

Martina and Nicolas Buzzi

Variations of a Paper Organ

Instrument Building Guide

Instrument Building Guide

To build a paper organ, you will need:

- At least one tube (three or four are suggested)
- Mounting system for each tube (eg. a lighting stand)
- At least one microphone (one for each tube)
- At least one loudspeaker (usually one for each tube) + amplifier (as many amplifiers as loudspeakers)
- An audio interface (with as many in- and outputs as there are loudspeakers and amplifiers)
- A laptop that can run Max 8 media software
- Max 8 software

Introduction

Many kinds of instruments rely on a combination of harmonic vibrations within different kinds of tubes, and the control of air and vibration is a cornerstone of instrument making. Woodwind and brass instruments, organs, mallet, and other percussion instruments are all built with tubular resonators. A great variety of sounds can be achieved by altering the dimension, materials, composition, and interactions with these different types of tube. For example, the Whirly Tube, also known as the corruga tube, sound tube, or Bloogle Resonator, is a musical instrument that consists of a corrugated plastic tube that one "plays" by swinging in a circle. The sound is created by air moving through the tube.

The paper organ moves air within tubes using electronic feedback. A system of microphone, amplifier, and loudspeaker is used to make a mixture of sound at a desired volume and for a given length of time. This system also allows a certain variation of partial tone tuning. A computer program is used to control the parameters so that by using multiband filters and controlling the phase and amplitude of the individual partials, it is possible to vary the sound responses of this relatively simple and easy-to-manufacture system. The electronics are used only to excite the acoustic resonator and control the sound, not to generate sound on their own, making the paper organ an instrument known as a computer-controlled, electromechanical sound generator.



What is a Paper Organ?

It is the construction of a system that can make sound in a variety of ways. The system can be built with an unlimited number of tubes, giving each paper organ an open range and number of sound sources. If several pipes are available, they can be distributed as desired across one or more rooms. The only thing to keep in mind is that there must be a sound transducer at each end of each tube, along with one microphone and one loudspeaker. You can also experiment with adding or subtracting the number of tubes and making your tubes out of different shapes and materials.

Note

We suggest you use this guide throughout the planning, selection of material, composition, and installation of making your own Paper Organ. Below are explanations for each acoustic component in the system. An arrangement with only one pipe is assumed, allowing for open scalability if more are desired.



Feedback

The audible sound element of the paper organ is created by the feedback between a loudspeaker and a microphone. When a microphone converts sound into a signal, and this signal is, in turn, converted into sound by a loudspeaker with a sound connection to the microphone, the acoustic coupling can be made to self-excite. To do this, all that's required is to adjust the amplification chain so that it's neutral at the location of the microphone, or rather that the volume increases with each iteration of the circuit. Once this occurs the increase in volume is exponential, which is why there is often a rapid, extreme increase in level; a process that, uncontrolled, is only slowed down by the limits of the material involved.

The qualitative manifestation of the sounds produced by the feedback depends on physical quantities and on the peculiarities of the components used. Something called a sinusoid is produced, which is then altered by distortions in the electronic components of the instrument. The time interval between the conversion processes of microphone and loudspeaker (in the case of digital components), the propagation time for the sound between microphone and loudspeaker, as well as their relative phase position, determines the number of possible frequencies. Other elements also affect this, such as coincidences, ambient sound, external occurrences and/or non-linear properties of the involved material. In most cases, one frequency prevails, and it takes a change in these parameters for a variance to occur. The paper organ is designed to influence these parameters. Feedback is used in both analog and digital sound processing and generation. Some filters, oscillators, and effects (such as echo) rely on it, as do some techniques of sound synthesis (such as no-input mixing). However, before being projected by a loudspeaker, these signal-based recursive systems have nothing to do with acoustics. The paper organ is different—the feedback relies on an air column (with all its peculiarities) for influencing the sonic results.

Loudspeaker/chassis

A loudspeaker chassis or driver takes an electronic signal and turns it into sound. It usually consists of a frame, a motor, a membrane, and a suspension system.

