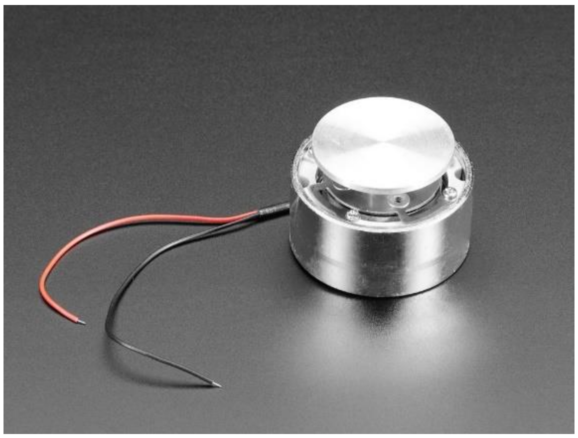


Figure 5.7: Adafruit Large Surface Transducer.[17]

When deciding on which one to use, it is important to take these specifications into account, as well as the frequency responses of the surface transducers. The human hearing ranges from 20Hz to 20000Hz, with the range deteriorating with age. As can be seen in Figure 5.8, not all frequencies are perceived at the same loudness when played at the same volumes. This graph has been the basis for the ISO standard ISO 226 until a revision in 2003, which sums up to a period of around 47 years.

[16] Source: https://www.adafruit.com/product/1785
[17] Source: https://www.adafruit.com/product/1784

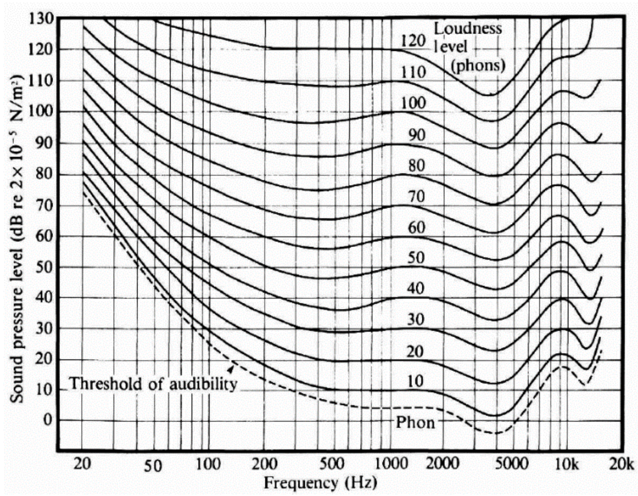


Figure 5.8: Equal loudness curves by Robinson and Dadson, 1956. [36]

Just like human ears have a different sensitivity to different frequencies, so do microphones and speakers. In an ideal world, the frequency response curve would be completely flat, so that every frequency would have the same perceived loudness to the listener. In reality, when two different frequencies are sent into a speaker at the same amplitude, the resulting output volume will likely not be the same. This is the case for almost any speaker. Figure 5.9 and Figure 5.10 show the frequency response curves for both of the Adafruit Surface Transducers.

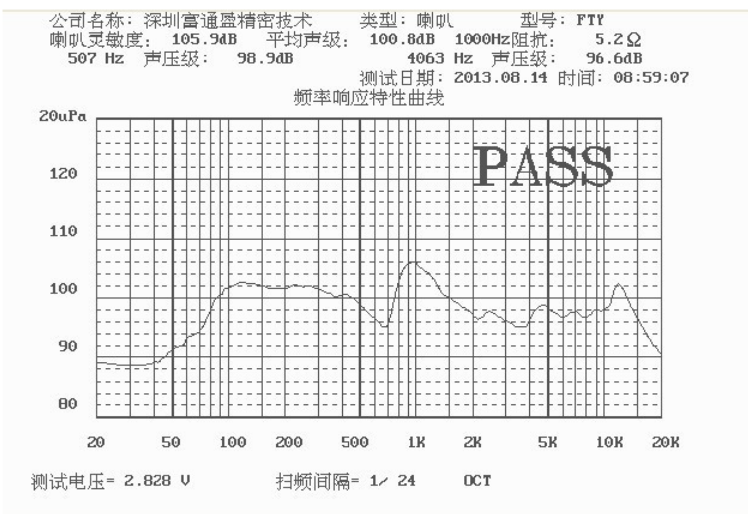


Figure 5.9: Adafruit Medium Surface Transducer frequency response graph [37]. *The large 'PASS' means that this transducer passed some sort of quality control, it has nothing to do with filtering.*

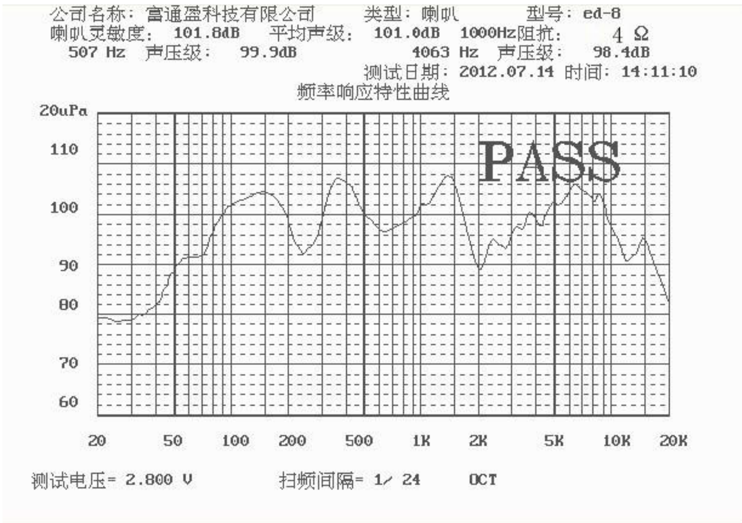


Figure 5.10: Adafruit Large Surface Transducer frequency response graph [38].

As can clearly be seen, neither of the surface transducer has a frequency response that can be called is anything close to flat. If one would want the curve to be flat, this could be accounted for by using a multi-banded equalizer. This allows for amplification of a certain band of frequencies and attenuations of others. The sound that needs to be played will get sent to the equalizer first, which can pass it on to the speaker after processing. This would theoretically make for a speaker that plays sounds of the same amplitude at each frequency at the same loudness.

However, even if it would be possible to obtain a completely flat frequency response curve, there are many other factors that influence sound along its way into your ears, such as reflections. These reflections can occur against things like the surface the speaker is placed on, or the walls of the room that it is placed in. For surface transducers, this is especially important to remember, as they rely on other objects acting as their diaphragm. Depending of the shape of this diaphragm, the sound signature that is eventually heard changes drastically. The speed at which sound travels depends on the medium that is used to transport it. That is why looking at Figure 5.9 and Figure 5.10 does not teach us anything. In this project, a variety of ceramic materials in a variety of shapes is tested, which all have different sound signatures when actuated. As long as there is not an extreme dip in the frequency response at the resonant frequency of the to-be sonified object, a completely flat frequency response is far from necessary. Of course, a frequency does need to be produced at the resonant frequency of the object but depending on the amplification factor of the sound that is fed into it, the surface transducer itself does not need to show an inherent resonance or even a flat behavior at the resonant frequency of the object. The aspect that matters most for creating an audio feedback loop is the loudness of the surface transducer, which makes the choice for the Adafruit Large Surface Transducer a logical one. However, both of them can be tested in experiments, as the resonant frequencies of ceramics are not known yet and either one may work better than the other for some objects.

5.3.3 Conclusion

Because in this project, one of the aims is to really let the characteristics of a ceramic object be the voice that the object produces, a surface transducer seems like a good choice for an exciter. Both the large and medium surface transducer by Adafruit will be tested in experiments.

5.4 Pickup

5.4.1 Introduction

The pickup is the part of a system that allows sound to be picked up, so that it can be used for further purposed, such as recording or further modulation by means of different audio effects for example. In the following paragraphs, a look will be taken at different means of picking up sounds.

5.4.2 Condenser microphone

The first thing that would come to mind for most people when they think of a means to pick up sounds is a simple microphone. Many different types of recording microphones exist, with the most important two being dynamic and condenser microphones. The main difference between the two is that dynamic microphones do not require external power, whereas condenser microphones do. This also makes for the fact that condenser microphones have a better frequency response than dynamic ones: condenser microphones tend to have quite a flat frequency response curve, such as the Studio Projects B1 (Figure 5.11), of which the frequency response curve can be seen in Figure 5.12.

Figure 5.11: Studio Projects B1 condenser microphone.[18]

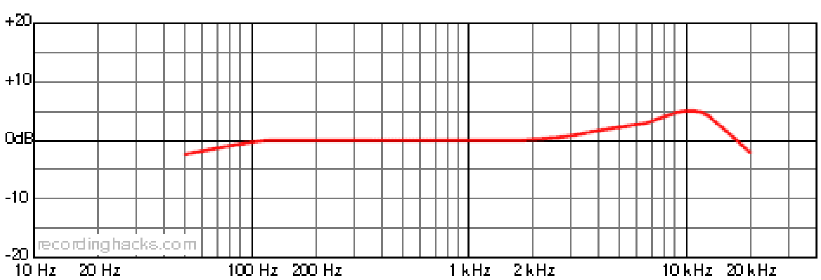


Figure 5.12: frequency response curve of the Studio Projects B1.[19]

5.4.3 Magnetic coil pickup

The main downside of using a regular microphone is the fact that it does not pick up just the part that needs to be picked up, it also picks up a lot of the timbre that the acoustics of the room it is used in can give to the sound. When looking at musical instruments, there are other ways sounds can get amplified. When looking at an electric guitar for example, the dry output signal is almost exclusively determined by the shape and type of wood, the strings and the pickups. This is of course before the output signal gets into contact with any effect pedals or amplifiers.

[18] Source: https://www.soundonsound.com/reviews/studio-projects-b1
[19] Source: http://recordinghacks.com/microphones/Studio-Projects/B1

The inside of a single coil guitar pickup is shown in Figure 5.13. When one of the six (ferromagnetic) strings of a guitar vibrates over one of the magnets, the magnetic field contracts and expands. This change of the magnetic field creates an alternating potential and with that, a signal.

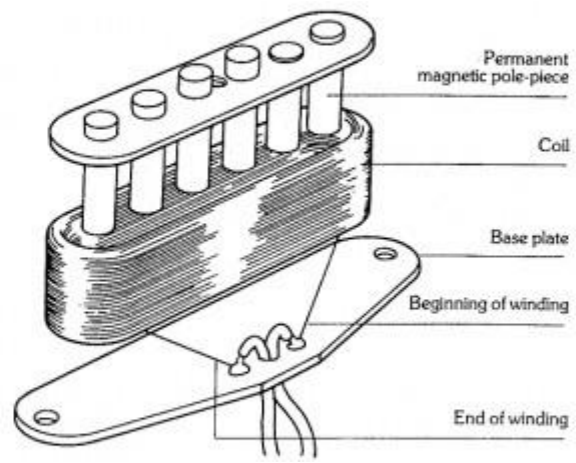


Figure 5.13: Inside of a single coil guitar pickup.[20]

Though ceramic materials may contain metals, this is likely not enough to produce a significant change in the magnetic field of the pickup. This means that a guitar pickup will not work for the amplification of ceramics as-is.

5.4.4 Laser microphone

However, principles similar to a guitar pickup can still be applied: a vibrating surface can still be detected. Lasers can be used for accurate proximity sensing purposes based on their reflections, but research has also indicated that by focusing a laser beam on an object at the right angle, it can be used as a microphone. This works because the changes in the reflection of the laser beam on the surface due to the movement of the plate, and therefore distance from the laser beam, creates a signal that can be amplified [39]. A diagram of an example setup can be seen in figure 5.14.

[20] Source: https://www.seymourduncan.com/blog/the-tone-garage/the-anatomy-of-single-coil-pickups

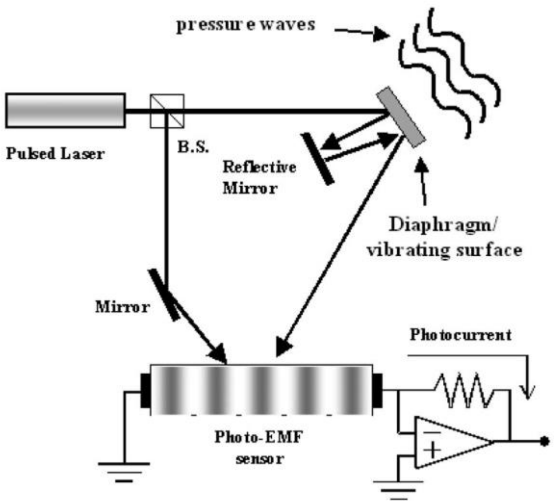


figure 5.14: Setup of a laser microphone [39].

