The experimental legacy

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an introduction

Landed in the second decennium of the 21st century we can start to look back on the 20th century as part of history. As usual long known problems are encountered. A fundamental question will always be what to preserve. Selections have to be made and criteria justified. Even if we don't do anything, leaving things as they are, entropy will guarantee a form of selection. Think about electronic music made on analogue tape. We know that preserving those tapes poses a manifold of problems. Not only the carriers deteriorate, but also the machines to play them and to make copies to other media are more and more difficult to find and to maintain in full operating condition. The technology of the pre-digital era is no longer commonly available to electronic engineers. Repairman are by now all if not dead, at least retired or no longer capable to properly do maintenance and repair. The components once used are since long no longer made nor findable on the market. Moreover, even if important tapes from the archives were transferred to CD's, we now observe that these CD's are rapidly becoming unplayable. Entropy cannot be circumvented it seems, at the most we can slow it down.
Looking at the music of the 20th century, the problems we encounter can be of very different nature:
1. Music that was composed and conceived for technological tools that at the time of the conception of their use just couldn't fill the promises.
Some examples:
From the first half of the 20th century:

- Georges Antheil, 'Ballet Mecanique' (player piano synchronisation, propellers, sirens, electric bells...)
- Igor Strawinsky ' Les Noces' (player piano synchronisation)
- Kurt Weill 'Dreigrosschenoper' (for barrelorgan as first proposed by Bertold Brecht)
- Leopold Stochowsky: theremin instruments in the orchestra

From the second half of the 20th century

- Karlheinz Stockhausen 'Solo mit Rueckkopplung'
- Alvin Lucier, pieces involving brainwaves
- Dick Raaijmakers 'Elektries Strijkkwartet'
- Conlon Nancarrow 'Player piano studies', percussion extensions

In these cases musicologists are often tempted to consider the work eventually revised later by the composer, as the definitive work. Here we can raise the question whether it wouldn't do more justice to the composer to get back to his original concepts and try to realize them with what technology permits us to do today. In this line of thinking, at Logos we did a version of Noces using only automated instruments, we realized Ballet Mecanique with real air plane propellers, modern player pianos, automated sirens and industrial bells and in 2016 we finished and presented a version of Kurt Weills Beggars Opera using our complete robot orchestra, thus realising the original concept of Bertolt Brecht. Other people have realized 'Solo mit Rueckkopplung' using digital technology, thus not plagued by the inherent instability of long analog tape loops on stage. As to Dick Raaijmakers 'elektries strijkkwartet', we made a working version whilst the composer was still alive. A report can be found here (as yet only in dutch). Some biographical data with regard to Raaijmakers can be found here.
2. Music making use of technology that is no longer available or in need of maintenance and repair.

The crucial question here is in how far the technology used is an essential part of the rhetoric of the performance. This is not always the case. For instance the thousands of compositions written for tape and musicians, do not require the reel to reel tape recorder as an essential component of performance and for such pieces, playing the on-tape sound track from just about any modern medium (CD or simply from the computers memory) does not change anything to the performance. However, there are many examples of pieces where the tape recorder becomes an instrument in itself, more than merely a reproducing device.