A conventional loudspeaker chassis (full-range or bass chassis) or a complete full-range loudspeaker is used as the physical driver for the paper organ. Since the tube functions as a resonator and thus has an amplifying effect, considerable results can be achieved even with small drivers. As a rule of thumb for your driver selection, the resonant frequency F_s of the driver, or the lower end of the specified frequency response of the loudspeaker (-3dB), should not be more than an octave higher than that of the lowest frequency you expect to generate. This frequency is defined by the length of the tube, as described in the next point. Linearity in the frequency response and its upper end play a subordinate role. The aim is that the frequencies reproduced are clearly defined and not distributed over the entire spectrum, are balanced in their amplitude by software, and end towards the top in a range that can be easily represented by any conventional chassis.

For pure sounds, the chassis should be mounted in such a way that no clattering noises are caused by the vibrations. The moving parts are particularly sensitive and should not touch anything, even when vibrating. In addition, it should be noted that chassis tend to radiate sound and should therefore be mounted at a certain distance from reflective surfaces. This avoids cancellations (caused by the phase-inverted sound from the rear of the chassis mixing with the useful signal). For ideal sound generation, place the tube about 50cm away from walls, ceilings, or floors (being prepared to experiment with the positioning of the tube if necessary).



Tube

A tube is a hollow body whose diameter is smaller than its length. In the case of the paper organ, this ratio should be roughly in the range of 1:5 to 1:30, corresponding to the width measurement of organ pipes. Tubes can be conical, cylindrical, straight, bent or coiled, round, square, open, or closed. Paper organ tubes are generally cylindrical, open at both ends, and otherwise airtight. Due to its physical properties this kind of tube forms a resonator, allowing pressure waves to propagate along the path of least resistance due to its shape forming a specific atmosphere. If sound enters the tube because of vibration by a reed or in this case a loudspeaker, the elongated shape determines the direction of travel, and it is this longitudinal direction that can propagate unhindered sound over a longer distance. When the sound wave reaches the end of the tube, a large part of it is reflected by the outer atmosphere and thus returns to the tube – a process that can be repeated at the other end. The non-reflected part propagates outside the tube, allowing us to hear the sound inside the tube. If the excitation remains, the oscillation behavior within the tube stabilizes after a few repetitions, with frequencies that roughly correspond in wavelength to the tube length. This results in a resonator or a filter with a harmonic structure. On the tonal level, these frequencies are called partials, on the spatial level, they are called modes.

The lowest partial produced in a tube is the fundamental. The length of the air column in an open tube corresponds to half its wavelength. Due to the elasticity of the atmosphere surrounding the tube, the air column and vibrating air within the system tend to protrude a few centimeters beyond the tube. Therefore, the physical tube is shorter than the wavelength and must be compensated for by what is known as an 'orifice correction'. The size difference between the tube length and air column depends on the diameter of the tube. (A possible procedure for determining the tube length of the paper organ is explained under TOF.)

The paper organ tubes are used exclusively to generate and control a column of air. Since the tube itself does not have a very significant sound and the timbre is mainly defined by the electronic and electromechanical components, its material plays only a minor role. It is advisable to look into the tonal nuances of different materials only once you've made your first instrument. Materials that are readily available and easy to work with are advised. For example, metal or plastic tubes are ideal because they offer so many possibilities for assembly. However, it is a good idea to make at least parts of the tube from a paper roll so that you create a telescopic system in which at least part of the tube length is variable. Tubes made entirely of dense, rigid paper rolls also work well. A key advantage of paper is the ability to easily determine the length and diameter of your tube. It is also relatively inexpensive, lightweight and easy to manufacture and store. With a suitable mounting fixture, paper tubes can be suspended above or adjacent to a chassis (as close as possible but without a closing connection) or be suspended or laid down independently of the chassis. It is also possible to experiment with a fixed, load-bearing connection between the two elements.



Microphone

Microphones are transducers which detect sound signals and produce an electrical image of the sound, i.e., they produce a voltage or a current which is proportional to the sound signal. A transducer is a device that converts energy from one form to another. Usually, a transducer converts a signal in one form of energy to a signal in another. As the loudspeaker, the microphone consists of a frame, a membrane, and a suspension. But instead of a motor, it has a similar device that produces electricity when the membrane is moved by sound.