The main downside of this technique is that the received signal will not be clear at all if the reflection angle is not exactly right. In a setup where objects need to be moved around a lot, such as in Maalman's installations, this might pose problems in getting consistent results from the laser microphone. Aside from all this, laser-microphone technology is still quite expensive and part of the challenge of this project is to still keep it affordable.

5.4.5 Hall effect sensor

A hall effect sensor (Figure 5.15) excels in its simplicity. The main working principle of the sensor is, as the name implies, the Hall effect, which basically makes it sense how strong the magnetic field around it is, to which its output voltage is directly proportional. The way it works can be seen more in-depth in Figure 5.16. There are two types of this sensor: threshold and linear. The threshold hall effect sensor acts as a switch as soon as a threshold in magnetic field is crossed, the linear hall effect sensor reacts linearly to a change in magnetic field around it. The linearity of the latter mentioned sensor makes it an excellent sensor for proximity sensing. Because Hall effect sensors can be completely analog, without any digital quantization or sampling required to use, they could theoretically be connected to an audio amplifier, with their output changing based on a vibration in the magnetic field within their range. In this project, a magnet could be attached to the rim of a porcelain plate with a hall effect sensor positioned above it (mounted on a separate, non-vibrating frame).

Figure 5.15: A linear Hall effect sensor.[21]

[21] Source: https://www.sparkfun.com/products/14709

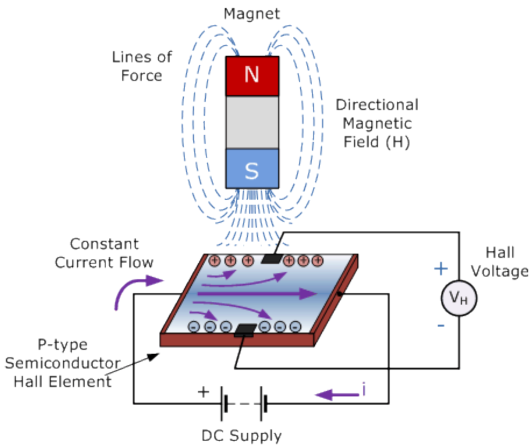


Figure 5.16: Diagram that shows how a Hall effect sensor works.[22]

The main downside of using this method, is that the output signal would be quite low depending on how much a porcelain plate would actually move when vibrating, and the fact that the placement of the sensor would have a huge impact on the characteristics of the sound, similar to what was mentioned previously about laser microphones. These characteristics may not make it the ideal candidate for audio purposes.

5.4.6 Contact microphone

5.4.6.1 Piezo

To capture the sounds from any object with as much detail as possible, a contact microphone is often used, as stated before. This is a microphone that is attached directly to the surface that it aims to capture the sound characteristics of. One type of contact microphone is a piezoelectric microphone or piezo. This type of microphone is relatively cheap and often found in acoustic guitar pickups and clip-on guitar tuners. On acoustic guitars, they are usually mounted by means of an adhesive such as double-sided tape, 3M, Blu-Tack, hot glue, epoxy or dedicated Shadow Glue for transducers. Though these means are sufficient for picking up sounds, Blu-Tack, hot glue and Shadow Glue are still thick, shock absorbent materials that come between the piezo disc and the to-be sonified ceramic plate. This lack of direct contact between the contact microphone and the surface that it should pick up the sound from, combined with the shock absorbent properties of the adhesive results in a to some degree attenuated version of the sound that is sent to the amplifier.

As for clip-on tuners or other devices that clip onto an instrument for pitch detection or amplification purposes, such as the Snark tuner (Figure 5.17 &Figure 5.18), consist of a solid body, through which the vibrations produced by an instrument travel, to which a piezo disc is directly attached. Between the clip and the body of the instrument, small pieces of rubber are places. This is meant to both get a better grip on the instrument, as well as to prevent any scratching damage to the surface of the instrument. A side effect of this rubber is that certain higher frequency components of the sound produced by the instrument will be absorbed and not make it to the piezo disc. For amplification purposes, this would pose a problem, as it is impossible to send an accurate representation of the sound to the speaker. For tuning purposes however, it is actually a desirable phenomenon as this means less high-frequency harmonics and noise present in the signal have to be analyzed. The fundamental frequency and therefore the pitch of the sound

[22] Source: https://circuits4you.com/2019/01/01/esp32-internal-hall-sensor-example/

will be more dominant to the tuner and thus easier to analyze and give visual feedback on using the display.

Figure 5.17: A Snark SN-8 clip-on tuner.[23]

Figure 5.18: The inside of a Snark clip-on tuner, with the location of the piezo disc clearly visible.[24]

Another way of mounting a contact microphone that came to mind during a brainstorm, is to use a magnet to keep the piezo in place and firmly pressed onto the surface that needs to be sonified. For example, a small magnet can be put on top of a piezo disc, which can then be put on the rim of a plate, which has a magnet underneath, causing the piezo disc to snap onto the rim of the plate because of the attraction between the magnets. As it turns out, this technique has actually been used before by sound artist Ed Devane. On the website for his company SoniPhorm, he sells a magnetic contact microphone (Figure 5.19), which he mentions being useful in scenarios where amplification of metal structures, such as gates or bridges or any other ferrous metal, is necessary [30]. The device has a 3D printed casing that has been filled with epoxy, a piezo disc and a strong magnet, which enables it to capture every last bit of detail as almost no damping occurs because of the lack of an absorbent material.

[23] Source: https://www.amazon.com/Snark-Super-Tight-Instrument-Tuner/dp/B00646MZHK
[24] Source: https://www.youtube.com/watch?v=Rq085KEhcdA

Figure 5.19: SoniPhorm magnetic contact microphone [40].

5.4.6.2 Accelerometer

Aside from piezo-based contact microphones, there are others types of contact microphones as well. One example of this is the Omnia by Heartsound which uses an accelerometer (Figure 5.20). Its focus is on amplifying acoustic instrument sounds in as much detail as possible, but they claim it works on every vibrating surface [41][42]. It comes with its own dedicated pre-amplifier, which can be tweaked to the liking of the user regarding gain, frequency response and volume. The pre-amplifier features a BNC connector for the contact microphone input and an XLR connector for the sound output. As both of these are balanced and electronically shielded connections, there will be very little interference present in the final signal.

Figure 5.20: Dedicated Omnia pre-amplifier on the left, accelerometer-based contact microphone on the right [41].

To attach the microphone to the surface that needs to be captured, a special type of adhesive paste of which they claim it will always stay sticky is pasted between the surface and the microphone. They do not mention anything about possible damping effects this adhesive may have, and the sound samples on the website display a lot of detail as well. They claim the use of an accelerometer rather than a piezo helps reduce feedback in acoustic instruments that are amplified electronically. For a device that is made for this purpose specifically, this is a great feature to have. However, audio feedback is seen as a desirable phenomenon in this project, so it might not be the best candidate for the job. This, as well as the fact that the device costs over €400,-, means that this is very likely not the best option for this project.

5.4.7 Conclusion

The best option to go with seems to be the piezo-based contact microphone, as out of all the mentioned options, it is by far the cheapest. Though they can be fragile, its low price tag means that if it were to break in any sort of way during testing, it would be very easy to get a replacement. As for mounting it, using a small magnet at either side of the plate to hold it down will ensure an undamped frequency response and an optimal sound transmission to the amplifier

5.5 Amplification

5.5.1 Introduction

To make and audio input and output come together, be it for simple amplification of a person speaking into a microphone or setting up an audio feedback loop that keeps on going without intervention, an amplifier is needed. In an audio feedback loop, the amplifier is the component that can most easily be swapped on the fly, as its positioning does not affect the resonance, unlike speakers and microphones. Different types of amplifiers will be discussed in the following paragraphs.

5.5.2 Audio interface

As mentioned before, audio interfaces are essentially external sound cards, used as both ADCs and DACs for computers. Because they are external devices, they do not necessarily have to take the space constraints that internal sound cards have into account. This allows manufacturers to use full-size 6.3mm jack ports, XLR ports and MIDI ports. Another advantage is that all these ports are very accessible as well, as an audio interface is often placed in a more convenient and accessible location than a traditional sound card, which requires the user to go the back of their desktop computer to operate it.

Typically, audio interfaces have a strong focus on transferring audio in a fidelity that is as high as possible, even more so than the sound card that is built into the computer in the factory, on which the manufacturer might have cut corners to save costs and run a profitable production line. Oftentimes the pre-amplifiers built in to the audio interface have a very flat frequency response, a very low total harmonic distortion (THD) and adjustable impedances. With all these advanced features, it will definitely get the job done, though it might be overkill.

5.5.3 DIY audio amplifier circuit

An audio amplifier can also be homemade relatively easily. The LM386 Low Voltage Audio Amplifier is a well-documented audio amplifier circuit with many pre-existing do-it-yourself projects all over the internet. One of these projects is the tin-can amplifier by Evan Kale (the one built specifically for this research can be found in Figure 5.21). It has a 6.3 mm mono jack input and output, a 2.5 mm barrel DC jack, an on/off switch and a volume control. It is built into a peppermint tin can for portability and rigidity, but the tin can also acts as a noise-reducing Faraday cage. The schematic can be found in Figure 5.22. As can be seen when comparing it to Figure 2.5, the tin can amplifier appears to be almost identical to the EBow, which means that it should work perfectly for sustaining a sound in a ceramic object.

Figure 5.21: A build of Evan Kale's tin can amplifier.

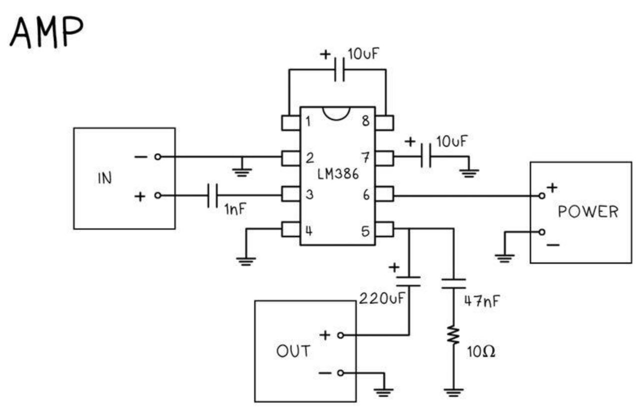


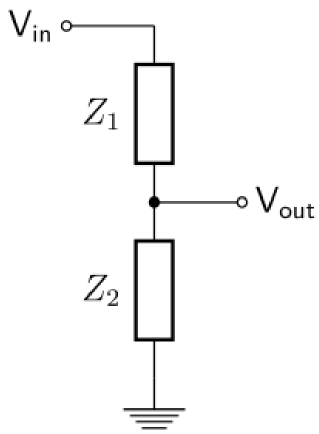
Figure 5.22: circuit of Evan Kale's tin can amplifier build.[25]

There are three basic types of the LM386 in DIP version (easier for prototyping), each with their own characteristics:

	Minimum supply voltage	Maximum supply voltage	Minimum output power	Typical output power
LM386N-1	4V	12V	250 mW	325 mW
LM386N-3	4V	12V	500 mW	700 mW
LM386N-4	5V	18V	700 mW	1W

What they all have in common is that they have an input impedance value of 50 kΩ. This amount is relatively little for an audio amplifier, which would ideally have an infinitely high input impedance. Applying the laws of a voltage divider (Figure 5.23), a low input impedance means that the signal is attenuated quite a bit before it can get amplified.

[25] Source: https://www.instructables.com/id/Post-Apocalyptic-Amplifier/



$$V_{out} = \frac{Z_2}{Z_1+Z_2} * V_{in}$$

Figure 5.23: A schematic of a voltage divider. In the case of an amplifier input, its input impedance is Z_2. This means that the lower its value, the lower V_{out} will be.

This can be remedied by adding an operational amplifier (Op-Amp for short) in buffer configuration (Figure 5.24). Without getting too in-depth about the inner workings of an Op-Amp, it tries to match the voltages of both its non-inverting (+) and its inverting (-) inputs. In buffer configuration, there are no other components required and the output simply needs to be connected to the inverting input. It allows for the creation of an input impedance in the order of MΩ's and very low output impedances. Both of those characteristics are ideal for increasing the input impedance of an audio amplifier. However, as no buffer was is in the EBow or in the tin can amplifier build, it might not be absolutely necessary to use one.

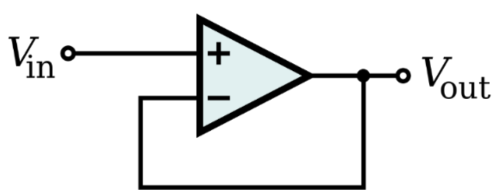


Figure 5.24: Operational Amplifier in buffer configuration.

