Master's thesis

NTNU
Norwegian University of Science and Technology
Faculty of Humanities
Department of Music

Christian Feess

Exploring Complex, Musical Audio-Feedback through the Design and Construction of a Sound Art Installation

Master's thesis in Creative Music Technology
Supervisor: Øyvind Brandtsegg
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Abstract

Audio feedback is the re-circulation of the output sound of a system as an input for the same system, creating a feedback loop. The behaviour of feedback can be immensely complex, shaped by various interdependent components. A feedback-based approach to sound generation requires specific methods and design techniques to achieve a rich and unique sound. Balancing instability (complexity) and controllability while avoiding saturating feedback is a characteristic challenge of feedback instruments. This work explores complex musical behaviour of audio feedback through the conceptualising and making of a sound art installation. With it, interference and interactions between several feedback loops are examined. The installation utilizes audio feedback to create a musical ambiance. Acoustical pipes serve as resonance attractors. Furthermore, methods are proposed to approach resonance equilibrium, leading to a maximum instability, allowing for small changes to influence the sound. The project is evaluated and discussed in retrospection, and possible future work is proposed.

Keywords: *Audio feedback - Complex musical behaviour - Feedback interference and interactions - Sound art installation - Resonances*

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This work would not be what it is without *you*.

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Chapter 1

Introduction

Audio feedback has been used in creative practices dating back to the 1960s (Sanfilippo & Valle, 2013). Artistic explorations include early usages in live performances, famously from Jimi Hendrix, sound art installations, such as *Empty Vessels* by Alvin Lucier, and more recently, feedback instruments, such as the *Halldrophone*, amongst many others. The amount of research and works within the field has expanded greatly since the first beginnings. Eldridge et al. (2021) showcase an extensive range of self-resonating vibrotactile feedback instruments and discuss the making, playing and conceptualising of such instruments. A brief history of feedback systems in music and an in-depth theoretical characterisation of them is given by Sanfilippo and Valle (2013). The *feedback musicianship network*[1] brings together researchers and artists to address and discuss the challenges of feedback instruments. The network gathers relevant work and research within the field.

Audio feedback systems require an entirely different approach to shaping sound than other common sound generation methods. They recirculate the output sound back into the system as an input to produce the subsequent sound, completing the feedback loop. This approach demands specific methods and design techniques to achieve a rich sound while also providing unique sound characteristics and potentially interactive possibilities. The behaviour of audio feedback can be complex, rich and unexpected. Subtle changes in the system can have a great impact on the processes within. It is often desired to steer the systems into a maximum dynamic complexity (maximum instability). However, achieving a richness in sound and variation can be a challenging task, as these systems tend to fall into saturating feedback. There is a fine balance between stability, the tendency to reach and remain in a stable state, and instability. Kiefer et al. (2020) explore this and propose a system for complexity-controlled gain dynamics.

With all the research and artistic works as a foundation, I am venturing into my personal explorations of feedback systems. In this thesis, I am describing a feedback based sound art installation that I developed over the course of this master's project. The source of

[1] See feedback-musicianship.pubpub.org

motivation and the artistic concept behind this project is to utilize audio feedback to discover musical expressions. The installation presented generates sound through several individual feedback loops that interfere and interact with one another. The feedback is generated with acoustic pipes[2] that are equipped with speakers and microphones. The behaviour of the installation is influenced by a great number of components, processes and attractors. These are described in detail in an attempt to achieve a better understanding of audio feedback behaviour. Furthermore, the work contains a foundation of relevant theory and related works. On an artistic level, I am interested in exploring feedback systems of intricately, interwoven and interdependent components musically. Feedback systems are usually a sum of a great amount of subtleties that define the behaviour of the whole. Changes to one component are affecting every other component, which often has a great impact on the whole. This is highly analogous to feedback mechanisms in nature, for example the climate. Communicating this through the installation and the sound is what I strive for.

This work is split into four main chapters. This chapter serves as the introduction to the work, followed by a chapter on related work in the field. The subsequent chapter contains and covers relevant theory, ranging from sound propagation, audio feedback, to concepts of cybernetics followed by a chapter describing the construction, signal processing and operation of the installation in detail. Finally, this work will be evaluated in a discussion and conclusion, where perspectives on future works and research are outlined and suggested.

1.1 Methods

The project presented in this thesis was approached through methods of *exploratory prototyping*, *experimental prototyping* and arguably *evolutionary prototyping*. Mayhew and Dearnley (1987) discussed these terms in detail in the context of software design. While closely related and based on these definitions, the terms find a slightly different use here. The process of exploratory prototyping serves as a starting point, where ideas and possibilities are creatively explored. From this process, clarification and requirements for the project might arise. Through experimental prototyping, a prototype is approached that might fulfil the requirements for the task at hand. Part of this process is the experimental use of the prototype and its evaluation, together with reflections and readjustments. Since the prototyping process here is not necessarily as streamlined as commercial product design processes, and the requirements might not be as definite, the final output might consist of various prototypes produced throughout. This process allows for creative exploration and provides room for maturing the artistic concept along with it.

[2] The term pipe was specifically chosen in this thesis. Pipes refer to hollow cylinders where the inside diameter is of importance, which is more relevant here compared to tubes (Engineering ToolBox, n.d.)

Chapter 2

Related Work

The landscape of audio feedback installations and instruments is rich. From its early experiments by Lucier and contemporaries in the 1960s, there has been continuous activity, with a marked increase of interest over the last decade (Eldridge et al., 2021; Kiefer et al., 2020; Sanfilippo & Valle, 2013). Works range from initial unintended, accidental discoveries of feedback phenomenons to thought-through, crafted feedback systems. Through these works, the practices, approaches and methods were refined or newly discovered. This serves as a foundation on which can be built upon. The combinatorial possibilities are manifold, and the sonic appearance can take various shapes. The aim of many audio feedback systems is to explore their behaviour in a musical way. Control is an important factor when constructing instruments, and especially so with feedback instruments. It can be challenging, as many feedback systems exhibit levels of unpredictability and self-organization.

The sound installation presented in this thesis finds itself in the landscape somewhere between autonomous audio feedback installations and interactive audio feedback instruments. It incorporates a multiplicity of the Larsen effect, utilizes pipe resonances as pitch attractors and explores interference in several layers. Certainly my work presented here is standing on the shoulders of a great amount of existing works on which I count myself fortunate to draw from. In this chapter a selection of related installations, performances and instruments, relevant in the context of this thesis, will be presented. These works contain significant thoughts and approaches and were a source of inspiration for my own work. The works presented below are listed in chronological order, as each is relevant in different aspects.

2.1 Empty Vessels (1997) by Alvin Lucier

The sound installation *Empty Vessels* by Alvin Lucier marks a significant audio feedback exploration. With this installation, Lucier explores the sound characteristics of different hollow glass vessels and their interaction with the surrounding room through an audio feedback system. Each vessel is placed on a pedestal. The pedestals are around 1.20 metre high. Microphones are positioned in the proximity of the opening of each vessel. Each microphone is connected to a speaker. The speakers are placed on pedestals as well, facing the corresponding vessel and microphone. The audience can walk in between the pedestals to explore the installation, influencing the microphone-speaker feedback flow (Atelier Nord, 2013). Figure 2.1 shows a simplified illustration of the installation.

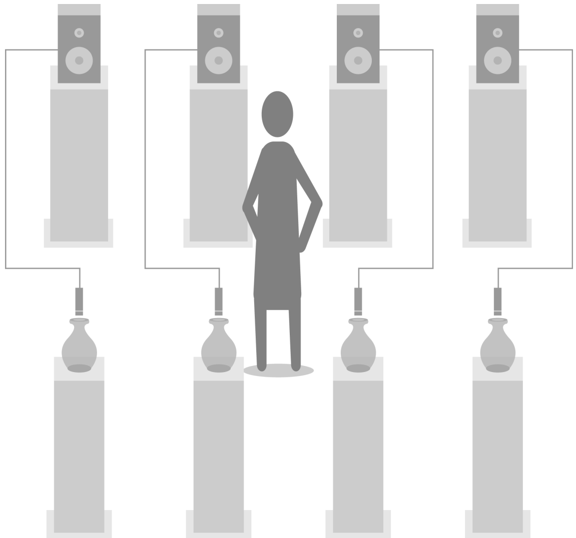

Figure 2.1: Simplified illustration of Alvin Lucier's 'Empty Vessels'.

To initialize the installation, the system is set into a sustaining feedback loop with the use of sufficient initial excitation. The microphones capture the sound from the room, which is filled with the sounds produced by the speakers, aside from potential noise produced by the room or the audience. The speakers reproduce the captured signals of the microphones. Every signal is provided with sufficient amplification to ensure that the feedback stays in a sustained loop. The microphones' position, the recorded signals and the processing is tuned in such a way that a physical change to the space and architecture (room acoustics) manifests in an audible change of the sound. This could be the placement of an object or the presence and movements of the audience. The resulting sound and its frequency spectrum are thus influenced by many different factors. This seems to be the core exploration of the installation. The resonances of the room, the positions of the pedestals, the positions of the microphones, the resonance frequencies of each vessel and the presence and movements of the audience are integrated components of the feedback loop. Therefore, they influence the system to greater or lesser extent.

Empty Vessels is closely related to the work presented in this thesis. The use of multiple individual feedback loops that interact and interfere with one another is an important aspect. Objects are used as resonance attractors and the Larsen effect is integrated through speakers and microphones. In addition, the integration of the immediate space surrounding the installation into the feedback system is of great importance in Luciers' installation. All these aspects are relevant for the installation described in this thesis.

2.2 Pendulum Music (1968) by Steve Reich

Pendulum Music is a sound installation by Steve Reich. It is based on loudspeakers and swinging microphones. Pendulous movements are incorporated into the sound generation process. Steve Reich described the installation as an *audible sculpture*:

> *"In many ways you could describe Pendulum Music as audible sculpture. The objects involved are the swinging microphones and the loudspeakers. I always set them up quite clearly as sculpture." (Reich, 1976)*

The setup is simple and consists of three or four microphones, amplifiers, loudspeakers, stands and cables. The speakers are laid flat on the floor with the membranes facing upwards, towards the ceiling. For each speaker, there is a corresponding microphone. The speaker reproduces the signal provided by the microphone. The signal of the microphone is run through an amplifier to provide the necessary amplification, ensuring that feedback can occur. The microphones are suspended from stands and are able to swing freely. Each microphone is put into motion, so it would swing back and forth in a pendulous manner. The trajectory of a moving microphone passes the corresponding speaker with just a little gap in between the microphone capsule and the speaker membrane (Reich, 1974). Figure 2.2 shows the installation at a performance in 1969.

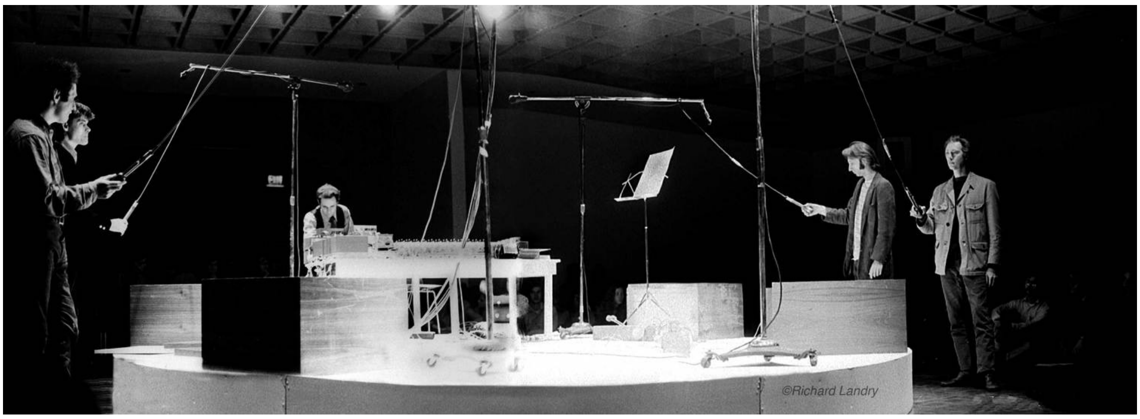

Figure 2.2: 'Pendulum' music performance at Whitney Museum of Art in NYC, 1969.

Source: Steve Reichs' Facebook profile

Each speaker-microphone pair will produce a pitched feedback tone when the microphone is in close proximity to the speaker. The distance at which this occurs is determined by the level of amplification. The feedback tone will fade in and out periodically as the microphone swings back and forth. Rhythms and varying patterns can emerge as there are up to four instances of individual feedback loops created simultaneously. Naturally, each microphone pendulum will lose momentum over time due to friction within the system. Each swing comes with a loss of momentum. The microphone spends more time in the close proximity of a membrane as its velocity decreases, which results in a continuous increase of the duration of the sustained tone. A constant loss of momentum will ultimately result in a state of nearly complete rest. Towards the end of a performance, the produced feedback tones will overlap more often. Thus, the individual feedback loops will influence one another more heavily. The most dominant resonances within the feedback system will take the precedence.

Pendulum Music explores not static but cyclical production of feedback tones. Nevertheless, it is still related to the work here. The idea of including hanging elements and pendulum movement is important here. So are the interactions of the individual feedback loops that happen more frequently at the end phase of a performance, as the microphone movement is comparatively small. The frequency spectrum is prone to change repeatedly at that phase. Additionally, the inclusion of sound propagation through physical space into the feedback system and the use of the Larsen effect are core aspects of this installation. The exploration of the interaction of multiple feedback loops, each with their own resonant frequencies, is a key aspect in the installation proposed in this thesis as well.