Example:
Alvin Lucier's pieces using slow sweep sine wave oscillators
Brian Ferneyhough and Karlheinz Stockhausen's pieces involving ring modulators
Pieces involving live manipulation of audio tape and tape recorders: Dick Raaymakers, Michel Waisvisz, Brian Ferneyhough, Steven Montague, Gordon Mumma, Pauline Oliveros, John Cage...
Pieces involving manual handling of electronic circuitry and components: John Cage, David Behrman, David Tudor, Gordon Mumma, Dick Raaijmakers, Michel Waisvisz, Takehisa Kosugi, Larry Wendt, Allan Strange, Ron Kuivila, Nam Yun Paik
In these cases it appears to us to be pointless to replace the technology used with modern alternatives. A laptop just cannot replace a vacuum tube oscillator for it would ruin the act of performing the piece on stage. It undermines the rhetoric and thus the intrinsic value of such compositions.
Obviously there are quite many cases were the technology is not as such at the focus of the work. The many pieces composed in live electronics using samplers for instance, can easily be performed using nowadays technology. However one has to be careful, as many composers have used equipment at or just over the border of their capabilities in which case the use of the original equipment is indicated.
3. Compositions that involve the construction of technological devices.
The devices can range from simple contact microphones and their pre-amps (David Behrman, Hugh Davies, David Tudor, Gordon Mumma, Richard Lerman, Mauricio Kagel...) , up to the most diverse sensing devices such as radar sensors (in the work of Jerry Hunt), light sensors (John Cage, Toshi Ichianagi), sonar devices (Alvin Lucier, Dieter Truestedt). Also electronic sound modifying devices, real circuits, often need to be build (Robert Ashley, Pauline Oliveros, Hans Otte). Ringmodulators and effects are examples. The performance of my own composition 'Logos 3:5' (1969) for instance, requires the performers to build an automated conducting device.
In all such cases, the realisation of the required technology should be considered to be part of the performance practice. More often than not, one has to go through the complete description by the author, try to fully understand (and reverse engineer) it and after that finding a working realisation with the same artistic result. I still remember some Cage performances, where Takehisa Kosugi was seated behind his performance table, handling a hot soldering iron.
The pursuit of new forms of and tools for musical expression in the second half of last century, has lead a substantial number of composers and performers to become acquainted with technology: at first mechanics and electricity, then soon after, electronics and since the late sixties, computer programming and software development.
In the post-WW2 period, we saw the upcoming of studios for electronic music all over Western Europe. Those were in general connected to either broadcasting stations or universities. The Koeln studio was part of the WDR, the Milano studio of the RAI, the Utrecht studio at the Conservatory, the Ghent studio at BRT-radio... At such institutions, it was common practice to have engineers and technicians available to work with and for the composers. In the America's this was more rarely the case. So composers that were deeply motivated to experiment with new technologies, got a stronger stimulus to study the technology for themselves. It is also of importance to mark the attention to the fact that all over the 'first' world, amateurism in electronics (in particular radio!) was highly popular. There were a lot of magazines around it, Wireless World, Circuit Cellar, Elektuur, Radio Bulletin, Elrad, containing worked out circuits and schematics. Often they even had kits available for their subscribers. Music related electronics was a constant theme covered in the magazine contributions: the theremin, electronic organs, loudspeaker enclosures, oscillators, amplifiers, mixers, filters, equalisers, effects, vocoders, synthesizers, tape recorders...
Composers with a substantial background in electronics are in the field of experimental music not at all rare: David Tudor, Gordon Mumma, Dick Raaijmakers, Allan Strange, David Behrman, Don

Buchla, Hans Otte, Jozef Anton Riedl, Albert Mayr, Richard Lerman, Trevor Wishart, Peter Singer, Gerhard Trimpin, Alec Bernstein, Larry Wendt, David Rosenboom, Joel Chadabe, Serge Tcherepnin, Darius Clynes, Warren Burt, Matt Rogalsky, Jacques Dudon, Dieter Truestedt, Jacques Remus, Jan Boerman, Larry Austin, David Wessel, Nicolas Collins... It was not without a reason, we ourselves always considered some training in electronics to be an essential component in the curriculum for musical composition at the conservatory. Electronics as well as programming have been compulsory components in my own teaching until I retired in 2015.
4. Compositions that depend on environmental circumstances and conditions.
A good example here is to be found in John Cage's 'Radio Music', wherefore short-wave receivers are used. Even if the difficulty in finding suitable short-wave receivers (not evident, as modern receivers use a lot of circuitry to suppress unwanted signals, circuitry that was not present in legacy receivers) is surmounted, it is a fact that the number of short-wave stations in the aether has tremendously decreased. So the multitude of morse-signals that could be picked-up up to the late seventies, is simply no longer present. Using samples in this case just ruins the piece as all adventure and all chance elements are missing. Using the FM-band instead, leads to an aesthetic that Cage would most certainly have rejected, as that radio band merely carries popular music stations.
In our opinion, such pieces may be lost forever in as far as their viable live performance is concerned. The recordings remain and the narrative around the original performances became 'history'.

In a following series of chapters, we will try to cover at least some of the most common practical issues with regard to the use and restoration of legacy electronics and technology.
1.- Tape recorders
2.- Sound and wave generators
3.- Voltage controlled equipment: synthesizers and modules
4.- Ring modulators
5.- Transducers and sensors
6.- Piezoelectric transducers
7.- Effects
8.- Power Supplies
- Saturated core supplies
- High voltage variable power supply

9.- Alarms: Bells, horns, buzzers and sirens [odt file] - drop this?
[10.- Electromechanics: automated acoustic instruments: drop this ?
- player pianos
- novel instruments and constructions: Christoph Schlaegel, Trimpin, Jacques Remus, Martin Riches
- Expression control in musical automata
- sound art installations

11.- Interactivity: Gesture sensing [pdf file]] drop this?
[12.- Amplifiers: voltage amp's and power amps.
13.- Sound producing devices and vibrators
- Loudspeakers (Raaijmakers, Cathy Van Eck,
- Megaphones (Charlie Morrow,
- Horns]

10.- Case Studies [to be expanded]
- John Cage
- Alvin Lucier
- Karlheinz Stockhausen
- Dick Raaijmakers

- Gordon Mumma
- David Behrman
- Robert Ashley
- Hans Otte
- Jozef Anton Riedl
- Dieter Truestedt
- Dieter Schnebel
- Michel Waisvisz
- Trevor Wishart

1.Tape recorders as musical instruments

The analogue reel to reel tape recorder

A detailed description of the analogue tape recorder would cover more than a single volume book. Moreover, such books are available in archives worldwide. Some references are given in the bibliography. The essentials might be worth recapitulating here though. Moreover, it is quite essential for musicians to have a good insight in order to be able to perform quite some pieces using analogue tape recorders.