All versions of the LM386 can also handle loads with an impedance of 4 to 32 Ω, which means that they will be compatible with both the large and the medium-sized surface transducer. They can also all be powered conveniently by means of USB (5VDC) or a compact 9V battery. They are all very similar in many other aspects too. However, as the final output volume of the circuit can simply be controlled by means of a logarithmic potentiometer placed at the input of the circuit anyway, selecting the version with the largest output power is the best choice.

5.5.4 Choice

As amplifiers can be swapped out the most easily out of all the components in an audio feedback loop, there will be experimented with both of the mentioned amplifiers. Swapping amplifiers allows for keeping the exciter and the pickup in the same place, so that if audio feedback occurred before the amplifier swap, it will likely occur again as soon as the new amplifier is plugged in.

5.6 Feedback control

5.6.1 Corpus-based method

Though an audible feedback sound should arise using the three aforementioned requirements for sonification, the sound will be quite hard to control, as mentioned in 2.1. The conclusion of the scientific literature review was that a DSP algorithm of some type had to be used. The best fit, as mentioned in [7], is the CataRT software, a corpus-based concatenative synthesis software that was developed as a plugin or external object for Max MSP by Cycling '74, a modular DSP environment software.

For feedback control using this software, [7] mentions using a bandpass filter in the feedback loop to determine which frequency component gets amplified the most and fastest in the transient state of the audio feedback, which is the target frequency of the resulting sound. This makes it more likely to be the fundamental frequency of the finally produced sound. They also mention AGC (automatic gain control) based on the difference between the RMS (root mean square amplitude) and the intended RMS amplitude. The more drastically the gain changes, the more directly the signal responds, though with oscillations. The smaller the difference in target gain is between the actual RMS and the intended RMS amplitude, the slower the signal responds (ramp), though with significantly less oscillation.

A set of target gain parameters, bandpass filter frequency combinations and their final sound output are recorded in clips consisting of 3 seconds of sound (1st second is discarded to only get the steady-state audio feedback sound), and 6 seconds of leftover room for decay. These recorded sounds are then analyzed, put into a large database also known as a corpus and used a lookup table. The audio files and control parameters are then imported into CataRT. The audio files can then be recalled from the lookup table via a Boolean query of sonic descriptors. The actual concatenation part of the software, which is the combining of segments of the audio samples into new sounds, is not used here. Only the corpus analysis, storage, data retrieval, and unit selection algorithm features are used.

In hopes of trying out this software, all required components were installed. However, even with the help of Diemo Schwarz, the developer of CataRT himself via email, it could not be gotten to work, not even with older versions of Max. Schwarz mentioned a standalone version of the software that was developed for Apple Macintosh, which would have been a valuable alternative, had it not been for the fact that access to this software requires a €1200,- subscription to all Forumnet ircam (Institute for Research and Coordination in Acoustics/Music) Centre Pompidou software. The University of Twente does not have this subscription, which means that this option is off the table. This means that the control of the audio feedback will simply have to be done by carefully changing the parameters mentioned in [17], though this was mentioned as very hard to do accurately: room reverberation for tempo control, ambient noise level for amplitude control and resonances of the acoustic environment for control of the timbre, which is the energy distribution over frequencies of a sound.

5.6.2 Analog method

Because audio feedback in itself is quite an unstable phenomenon as mentioned before, it would be very beneficial to have at least some way of steering the sound towards a certain tendency instead of letting it ring out at and clamp to a mostly random resonant frequency that may sound very dissonant in a given context, such as a musical one. Because the corpus-based method essentially works by firing specially filtered samples of a sound of a certain fundamental frequency into a feedback loop, it could be that this also works without the use of samples. Instead of the samples, a simple sine wave on an arbitrary waveform generator will be used in experiments. The arbitrary waveform generator that will be used in this specific case is the HP 33120A function generator (Figure 5.25).

Figure 5.25: HP 33120A function generator.

5.7 System requirements

These requirements are similar to the ones mentioned in the ideation phase. However, they will be further defined here, and separated into functional and non-functional requirements. The functional requirements are in **bold** and will be quantified as much as possible.

Musts:

- **Plug & play for existing Lost & Found Orchestras**
    - This means that safe and standard voltages need to be used, likely in the range of 5 to 12 V. It also needs to be capable of being interfaced with Ableton Live, be it for providing the control signal or via MIDI control of different aspects of the amplifier.
- **Use the characteristics of the object itself in the produced sound**
    - The produced sound in the audio feedback loop should not be inherent to which specific excitation, pickup and amplification components are used, though these can be used to filter out certain frequencies. To check if the feedback loop is producing a sound that is based on the characteristics of the object, the pitches that are produced when simply tapping the object need to be compared to the sounds that are produced when an audio feedback loop is applied to the surface.
- **Accurate pitch control**
    - Because objects almost all have multiple inherent resonant frequencies, it needs to be possible to address more than one of these resonances. This can be aided by using a control signal before setting up an audio feedback loop.
- **No damage to the ceramic object**
    - This means that there should be no aggressive, undamped impact by means of an object more rigid than the to-be sonified object applied to the surface of the to-be sonified object. Possibly damaging impact by a vibrating surface transducer for example can be prevented by putting a dampening material such as rubber between the two. However, this layer should not be too thick, as it might then dampen too much of the higher frequencies.

Shoulds:

- Affordable
    - Given that the user already owns a computer and a ceramic object, it should not cost more than €75,- to make replacing components that may have broken during testing not a large dent in the finances of the user.
- **Expandable**
    - The system should remain modular enough to swap out components such as amplifiers, different surface transducers or other exciters and pickups. This is why the system will not operate at a too high wattage, as this may damage some components.
- **Easy to move around and remove**
    - To discover as many resonant frequencies as possible in objects, it is crucial to be able to move around the components, especially the pickup

Coulds:

- **Intuitive user interface**
    - This can already be done simply by making the system interfaceable with Ableton Live, but also by means of a separate display on the amplifier itself, for example. This display could then indicate the current positions and values of the potentiometers.
- Portable
    - To make the system installable almost anywhere to sonify as many objects as possible, it would be an added convenience if it could fit into a backpack.

6 Realization

6.1 Introduction

In this chapter, the components selected in the specification phase will be put together in the process of building prototypes. Different setups of the prototypes will be built to get insights in different aspects of this research project. However, there are a few components that they will all have in common. Those components will be treated here and shown in the different configurations, both graphically and in block diagrams. Lastly, the amount of voltage that can be used across the surface transducers is calculated, so that the experiments can be executed.

6.2 (Pre-)amplification

The most critical part of using a contact microphone is that the pre-amplifier that it is connected to, specifically its input impedance. Something that happens often is that a contact microphone lacks lower frequencies in the output signal because of an impedance mismatch that causes a high-pass filter effect [43]. In musical instruments, four ranges of impedances are used. These are low (0Ω-1kΩ), medium (10kΩ-100kΩ), high (electric guitar & bass, 1MΩ) and ultra-high (1MΩ+). The ultra-high range belongs almost exclusively to piezo pickups, which have an output impedance of around 10MΩ. This means that an appropriate pre-amplifier needs to be used in testing. The choice available at hand is a Focusrite Saffire PRO 14 audio interface (Figure 6.1), which can be connected to a computer via FireWire, but can also operate standalone when just connected to a 12V DC power supply.

Figure 6.1: Focusrite Saffire PRO 14 FireWire audio interface.[26]

The Saffire has 6 analog audio inputs, which have the following impedances:

- 2x microphone input, 2kΩ
- 2x line input, >10kΩ
- 2x instrument input, 10MΩ

This means that an instrument input has to be used to avoid loading effects on the piezo and get the full range of sounds that it can produce. In this case, the impedances are a perfect match. However, if the input impedance of the Saffire would have been higher, no problems would occur either. The only

[26] Source: https://www.gear4music.com/Recording-and-Computers/Focusrite-Saffire-Pro-14-Firewire-and-Thunderbolt-Audio/DSV

downside would be that the sound from the piezo would be somewhat quieter, but it would still have a similar frequency response. This is the case for all impedances: a lower impedance output can work fine on a higher impedance input, albeit quieter, but a higher output impedance connected to a lower input impedance creates loading effects and distorts the frequency response.

For testing purposes, the Focusrite Saffire PRO 14, or any other decently advanced audio interface would seem like it would do the trick for the regular amplification part of the feedback loop as well. It will likely give the most consistent result as the only hardware errors that can occur are derivative of production errors, which is very unlikely for a good quality machine-manufactured piece of electronic equipment.

However, this is quite an expensive route to take: first of all, a computer is required to set up the audio interface, regardless of whether it will be used in conjunction with software or standalone. For the Saffire specifically, the computer also needs to have a port that is compatible with FireWire 400 or higher, which most computers do not have any more nowadays. This usually means a Thunderbolt-to-FireWire adapter or a FireWire ExpressCard needs to be bought as well. Aside from all this, an audio interface is quite a large and vulnerable piece of equipment that is not easy to travel around with as it is not designed to take a beating and to be repaired easily. Though plenty of companies do make pocket-sized audio interfaces meant for use during travel, these are often still very expensive.

During testing, it turned out not to be possible to get a feedback loop started using the Focusrite Saffire PRO 14. Troubleshooting was done for every aspect of the feedback loop: the piezo disc and surface transducer were swapped out, an extra amplifier was added into the loop for extra gain and the gain and output volume settings of all the different inputs and outputs of the Saffire were maxed out. After all this, there was still not a feedback sound to be heard. Soon after, it was discovered that the Saffire has built-in feedback prevention measures. Normally, in a professional audio processing context, such as a studio, this is a great feature to have on your equipment and will save you many headaches. In this specific context however, these measures rendered the Saffire useless, as there was no way to turn the feedback prevention measures off. This means that using the Saffire is off the table and the earlier mentioned tin can amplifier needs to be built and used for all experiments. This will result in a lower-than-ideal input impedance, but it should work just fine.

6.3 Shock mount design

To ensure that the ceramic object can optimally resonate while making surrounding objects vibrate with it in the process, the ceramic object has to be isolated from the ground as well as possible. To accomplish this, the setup from Figure 6.2 was built, inspired by a condenser microphone shock mount (Figure 6.4). It was designed in CorelDRAW X7 and constructed from laser cut 4mm poplar multiplex (Figure 6.3). There are holes in the frame to allow cut-open rubber bands to be pulled through. These rubber bands allow for suspension of a round platform on which the ceramic plate can rest. One drop of hot glue was placed on the bottom of each corner of the frame to keep the plate on the platform and to keep the plate stable, an additional platform was attached on top of the other. This additional platform has a diameter of exactly 145 mm, which is also the diameter of the underside of the plate, so that the plate cannot slip off.

Figure 6.2: shock mount test setup for ceramic plate.

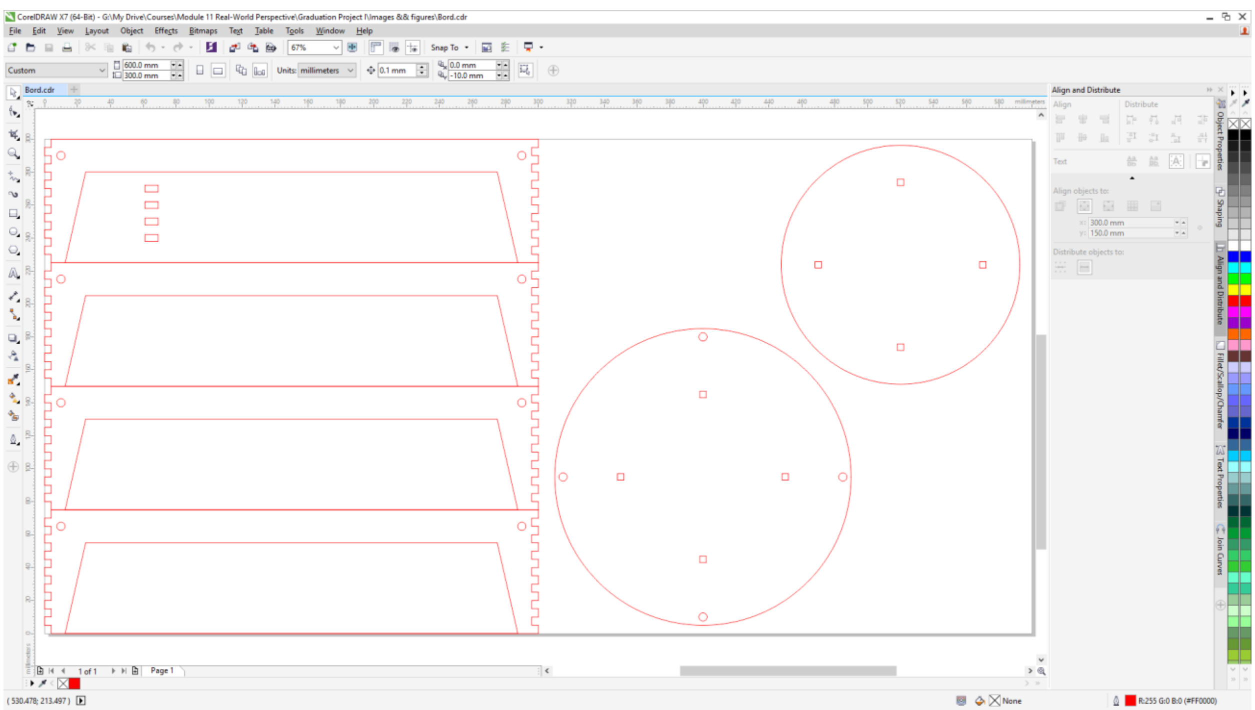


Figure 6.3: CorelDRAW X7 design file of the shock mount.