2.3 Amplifications (2013) by Lesley Flanigan

Amplifications is an album by the experimental electronic musician and vocalist Lesley Flanigan. Flanigan builds and plays her own instruments for her music and her performances. In most of her work, there is a strong interplay between her vocal performance and electronic instruments. She appears to have a special interest in audio feedback instruments. For the *Amplifications*, which she performed in 2013 at the NOISE festival, she played several feedback instruments of her own design. All of them are similar in principle, with slight variations, and made of few, reclaimed elements.

The feedback instruments she uses consist of one or several speakers, one contact microphone and a wooden housing. The microphone has stiff, yet bendable wires connected to it, enabling Flanigan to position it as needed. There is a switch and a potentiometer on the devices, to power them on and off and to adjust the gain of the microphone signal. Each instrument, appearing as a wooden rectangular box, is laying on the ground, with the speaker membrane facing upwards and the microphone wires emerging from the box (Flanigan, n.d.).

At her performance, Lesely sits in between the four instruments, with a vocal microphone in one hand, while singing and playing the four feedback instruments (see figure 2.3). She shapes the sounds produced by the instruments by changing the position of the small contact microphones. With a mixer in front of her, she is able to balance the various audio signals. An additional loop station enables her to record and loop sequences. It appears that the vocal microphone signal is fed into the instruments' speakers, as she frequently uses it to generate feedback with the instruments (Flanigan, 2014).

The minimalist design of her feedback instruments and her raw approach to feedback inspired the approach to my work. Utilizing different speaker drivers results in different sound characteristics, and the modularity of her system enables her to quickly rearrange and position each instrument. Her form of interaction with the instruments is simple yet effective, providing a diverse sonic palette. Using reclaimed materials for the instruments also strongly resonated with how I wanted to construct my own installation.

Figure 2.3: Lesley Flanigan performing 'Amplifications' in La Sala (Brooklyn), 2012.

Source: Flanigan (n.d.)

2.4 circuit - Feedback Studies (2015) by Jeroen Vandesande

The exhibition *circuit* by Jeroen Vandesande is related to the installation presented here in several aspects. Vandesande is a sound artist, creating sound installations as well as giving performances. With *circuit*, Vandesande explores audio feedback and room acoustics in a musical way. It is a result of his previous experiments with audio feedback.

The installation consists of several tuned, open-ended aluminium pipes, small motors, a pair of loudspeakers and small microphones. Each pipe hangs horizontally on two ropes from the ceiling. The pipes are located at various positions and heights in the room. The pipes are constantly rotating horizontally around their own axis. Within the pipes sit microphones, and each pipe has a hole that can be closed by a small motor contraption placed just above the hole. A set of speakers is placed at different positions in the room. Judging from the resources available, there are only two speakers, while there are at least six pipes at the setup at *singuhr* in Berlin. It is probable that the microphone signals of all pipes are summed on these two speakers. The audience can move freely through the room (Vandesande, n.d.).

The installation is a combination of composition and autonomous sound generation. Once the feedback system is running and in a sustaining loop, the speakers produce pitched tones due to the Larsen effect. The core exploration is, arguably, how the system and the sound behave due to the properties of the room and the pipes. Many factors influence the frequencies of the produced tones. Each pipe has different dimensions, therefore different resonance frequencies, determined by the length of the pipes and their opened or closed holes. The microphones capture sound that has travelled through the pipes. The resonance frequencies are therefore accentuated. The microphone signals reproduced by the speakers. While the speakers' frequency response plays a role in the systems' behaviour, more significant determinants are the spatial locations of the speakers within the room and the room properties. In Vandesandes' exhibit at *singuhr* in Berlin, the surrounding surfaces (walls, ceiling and floor) are hard and likely reflect the sound, spreading it in the room. The holes in the pipes are opened and closed with remotely controlled motors. This adds an additional layer of composition, changing the fundamental and the harmonics of the pipes.

This installation is in many aspects connected to the work presented in this thesis. There is a great focus on the inclusion of acoustic pipes as resonance attractors. The exploration of different pipe rotations and positions and their interaction with the room acoustics are main aspects here. The ideas of the hanging system are closely related. *circuit* has a more composed component, however, and the movement of the pipes is strictly automated.

Figure 2.4: Installation 'circuit' by Jeroen Vandesande.

Source: Vandesande (n.d.)

Chapter 3

Theory

This chapter holds relevant theory which is of use in later explanations and provide a basis for assumptions. Important fundamental acoustic models and behaviours will be covered, followed by a section on audio feedback, before concepts and terminologies specific to complex systems will be outlined. The topics and thoughts elaborated on here require the explanation, as they will aid significantly in piecing together parts of the mosaic that shape the behaviour of the installation presented in this thesis.

3.1 Acoustics

This chapter covers a foundation of relevant terms, models and theories in the *acoustic* realm. Therefore, starting off with a brief definition of what *acoustic* means is of use. Beranek and Blackstock (1988) define *acoustic* as:

> *"[...] intimately associated with sound waves or with the individual media, phenomena, apparatus, quantities, or units discussed in the science of sound waves." (Beranek & Blackstock, 1988, p. 9)*

With the term *acoustics*, the science in relation to sound is implied. Fundamental scientific models of acoustics are of great use when designing an instrument or a sound installation. They offer an approach to describe and understand the behaviour of the work presented in this thesis. This section ranges from the general idea of sound propagation, resonance, pipes, relevant speaker and microphone properties to room modes. All have an effect on the installation in one way or another.

3.1.1 Sound propagation

Wherever there is a source of sound, there will be emitted sound waves. Sound waves are the physical, measurable means of transferring sound. The reason sound waves are named as such is due to their property of propagating energy from one point to another through a medium. This is what all types of waves have in common. While the waves transport the information and energy through the medium, the medium itself is not being transported. Here we are focusing on air as the medium (Rossing et al., 2002).

When sound gets produced by a vibrating source (like a loudspeaker), it propagates through air, travelling with a speed of around $343 \frac{m}{s}$. The dimensions and the shape of the sound source is a decisive factor in how the sound will radiate. An ideal spherically symmetric point source would radiate sound spherically, while an ideal cylindrical, symmetrical line source would radiate sounds cylindrically. Practically, however, such sources do not exist and can only be approximated. A loudspeaker, small in size compared to the wavelengths it emits, approximates the spherical radiation behaviour of a point source for those wavelengths. An array of small speakers in a line approximates cylindrical radiation for low frequencies (Rossing et al., 2002).

Below is a description of how sound behaves over distance, together with three specific behaviours of wave propagation: reflection, diffraction and refraction.

Distance and sound pressure

With the foundations given so far, it can be established that sound waves are pressure waves that compress and decompress air. The sound waves create periodic fluctuations in the air pressure. These can be calculated and measured, and the pressure is often expressed as *sound level* or *sound pressure level (SPL)*. As sound travels a distance, the sound pressure will decrease. The amount of reduction in sound pressure in free field is given with the inverse distance law $1/r$, where r is the distance from the sound source. With each doubling of the distance, the sound pressure is halved. This results in a sound pressure level drop of 6 dB per doubling (Rossing et al., 2002).

Reflection

When sound waves hit a hard surface, reflections will occur. The reflected sound waves are then travelling away from the wall. How the reflected sound waves radiate can be described with an additional imagined sound source, the so-called *image* source. This *image* sound source is opposite of the reflecting surface, located at the same distance from the reflector as the original source is, as shown in figure 3.1. Reflections are the root cause for reverberation and echoes (Rossing et al., 2002).

Diffraction

Diffraction can occur when sound waves encounter a physical obstacle. If the obstacle is small compared to the size of the wavelength, the wave will be *diffracted* and will essentially bend around the obstacle. This diffraction can also occur when the waves travel through a narrow opening. The extent of the diffraction is determined by the relation of the size of the obstacle or the opening to that of the wavelength. Waves with bigger wavelengths diffract more than ones with smaller wavelengths when encountering the same, comparatively medium-sized obstacle. This significantly influences the frequency dependent directivity of loudspeakers. Generally, lower frequencies will easily bend around the speaker, while higher frequencies will not bend to the same extent. This results in a more uniform, spherical radiation for lower frequencies and a focused, louder sound of higher frequencies in a cone like form in front of the speaker (Rossing et al., 2002).

Refraction

The speed at which waves travel at is mainly determined by the medium through which they travel. A change in speed causes a phenomenon known as refraction. The result of this phenomenon is a change of direction in which the wave propagates. A common visual effect of refraction is light passing from air to glass. This also happens to sound waves, as changes in air temperature affect the speed of sound. This change in speed would also result in the sound waves' change of direction (Rossing et al., 2002). Even though this phenomenon does not find the same significance in this work, it is worth mentioning as it completes the list here.

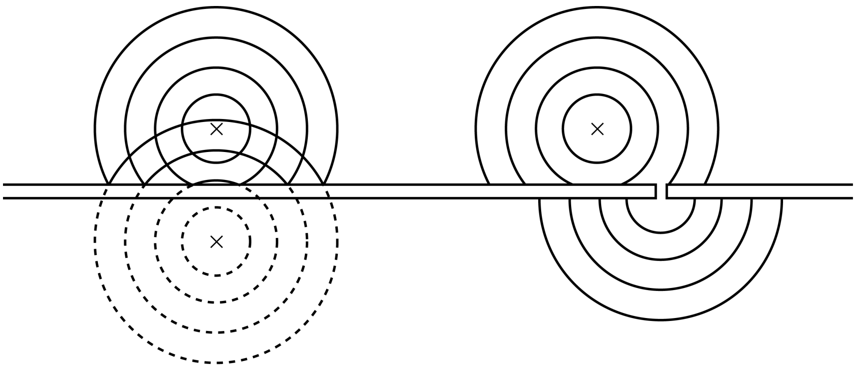

Figure 3.1: Visualization of reflection (left) and diffraction (right).

Superposition, interference and standing waves

When two or more sound waves overlap, such as by travelling in opposite directions or due to a reflection, the sound waves will follow the principle of linear *superposition* and create what is called *interference*. It is important to note here that sound waves can pass through one another while keeping their original identities. In a simplified, two-dimensional situation where two sound pulses are interfering with each other, two observations can be made. If both pulses are of the same phase, they will add up. This is called a *constructive interference*. If the pulses are of opposite phase, they will subtract, called a *destructive interference* (see figure 3.2) (Rossing et al., 2002).

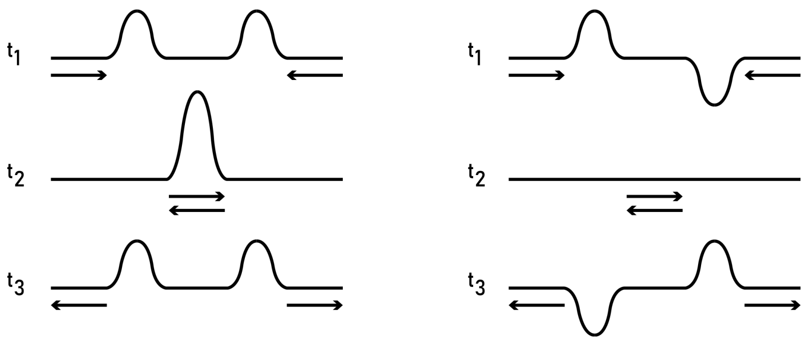

Figure 3.2: Constructive (left) and destructive (right) interference at time steps t1, t2, t3.

Interference is important when considering sound in an enclosed space due to potential reflections. As described previously, sound will reflect off a hard surface. As soon as a sound wave hits the surface, the situation changes from one to two waves travelling in opposite directions to each other. They are implied to overlap and cause interference. Now, considering a continuous cyclical sound wave, instead of a single pulse, in a scenario where there are two parallel hard walls. The sound will be reflected cyclically, back and forth, from one wall to the other (Rossing et al., 2002). This essentially means that there are sound waves with specific phase offsets travelling repeatedly in opposite directions. This is important to be aware of as it can cause standing waves, as explained below.

An enclosed space, such as a room, might reflect sound from the walls, floor, ceiling and other surfaces. If the distance between any two given parallel surfaces is a multiple of half the wavelength of a produced sound, then the reflected sound will travel opposite to the original sound and is exactly phase-inverted to it. The acoustic result is called *standing waves*. Standing waves exhibit both constructive and destructive interference effects, therefore amplifying the sound at points and cancelling out at others. The wavelengths λ at which standing waves occur, of the order n (n being an odd natural number) is given by the equation (3.1) with L as the distance between two parallel surfaces (Rossing et al., 2002; Sengpiel, 2012).

$$\lambda_n = n * 2 * L \tag{3.1}$$

Standing waves are characterized by so-called *nodes* or *minima*, which are zero crossing points and their opposites, *antinodes* or *maxima*, which are located at the centre of two nodes. A node is the location at which the sound pressure of the original sound wave and of a reflection effectively cancel each other out. Thus, at nodes the sound pressure does not change and at this location no tone is produced. At the locations of the antinodes, the source and reflection overlap, resulting in a constructive interference and adding to a maximum movement (Rossing et al., 2002; Sengpiel, 2012).