The microphone has the task of converting the sound filtered by the modes in the tube into an electrical signal so that it can be fed back to the chassis/speaker. As with the chassis/speaker, special requirements so that this can occur, apply here. Neither signal-to-noise ratio, directional characteristic nor linearity are vital in selecting which type you use. Instead, the most important parameters are frequency response, maximum sound pressure level and, if necessary, weight and mounting options. Again, as with the chassis, the upper cutoff frequency is negligible, but the lowest modes should be easily represented. In addition, the microphones should be able to handle the volume generated in the tube. The cheapest available measurement microphones are a reasonable choice, with the most important thing being the mounting location of the microphone. All partials must be represented and all modes able to be recorded.

Modes

Modes are colloquially also called standing sound waves—a term that can be used to illustrate what is happening inside the tube/s of a paper organ. The standing sound waves exist in relationship to the length of the tube, and mean that the phase positions of the modes are coupled to certain physical positions within the tube. (eg. the deflection of the modes reaches its highest or lowest values at the same place in the tube. The nodes and anti-nodes of the individual partials are distributed over the entire tube - the only deflection that is consistently the same over the entire spectrum is located at the ends of the air column.) For the system to function properly it is critical that the microphone be positioned at a location where none of the modes are at a null point. To ensure this, it is a good idea to position the microphone at a point where all modes have an anti-node. While one end of the tube is occupied by the chassis, the other is free for mounting the microphone. Even though the air column is a little longer than the tube itself, positioning the microphone capsule exactly at the end of the tube still allows for the successful creation of feedback.

(Please note that in addition to its function as a component of the instrument, the microphone signals can also be tapped for possible amplification by external loudspeakers or for recording the instrument)

Amplifier

Amplifiers are needed to boost a current source because the current originally carrying the signal is too weak to drive the loudspeaker's motor and render audible sound. Amplifiers use transistors to allow a weak signal to depict the form of a much stronger signal.

The amplifier should correspond to the chassis. Since high, unpleasant, or even destructive levels can be reached, it is advisable to remain sensitive to volume control if possible. If necessary, the use of a limiter is suggested so that the task of limiting isn't solely left to the chassis.

Microphone preamplifier, ADC/interface

When converting the microphone signal into a digital signal and transmitting it to a computer, the two things that need to be considered are sensible gain control and a fast, reliable connection. In a recursive computerized system that works between transducers, latency results in a change of phase, which has a direct influence on the sound generation. The existence of this change is not disastrous, but a correction must be made to the phases of all partials. The delay between the ADC and the DAC is also relevant in that it creates the smallest possible delay. (Larger delays may also cause difficulties in determining the partials.) It is better not to have to change the buffer settings in hardware and software after the system has been set up, as this could cause the pipes to be reset. The selected values should therefore be as small as possible, but as large as necessary for an error-free system function.

MAX-Patch

The MAX patch is deliberately kept simple and user friendly. The types of control are as varied as the possibilities in the structure of the instrument, so it is assumed that these will be implemented individually according to specific needs. The structuring of the patch and suggestions for building a system should be obvious from the instructions. The features necessary for the functioning of the paper organ in relation to this are explained under the following points.

MAX Input

The MAX patch is deliberately kept simple and user friendly. The types of control are as varied as the possibilities in the structure of the instrument, so it is assumed that these will be implemented individually according to specific needs. The structuring of the patch and suggestions for building a system should be obvious from the instructions. The features necessary for the functioning of the paper organ in relation to this are explained under the following points.

TOF (Time of Flight)

The first component serves two purposes. First, it can be used to determine the roundtrip latency of the interface, drivers, and software for correction purposes. Secondly, it is used to determine the distance between the loudspeaker and the microphone (i.e. the approximate tube length). This ensures that a preset of the tube's pitch can be made and the determination of the air column's length is simplified. When the module is activated, an impulse is generated, which is simultaneously recorded and output—to be recorded again after output, this time via the microphone input. This process can be used to read out the delay between program output and input. When setting up a system for the first time, the roundtrip latency must first be determined. The simplest way is to position the microphone closest to the loudspeaker before the pulse is triggered. It is advisable to make sure beforehand that the microphone signal is not added to the loudspeaker in any way. You should trigger the impulse in a space where there is no ambient noise and confirm the result by clicking on "Set Latency". (It would be good if the latency was as low as possible.) Once the latency is determined, the setup of the tube can be started. After the chassis, tube, and microphone have been set up according to the instructions, the measurement process can be performed again. This time, however, the aim is to determine the fundamental frequency of the tube. To be able to do this reliably, the tube diameter must first be entered into the TOF module. This enables the orifice correction (i.e. the mathematical compensation of tube length and air column). After that the process should be started after the system has been set at "Set Fundamental".