Figure 6.4: A condenser microphone shock mount.[27]

However, it would later turn out that this additional platform prevented the plate from resonating optimally. This is why three drops of hot glue were put at 120° intervals on the larger platform, so that the plate would not touch the upper platform with its bottom, minimizing the contact area to only three tiny points. Three points is the minimum amount of points for the ceramic plate to stand stable, as it would wobble if placed onto two points and four points would create unnecessary contact area.

6.4 Setups

The aforementioned shock mount will be used in all the experiments, just like the ceramic plate, the piezo disc, the surface transducers and the tin can amplifier. In the experiments, multiple different configurations of audio feedback loops will be used. In order to keep a clear overview of which setup is used when and where and to not get confused by pictures, diagrams are a very useful tool.

6.4.1 Experiment 1

The first experiment is further explained in Appendix I, the measurement setup is displayed graphically in Figure 6.5 and as a block diagram in Figure 6.6. The focus of this experiment is on discovering the resonant frequencies produced in the audio feedback loop that goes through the plate.

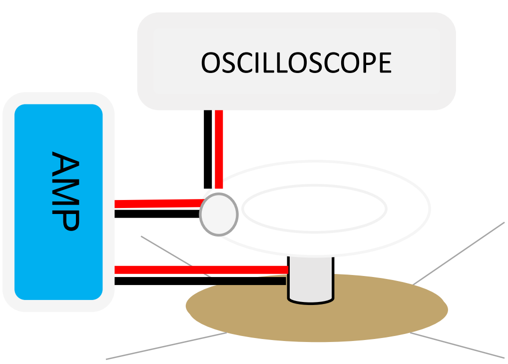


Figure 6.5: Graphical overview of the block diagram of the setup in experiment 1.

[27] Source: http://www.soundtown.com.au/products/isk/isk-shm-20-large-condenser-microphone-shockmount/

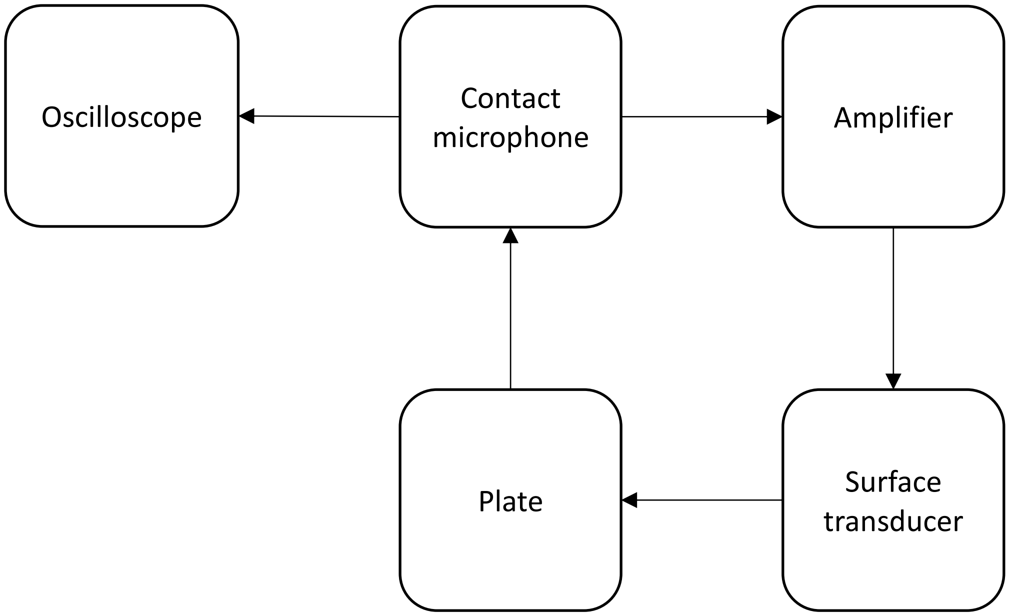


Figure 6.6: Block diagram of the setup of experiment 1, the arrows signify the travel of sound.

6.4.2 Experiment 2

The second experiment is further explained in Appendix II, the measurement setup is displayed graphically in Figure 6.7 and as a block diagram in Figure 6.8. The focus of this experiment is on creating a bode plot of all the resonances of the plate.

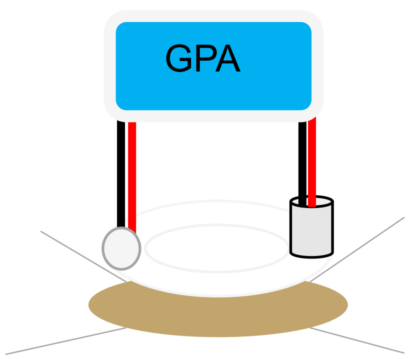


Figure 6.7: Graphical overview of the block diagram of the setup in experiment 2.

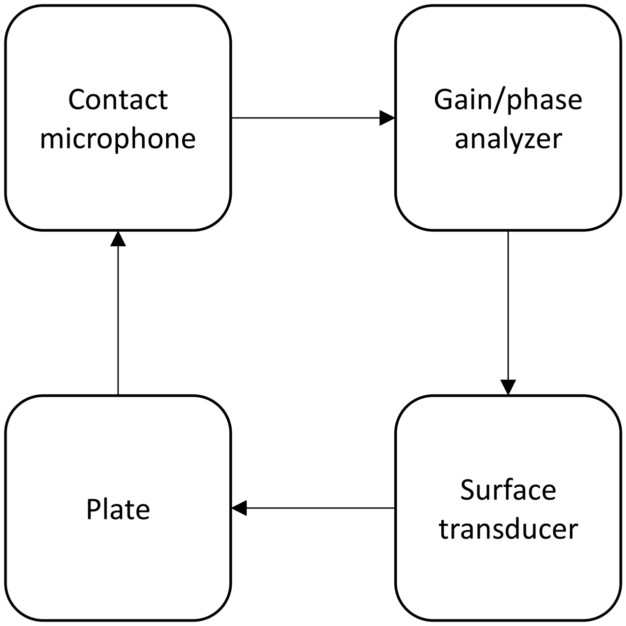


Figure 6.8: Block diagram of the setup of experiment 2, the arrows signify the travel of sound.

6.4.3 Experiment 3

The last experiment is further explained in Appendix III, the measurement setup is displayed graphically in Figure 6.9 and as a block diagram in Figure 6.10. The focus of this experiment is on inputting a control signal into an audio feedback loop to change the pitch of the resonance.

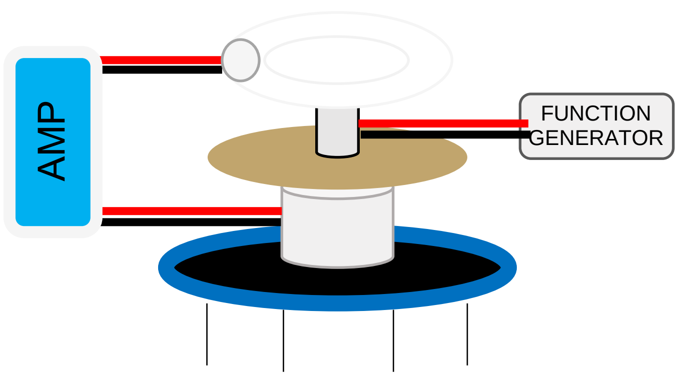


Figure 6.9: Graphical overview of the block diagram of the setup in experiment 3.

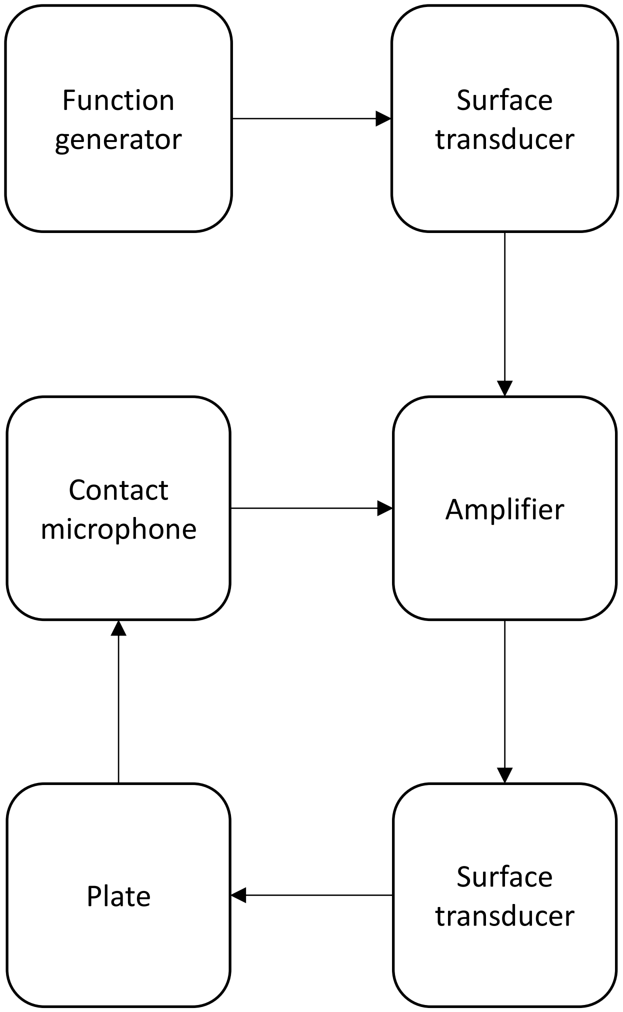


Figure 6.10: Block diagram of the setup of experiment 3, the arrows signify the travel of sound.

6.5 Characteristics of surface transducers

The means by which the plate will be excited and therefore be capable of producing sounds has been determined to be a surface transducer, as this is the best documented option for getting surfaces of most materials to resonate and produce sounds. This is necessary as there is an almost infinite amount of possible ceramic compositions. This was already shown in [34], where Maalman used artifacts made of three types of ceramics. Some contained a metal to influence the properties of the ceramics. Both the Adafruit Medium Surface Transducer and the Adafruit Large Surface Transducer will be assessed here.

The Adafruit Medium Surface Transducer has a wattage of 3W an impedance of 4Ω. The latter is the load that a speaker places on the amplifier it is connected to. Because the output of an amplifier is an AC signal and AC resistance is also called impedance. The impedance value mentioned on a speaker, or 'nominal impedance' is usually a value of either 4Ω, 8Ω or 16Ω. As can be seen in most speaker datasheets however, the mentioned impedance actually varies based on the frequency of the input.

The wattage rating means the maximum amount of watts a speaker can handle. For guitar amplifiers, a speaker rated at least twice the amplifier wattage is typically recommended. The nominal impedance Z of a speaker can be thought of as resistance R. Ohm's Law:

$$V = Z * I$$

$$P = V * I$$

$$P = V * \frac{V}{Z}$$

$$P = \frac{V^2}{Z}$$

For the medium transducer, this means the following:

$$P_{RMS} = \frac{V^2{}_{RMS}}{Z}$$

$$V_{RMS} = \sqrt{3 * 4} = 2\sqrt{3} = 3.46V$$

Because a sine wave is used, this means that the V_{PP} that should be used is calculated as follows:

$$V_{PP} = V_{RMS} * 2\sqrt{2} = 2\sqrt{3} * 2\sqrt{2} = 9.80V$$

And for the large surface transducer:

$$V_{RMS} = \sqrt{5 * 4} = 2\sqrt{5} = 4.47V$$

$$V_{PP} = V_{RMS} * 2\sqrt{2} = 2\sqrt{5} * 2\sqrt{2} = 4\sqrt{10} = 12.65V$$

Using these calculations, the function generator can be set up for use on the surface transducers. Once its output is set to HIGH Z (high impedance), testing can begin. The experiments that were done can be found in Appendices I to III.