3.1.2 Resonance

Resonance is of great importance when working with audio feedback systems. It can occur within various different components and can influence a system's behaviour to a great extent. Therefore, an understanding of what resonance is can be helpful when making a feedback system. Here, the concept of resonance will mainly be used in the context of mechanical systems. The term *resonance* literally translates as *audible repercussion*. According to the Encyclopedia Britannica (1998), resonance, in the field of physics, is understood as a relatively large selective response of an object or a system that vibrates in step or phase to an externally applied oscillatory force. This can be exemplified with a simple pendulum. Consider the mechanical system of a pendulum, as in figure 3.3: a weight m is hanging at the end of a rope that has the length L. A pendulum like this has a natural frequency f determined by the length of the rope. The natural frequency is given by the equation 3.2, where g is the gravitational acceleration. If a force (F_1) is repeatedly applied each time the pendulum is at its highest amplitude, thus following the pendulum's natural frequency, and considering the applied force is larger than the loss due to factors such as friction and air drag, the amplitude will increase with each cycle (Rossing et al., 2002). If the same force were to be applied at a different frequency, not in resonance with the pendulum's natural frequency, it would interrupt the movement and the system would react with a loss of energy (amplitude).

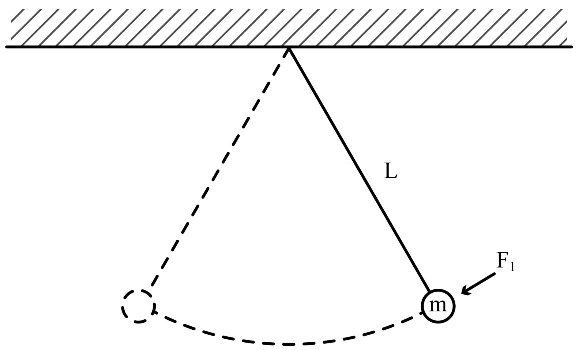

Figure 3.3: Pendulum with external force.

$$f = \frac{1}{2\pi}\sqrt{\frac{g}{l}} \tag{3.2}$$

The phenomenon of resonance is closely related to sound. Sound is usually generated by a vibrating, moving object, such as a membrane or a string. A string generates sound waves through cyclic movements and has a natural frequency of its own, much like the pendulum. Therefore, it also vibrates more intensely when driven by an external oscillation at the frequency of one of its natural modes. Resonance and material are of importance when constructing and working with musical instruments, and likewise with feedback systems. In a feedback loop, it will steer and determine a great deal of the tonal movements of sound within the system. Balancing the influencing resonances can contribute to a more complex and varied sound.

3.1.3 Pipes

Pipes have been very commonly used in the construction of musical instruments. From woodwind instruments, brass instruments to organ pipes among many more. This is most likely due to the controllable, reproducible tonal behaviour of a pipe. Consider a simplified version of a trombone, for example. Essentially two bent pipes, one fit into the other, with a mouthpiece. This simple instrument enables the player to control and play a wide range of tones by extending and retracting the acoustical pipe.

Harmonics

Pipes have a very specific tonal characteristic. The length, the diameter, holes and whether one end or both ends are closed or open influence the set of resonance frequencies of the pipe. The resonance frequencies (modes of vibration) are called *harmonics* if the modes are whole-numbered multiples of the fundamental. The term *overtones* is used for all higher modes of vibration, including not whole-numbered ones (Rossing et al., 2002). How sound waves propagate within a pipe can be closely modelled. Utilizing these models allows for the creation of precise, controlled, pitched tones.

There is an important distinction between two kinds of pipes. The pipe can be open or closed. A closed pipe is closed at one end and open at the other, while an open pipe is understood as being open at both ends or closed at both ends. This differentiation matters as sound propagates differently in each. An open pipe has both odd- and even-numbered harmonics, while a closed pipe includes only odd-numbered harmonics (Rossing et al., 2002). The n-th frequency resonance of a straight pipe is given with the following equations. Equation (3.3) is used for closed pipes and equation (3.4) is used for open pipes. F_n is the resonance frequency, n (a positive integer) is the order of the harmonic, c is the speed of sound of the medium the sound propagates through, and L is the length of the pipe. For closed pipes, n is required to be an odd number (Kinsler et al., 1982).

$$F_n = \frac{n * c}{4L} \tag{3.3}$$

$$F_n = \frac{n * c}{2L} \tag{3.4}$$

Assuming sound propagates through air with around $c = 343\frac{m}{s}$ and the length of the pipe is $L = 2m$, then the lowest frequency resonance of a closed pipe with these properties would be ≈ 42.875 Hz. The lowest frequency resonance of an open pipe with the same values would be ≈ 85.75 Hz, therefore one octave above.

Sound radiation of pipes

In section 3.1.1, sound wave propagation was described. Some of the aspects find relevance in context of pipe sound radiation. Of particular importance in this paragraph will be diffraction. When sound travels through a pipe, it will radiate and diffract at the opening of the pipe. Considering the sound source is a loudspeaker, placed at one end of the pipe, then the directivity of the sound will be affected by this. It is worth mentioning that the pipe itself can vibrate and radiate sound outwards, however this is will not be examined here. Beranek and Blackstock (1988) examined the radiation of loudspeakers in a theoretical study, represented with a vibrating rigid piston. One scenario describes a piston mounted in the end of a long tube. The model proposed gives information on the directivity of the sound propagation. Figure 3.4 shows polar diagrams depicting the radiation power in dB at specific frequency-cone ratios ka, where $k = \frac{\pi}{\lambda}$ and a is the radius of the cone in meter. ka is the radius divided by the wavelength (Eargle, 2003).

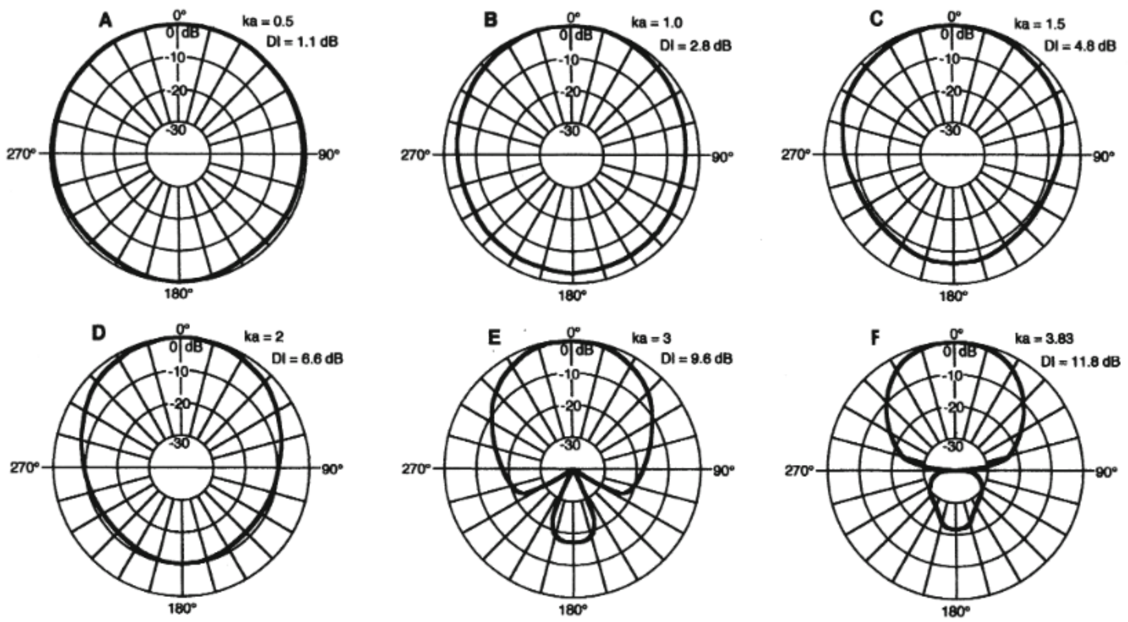

Figure 3.4: Directional response of a piston mounted in the end of a long tube.

Source: Eargle (2003) with data after Beranek and Blackstock (1988)

The directivity levels for the pattern are determined at a distance comparatively large to the piston diameter so that the pattern applies to the far-field. The observation can be made, that the sound radiates more omnidirectional at lower ka and more directional at higher ka. The effects of diffraction play an important role here.

3.1.4 Speakers and microphones

In this section, some fundamental properties of speakers and microphones will be examined. They are of relevance for feedback systems that make use of the Larsen effect.

Speakers

A dynamic loudspeaker converts electricity into the kinetic movement of a cone (sometimes called a diaphragm or membrane), that in turn changes the pressure of the medium surrounding it, which is most commonly air. This conversion takes place within the speaker system with the use of magnets and a coil. The coil conducts the electrical audio signal that the speaker system is provided with. The cone of a speaker is connected to one end of the coil. This bare-bones speaker setup can be resembled in a simplified mass-spring system. The mass consists of the combined mass of the coil, the cone and the air moving with it, and the spring constant is determined by the suspension elements (Rossing et al., 2002).

Frequency response While a dynamic loudspeaker is more complex than the models I use here, the models are important in this context and help to outline frequency response behaviours of speakers. When examining the behaviour of a dynamic loudspeaker, it can be observed that it exhibits a non-linear frequency response. This means, that the speaker system reacts more readily to specific frequencies compared to others. Deriving from this observation combined with the mechanical mass-spring system model, it can be concluded that the driver system likely has a resonance frequency as well. This is the frequency at which the loudspeaker most readily responds to an electronic signal. However, further components and factors not covered here can influence the frequency response (Eargle, 2003).

The natural, undampened resonance frequency f_d of a loudspeaker can be calculated with the equation (3.6), where K is the spring constant and m is the mass. The spring constant can be derived from the speaker's compliance C reciprocal, as in equation (3.5) (Rossing et al., 2002).

$$K = \sqrt{\frac{1}{C}} \quad (3.5)$$

$$f_d = \frac{1}{2\pi}\sqrt{\frac{k}{m}} \quad (3.6)$$

Directivity and acoustic short-circuiting Speaker systems are usually mounted inside housings. The housing assists to control and direct the radiation of the sound produced by the speaker and reduces the effects of a phenomenon called *acoustic short-circuiting.* When a dynamic loudspeaker is producing sound, the movement of the cone produces a change in pressure in the air surrounding it. While this is desired to produce sound, it is important to note that this will happen on both sides of the cone, in front, and on the rear. However, the pressures oscillate inversely to one another: when the cone moves forward, the air pressure is increased in front of the cone and decreased in the rear and vice versa. Sound will be generated on both sides of the cone, and the speaker is acting in a dipole manner. Therefore, when a loudspeaker driver is used in a setup without a housing (*unbaffled*), the sound waves produced at one side of the speaker will bend around the edge of the speaker and meet the sound waves produced at the other side of the speaker. When the sound waves radiate spherically, they will meet along the frontal plane of the cone. In a polar diagram depiction as in figure 3.5, the short-circuiting would occur at 90° and 270° relative to the cone direction. On this plane, the sound pressure produced in the front and back effectively cancel each other out, creating the so-called *acoustic short circuit* (Rossing et al., 2002). This can be very effectively avoided by mounting the speaker in a large wall or enclosing.

Figure 3.5 shows the directional response of an unbaffled piston, resembling the loudspeaker behaviour. It can be observed that the directivity changes for different values of ka: higher frequencies with shorter wavelengths radiate more directional than lower frequencies. This is due to the effect of diffraction, as elaborated on in 3.1.1.

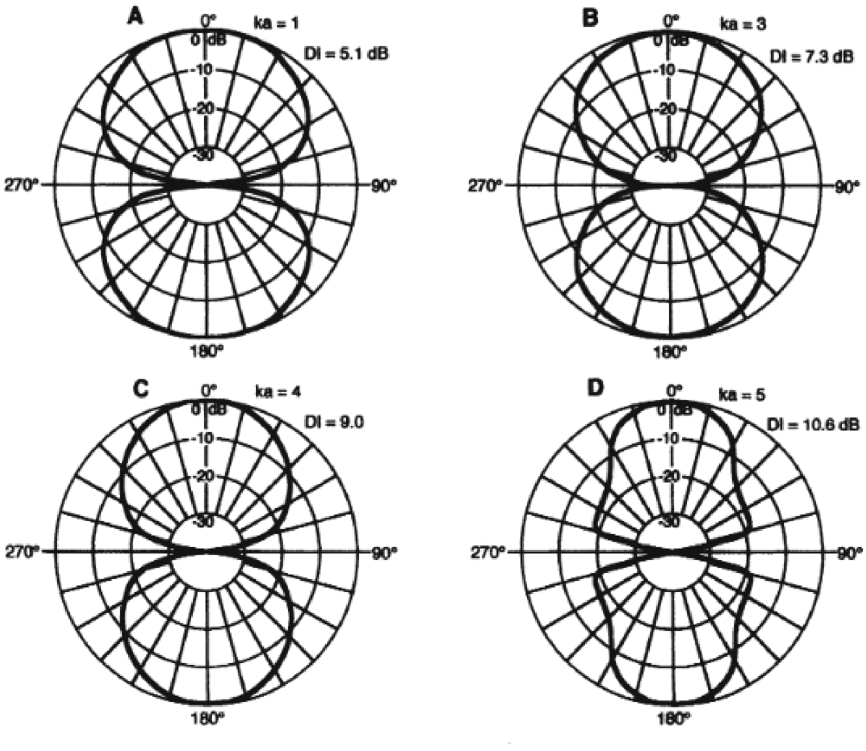

Figure 3.5: Directional response of an unbaffled piston.

Source: Eargle (2003) with data after Beranek and Blackstock (1988)

Microphones

Microphones are transducers that produce an electrical signal when actuated by sound waves (Rossing et al., 2002). There are several types of microphones that incorporate different techniques to capture sound waves. In the context of this work, I will focus on condenser microphones, as they are the most relevant due to their generally high sensitivity, frequency response and compact design. Condenser microphones make use of capacitance to generate an electrical signal. They commonly consist of a thin, electrically conductive, movable membrane (*diaphragm*), and a separate solid metal plate, located close to it. These two components act as a *capacitor*, also called *condenser*, when provided with the necessary power supply to polarize the metal plate. When the diaphragm gets actuated from sound waves, it moves back and forth. The movement causes changes in the distance between the diaphragm and the metal plate, which results in a small flow of current in the circuit. The changes in capacitance results in a change of voltage, which serves as the audio signal (Rossing et al., 2002). Though this may come as no surprise by now, it is worth mentioning that microphones exhibit non-linear frequency responses as well.