Multiband filter, volume control

After the system has been calibrated using TOF, sound control and stability are obtained by passing the signal through 16 bandpass filters with separate controls for Q and Gain, whose frequencies correspond to the 16 lowest modes of the tube. This component separates the microphone signal into 16 filtered signals (filter refers to the component in its entirety), thus doubling the natural filtering effect of the tube, which in turn contributes to the overall stability and controllability of the instrument. Added to that, with the separation of the signal into 16 bands, the filter can also form the basis for volume control of the first 16 partials.

The reduction to the first 16 partials (or a range of four octaves) is based on the fact that the intervals between them become so small that reliable control soon becomes impossible. Still, the number of bands can be extended by adapting the control software if this should be required. It is, however, generally advisable to add more tubes, (eg. in octave steps) to the same speaker to expand the range and thus create a system with multiple stops.

The filter has two variable parameters per band: preamplification and Q. Neither of these is intended for direct sound-shaping; but are stored as presets for the purpose of balancing partials (and matching multiple pipes to each other, if necessary). While the Q parameter can be left at its default value in most situations, it is important to set up the pre-gain according to your specific needs.

Neither chassis, tube nor microphone needs to have a linear frequency response. Since the frequencies produced by the paper organ are limited, unevenness can be easily corrected. The level adjustments of the separate bands are performed after or during the actual tuning of the phase adjustment (explained below). At this point, only the preamplifier gain needs to be set. Generally, this can be leveled in a high range since the highest possible gain is predictable. It is also recommended that you make sure the overall level of the microphone preamplifier can be adjusted in both directions, (i.e. amplified or attenuated.) As a reminder, noise levels are of little importance. After calibration (by means of the level meter integrated within the filter) it should be possible to read off approximately the same value for all partials. Alternatively, a level meter at the line-out of the interface could be used for the same calibration.

Bias

Bias allows a continuously adjustable weighting of the higher or lower partials in real-time. The component is deliberately kept simple and is intended as a means of directly controlling the timbre of mixed sounds. It allows making sudden or flowing changes of the timbre with only one value, without having to fundamentally reshape the spectral components or affect the integrity of a sound. If necessary, the component can be extended (eg. with the possibility of creating formants). Remember that when setting up and calibrating the system, bias should be in the center position, thus remaining neutral.

Phase correction

Due to the distance between loudspeaker and microphone, the sound propagation time in the tube, and the latency of the system, it is expected that there will be a phase shift of the signal between loudspeaker and microphone that changes the phase positions of the partials at the nodes. This means that certain partials can't be established, meaning the phase position must be adjusted by further delaying the signal. Since all partials have the same delay due to propagation time and latency but their phases change at a different frequency, this correction must be determined separately for each individual partial. This is done by an array of sample-exact delays, which are applied in the signal chain after filter and bias.

If the phase is not correct it might mean that no sound is produced at low volumes, and/or the sound produced does not correspond to the frequency of the respective partial. This means that with the change of the delay the system must be tuned. Of course, it is also possible to detune the partials individually within a certain range, which makes it possible to shape the overall sound of a tube when several partials interact.

Reference tone generator

To facilitate the tuning process, the MAX patch contains a tuning aid in the form of a 16-fold additive signal generator. This allows each partial, controlled separately, to be added to the feedback loop so that the system can be tuned by ear. Although the reference tone generator is intended only for the tuning process to achieve a controlled harmonic spectrum, it is unlikely that anyone will be able or willing to make sure that it is used exclusively for this purpose.

Automatic amplitude control

When the system is made to self-excite as described in the feedback section above, the signal must be electronically limited so that this does not have to happen by mechanical limitations. Furthermore, controlling the amplitude allows very quiet tones and sounds to be produced.

Amplitude control is very simple. The absolute value of the signal strength of each partial tone is given a low-pass filter for smoothing, before the signal is multiplied by the smoothed, inverted result. The result is that a signal with a low level is amplified, but a strong signal is attenuated; meaning that when the amplification is controlled manually, a stable tone of any loudness can be produced.

The automatic amplitude control can also be used creatively. (eg. If the speed of smoothing is varied, a certain roughness can be added to the sound.) It is also possible to create pulsating or slowly rising and falling textures, especially if the rate of level reduction does not match that of level increase.