7 Evaluation

7.1 Introduction

In this chapter, the results of the experiments that can be found in the appendices of this paper will be summarized and discussed. Furthermore, the non-functional requirements as set in the ideation phase and the functional requirements as set in the specification phase will be evaluated and checked to see if they have been met, to see if the project can be counted as a success or not.

7.2 Discussion of experiments

7.2.1 Experiment 1

See Appendix I. The resonance the audio feedback sticks to is not predictable at all, though it does seem to have a slight preference depending on which surface transducer is used. The large one prefers higher resonances, whereas the medium one sticks to lower resonances more often. It also seemed that the lower resonance of 750 Hz had a waveform that looked like a perfect sine wave, whereas the higher resonance showed some deformations, likely due to higher harmonics of this resonance also being quite prominently present in this signal, indicating that there are more resonant frequencies than just the two mentioned ones. Nevertheless, the frequencies both surface transducers stuck to do seem to be the dominant frequencies that can also be heard when tapping the plate as it is suspended, which confirms that the audio feedback loop will find the resonant frequencies of the plate, rather than the resonant frequencies of the piezo and surface transducer. The fact that the surface transducers have different preferences when it comes to frequencies they stick to in a feedback loop is something that can still be improved upon.

7.2.2 Experiment 2

See Appendix II. In order to further explore the other resonant frequencies that the dinner plate has to offer, it was decided that a having a bode plot of the frequency response was useful to have. For this purpose, a gain/phase analyzer was used. After many measurements, of which none seemed to even have the originally determined resonant frequencies of 750 Hz and 1730 Hz, it was discovered that placing the surface transducer on the edge of the plate heavily influenced its resonant frequencies. Sadly, all measurements had already been made when this was discovered and because of the long time it takes to make these gain/phase measurements and the limited availability of the laboratory the gain/phase analyzer was located in, it was not possible to re-do the measurements with the surface transducer at the bottom of the plate. This is most definitely a recommendation for further research, as an accurate bode plot was in fact obtained, though not with the right resonant frequencies in it.

7.2.3 Experiment 3

See Appendix III. In the end, the prototype worked: a steady and audible feedback loop occurred and when the surface transducers were stacked in a specific way, the pitch was controllable as well. For this pitch control however, a sine wave with a frequency close to one of the resonant frequencies had to be input first, after which the amplifier could be turned on, after which the control signal could be stopped. Though it did work accurately, it is quite a hassle to repeat this process every time you want to change the pitch. Luckily, it should be relatively easy to automate this process using an audio interface, MIDI commands and short waveform samples from Ableton Live and an Arduino. The Arduino could turn the amplifier on or off with a relay or mute and slowly turn up the volume with a digital potentiometer. This can be done using MIDI commands from Ableton Live. The control signal can then also be provided by Ableton Live.

7.3 Requirements check

It is important to keep the goals of the project in the back of your head throughout all of the development process and to check which aspects are perfectly taken care of in your design, but also which aspects still leave something to be desired. All in all, after evaluation of all experiments, it can be concluded that the following requirements were met to some degree:

Musts:

- Plug & play for existing Lost & Found Orchestras
    - Not quite there yet, though it should be relatively easy to do so by simply adding an Arduino and either a relay or a digital potentiometer to the amplifier. However, the chosen exciter and pickup are in fact compatible with almost any surface, so that part of the requirement is met.
- Accurate pitch control
    - Met, though not all the resonant frequencies are known yet. Those can easily be found out after new measurements using a gain/phase analyzer.
- No damage to the ceramic object
    - Met, absolutely no damage whatsoever has been done to the plate

Shoulds:

- Affordable
    - The entire setup costs around €60,-, with the most expensive part being the surface transducers, so affordability has been achieved
- Expandable
    - The system is almost completely modular and almost every component can be swapped out, so this requirement is definitely met.
- Easy to move around and remove
    - The contact microphone is mounted by means of magnets and the surface transducer can be mounted magnetically as well using an M5-threaded magnet, which are all very easy to move around, yet also capable of staying in place when left at rest.

Coulds:

- Intuitive user interface
    - Right now, the only instances of a user interface being present are the one of the function generator, which is very intuitive as it is a finished commercial product, the one on the amplifier, which is very intuitive only consists of a volume knob and a power switch, and the moving around of the exciter and the pickup, which yields quite unpredictable results. To make for a more intuitive user interface, the exciter and pickup should be kept in the same place when making adjustments using the other two interfaces. This requirement is partly met.
- Portable
    - Though the function generator does fit in a backpack, it can and should definitely be replaced by a laptop running Ableton Live. The shock mount and trampoline together could be replaced by a slightly taller shock mount to prevent anything from touching the ground because of weight and creating unwanted resonances, which will have to be

made. Aside from this, all the components fit in a backpack, so this requirement is as good as met.

To summarize, all the requirements that have not yet been met should be relatively easy to meet using some slight improvements to the system. After all experimentation, a working prototype finally came out. A video demonstration of the prototype in action can be seen by entering the following link into an internet browser:

https://drive.google.com/file/d/1cjNhRFC-aboTYds2x_oyb8ue4jsjjz-E/view?usp=sharing

8 Conclusion

After all theoretical and empirical research, the first thing that can be concluded is the fact that a sustained, non-percussive sound be produced from a ceramic object, in which only the characteristics of this specific object determine the tone can be obtained by means of an on-surface audio feedback loop. This audio feedback loop is produced by means of 1: a large surface transducer to output sound, 2: a contact microphone to pick up the sounds produced by the surface transducer, and 3: an audio amplifier circuit to amplify the output of the contact microphone, which is then output to the surface transducer again, completing the loop. This audio feedback loop will automatically produce an audible oscillation. The produced tone is at one of the resonant frequencies of the sonified object and is therefore completely reliant on the characteristics of the specific object, rather than at a resonant frequency dependent on the frequency responses of both the contact microphone and the surface transducer.

Which resonant frequency of the sonified object gets addressed in the audio feedback loop turns out to be heavily influenced by the choice and placement of the source of excitation and the pickup. In the case of the experiments with the dinner plate, the ideal pickup position for the contact microphone for getting a pleasant-sounding oscillation was exactly in the middle of the rim of the plate, kept in place by means of light, yet strong magnets. As regards the surface transducer, it does produce an audio feedback loop more easily when mounted at the other side of the plate using a magnet. However, the extra weight on the rim of the plate brings it out of balance in the shock mount and completely changes the resonance of the construction and therefore does not accurately showcase the characteristics of the sonified object.

Because in a plain audio feedback loop it is not possible to accurately control the pitch, a measure to enhance user control was developed. Rudimentary pitch control was achieved by placing a second surface transducer directly underneath the middle of the dinner plate, which has to output a waveform at one of the resonant frequencies of the plate before the audio feedback arises, i.e. turning on the amplifier. After turning on the amplifier, the audio feedback will clamp its pitch to the resonant frequency that is being played over the second surface transducer. The pitch that was input as a control signal through the surface transducer will continue indefinitely after ceasing playback of the control signal. To change the pitch, the audio feedback loop needs to be broken, i.e. the amplifier needs to be turned off, after which a new control signal can be fed into the plate via the surface transducer.

For objects of shapes different than dinner plates, the locations of the contact microphone, the feedback surface transducer and control surface transducer need to be investigated for each object separately by means of trial and error. The resonant frequencies can be determined by suspending the object, tapping it and listening to and measuring the most dominant of the produced frequencies.

There are a few recommendations for small improvements that could be made to the different setups of the different experiments. The first one that definitely needs to be made is better shielding of the cables used in the setup. The amplifier casing is already shielded, but the unbalanced cables can still pick up a lot of interference, which could have influenced measurements.

For the excitation part of the feedback loop, different surface transducers with a flatter frequency response could be used to see what their impact is on both a naturally occurring audio feedback sound and a controlled one in a stack. Aside from this, an arbitrary waveform generator has been used up to this point to supply the feedback loop with a control signal. This could also be done by simply importing a short fragment of a waveform at the target frequency into Ableton Live, which can then be output to the smaller surface transducer in the stack via the output of an audio interface. This would make the setup more portable and less cluttered, as it saves the usage of one whole separate device.

To ensure the flatness of the frequency response of these transducers, an audio amplifier with a flatter frequency response could also be used. This same amplifier could also have a higher input impedance to prevent loading effects of the contact microphone, which could be done by means of a simple Op-Amp in buffer configuration.

For the pickup part of the feedback loop, there is not much to be improved upon when compared to the used setups. What could be tried is using the Omnia by Heartsound that was mentioned before, as it will give a more faithful reproduction of the produced sounds. Its included matching pre-amplifier also allows for a lot of customization by means of gain and filter adjustments, which could accentuate the relevant frequencies and reduce those less relevant for better capturing the sonic essence of the ceramic object.

Lastly, there are many different types of other materials than ceramics that could be used for the purpose of sonification, such as glasses, crystals and metals. Using these different materials could expand the sonic palette of a sound artist even further.

9 Future Work

Aside from the previously given recommendations, there is potential for further studies in this project as well. What was already briefly touched upon was the usage of materials other than ceramics to see what the effect of that would be on the resulting sound. Aside from these rigid materials, it could be interesting to study the behavior of the resonances of flexible materials. Recent technological developments have made the production of flexible ceramics possible at a small scale [44]. Watching for further development of this technology is worthwhile, as it could allow for changing resonances of a ceramic object on the fly, which would be a very interesting sound art tool. However, instead of waiting for this technology, sheets of different types of metal could also be bent to change their resonant frequencies, so studying that phenomenon can be done in the meanwhile.

Secondly, options for an amplifier circuit dedicated to audio feedback control could be explored. As of yet, there are two separate surface transducers required in order to gain control over the audio feedback frequency. If the building of a dedicated audio feedback control amplifier were to be done right, it would be one separate device with one audio input and one audio output to set up the feedback loop, with the control signal being selectable by means of a user interface on the amplifier itself or maybe even via MIDI or Ableton. This would reduce the amount of required appliances and make the entire setup easier to move around and customize. As the parameters of an amplifier, such as the equalizer settings, can be tweaked to influence the resulting sound, an amplifier with digital potentiometers for all the equalizer parameters (treble, mid, bass, presence) could be built. The digital potentiometers can then be controlled by a user through a computer and a microcontroller such as an Arduino.

Thirdly, while the corpus-based method of feedback control has been proven to work on guitar feedback [7], it was not managed to get it to work in the system produced in this research, mostly due to budgetary limitations. If a license for CataRT were to be acquired, the experiments as performed by Ferguson, Johnston and Martin could be conducted on the setup that was created for this research, albeit with slight adjustments. This method could provide the current system with a more natural sounding control signal, as it is simply based upon samples of the sound of the object of which the audio feedback needs to be controlled.

Lastly, new experiments could be done with the gain/phase analyzer. This time, the surface transducer would have to be located at the bottom of the plate instead of on the rim. This would produce an accurate bode plot of all the resonant frequencies of the plate, which is useful for determining which other frequencies could be used as a control signal.

Appendix I

Experiment 1: determining the resonant frequencies of a feedback loop

Introduction

The goal of this experiment is to discover the dominant frequencies in the audio feedback loop that occurs in a ceramic object.

List of components

Electronics

- Tektronix TDS 2022B oscilloscope
- Tin can amplifier & wiring
- 15mm piezo disc
- Adafruit 4Ω 3W surface transducer
- Adafruit 4Ω 5W surface transducer
- Probe

Other

- dinner plate
- small piece of rubber
- 3x small magnet
- Shock mount

The diagrams can be found in Figure 6.5 and Figure 6.6.

Method

Start by mounting the piezo disc on the rim of the dinner plate by placing it on top of it and clamping it in place with two magnets by putting one of the small magnets on top of the piezo and the other one underneath the rim (Figure A1.0.1). The surface transducer will be placed under the middle of the plate (Figure A1.0.2). The surface transducer on its own would vibrate the plate off of itself, so a piece of rubber was put between the surface transducer and the plate. This rubber also provides some additional grip.