A special type of condenser microphone is the electret microphone. The difference to other condenser microphones is that the metal plate of an electret microphone is pre-charged. Therefore, there is no need of a polarizing power supply to operate electret microphones. However, as they usually include an integrated preamplifier, the microphone still needs to be supplied with a small amount of power.

Pickup patterns The directivity of a microphone is commonly visualized through *pickup patterns*. These polar patterns give information about the sensitivity of a microphone relative to the direction. The microphone response in decibel is usually expressed through the length of the radius. The most common pickup pattern types are *omnidirectional*, a uniform sensitivity of the microphone to all sides, *unidirectional*, a maximum sensitivity in one direction, and *bidirectional*, a maximum sensitivity in two directions. One of the most common unidirectional pickup patterns is the *cardioid*, exhibiting a kidney-shaped curve sensitivity pattern (see 3.6) (Sengpiel, 2002).

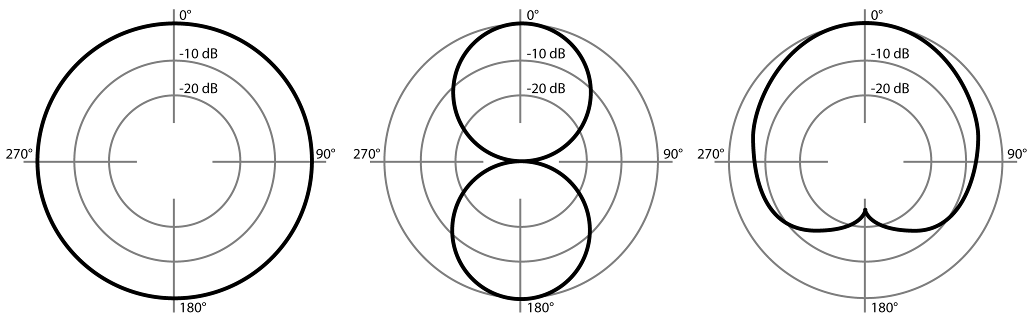

Figure 3.6: Microphone pickup patterns, from left to right: omnidirectional, bidirectional and cardioid.

3.2 Audio signal processing

Audio signals can be represented both analogue and digital. In both domains, the signal can be processed with various techniques to alter it for different purposes and results. The most common musical applications are processes that manipulate the amplitude, the dynamics of a signal, or the frequency spectrum. In the digital domain, the signal manipulation is called *digital signal processing* (DSP) and is very commonly done in a *digital audio workstation* (DAW), sometimes with the use of external plugins.

3.2.1 Compression

Compression is a process that changes the dynamics of an audio signal. A compressor allows for altering the difference between loud and quiet segments in a signal. Usually, it is used to reduce the amplitude of a signal when it crosses above a *threshold*. The threshold is expressed in negative decibels. The amount of reduction (or expansion) is determined by a *compression-ratio*, also sometimes simply referred to as *ratio*. When the signal crosses the threshold, the compression kicks in. An *attack* parameter, which is usually expressed in milliseconds, determines how quickly the compressor starts to affect the amplitude. How fast the compression is disengaged, after the signal is below the threshold again, is determined by the *release* parameter, also usually expressed in units of milliseconds. A *knee* describes the curvature of the threshold. A hard knee is essentially a point after which the compression is fully engaged, while a softer knee can affect the signal earlier, yet to different amounts. A specific form of the compressor is the *limiter*. A limiter has essentially a compression-ratio of ∞:1 and usually a very short attack time. The limiter helps in keeping the amplitude of the signal strictly limited to the set threshold.

3.2.2 Filtering

Filtering is the process of attenuating frequencies in an audio signal. The three basic types are *low-pass*, *high-pass*, and *band-pass* filters. The low-pass filter allows for high frequencies to be filtered out and the high-pass filter for low frequencies to be filtered out. They are also sometimes called high-cut and low-cut. The *band-pass* filter passes the signal within a frequency range (band) and attenuates all other frequencies. The opposite is a *band-reject* filter, also called a *notch* filter. A frequency band of the signal gets attenuated, while all other frequencies pass. The *cut-off frequency* of a filter is the frequency at which the attenuation of a low- or high-pass filter fully kicks in. At this frequency, the response will have dropped to -3 dB (71%) of the input value (Rossing et al., 2002). The *centre frequency* is the frequency in the centre of a band of frequencies and is used for band-pass, and -reject filters, for example. The *Q factor*, also sometimes called *resonance*, describes how narrow or wide the bandwidth around the centre frequency is. A higher Q results in a narrow band and a lower Q in a wider band (Truax, 1978). Additional filter types are *high-* and *low-shelf* filters, which attenuate all frequencies higher or lower than the cut-off frequency.

Equalisation is a method of filtering that utilizes the types described above, yet also allows for boosting frequencies. A parametric equaliser allows for applying filters at variable frequency positions. It usually provides *peak* filters, a type of filter which attenuates or boosts frequencies around a frequency centre. A special type of equalisation is *dynamic equalisation*, which essentially combines compression and equalisation. With a dynamic equaliser, the amount of attenuation or boost applied by the filter is reactive to the signal amplitude at that frequency band (Rossing et al., 2002; Truax, 1978).

3.2.3 Reverb

Reverberation is an acoustic phenomenon that occurs due to reflections of sound. Compared to an echo, reverb does not necessarily have distinguishable reflections and is usually diffuse due to a great number of reflections. Closed structures or rooms with hard surfaces and little sound produce reverberation. This can be commonly observed in large halls or churches. In audio signal processing, reverb effects usually aim to simulate the behaviour of acoustically reverberated sound. One characteristic of reverberation is the reverb time T_{60}, expressed in seconds. T_{60} is defined as the time it takes for a reverberated sound to decay by 60 decibels after the initial sound stopped (Rossing et al., 2002). Furthermore, a *pre-delay* time is common. It describes the time between the initial sound and the reverberation.

3.2.4 Latency

It is important to mention that the processes described above and all other digital audio signal processing methods are bound to cause delay in a signal flow. The term *latency* is used to describe this kind of delay. The latency might be very small for an individual process, though the delays can quickly add up. The reason why latency occurs is that the necessary computation, as any other computation, is never immediate. The more computation intensive the process is, the longer the latency will be. Main contributors in audio systems are usually the analogue to digital and digital to analogue conversion and audio buffering to optimize processing routines.

3.3 Audio feedback

Feedback describes the process of utilizing some of the output of a system as some of the input of the same system, hence *feeding* the signal back into itself. The system can consist of several components, all of them interdependent. This concept finds its relevance in a wide variety of fields and disciplines. It can be observed in climate, for example; an extremely interwoven network of a great amount of interdependent actors and components forming a network of reactive feedback connections (Sanfilippo, 2020).

In this work, feedback is viewed mainly through the lens of information theory and control systems. Hence, it is the product of an output signal and an input signal forming a loop within a signal flow, as illustrated in figure 3.7. In a theoretical world, we can dismiss any delay in the process. However, due to the physical properties of the real world, delay is implied to occur in any practical application of feedback systems. The usage and behaviour of a feedback loop, of course, depends on the design of the system and its components.

Feedback can be divided into two types: *positive* feedback and *negative* feedback. Positive feedback is defined by a direct input-output relation: when the output increases, the input increases as well. Vice versa, when the output decreases, the input decreases as well. This growth or decay response can be, and commonly is, exponential. Negative feedback loops on the other hand exhibit an inverse input-output relation. An input increase causes an output decrease and vice versa (Ashby, 1956; Heylighen & Joslyn, 2003; Sanfilippo, 2020; Sanfilippo & Valle, 2013).

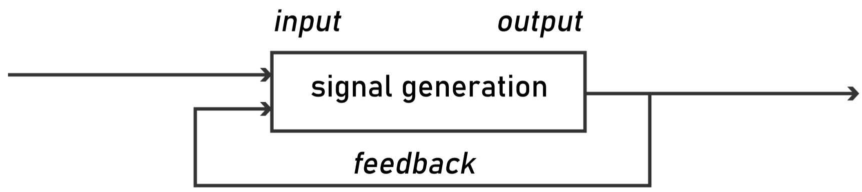

Figure 3.7: General feedback signal flow.

In the 1960s, the idea of feedback systems, cybernetics and closed, autonomous systems gained popularity. Alongside, audio feedback was utilized in musical applications. One of the most iconic and popular examples is Jimi Hendrix' usage of a guitar and an amplifier to create feedback. The range of works utilizing feedback methods musically has expanded greatly since (Sanfilippo & Valle, 2013).

To describe audio feedback, we can draw from the generalised definition of feedback and view it in the context of audio. Hence, an audio feedback system is one that ultimately generates an audio signal and uses some form of this audio signal as some of its input signal. This loop can result in self-organizing and self-oscillating, cybernetic audio generation. Often, audio feedback is an unintended effect to be avoided or diminished. However, different techniques can be utilised to generate and shape sound utilizing unique behavioural characteristics that come with the specific feedback system. The combinatorial application of this rather simple concept can take numerous and complex manifestations, which shows in the variety of produced works and artistic practices.

3.3.1 The Larsen Effect

Likely, the first connotation that comes to mind when thinking of audio feedback is the loud howling of an amplified microphone signal. This howling is the result of the *Larsen effect*. How this effect comes about can be illustrated with a simplified microphone-speaker signal (see 3.8). A microphone captures sound, creates an electrical audio signal that recursively serves as the input signal for the system. The signal runs through an amplifier, and a speaker transforms the audio signal back into sound waves, forming the output of the system. Supposing there is sufficient amplification of the signal and that the distance between the speaker and microphone is small, the sound produced by the speaker will be picked up by the microphone. Above a certain threshold of amplification, the signal amplitude will continuously increase as each iteration is louder than the previous (Boner & Boner, 1966). Much like the pendulum described in the section on Resonance (3.1.2), the amplitude is increasing with each cycle when the "force" (in this case the re-captured output sound) is bigger than the loss in the system. However, in this case, the re-captured output ('force') is exponentially increasing as it results from the input. This system will also have a natural frequency (resonance frequency), similar to the pendulum, at which the increase most readily occurs. The sound will eventually saturate, overloading a component in a system. The result is a pitched, sustained feedback tone (Sanfilippo & Valle, 2013).

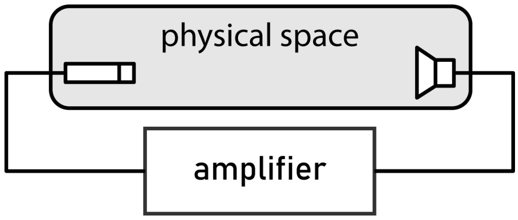

Figure 3.8: Microphone-speaker feedback loop.

Amplification plays an important role in most audio feedback systems. Sufficient amplification is required for sustained feedback. When there is too little amplification, the signal of every iteration will have a smaller amplitude than the one from the previous iteration. Therefore, no sustained feedback can occur, and any feedback tone will eventually ring out. Additionally, some initial excitation is necessary to get sustained feedback going. When the excitation is achieved through idle ambiance sound combined with a steady increase in amplification, the result will eventually be a very loudly amplified tone. Sustaining audio feedback requires less amplification than exciting it, with this approach. If a sustained feedback tone is achieved, then the amplitude will increase with each iteration, as described above (Boner & Boner, 1966). Theoretically, the increase would never halt; practically, however, the signal amplitude will eventually be limited by saturation in one or more system components. It will remain in a stable state due to the real world limitation given. In applied audio feedback systems, these limitations can be set by the power provided by the amplifier, for example.

3.3.2 Resonances

Unlike the pendulum resonance example in 3.1.2, it is not as straightforward to define a natural frequency for audio feedback systems. There are usually many factors influencing the resonance state. Since it is challenging to determine the resonances of a feedback system due to the sheer complexity, it is therefore useful to look at components in the system that are contributing to resonant states. Some components affect the system more than others. Eldridge et al. (2021) use the word *attractors* to describe the components that attract the system to a specific state. Even in a simple scenario as illustrated in figure 3.8, there is a culmination of various attractors: the frequency response of the loudspeaker, the frequency response of the microphone and of the amplifier as well as the delay caused by the system are the most apparent ones. They steer the feedback to varying extent, though all of them are part of the whole.

3.4 Complex systems

As the name suggests, complex systems include those systems that exhibit processes and behaviours of high complexity. Therefore, it is relevant to further inspect what complexity means. According to Weaver (1991), a system can be classed as *simple* when its behaviour can be described statistically with few variables. On the other end, a system behaviour determined by many combinatorial states, variables, processes and interactions can be classed as *fully complex* (Walker, 1971). Quantity and aggregation appear to be significant factors that contribute to complexity. The mere multiplicity of very simple processes and components interacting with one another can result in high complexity. The observation of potentially intricate behaviours and properties of a system can give indicators about the complexity of the whole. Sanfilippo (2020) describes that there is a strict relation between emergence and complexity, concluding that the observer needs to be taken into account as well.