Partial tone volume control

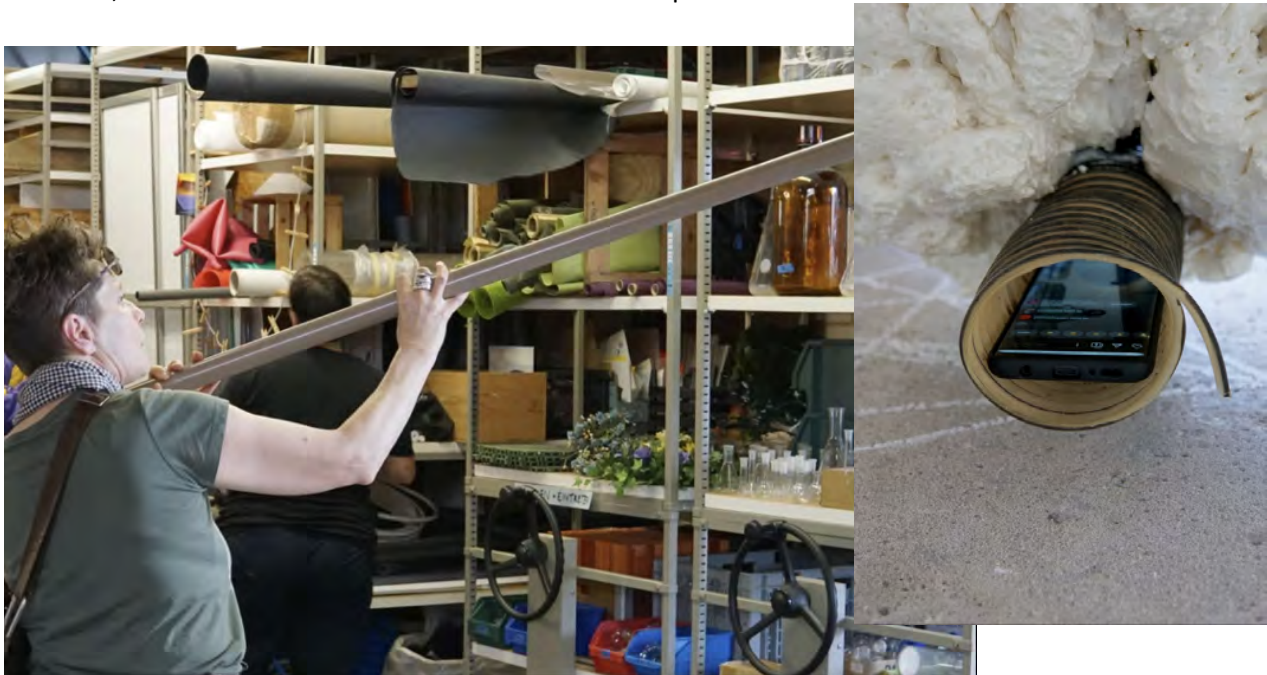
The combination of filter, phase correction, and automatic amplitude control makes it possible for the feedback system to generate several partials at the same time, (eg.. one sound that allows very specific control of its harmonic spectrum.) This component is simply 16 amplifiers with controls that can determine the volume of the partials. Although they can be operated manually, it is recommended to add an automation or playing method appropriate to the intended use. With this device, single tones, harmonic clusters, or sequences of any combination can be designed and played.

Overall volume control

The last control element is a slider for simultaneous control of all partials. The dual design of the volume control serves to expand the playing possibilities. (eg. several partials can be easily controlled in a constant ratio, or presets can be made in a hidden manner.) Bearing in mind, of course, that control can be automated or controlled with any control interface.

MAX Output; Interface/DAC

For the sake of completeness, the output from the program is listed here. This should be connected to the interface, and the interface in turn connected to the amplifier.



Learning kit by Martina and Nicolas Buzzi

Produced during Empowering Tools residency
at Nida Art Colony of Vilnius Academy of Art
Commission: Association "RĪGAS STARPTAUTISKĀ
BIENNĀLE"

The Erasmus+ project "RIBOCA Repository of
Knowledge" is co-financed by the European Union. The
European Commission is not responsible for the content
of this publication.



Co-funded by
European Union