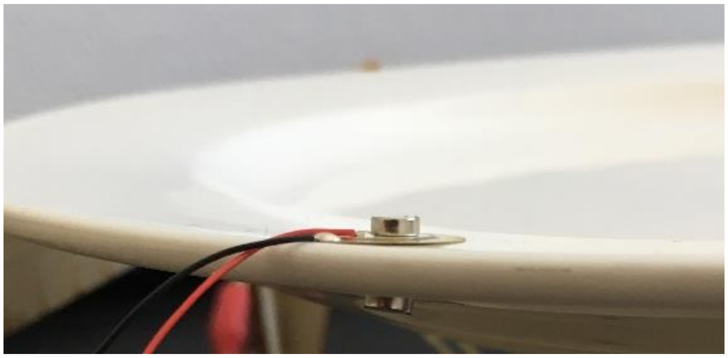


Figure A1.0.1: Mounting the piezo disc mounted on the rim of the dinner plate.

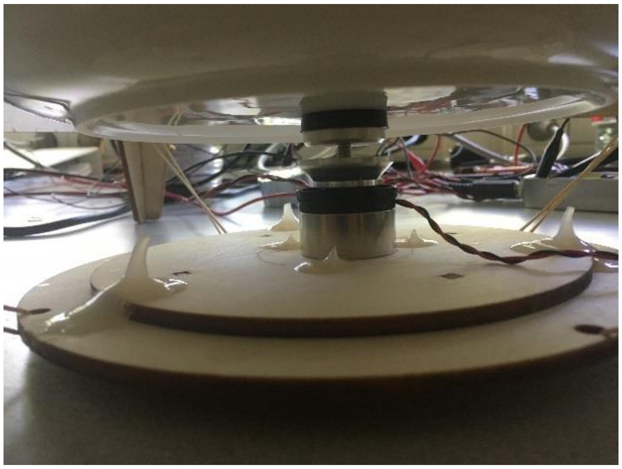


Figure A1.0.2: Placing the surface transducer under the plate.

Results

First, when lightly tapping the plate, the inherent resonances can be heard. These frequencies are measured using IK Multimedia’s UltraTuner iPhone application. They turned out to be around 765 Hz and 1730 Hz. These frequencies will be referred to as the natural resonances from now on. Then, the amplifier was turned on to create a natural audio feedback loop. When connected to an oscilloscope using a probe, it was revealed that the medium surface transducer made for a sound with a frequency of around 750 Hz. When swapped for the large surface transducer, the produced frequency was around 1730 Hz, as can be seen in Figure A1.0.3 and Figure A1.0.4 respectively. Note that the pictures were taken with a phone camera instead of using the built-in screen print functionality, because it could not be gotten to work.

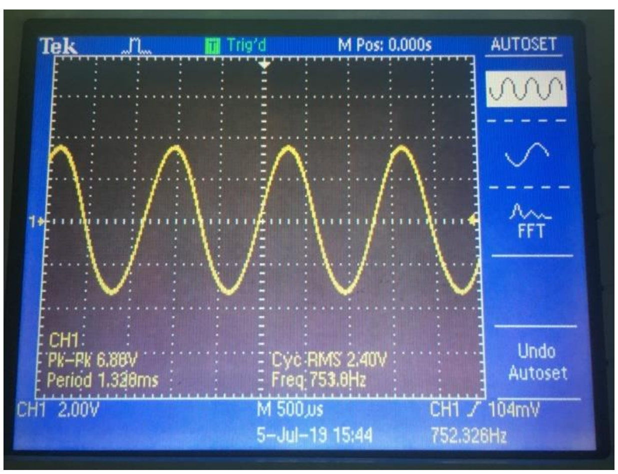


Figure A1.0.3: Oscilloscope view of the produced audio feedback using the medium surface transducer.

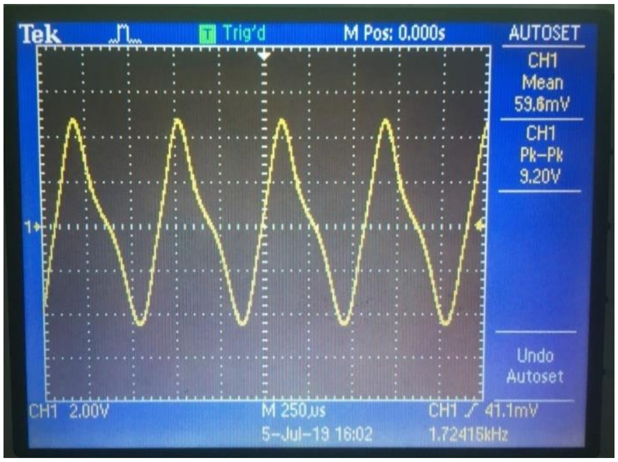


Figure A1.0.4: Oscilloscope view of the produced audio feedback using the large surface transducer.

Conclusion

It can be seen means that both of the dominant main natural resonances of the plate are addressed by using the two different surface transducers when a feedback loop is applied to the surface of the plate.

Appendix II

Experiment 2: resonant frequencies & gain phase analyzer

Introduction

List of components

Electronics

- HP 4194A Impedance/Gain-Phase Analyzer
- 15 mm piezo disc
- 35 mm piezo disc
- Adafruit 4 Ω, 3 W surface transducer
- Adafruit 4 Ω, 5 W surface transducer
- Probe
- BNC-to-banana plug adapter
- BNC splitter
- 2x banana cable
- Alligator clips

Other

- 1x porcelain dinner plate
- Magnet with M5 screw thread
- 2x small magnet
- Small metal object (3 cm diameter)
- Shock mount
- Piece of rubber

The diagrams can be found in Figure 6.7 and fFigure 6.8.

Method

The goal of this experiment is to get a bode plot of the frequency response of one dinner plate and Daniël Maalman's porcelain artifact. This will show in which spectra the resonances are located, but this time more precisely and with a better visual representation than in the previous experiment. To accomplish this, a different machine will be used, which is the HP 4194A Impedance/Gain-Phase Analyzer. Its output will send a logarithmic sine sweep from 100 Hz (as low as it goes) to 20000 Hz through the surface transducer, which is attached to the plate the same way as in the first experiment. The plate, placed on its suspension system, will vibrate and send these vibrations to the piezo that is also attached in the same way as in the first experiment, which is connected to the reference input of the HP 4194A via a probe. The same thing will be done to the artifact. The impedance analyzer draws up a bode plot, of which conclusions about the resonant frequencies will be drawn. Note that the bode plots are not screen captures but rather actual photos of the display, as the device is from 1985 and does not have a conventional means of making screen prints

An example test setup can be found in **Error! Reference source not found.**. The output of the HP 4194A is connected to the surface transducer using a BNC-to-banana adapter and to banana clamps. The surface transducer is screwed in to a magnet with M5 threading. The magnet attracts a small metal disc located at the underside of the plate, causing the surface transducer to clamp to the plate tightly to prevent any rattle. At the other side of the plate, the piezo is connected to the plate using a different magnet of the same kind, with a metal washer at the underside of the plate as well. This setup is meant to put as much ceramic material between the piezo disc and the surface transducer as possible, so that the produced bode plots will hopefully reflect the characteristics of the ceramic material best. For the tests with the large surface transducer, two of the legs on the shock mount are raised up slightly to prevent the bottom of the plateau the plate is resting on from touching the surface the frame is standing on. Carefully, it is also made sure that the plate does not touch the rubber bands that suspend the platform it its standing on, nor the smaller feet that are meant for the artifact to stand on later.

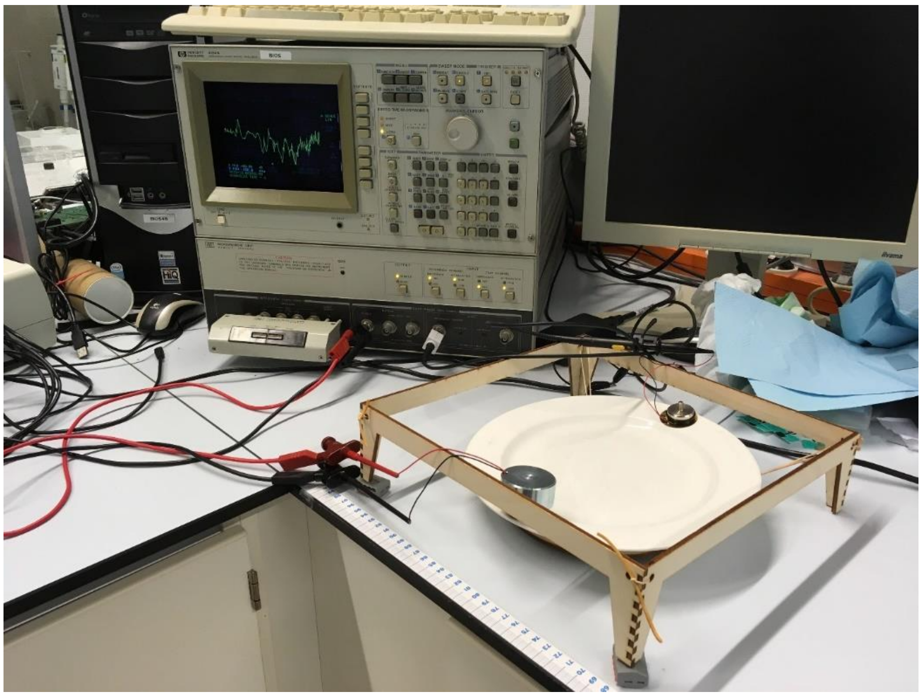


Figure A2.0.1: One of the used test setups, this iteration with a large surface transducer and a large piezo disc.

On the HP 4194A, the oscillation level is set to 1.26V or +15dBm, which is the maximum amplitude possible on the *single* output. A logarithmic frequency sweep is done from 100Hz to 20kHz, as 100Hz is the lowest possible frequency with the gain-phase analyzer and 20kHz is the highest possible audible frequency for the human ear, as mentioned earlier. The integration time is set to 'long' and the averaging to 4 to ensure accurate readings and graphs. A higher averaging value could have been selected for even more accurate results, but other people working in the laboratory need to be considered when doing experiments that involve eardrum piercing frequencies played loudly. That is why it was decided to not to take an averaging value of 8 (next possible option) or higher, as this takes the machine (at least) twice as long to complete, annoying people in the laboratory for longer than strictly necessary. 4 was decided as being a merciful value.

Because of unfamiliarity with the HP 4194A, tests were done at both 0 dB and 20 dB attenuation of the input of the machine. This was to see if this setting would have any impact on the frequency response of the piezo discs. Furthermore, tests were also done with both a smaller piezo of 15 mm in diameter (such as the one used in experiment 1) and a larger piezo of 35 mm in diameter, to see what the impact would be on the produced bode plot.

Results

The first bode plots can be seen in Figure A2.0.2 and Figure A2.0.3, which were done using the large surface transducer and the large piezo disc. Aside from the higher peak around 1000 Hz in Figure A2.0.3, the frequency responses seem to be very close, if not almost identical.

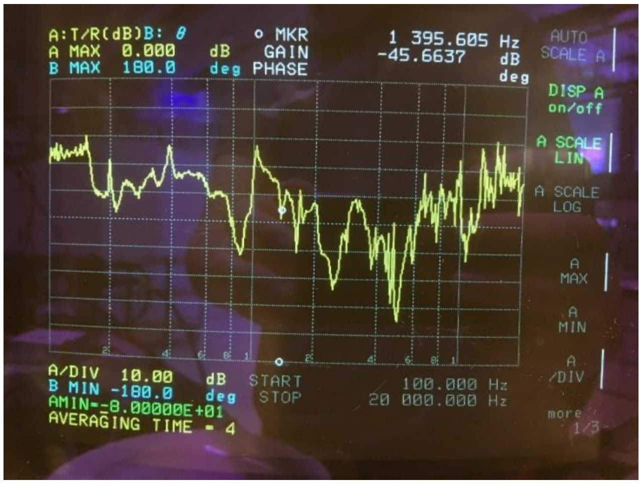


Figure A2.0.2: Large Surface transducer on plate at 20dB attenuation, large piezo.

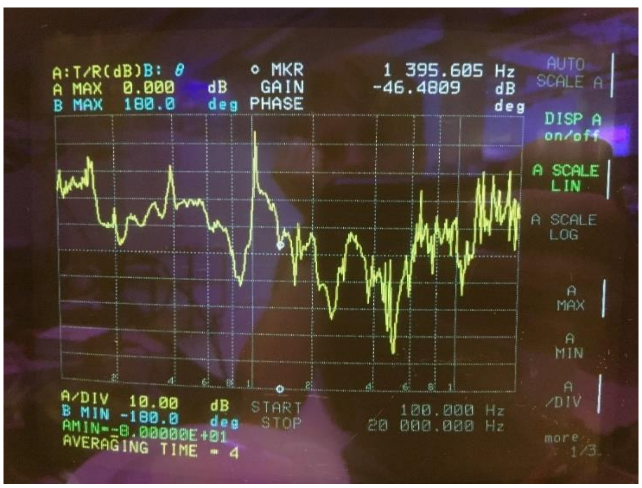


Figure A2.0.3: Large Surface transducer on plate at 0dB attenuation, large piezo.