Feedback systems fall under the category of complex systems. They commonly exhibit simple processes culminating in complex behaviours (Sanfilippo & Valle, 2013). Depending on the design of an audio feedback system and the layer of observation taken, the complexity will vary. However, feedback based audio instruments and installations, such as the Halldrophone (Úlfarsson, 2013), the Proto-Langspil (Armitage et al., 2022) or the installation *Empty Vessels*, are characterised by their intricate and complex behaviour. The complexity of these systems, arguably, provides most of their intriguing and interesting musical value. Eldridge et al. (2021) describe and analyse self resonating vibrotactile feedback instruments. They suggest a core design principle of multiple attractors for such feedback instruments. This design principle inherently entails the increases in the complexity of the instrument behaviour. When a great variety of components, attractors and variables are within an audio feedback loop, the processes will be more complex. The study and analysis of feedback based installations and instruments can be a challenge. Traditional models and methods can provide a good starting point and reference for some aspects, though they might fail to describe the observable behaviour extensively and appropriately enough, according to Eldridge et al. (2021).

3.4.1 Cybernetics

The term cybernetics is a neo-greek expression, formed from the Greek word *kybernetikos*, $\kappa\upsilon\beta\epsilon\rho\nu\eta\tau\iota\kappa\acute{o}\varsigma$, which translates to *steersman.* It was coined by Wiener (2000) in 1947. They intended to give a name to the entire field of control and communication theory regarding both machine and animal. Feedback and cybernetics are closely related in their definitions. This influenced their choice of nomenclature, and Wiener mentions in his book *"Cybernetics"* that:

> *"[...] the first significant paper on feedback mechanisms is an article on governors, which was published by Clerk Maxwell in 1868, and that governor is derived from a Latin corruption of* $\kappa\upsilon\beta\epsilon\rho\nu\eta\tau\iota\kappa\acute{o}\varsigma$*. We also wish to refer to the fact that the steering engines of a ship are indeed one of the earliest and best-developed forms of feedback mechanisms." (Wiener, 2000, pp. 11-12)*

Since then, research, studies and teachings in the field of cybernetics have grown immensely. The term cybernetics is commonly used for autonomous, non-humane systems. A thermostat, for example, as Wiener also describes in his book, is a feedback-based, cybernetic system: the temperature it reads is used as an input, based on which it will regulate the temperature (Wiener, 2000). The autonomous nature of this system implies a form of self-organizing system. A cybernetic system used for music might be a generative system. However, not all generative music is of cybernetic nature.

3.4.2 Chaos and entropy

Though an understanding of the idea of *chaos* is so widespread that an explanation might seem trivial, there are still some aspects I want to magnify. Especially the relation to entropy. Early understandings of the idea of *chaos* regarded it as the *"[...] formless void from which order springs"* (Skarda & Freeman, 1987, p. 173). In that definition, we can find a relation between past and future states. Chaos might more commonly be understood as the state of disorder, unpredictability and lack of regularity. However, there is a distinct mathematical theory for chaos which relates past, present and future. The definition additionally describes systems sensitive to their initial states (Sanfilippo, 2020). This applies to feedback systems and suggests the possible emergence of chaos within them. It is also important to note that there is a distinction to be made between randomness and chaos. Randomness does not relate to previous states and the outcome does not affect its stochastic elements (Sanfilippo, 2020).

Entropy can be seen as a unit and descriptor of the level of chaos. Entropy is a concept with a long history and differing meanings and teachings depending on the field. Entropy in the context of information theory can be understood in the following sense, as Meyer (1957) described: for a highly organized situation with high probabilities of outcome the entropy of the information is understood as *low*, while for the opposite, in a situation with high disorganization and unpredictability, the entropy is understood as *high*. The concept can be applied and utilized in music and musical instruments. Feedback instruments are often characterized by complex behaviour and uncontrollability (Kiefer et al., 2020). This is a key factor to be taken into account when designing a feedback instrument, as there is a fine balance between controllability and instability. Arguably, complete mastery of an instrument might be what an instrumentalist strives for. Yet an instrument might also be compelling when the entropy is balanced such that there is enough controllability paired with unpredictable, unexpected behaviour.

Chapter 4

The sound installation

In this chapter, I will present a sound art installation that uses audio feedback to generate a musical ambiance. The focus of this chapter is to give an in-depth insight into the systems' workings and a thorough description of the processes utilized. Alongside, the reasoning for my decisions will be elaborated upon. It starts with an overview of the project with an outline of the concept, followed by a detailed description of the construction and its components. The installation as a whole is the sum of numerous individual components. Examining them individually provides a better understanding of their purpose, contribution and function within the bigger picture. Different signal processing methods are described as they are utilized to steer the installation to behave in a compelling manner. Balancing the resonances and attractors within the system plays an important role in that. Therefore, the tuning process for the system will also be described in detail. Attached to this thesis are audio and video snapshots from prototypes throughout the process.

4.1 Overview

Exploring complex musical behaviour through the process of designing and constructing an audio feedback sound art installation is the foundational concept of this work. Throughout the exploratory prototyping process, specifics and desired properties for the installation emerged. The idea of exploring multiple individual feedback loops interfering and interacting with one another (therefore creating overarching loops) became an essential aspect early on. Likewise, did the use of the harmonics of pipes as dominant attractors. The installation should be sensitive to changes in the system and capable of communicating this through the sound. This goes hand in hand as the sound is integral to the feedback system. The aspiration of a sonically rich and varied ambiance was constant throughout the process, alongside the inclusion of some interactive component. Towards later prototypes, a more significant embedding of the room and the audience into the feedback system became of interest. Even though the sound generation can be understood as autonomous, the audience should have some ability to influence it by playing a role in the sound generation. Another aspiration, driven by aesthetical and personal motivations, was to construct the installation from as many reclaimed materials as feasible.

For the installation, several pipe modules are hung freely in stacks at different locations in an enclosed space. Each pipe module has a microphone and a speaker placed at the pipe openings. The sound is generated through multiple feedback loops, utilizing the Larsen effect. By using pipes, there are strong resonance frequencies acting as attractors in the system. The generated sound radiates from the speaker, through the pipe openings and propagates through the surrounding space. Therefore, the spatiality of the installation, as well as the properties of the surroundings, play an integral part in the feedback behaviour. The feedback system can be influenced by rotating the pipes, changing their vertical angle, by changing their positioning and, through the presence and movement of the observer within the room. As the surrounding acoustic properties are embedded in the feedback system, changes to the sound propagation within them should affect the generated sound.

4.2 Construction

To realise this installation, it is necessary to build and assemble several components. This chapter serves as a thorough description of the individual components. An overview and a list of all utilized components is provided. The relevant properties of the audio-equipment are stated, including the speakers, microphones, interface, amplifiers and the DAW. The pipe modules are described, followed by a description of the hanging system and is functionalities. Everything is put into the context of the installation as a whole.

4.2.1 Components

Each of the pipe modules described below houses a condenser microphone and a dynamic loudspeaker. The microphones and speakers are wired to 3.5 mm and 6.2 mm audio jack sockets for an easier, more modular setup. Six metre long cables connect the microphones to an interface. One end is equipped with a XLR connector together with 27 $k\Omega$ resistors, the other with a 3.5 mm audio jack plug. Likewise, cables of the same length, with two copper strands at one end and a 6.2 mm audio jack plug at the other, connect the loudspeakers with the amplifiers. Four small stereo Class-D amplifier chips provide 20 Watt output per channel on terminal blocks. Each of them requires a power supply that provides 60 Watts, to ensure enough buffer. The audio input for the amplifiers is provided through a 3.5 mm stereo socket. Therefore, the amplifiers are connected to the interface with four cables, with a 3.5 mm stereo audio jack plug and two 6.3 mm mono audio jack plugs. The interface is connected to a computer via USB. Strings are used to hang the pipe modules in place. Table 4.1 below holds a complete list of all components.

Component	Amount
Pipes (cardboard and metal)	8
Electret-condenser microphone capsule (EMY-63M/P)	8
Speaker drivers (various types)	8
Audio jack sockets (3.5 mm and 6.2 mm, each)	8
Microphone cable, 6 metre (XLR to 3.5 mm jack plug)	8
Speaker cable, 6 metre (6.3 mm jack plug to copper strands)	8
Amp-Interface cable, 3 metre (3.5 mm stereo to 6.3 mm mono jack plug)	4
Amplifier, stereo 20W Class-D board (Adafruit MAX9744)	4
Power supply (60W, 10 V, 5 mA)	4
Interface, 8 input (phantom powered), 8 output (Focusrite 18i20)	1
Computer with DAW	1

Table 4.1: List of all components.

4.2.2 Speakers

The speaker drivers used in this installation are reclaimed from unused or thrown-away equipment. Therefore, there is no uniformity of build type. The variation in size and build across the speakers adds an interesting factor to the system, as each type contributes with its unique characteristics. All speakers are removed from any housing. As described in 3.1.4, dynamic loudspeakers exhibit non-linear frequency responses. Having speakers with distinct frequency responses adds to the complexity of the system and increases the variety of characteristics of the individual feedback loops. All speakers utilized have an impedance of 4Ω. The cone diameters are fairly small and range from around 3 mm to 10.5 mm.

4.2.3 Microphones

The microphones are all the same build type: an electret condenser microphone capsule with omnidirectional pickup pattern. The capsules are 5 mm high, 6 mm in diameter and contain a small pre-amplifier. The signal-to-noise ratio is about 60 dB, which becomes relevant for the signal processing described later. The electret condenser microphones need to be supplied with sufficient power to operate properly. The specific capsules used for this installation require between 1.5 Volts and 10 Volts. To be able to supply the right voltage using common phantom power, which is 48 Volts, a resistor with 27 $k\Omega$ was added to the microphone cables. This brings the voltage down to a sufficient amount of around 6 Volts.

4.2.4 Interface and amplification

Two components are essential to process the microphone signals digitally and to amplify them for the speakers: an audio interface and an amplifier. The interface used is a first generation Focusrite 18i20. It provides eight phantom powered inputs with pre-amplification and gain level control and ten balanced analogue outputs. The interface handles the analogue to digital (ADC) and digital to analogue (DAC) conversion. Via USB and the Focusrite ASIO protocol, the interface communicates to a computer, sending and receiving the digital signals. A sample rate of 48 kHz is used, and the buffer size is set to 128 samples, resulting in a round-trip latency of around 15 ms, according to the Focusrite driver.

The analogue output signals of the interface require to be amplified in order to drive the speakers. Four separate Class-D amplifier boards from Adafruit are used here to fulfil this task. At the hearts of these boards are MAX9744 amplifier chips. The chip has a wide power supply voltage range from 4.5 Volt to 14 Volt. Each board is able to drive two channels of 4 to 8 Ω speakers at 20 Watts each. I utilized four 60 Watt power supplies that provide 12 Volts (DC) each. The efficiency of the amplifier is between 88% and 93%. The input signal is taken from a 3.5 mm stereo jack socket, while the output is provided on two separate terminal blocks. As a Class-D amplifier, the MAX9744 makes use of pulse-width modulation (PWM). Therefore, the output is a high frequency pulse-width modulated square wave, relying on the inductance of the speaker to smooth out the signal. There is an integrated volume control that can be either used analogue or via I^2C. The output level can therefore be adjusted analogue with a 1 $k\Omega$ potentiometer. To keep the boards close together and somewhat protected, I repurposed an old small wooden box as a housing. The amplifiers are fit in place and holes in the box allow for all cables to be connected from the outside (see figure 4.1).

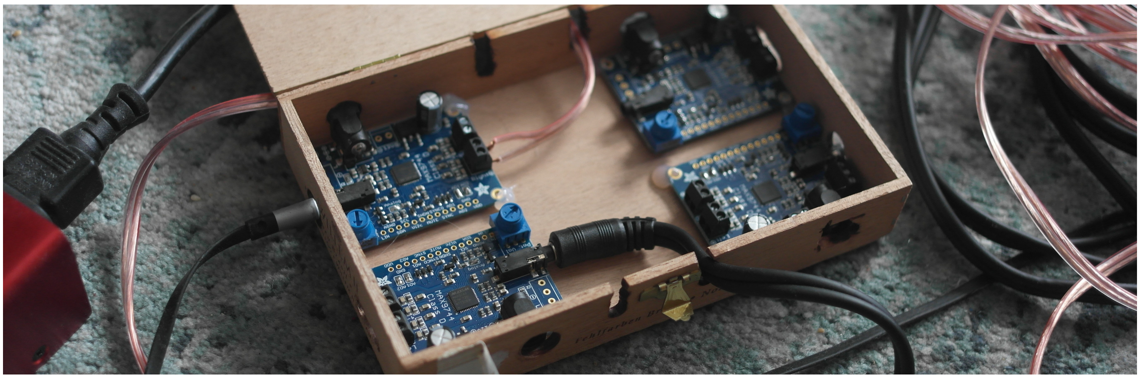

Figure 4.1: Amplifier boards and their housing.

4.2.5 Pipe modules

The pipe modules are arguably the heart of this installation. In terms of visual presentation, the modules are most likely what catches the eye first. One module consists of a pipe, a microphone and a speaker, along with the necessary connectors. There is a total of eight modules from reclaimed pipes, consisting either of cardboard or of thin metal. The pipes have different lengths, to cover a range of different harmonics and registers. The lengths were specifically chosen to be prime numbers or close to a prime number. The idea behind this is to reduce the number of similar higher harmonics (divisibility). Table 4.2 lists the pipes and their specifications: the sizes, the material, the calculated fundamental frequency resonance in a closed pipe scenario and the equivalent, approximated note. The horizontal break-lines in the list give information on which pipes are paired in stacks. The stacks will be described later in the hanging system sections 4.2.6. From this point onward, I will refer to individual pipes by their number (ID) assigned in this table.

Pipe ID	Length	Diameter	F_1, closed ($343\frac{m}{s}$)	Note
1	97 cm	10 cm	≈ 88 Hz	$\approx F_2$
2	73 cm	6.5 cm	≈ 117 Hz	$\approx A\#_2$
3	89 cm	6.5 cm	≈ 96 Hz	$\approx G_2$
4	64 cm	6.5 cm	≈ 134 Hz	$\approx C_3$
5	83 cm	6.5 cm	≈ 103 Hz	$\approx G\#_2$
6	69 cm	6.5 cm	≈ 124 Hz	$\approx B_2$
7	79 cm	6.5 cm	≈ 109 Hz	$\approx A_2$
8	59 cm	10 cm	≈ 145 Hz	$\approx D_3$

Table 4.2: Pipe groupings and specifications.

Plotting the calculated harmonics of the pipes provides a reference for the distribution of harmonics and the locations of possible clusters. This can be an assistance in choosing the pipe lengths. The top plot in figure 4.2 shows the harmonics for the utilized pipe lengths. The lower plot shows all harmonics in one dimension, with 30% opaque plot-points. Both plots range from 80 Hz to 3000 Hz.

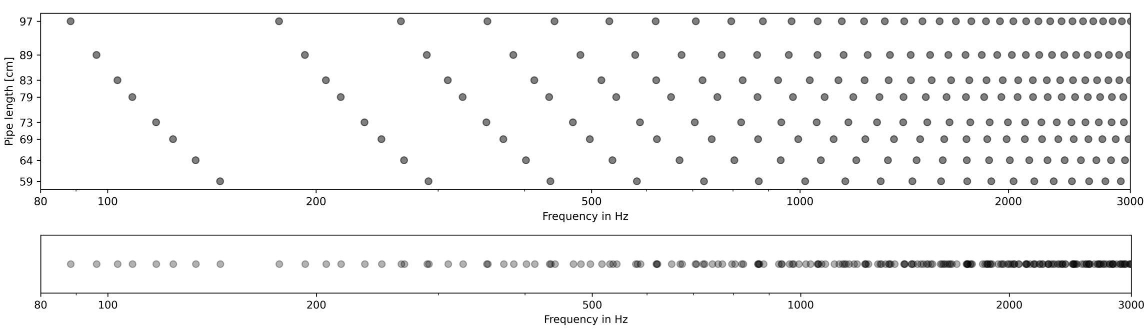

Figure 4.2: Plots of the pipes' harmonics.

Each module is equipped with two jack sockets that are mounted into the pipe wall or taped to it. They are located close to the openings of the pipe. A 3.5 mm jack socket at one end and a 6.3 mm jack socket at the other. The soldering pins of the sockets extend to the inside of the pipe to allow for the cables to be plugged in from the outside. A microphone capsule is wired to the 3.5 mm socket and a speaker driver to the 6.3 mm socket. The different connector types help with visual differentiation and ensure that the cables cannot be mismatched by accident.

The microphone capsule is placed a few centimetres outside the pipe opening. It is connected to two stiff, yet bendable, conductive wires, that are soldered to the 3.5 mm socket pins on the inside of the pipe, as shown in figure 4.3. The wires allow for adjustments of the microphone position. This is quite useful, as the position of the microphone has a drastic impact on the behaviour of the feedback loop. It impacts the balance between the pipe module sound and surrounding sound, alongside more complex relationships like reflections and frequency response.

The position and angle of the capsule determines the loudness relation between the sound captured coming from within the pipe and all other surrounding sounds captured. The surrounding sound includes the sound produced by the other pipe modules. The closer the capsule is positioned and angled towards the pipe opening, the louder the sound recorded of the pipe will be compared to all surrounding sounds, and vice versa.

Figure 4.3: Pipe module microphone placement.

The speaker is located at the opposite end of the pipe. Unlike the fairly uniform placement of the microphones across all pipe modules, the speakers are placed at one of two possible locations for each pipe module: either inside the pipe, near the opening of it or attached from the outside (similarly to a lid) closing the opening of the pipe. The cone is facing towards the inside or the outside of the pipe, depending on the pipe module. The different positions have an effect on the sound propagation. Some pipes are essentially *closed* (see 3.1.3) by the speaker as it covers the full diameter. Other speakers have a smaller diameter than the pipe, only resembling an obstruction instead of fully closing the pipe. Since the speaker drivers are mounted without housings, they will freely radiate sound from both the front and the rear of the cone (see 3.1.4 'Speakers and microphones').

Figure 4.5 below illustrates a cross-section view of a pipe module. A pipe module has a total of four holes in the pipe wall, that are equally spaced from the centre of mass. Through these holes run the strings necessary for hanging the pipe, a system which will be explained in detail below.

4.2.6 Hanging system

The hanging system dictates the modules' locations, their arrangement, the spacing to one another, the positioning possibilities, how the installation can physically be interacted with and how the observer can move through the space. These are among the most important aspects to consider in this installation due to their substantial impact on the feedback behaviour. Additionally, the visual presentation of the hanging system hints and guides interactive possibilities. It may communicate, for example, where to move and stand and where the main focus of the installation is. Therefore, the collection of pipe modules requires a purposeful, spatial positioning system.

Combining musically interesting movement possibilities, flexible positioning, with simple, non-intimidating interaction while also respecting physical and practical restrictions such as cable lengths, and accessible material was a core requirement. The method presented here consists of several module stacks that are arranged in an overarching hanging system. The system is expandable so that the amount of pipe modules within a stack and the amount of module stacks can be changed to experiment with different combinations. Figure 4.4 illustrates the proposed concept.

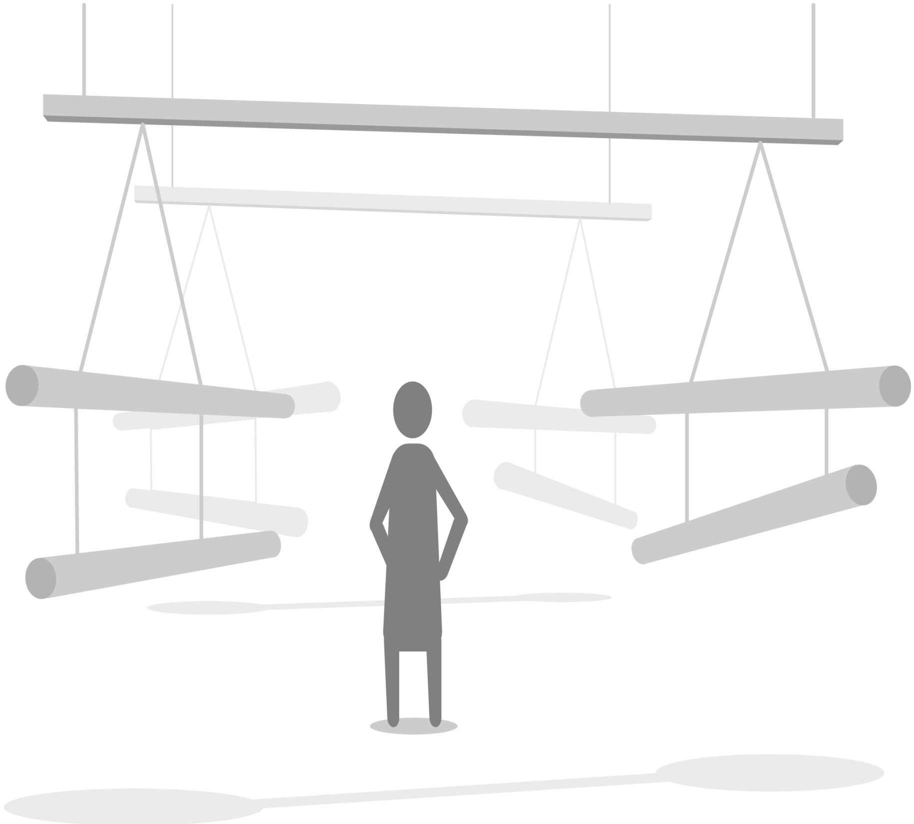

Figure 4.4: Hanging system illustration.

Module stack

The principle behind the pipe module mount is as follows: two pipe modules are vertically stacked with strings that run through them and hang them in place. A pipe module has a total of four small holes (∼3 mm diameter) in the pipe wall, purposed for the strings. One pair of holes is located on a line perpendicular to the pipe openings, on the side opposite to the connectors. The holes are equally spaced apart from the centre of mass of the pipe module. The other two holes are just opposite to them, as visualized in figure 4.5. I'll refer in the following to the side with the connectors as the *bottom* side and the opposite as the *top* side (see figure 4.5).

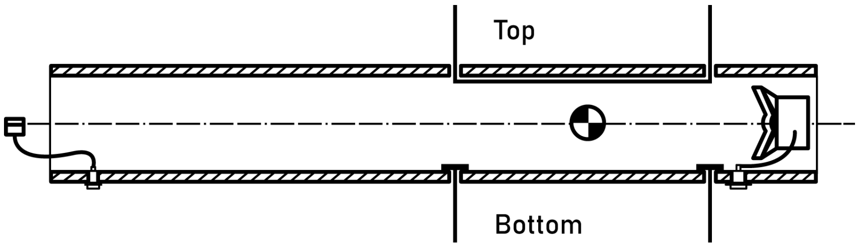

Figure 4.5: Cross-section view of a pipe module.

A string enters a pipe through one of the top holes and exits out again through the other (see figure 4.5). On the bottom side of the pipe, a separate string runs through one of the bottom holes. Here, a metal hook is knotted onto the end of the string, inside the pipe, to prevent the string from slipping back out. The other end of the string runs through the top holes of the pipe module underneath. The amount of slack determines the distance between the modules. The end of the string is fastened into place inside the upper pipe module with a small metal hook (see figure 4.5 and 4.6). If there is no pipe module underneath, then the string hangs loose, with both ends fixed into place inside the pipe. The system allows for as many modules stacked as needed. A weight is hung from the most bottom pipe module (see figure 4.6).

This method allows for the adjustment of the vertical angle of a pipe module by raising or lowering the ends of the pipe. The pipe can rest steady at various positioned angles. This is due to the friction between the string and the pipe wall combined with the careful balancing around the centre of mass. It allows for various angle combinations within a module stack (see figure 4.6). There is a maximum achievable angle, however, at which the friction does not suffice to hold the pipe in place.

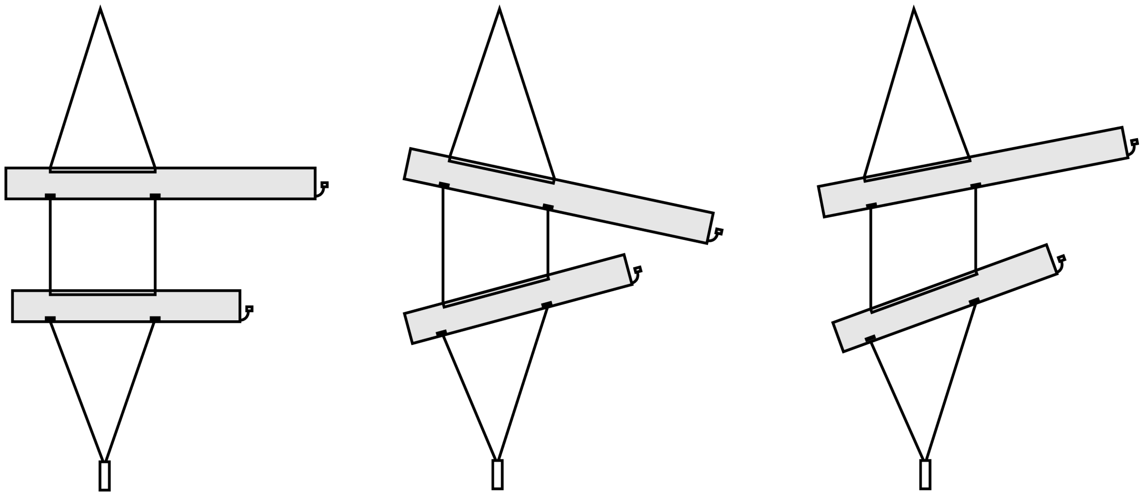

Figure 4.6: Possible module angles.

Stack arrangement

As each stack consists of two modules, there are four separate stacks that require a purposeful arrangement. There are a few factors that set boundary conditions for the stack placements. The pipe modules should not collide with each other. This would cause mechanical noise and movement that is unwanted in this context. Therefore, the stacks need to be arranged so that there is enough space for each pipe module to rotate fully around its vertical axis at the centre of mass (see figure 4.7). An additional space around that radius is necessary to take the pendulum motions into account. The extent of the swinging motion is intended to not be far. Furthermore, there should be enough space to allow for the audience to move in-between the module stacks. The arrangement might be further restricted by the availability of some resources, for example by the lengths of the cables.

A rectangular arrangement of the module stacks is used in the installation. Two stacks are hung from the ends of a wooden plank if there are insufficient hanging anchor points available. This establishes a static distance between them. The plank is placed in a fixed position, hung from the ceiling or mounted on a high stand. The top pipes of the stacks are placed at a height of around 1.70 m. Since eight pipe modules are used in the installation, two planks are necessary, holding two module stacks with two pipe modules each.