The same goes for Figure A2.0.4 and Figure A2.0.5, which used the medium surface transducer and the large piezo. While the results of both figures are different from those produced by using a large surface transducer, they are again almost identical to each other.

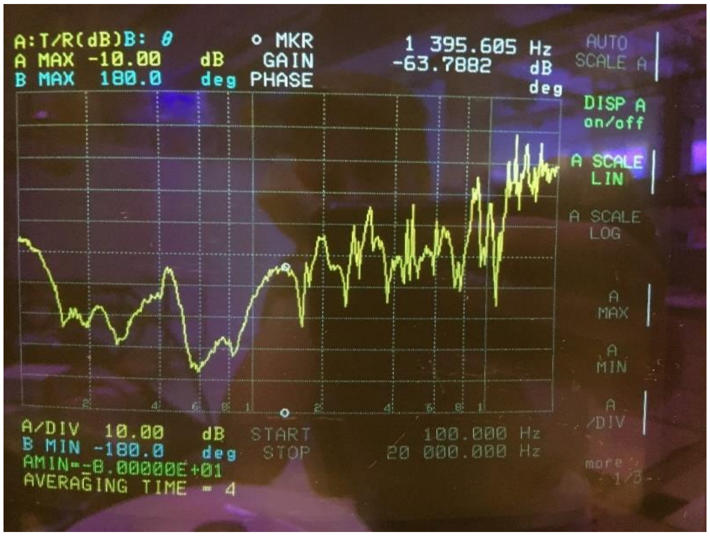


Figure A2.0.4: Medium surface transducer on plate at 0dB attenuation, large piezo.

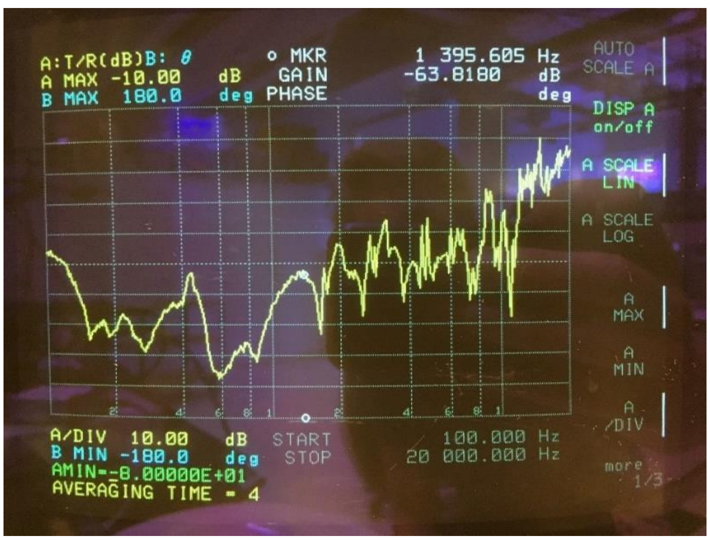


Figure A2.0.5: Medium surface transducer on plate at 20dB attenuation, large piezo.

When using the smaller piezo disc and the large surface transducer, the HP 4194A creates the bode plots that can be found in Figure A2.0.6 and Figure A2.0.7, which are very close to each other yet again. When comparing them to the results of Figure A2.0.2 and Figure A2.0.3, there are visible differences, but also many similarities, which can be found in the table.

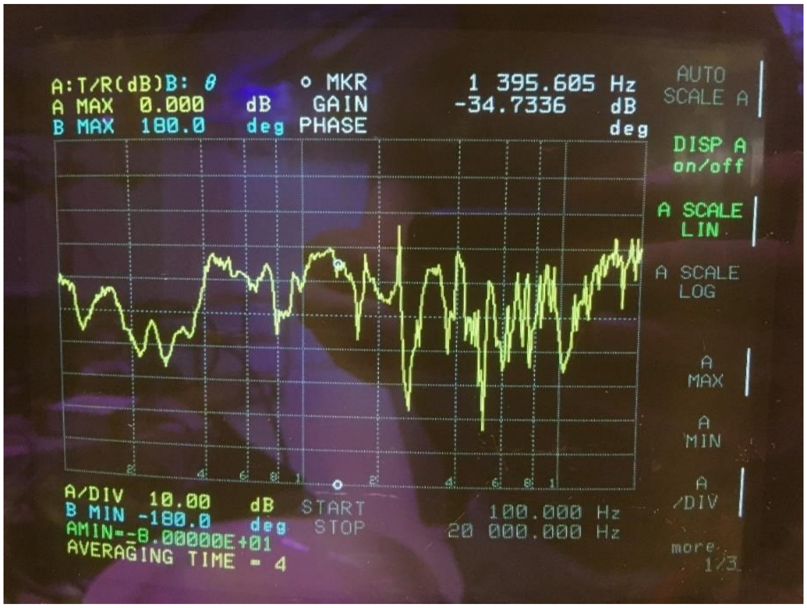


Figure A2.0.6: Large surface transducer on plate at 0dB attenuation, small piezo.

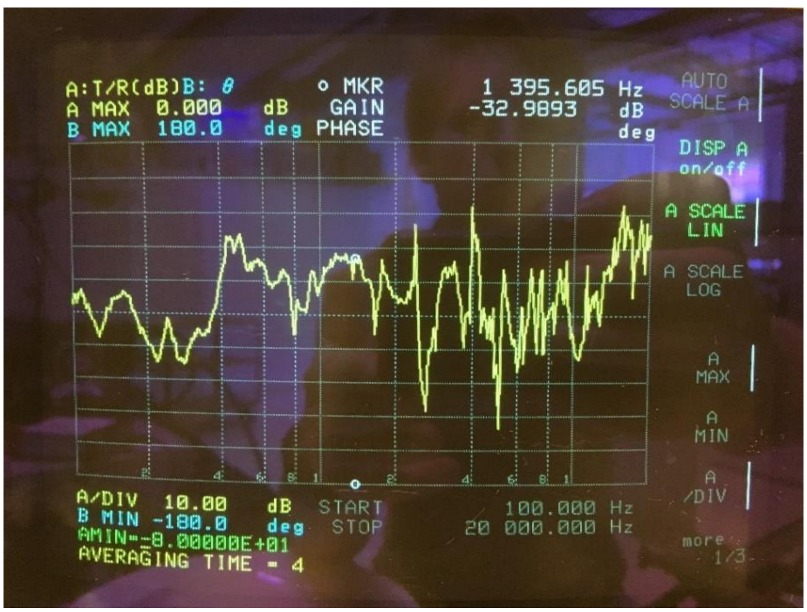


Figure A2.0.7: Large surface transducer on plate at 20dB attenuation, small piezo.

This means that though the size of the piezo has an impact on the perceived frequency response, the picture is not completely distorted by simply using a piezo of a different size.

As can be seen in Figure A2.0.8 and Figure A2.0.9, a difference in input attenuation does not make a difference there either. The similarities of these two figures compared to Figure A2.0.4 and Figure A2.0.5

can be found in the table below. There seems to be some correspondence in peaks, but nothing much to write home about.

Large surface transducer peaks for both sizes of piezo (Hz)	Large surface transducer dips for both sizes of piezo (Hz)	Medium surface transducer common peaks for both sizes of piezo (Hz)	Medium surface transducer dips for both sizes of piezo (Hz)
250	300	230	300
350	370	400	600
400	600	700	830
700	800	1400	1600
1000	1700	2000	2500
1800	2700	3200	3900
2500	3900	4000	4100
3500	5000	5200	5900
4000	8000	6000	7000
5200	10000	9000	9500
		10000	11000

Figure A2.0.8: Medium Surface transducer on plate at 0dB attenuation, small piezo.

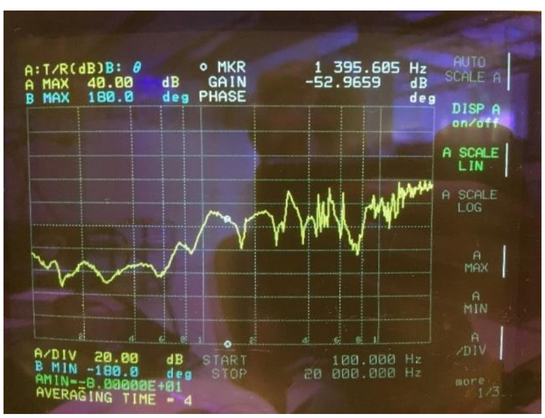


Figure A2.0.9: Medium Surface transducer on plate at 20dB attenuation, small piezo.

It can now be said with absolute certainty that the attenuation on the 4194A input makes absolutely no difference for the produced bode plot. Now, the impact of the location of the surface transducer on the rim of the plate on the produced bode plot can be investigated. The results can be found in Figure A2.0.10.

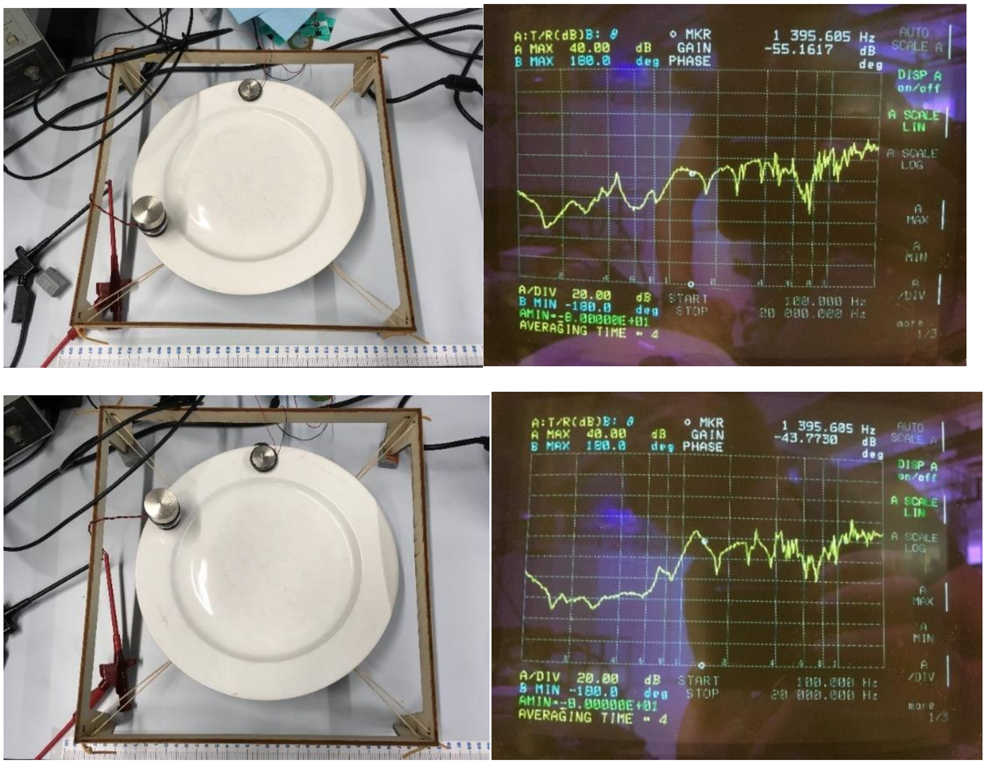
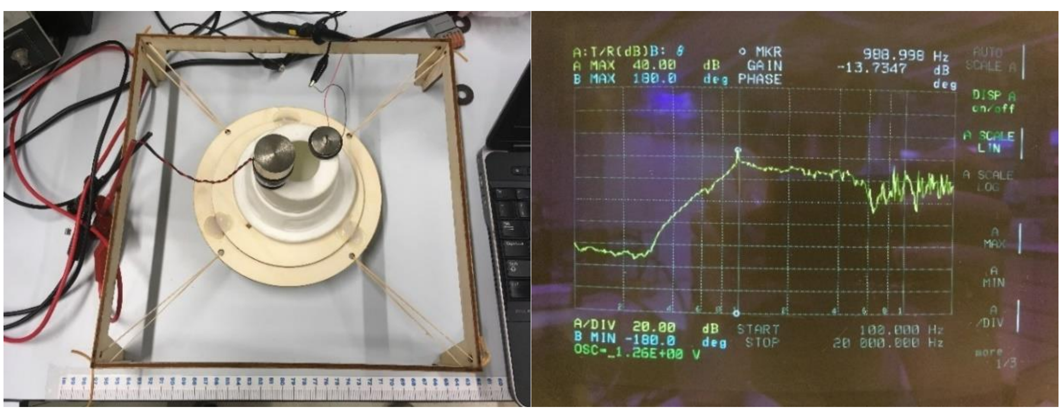


Figure A2.0.10: Influence of placement of the surface transducer on the produced bode plot.