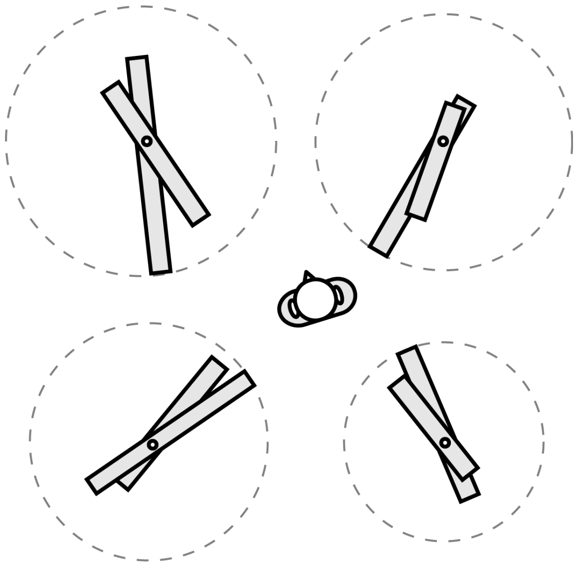

Figure 4.7: Top-Down view of the module arrangement.

Cabling

Each pipe module requires two cables two operate. The cables of all modules meet at the interface and amplifiers. Therefore, the location of that equipment in relation to the installation needs to be taken into account. Ideally, the cables should not create an obstacle for the audience and should not disturb the movement of the modules. One solution is to route the cables upwards, towards the ceiling, then out of the perimeter of the installation towards the audio equipment. This solution is only feasible if the ceiling height allows for it, or if another stand is available to serve as a 'bridge' for the cables. Another option is to simply let the cables hang down from the pipe modules and tape them to the floor. Cabling via the floor can save precious cable length, depending on the ceiling height of the space. The cables of an upper pipe are then taped to the lower pipe to prevent cables from colliding with the microphones or modules, creating unwanted mechanical noise. In both cabling scenarios, it is necessary to leave enough slack to allow for the tilt and rotation of the modules. The audio equipment is placed outside the hanging system to not interrupt the explorative space. I chose the downward-cabling solution that allowed for the furthest placement of the audio equipment away from the installation.

Space, room and surroundings

The physical space surrounding the installation is integral to the signal flow of the feedback loop. The sound has to move through it to complete the cyclical flow. The intricacies and the extent of influencing factors from the space and architecture around are complex. This installation is not intended with a specific space in mind. However, a few aspects deserve consideration. The sound propagation of the installation will be affected as soon as there is any structure or obstacle in the proximity of the installation, for example surfaces, walls, objects or people. Some frequencies might be reflected, some might diffract, sound might be absorbed and standing waves can occur. This colours the sound captured by the microphones. The extent to which the surroundings influence the system depends on many factors: for example on the distance between the surfaces and the installation, how surfaces are angled towards one another and how the microphones are positioned and angled. A position close to a wall or a corner can accentuate lower frequencies due to the reflections from their more omnidirectional propagation compared to higher frequencies. The closer the installation is to a wall, the earlier will first reflections occur with less loss of energy. A more central position will result in more uniform early reflections in all directions. The greater the distance to the walls and ceilings, the more amplification is needed to excite the reflections and room acoustics sufficiently to have significant effect on the system. Depending on the mode of operation the installation is in, which is described in 4.4.2, the room will influence the feedback behaviour to a greater or lesser extent.

4.3 Signal processing

In this section, an overview of the signal flow of the system is given. The *digital signal processing* (DSP) is described along with its use in the context of the feedback system. Compression, filtering, reverb, limiting and gain are utilized. The central aim of the applied signal processing is to decrease the stability of the feedback system and in turn increase its sensitivity (instability) to changes within the signal flow. The processes provide some degree of control over this.

4.3.1 General signal flow

In a running feedback loop, as the continuous cyclical flow takes place, it is hard to pinpoint a root of the signal. Here, I will trace the signal cycle, starting from the microphone capsule (see figure 4.8). The microphone of a pipe module captures sound waves in a physical space. The signal is received by an audio interface. The interface provides the necessary phantom power and converts the analogue signal into a digital signal. In the digital domain, the signal is passed to a computer via USB. There, the signal is processed in a DAW. Digital audio signal processing is applied inside the DAW and the routing of the channels to the outputs of the interface is set. The processed signal is sent back to the interface, where it is converted into an analogue signal again. The signal is then fed into an amplifier, which drives a speaker with the signal providing the required power. For simplicity, we can assume the amplifier as a single unit here, despite the four separate amplifiers used in the installation. The speaker produces sound waves in the physical space. Therefore, the signal cycle completes itself here, creating a feedback loop as the sound waves reach the microphone.

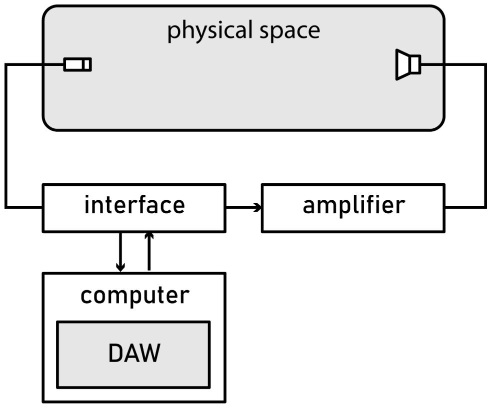

Figure 4.8: General signal flow of a single feedback loop.

This is the basic signal flow of one feedback loop. A lot is happening within the physical space that influences the signal, especially when multiple instances of feedback loops are taken into account. Within the digital domain, in the DAW, the signals are passed through a chain of processes individually, in separate channels. Each channel is equipped with a compressor, an equalizer, a dynamic equalizer, reverb, a limiter and a gain stage (see figure 4.9). In the following, the individual processes are described.

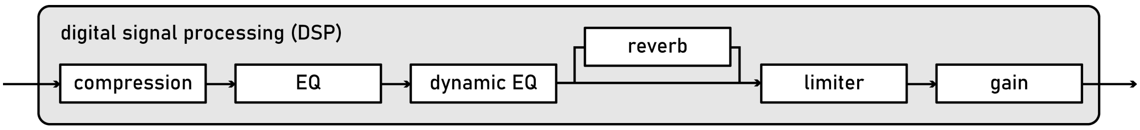

Figure 4.9: Digital signal processing chain in the DAW.

4.3.2 Compression

In musical applications, compression is commonly utilized to reduce dynamics. Here, compression is used to gain more control over the output level and the sustain of the feedback signal. As the volume of the feedback loop increases exponentially, it would eventually overload a component in the chain and saturate. This can be avoided with compression, providing a more nuanced level control. The compression allows for feedback to occur at quieter playback levels. The reason for this is that the input level can be increased, and therefore quieter signals are essentially played back louder. The feedback is falling into a sustained state, and the compression counteracts the exponential volume increase. The compression parameters are generally set quite drastically here: a very low threshold level, at around 30 dB, combined with a moderately high compression ratio of around 10:1. The lower the threshold, the more reduced will the signal-to-noise ratio be, considering the resonance frequency is the 'signal'. However, a low threshold also allows for quieter feedback. The attack time of the compressor is at around 10 ms and the release time at around 50 ms. Different release times can cause a form of oscillating amplitude modulation in the feedback loop at specific resonance states. This can be counteracted somewhat with shorter times, though this depends strongly on the scenario. The specific compression parameter values differ slightly in each signal chain as a result of the fine-tuning process.

4.3.3 Equalisation

The resonance frequencies of the pipes can become quite pronounced. Not all harmonics of the pipes and of other components are equal in their attraction force in the feedback loop. Some frequencies are far more dominant than others. This is counterproductive to the aim of inducing instability. Therefore, careful filtering with a parametric EQ is done to counteract and balance out some of the unevenness between the resonance frequencies. The frequencies are attenuated with several peaking filters with a narrow bandwidth (Q of around 13). They are applied at the most dominant resonance frequencies that arise. The gains of the filters are repeatedly and are re-adjusted by a simple, yet thorough, process that is an essential part of the tuning described in 4.4.3. It can be observed that, generally, the more iterations of the filtering applied, the less the difference between the attraction forces of the resonance frequencies will be. This increases the instability of the system along with the sensitivity to changes within the feedback loop. However, this process also heavily reduces the signal-to-noise ratio with each iteration. See 4.4.3 for more details on this.

Dynamic equalisation Additionally, dynamic equalisation is utilized. Unlike the static gain of the filtering described above, dynamic equalisation offers the possibility of a reactive gain level. This allows for another level of adjustment and affects the system differently than the filtering described above. For the dynamic filtering process, the attenuating peaking filters are placed also at the most dominant resonance frequencies in the same way as described above. The threshold is adjusted, so that the frequency is reduced by an amount that allows for further frequencies to occur and be sustained. The behaviour of dynamic equalisation in the context of feedback is quite different from the parametric equalisation described above. The frequencies can essentially be 'mixed' as they steady around a certain threshold level, not ducking away as quickly as with a common EQ. The process therefore feels more like setting the levels of the specific frequencies.

4.3.4 Reverb

Reverb further assist in steering the system into a sustained feedback loop, similar to the compression. However, it does so in another fashion, with different results and side effects. Unlike the other methods in the processing chain, the processing for the reverb is done in parallel (see figure 4.9). This means that the reverb is applied on a duplicate of the signal, while the original signal stays unaltered. Both signals are then mixed together. The reverb essentially sustains the audio signal artificially. The system is more likely to react to an excitation and tune into a resonance frequency, due to the longer sustain of tones. The reverberated signal can also influence the signal flow as an attractor to a previous signal state. The reverberated sound lags behind when there are changes in the frequency spectrum. However, the gain of the reverb signals is reduced by around 12 dB and therefore the influence is low compared to the original signal. A side effect of the reverb is, that it constantly colours the sound. This can be aesthetically interesting, yet it also stabilizes the system somewhat. A side-chain compressor was used to attenuate the reverberated signal when the original signal is above a set threshold. Therefore, the reverberated signal is quiet when the signal is in a sustained loop.

4.3.5 Limiting and gain

A limiter is utilized to ensure that the audio signal level stays in controlled and strictly confined boundaries. Because of the gain reduction caused by the above-mentioned signal processing methods, the overall signal level can be reduced greatly. The gain inside the limiter is adjusted to boost the signal while leaving a few decibels of headroom. This is done uniformly across all channels to ensure an even level in the digital domain. The release times are generous, to avoid unwanted distortion. The ceiling is set to -1 dB and every signal potentially exceeding that will be limited thusly.

4.4 Operation

In this section, the operation of the sound installation will be described. Procedures, preparations, and modes of operation will be covered before the tuning of the system is described in detail.

4.4.1 Preparation

Before the installation can be properly operated, a few preparations need to be made. The module stacks need to be positioned in the space first. It has been established in previous chapters how and why the position of the installation in the space influences the behaviour of the sound. Ideally, the stacks should be hung individually and evenly spaced apart, though this depends on the available ceiling anchor points and the space. The wooden planks allow for halving the anchor points needed if there are not enough available.

With the module stacks in place, the rest of the equipment can be positioned. The interface and amplifiers are placed close together, outside the perimeter of the installation. The cables should be connected accordingly to the pipe modules, interface, amplifiers and computer before the equipment is powered.

Before the tuning can begin, the amplifiers' amplification level is adjusted. The level of amplification is chosen to allow for interesting interference and interactions between the feedback loops. Furthermore, the loudness should be within listening levels, to ensure that the experience is not uncomfortable when in the proximity of the installation. By limiting the amplification at the amplifier stage, it can be assured that the set maximum loudness will not be surpassed, avoiding unintended loud feedback from a potential error in the DAW. As the loudspeakers utilized in this installation are of different built, an even loudness across all speakers is also aimed for when setting the levels of amplification. The level of the digital signal is adjusted in the tuning process described in the section 4.4.3 below.

4.4.2 Modes of operation

Two modes of operations are explored and described here. The difference between the modes lies in the routing of the microphones and speakers. I propose and use the terms *module-internal* or *cross-module* to refer to the modes. When a microphone of a pipe module is routed to the speaker of the same pipe module, the routing is *module-internal.* If the microphone of one module is routed to a speaker of another module, it is in a *cross-modul*e routing (see figure 4.10). Cross-module routing would theoretically allow for the routing of multiple signals to the same output, though this is not within the focus of interest in this work. Furthermore, a mix between the two types is not explored. Either all pipe modules are routed module-internal or all are cross-module routed.

The modes change the system behaviour significantly. This is due to the positional relationship between the speakers and the microphones. In a module-internal mode, the positional relation between the microphone and speaker is fixed. The distance is set, and so is the path in between that the sound travels. In a cross-module routing, this relation changes with the movement of the corresponding pipe modules. The path of the sound travelling from the speaker to the microphone differs vastly. It instead travels from the speaker of another module, through the pipe of that module, past another microphone and through the space between the module and the microphone. Furthermore, a microphone accentuates the sound produced by the speaker of its pipe module, unaffected by the mode.

In a cross-module routing, the feedback loop created by the speaker-microphone pair is always interfering with another feedback loop, as the sound has to pass another microphone. The modules' feedback loops are therefore inherently more intertwined. Therefore, the balance between the signals is also very different between the modes. The cross-module routing mode adds another layer of complexity. In that mode, the installation is also more sensitive to the room sound.

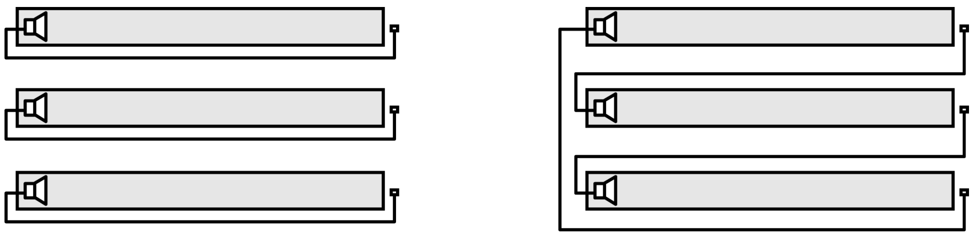

Figure 4.10: Illustration of module-internal mode (left) and cross-module mode (right).

4.4.3 Tuning procedure

To operate the installation as intended, a specific tuning procedure was developed. The procedure aims to achieve as much equilibrium in the system as possible, leading to a maximum instability and rich, complex behaviour. The system needs to be tuned for the module arrangement, for various possible system states and with the surrounding space taken into account. Because of the influence of many small components, every setup and every location provides slightly different conditions. The tuning process can start when the whole installation is hung into place and resting in an idle position.

Tuning a microphone-speaker pair

Depending on the mode of operation, a microphone-speaker pair might share the same pipe module or not. Therefore, to include both operation mode scenarios, I will refer here to tuning microphone-speaker pairs instead of tuning pipe modules. The microphone-speaker pairs are first tuned in isolation before all pairs are tuned together.

An important preparation is to adjust the microphone positions. While this can be done throughout the tuning, handling this beforehand can result in a cleaner process, as it does have a great effect on the resonance frequencies. The tuning can require thorough readjustments after the microphones have been adjusted. The position and its effects on the system are probably best explored with all microphone-speaker pairs active. With enough experience, the desired position can be estimated without the system being active. I utilized a position quite far away from the pipe opening to ensure that the surrounding sounds are comparatively not too quiet. The fine adjustments also depend on the mode of operation.

Before the resonance frequency equalisation starts, the respective channel is prepared in the DAW with the above-mentioned processing. A compressor with preliminary settings is enabled, a reverb is applied as well as a limiter with a gain parameter, to set a maximum level and make-up gain. To tune a single pair, the corresponding microphone and speaker are enabled in the channel in the DAW. The level is increased until the system falls into a sustained feedback loop. Utilizing a parametric equaliser, the process of equalising the resonance frequencies can begin. The process is as follows:

First, the most dominant, sustained frequency in the signal is determined. If there is no filter in close proximity of that frequency yet, a new peaking filter with a narrow bandwidth is applied at that frequency. While paying attention to the frequency spectrum, the gain of that filter is turned down slowly and continuously. It is reduced until a new resonance frequency emerges, taking precedence over the previous one. If no frequency arises, the level of the signal needs to be increased until the feedback is sustained again. Then, the process starts anew (see figure 4.11). This process is repeated as often as needed or desired. The adjustments can become quite intricate (see figure 4.12). Sometimes, it might be useful to use a shelf filter or a broader bandwidth to reduce a greater range of frequencies simultaneously. Especially high frequencies tend to clutter (see figure 4.2).

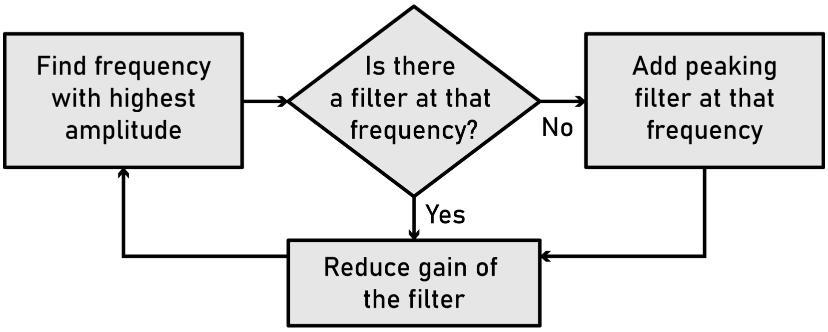

Figure 4.11: Resonance equalisation process flowchart.

The dynamic equaliser is adjusted in almost the same procedure. However, instead of turning down the gain of the filter, the threshold is lowered until another frequency is more present. Potentially, no additional frequencies arise. In that case, all thresholds are reduced simultaneously until one or several new frequencies appear.

Tuning all microphone-speaker pairs

The filtering process becomes more complex once more instances of the microphone-speaker pairs are added. All feedback loops are intended to interact and interfere with one another. The number of combinatorial states of the system increases and combined resonance frequencies can occur. These frequencies also need to be taken into account in the tuning process. Therefore, an additional equalisation process is necessary to even out the resonances in the relevant signals. The process is as follows:

First, the digital output levels of all channels are adjusted in the DAW. Each channel is briefly monitored in isolation to achieve an even loudness across all microphone-speaker pairs. This assures that the influence of the feedback loops on each other is as evenly balanced as possible. A new parametric and dynamic EQ is applied on each channel. The subsequent equalisation process is in essence the same as described above. However, the shared resonance frequencies will occur strong in multiple signals simultaneously. Typically, one channel has a greater influence on the resonance. Therefore, determining this channel is helpful as filtering is more effective on that channel. This can be done by comparing the level of the resonance frequency across the channels. Alternatively, by testing and comparing to what extent a gain reduction through the filter affects the resonance frequency. The equalisation procedure steps are the same as described above. Additionally, regular checks and readjustment of the loudness of the speakers are done, to attain a constant evenness across all pairs as best as possible.

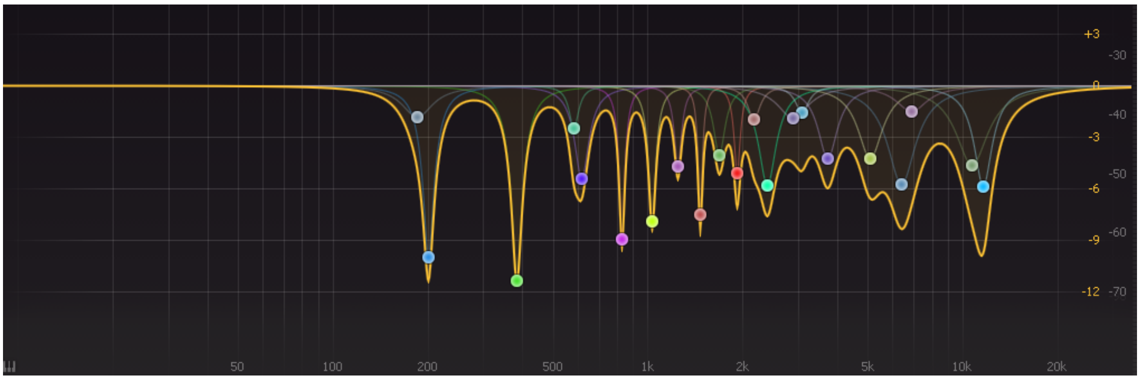

Figure 4.12: EQ setting after a resonance equalisation process.

Tuning for different states

Until now, the tuning was described with the installation in an idle state. The installation can be in various positional states, creating a range of resonance frequencies for each state. Therefore, additional tuning can be done by putting the installation into most common states and repeating the tuning processes described above.

4.4.4 Interaction

When the tuning is done, the installation is ready to operate. For the audience, there are a number of intended possibilities to interact with the installation. The weights hanging from the module stack serve as an interactive element. By pushing or pulling them, the whole stack moves in a pendulous motion. The audience can rotate the module stacks with either the help of the strings that connect the modules or by briefly pushing an end of a pipe. With the hanging mount of the pipe modules, the angles can be adjusted, as described in 4.2.6.

Chapter 5

Discussion

In this project, I have explored complex musical behaviour of audio feedback through the conceptualising and making of a sound art installation. I approached this through an iterative prototyping method. The by-products of this process are various exploratory prototypes. This method was time-intensive, as the exploratory prototypes often turned out to be unfruitful or insufficient. The systems were usually not sophisticated enough, or the feedback did not behave interestingly enough. However, the prototypes hinted possible directions and uncovered various further challenges. The project and my understanding of feedback greatly benefited from this. The experiences gained from the work and time spent on the prototypes provided a crucial framework for the artistic concept and the whole project to mature. The details, clarity and specifics of the project crystallized alongside the prototypes. An alternative approach to this project could be an analytically focused process. A complex system could be achieved through maximizing complexity indicators based on measurements, for example. While this is interesting in retrospection and definitely holds value, this approach would have influenced the exploration and experimentation integral to this project.

The presented theory aided in the process of evaluating and refining the system. The models on pipe resonances provided a basis for deciding on pipes as core attractors early on in the project. The pipes are the core attractors of the system, adding a characteristic musical layer. Theory on sound propagation found use in various aspects, such as balancing sound levels through the positioning of the microphones and the pipe modules. Towards later prototypes, a more even distribution of resonance frequencies was approached by calculating the pipes' harmonics and adjusting the pipe lengths based on the results. The theory on frequency dependant directivity of speakers sparked the idea of flipping the speakers to affect the frequency spectrum at the microphone position. Applying this, results in the attenuation of higher frequencies.

The embedding of physical space into the feedback loop was essential. Therefore, several hanging systems were prototyped. The hanging system became an important level of interfacing with the feedback loops through physical interaction. Of all the methods explored, the presented method turned out to be the most useful for the task at hand. It provides a foundation for interesting positional and harmonic combinations. The possibility to angle and rotate each pipe module is purposeful in the context of the system and enables the experimentation with many different states. The modular aspect of it allows for the easy expansion of the total number of modules and stacks, increasing the complexity of the system. However, sometimes the interactions do not affect the feedback system sufficiently and can therefore feel unresponsive. Consequently, the readability and simplicity of interaction possibilities for the audience suffers a little with this method. Nevertheless, the flexibility of the system is of great use for the tuning, while simple interactions, such as pushing the weights, are still approachable for the audience.

A goal was to achieve a rich and varying sound with the sound installation. This turned out to be a major challenge of this project and became an act of balancing instability and controllability. The behaviour exhibited by the system was set out to be complex, yet coherent and graspable. Therefore, a lot of time was dedicated to experimenting, examining and studying how basic signal processing affects feedback. While experiments with more complex signal processing proved interesting and intriguing, I focused on the fundamental methods, to attain a thorough understanding and feeling for their effects and behaviours. Drawing from this, a tuning process was developed. This process plays an important role in inducing complexity into the system while avoiding states of continuous feedback saturation. The procedure aims to attain as much equilibrium as possible. This leads to a maximum instability, allowing small physical changes to influence the sound. The procedure approaches this state through careful and thorough equalisation of resonance frequencies of the microphone-speaker pairs. A variety of examined signal processing methods further assist in this goal and provide control over different aspects. Finding a balance between tuning individual microphone-speaker pairs and all pairs, and when to proceed from one to another, can be challenging. Sometimes it is not clear which signal should be equalised and when to move on to dynamic equalisation.

When the installation is operating and tuned sufficiently enough, it exhibits the behaviour aimed for. The sound is varying and interactions are meaningful as they become audible in the sound. The amount of influence of physical changes on the system can vary greatly from one setup and tuning scenario to the next. Sometimes, the result is not as pronounced and nuanced as aimed for. This might be due to the challenge of taking a wide range of resonance frequencies and scenarios into account with the proposed tuning process. It can also be due to the physical construction of the system. Often, the system can fall back into a fairly steady state, that was not taken into account for in the tuning process. This might indicate that the tuning process is not effective enough for the task, or that it might have not been executed thoroughly enough. Nevertheless, the method does ensure that the system does not idle at one sustained saturated feedback frequency.

Chapter 6

Conclusion and future work

In this thesis, I have described the making of a sound art installation. With it, I explored possibilities of complex musical behaviour with audio feedback. The concept, components, construction and operation of the installation are described in detail and relevant theories are used in support. The system presented generates multiple interacting audio feedback loops through acoustical pipe modules. A module is equipped with microphones and speakers. A hanging system was described that arranges the modules spatially and allows for physical interaction. The interactions and interference between several feedback loops were a focus in this work. Another important aspect is the integration of the surrounding physical space into the feedback system. Methods were proposed and utilized to balance and tune the system while aiming to keep the system from falling into saturating feedback.

This work raises considerations for future work. The approach to complex musical behaviour through audio feedback given here is one among countless. Creating high complexity in a feedback system is a great challenge. Frequency specific gain controls (filtering) and dynamic equalisation appear to be of great relevance in maximising the dynamic complexity. Further audio signal processing methods could be studied in this context. An automation of the proposed filtering process could be considered, as well as a reactive filter process that adapts to the situation in near real-time. One that dynamically counteracts stabilisations of the system and therefore increases the complexity for a given situation. Kiefer et al. (2020) propose a method for complexity-controlled gain dynamics for feedback systems.

The filtering could be approached through analytical methods, similar to measuring frequency responses of speaker systems. Test signals such as sine-sweeps could be utilized to measure the frequency response of the system. Based on these measurements, filter settings could be determined to even out the spectrum. Furthermore, additional hanging systems and movement possibilities could be explored to include the physical space in greater amounts and to allow for different interactive explorations. A variable position of module stacks could enable more arrangement combinations and reflection patterns.

More complex signal processing methods could also be explored in detail, as experiments with frequency shift and grain delay yielded intriguing sonic results. These experiments hinted that retaining instability might be very challenging, though further studying of the effects may uncover interesting behaviours. The presented system could also be refined with a focus on performance, utilizing it as a musical instrument instead of an installation. Possible interactions and interfaces could be explored further with playability in mind.

The complexity, the subtleties and the depths of audio feedback are great. This work aimed to convey some of it through artistic practice, gaining a better understanding of it along the way. As the world of feedback is vast and intricate, this thesis can only provide a snapshot into that world. However, it does not aim to, nor possibly can, explain its entirety. The beautiful intricacy and interconnectedness of everything in this world are sources of curiosity and inspiration. The installation serves as another window into this wonderful landscape and finds itself somewhere between the many other artistic explorations in the field.

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