It does impact the bode plot a little, but the peaks are still in roughly the same places in both plots. Doing the same thing for the ceramic artifact results in Figure A2.0.11.



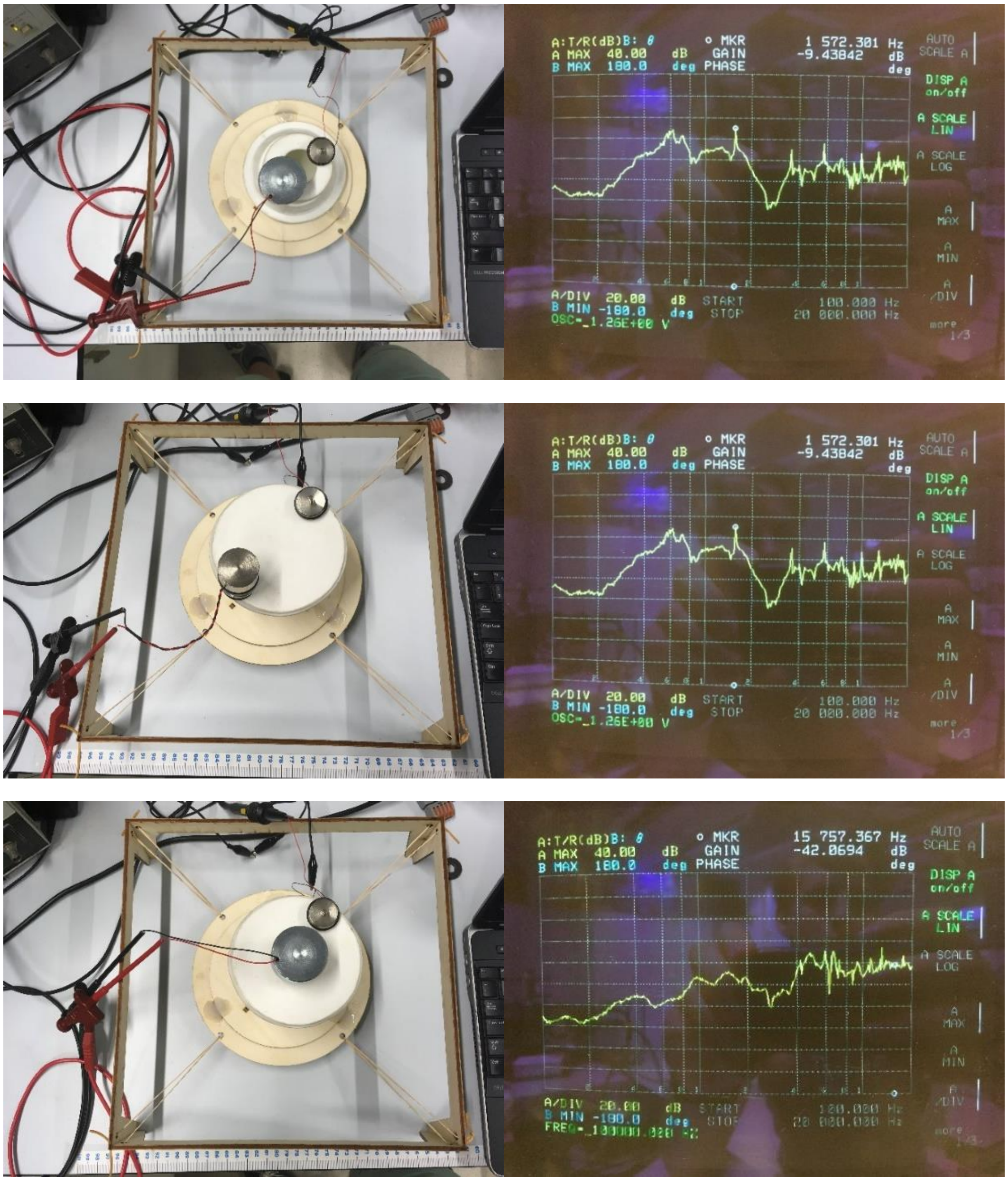


Figure A2.0.11: Impact of placement of surface transducer on produced bode plot for the ceramic artifact.

What can be seen here is that the placement of the surface transducer has quite little effect as well, with some peaks being evened out a little, but not much else.

Conclusion

First of all, there seems to be no correspondence whatsoever between the perceived loudness of frequencies and the values of these frequencies indicated on the display of the 4194A. This could have little to do with the resonances and more with the equal loudness curves mentioned in [36], but it is still worth mentioning.

Secondly, the frequencies indicated on the display of the 4194A hardly correspond to the resonant frequencies that can be heard when tapping the rim of the plate either (750 and 1730Hz). What was discovered after all the measurements were already done and all the hours were already put into it, is that this has to do with the fact that placing a surface transducer on the rim completely changing the resonant characteristics of the plate. This is because it adds to the total weight of the construction that needs to be kept in movement by the audio feedback loop to move air and therefore produce sound. The fact that in all of the setups, the surface transducers were clamped to the rim tightly using magnetic force makes this effect even stronger. For now, there was no more time to execute these tests with the surface transducer located in a place that would not affect the capacity of the plate to vibrate, i.e. the bottom. What will be done in further testing is placing the surface transducer at the bottom of the plate and try to obtain an audio feedback loop once more.

Lastly, neither the attenuation of the machine, the size of the piezo disc or the placement of the surface transducer seemed to have much effect on the produced bode plot. The two different types of surface transducers do have somewhat different frequency responses and therefore bod plots, but that is always to be expected with different speakers. Neither of them have a flat response, so it would simply be best to use the one that can accentuate the most relevant frequencies, which can be determined by looking at Figure 5.9 and Figure 5.10.

Appendix III

Experiment 3: controlling feedback with control signal of desired pitch

Introduction

The goal of this experiment is to try to control the fundamental frequency of the audio feedback that occurs in the system. This will be done by means of a control signal at one of the resonant frequencies of the plate, which that is injected into the feedback loop using a function generator.

List of components

Electronics

- HP 33120A function generator
- Tin can amplifier & wiring
- 15mm piezo disc
- Adafruit 4Ω 3W surface transducer
- Adafruit 4Ω 5W surface transducer
- Probe
- BNC-to-banana plug adapter
- BNC splitter
- 2x banana cable
- Alligator clips

Other

- porcelain dinner plate
- 2x small magnet
- Shock mount
- Trampoline

The diagrams can be found in Figure 6.9 and Figure 6.10.

Method

First, it was decided that for the control signal to be most effective, it needed to be input into the middle of the feedback loop for it to have the most influence on the resonance the sound would clamp onto. The plate would already have to be in movement at one of its resonant frequencies before the feedback loop gets activated. The medium surface transducer is placed directly underneath the plate, as it had already become clear in experiment 1 that it was more capable of getting the plate to resonate at a lower resonant frequency than the large surface transducer. Because a feedback loop still needs to occur, a surface transducer still needs to be placed under the surface transducer that provides the control signal, so that they form a stack. Placing this transducer directly underneath the control signal transducer would mean that the plate would have to be balanced even more carefully than before and be at a higher risk of falling and breaking. Therefore, it needs to be placed underneath the shock mount. This could cause the ground underneath to influence the resonance of the system, though. That is why a second suspension mechanism needs to be used for the bottom surface transducer to lay upon, so that it does not touch the ground, but it does touch the platform the control surface transducer is resting upon. A small trampoline was used for this purpose (figure A3.0.1). As can be seen in figure A3.0.2, the weight of the plate and the control surface transducer make the platform go down, so that it can perfectly rest on the bottom surface transducer. As there is quite a lot of material between the surface transducer that is part of the feedback loop and the plate, a large surface transducer was put at the bottom for an increased output of the feedback loop.

The control signal is applied before the amplifier gets turned on, after which the control signal will be turned off as soon as possible.

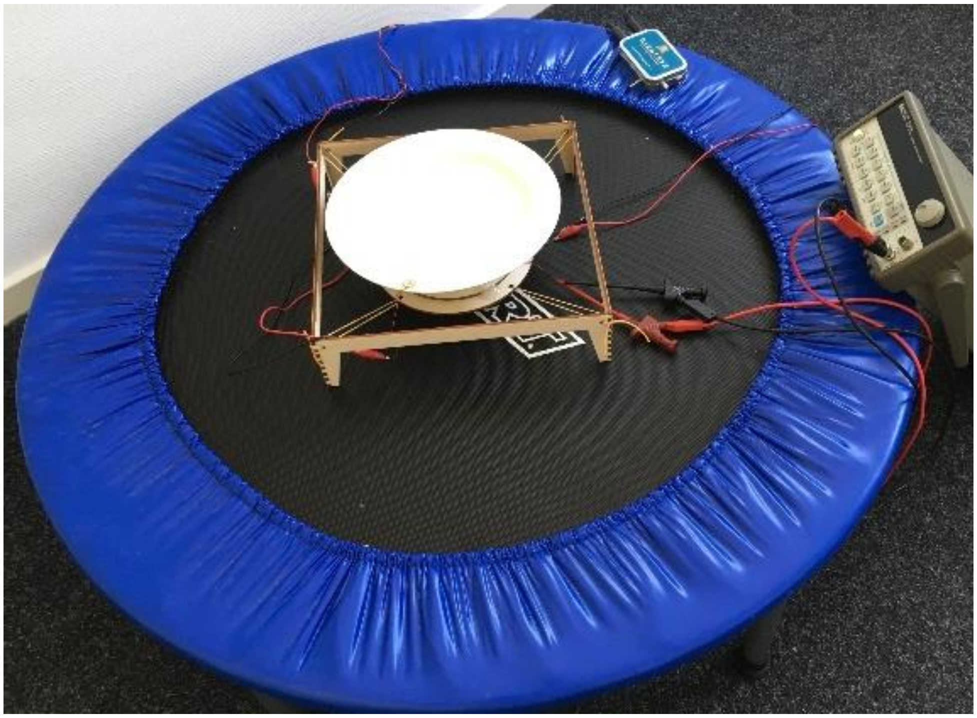


Figure A3.0.1: Setup of the working prototype.

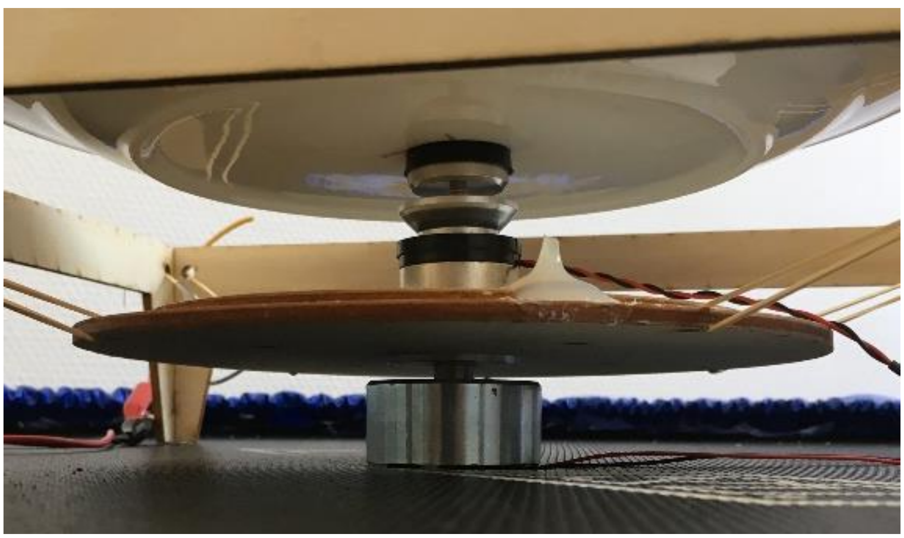


Figure A3.0.2: Stack of surface transducers, with the bottom one resting on the trampoline.

Results

As mentioned in 7.3, this prototype worked exactly the way that it was supposed to: the frequency of the audio feedback would assume the same value as the frequency of the control signal if it was at one of the two main resonant frequencies of the plate. When using other frequencies as a control signal, the outcome of which of its resonant frequencies the audio feedback loop would clamp itself to became seemingly random. The prototype can be seen working at this link:

https://drive.google.com/file/d/1cjNhRFC-aboTYds2x_oyb8ue4jsjjz-E/view?usp=sharing

Conclusion

It can be concluded that the frequency of an audio feedback loop can be accurately controlled using a control signal injected into the feedback loop using a second surface transducer. The surface transducer does need to supply one of the resonant frequencies, otherwise the outcomes become unpredictable.

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