Historical introduction

The historical invention of the magnetic recording principle is generally attributed to Valdemar Poulsen who invented and made a wire recorder in 1898 capable of recording telephone conversations. It used rolls of very thin steel wire and magnetization was straightforward, using a ring shaped electromagnet whereby the core ring was opened with a small gap through which the wire was lead with constant speed. The same electromagnet could be used for recording and reproducing.

It took until 1927 before this technique was improved with the invention of the bias current. Hereby an ultrasonic audio frequency was mixed with the signal to be recorded causing the magnetic poles on the wire to switch much faster and thus enabling even the recording of musical signals. This bias current has been preserved throughout the entire history of analogue magnetic recording technologies.

Only in 1935 the flat magnetic tape was invented and produced in Germany by BASF. An acetate tape was used, coated with a layer of magnetisable particles. This tape was used on the first German 'magnetophons' and used by their broadcast stations. The audio quality was much better than that of the shellac records used worldwide. At the end of WW2, a few machines were smuggled out of Germany to the US, where they started copying them and producing similar tapes on their own (Ampex, 3M). Only in 1947 they succeeded in producing workable copies of the German invention.
From that moment on, the magnetic recording technology soon became the universal standard for professional audio recording all over the world.

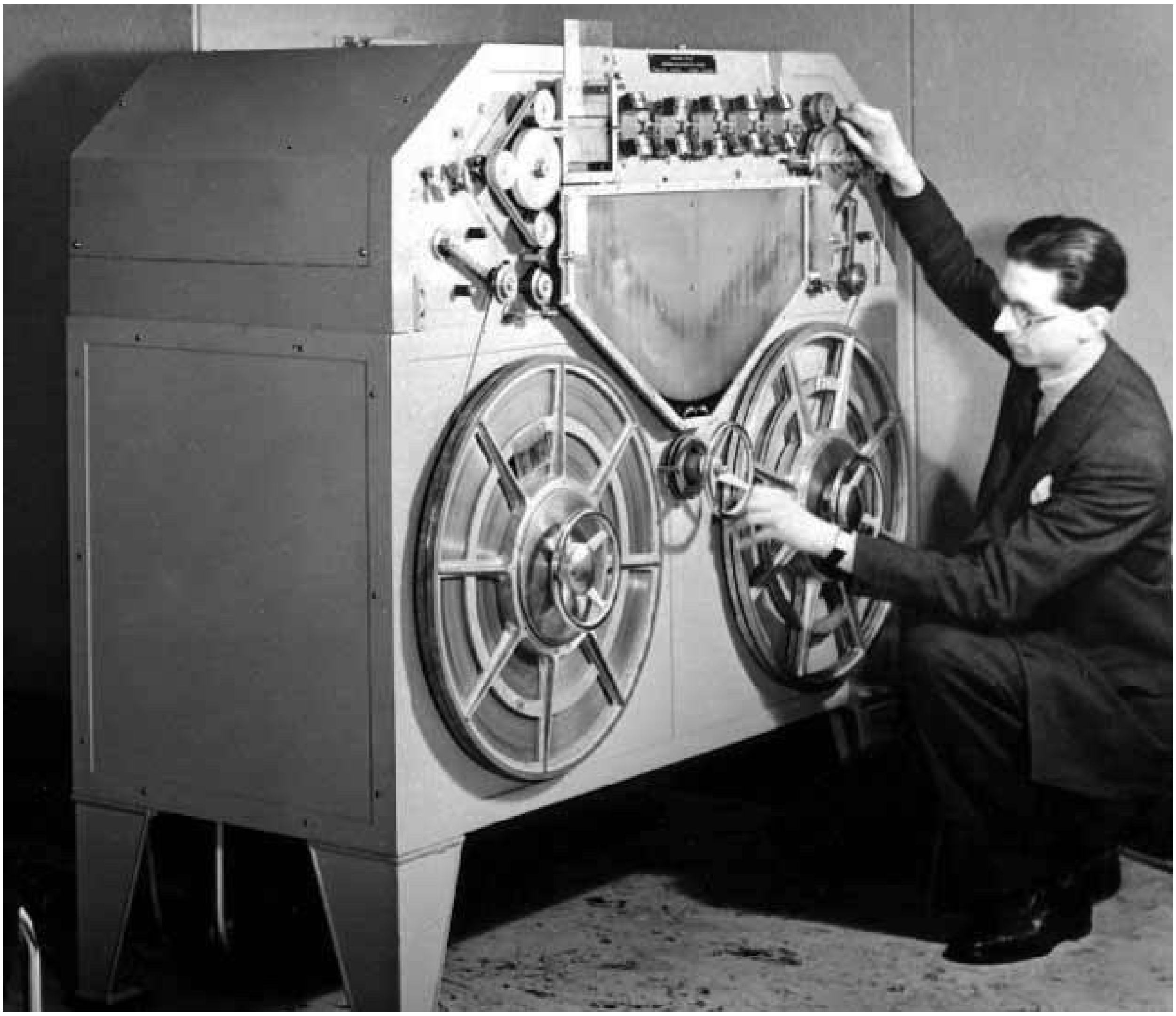
All tape recorders of professional quality consist of three motors: one for driving the capstan at a very constant speed (78cm/s at first, later 38cm/s), one for rewinding and one for fast forward and pickup force. Furthermore there are invariably three heads in the following order: erase head, recording head and playback head. The electronics consist of following essential circuit blocks:

- a bias oscillator (50kHz up to 180kHz, sine wave)
- a recording amplifier with proper equalization response
- a reproduction amplifier with an adapted equalization response (IEC or NAB)
- motor control circuitry
- a power supply

The most common tape format has always been 1/4" wide. For studio multichannel recordings 1/2", 3/4", 1" and 2" tapes were produced and used as well. Tape thickness had quite some variation as well. Long play tape, on a large reel, had a playing duration of some 45 minutes. Standard tape could hold only 30 minutes. Double play and triple play tapes have been produced but never found applications in the professional audio scene as the effects of magnetic print-through were getting more and more severe with decreasing thickness of the tape. On professional machines, the bias current and the magnetisation level were to be precisely adjusted to the characteristics of the tapes used. Most studio recordings were kept and stored 'tail-out' (with the end of the recording at the start of the reel). The reason for this at first sight odd habit was that if print-through happened, it would come after the first layer of recording on the reel and not anticipating it. Another reason was that fast rewinding a tape after use -as required for 'tail-in-front' storage- was always a bit more rough then taking up the reel after normal playback.

Professional equipment

For studio use: Telefunken, Philips, Otari, Studer-Kudelski. These machines, mounted on large consoles, were rarely leaving the studio. There are 1/4" as well as 1/2" and 1" types, 2 up to 16 tracks. Here is one:



And here we have Delia Derbyshire at work in the BBC radiophonic studio:

For mobile use: Nagra, Stellavox. These were the machines of choice for radio reporters on professional broadcast stations around the world. The Stellavox machine was commonly used in the film industry.
Uses in live electronic music: These machines had to be 'portable'.

Common types:

- Revox G36: Vacuum tube type first introduced in 1956. (service manual). Called 'portable', although with it's 22kg not a lightweight.
- Revox A77: Transistorized version (1967 - 1977)
  - Service manual
- Revox B77 (introduced in 1978)
- Revox PR99: Latest professional model produced.
- Fostex R8
- Nagra: portable equipment, top quality but rarely used in live electronics because of the limited motor force and the extreme high cost.

•Stellavox: highly popular in the movie industry.
•Tascam
•Uher portable (The 'poor-man's' professional recorder...)
•Carad: A Flemish brand, using vacuum tubes. The most common model was full-track mono.
•Hencot, a French professional brand known for its unstable behaviour
•Phillips: Phillips produced professional audio equipment and taperecorders for studios up to the late sixties. Henk Badings and Dick Raaijmakers used them and also they were used to run Varese's Poeme Electronique in the Phillips pavilion on the World Exhibition of 1958 in Brussels.

Essential features: 3 motors, 3 heads. Speed: 19cm/s, 38cm/s, 76cm/s

Track format: Full Track (mono), Two-track (stereo)

Quarter track (stereo) models were reserved for the amateur market and found little application in studios or live electronics.

Multichannel recorders (starting from 4-track on 1/4" tape) never really found applications in real-time performances. At the most, every so often a Teac, Fostex or Tascam 4-channel recorder was used for reproduction of pieces using quadrophonic audio.

On this picture: Pierre Henry in his studio between 1964 and 1971:

And here we have a view on the WDR studio at the time Stockholm composed his hymnen:

The tape recorder here is a 4-track 1" machine made by Telefunken.

Correction networks and standards:

Tape recorder heads are inductive devices and thus exhibit a non-linear frequency characteristic: the impedance goes up with frequency. Therefore at recording, a special equalisation is used compensated by a more or less inverted equalisation on playback. Europe and the USA used different standards: in the US, the NAB standard was in use and in Europe the CCIR (or IEC) standard.

European CCIR equalisation standards:

tape speed		
76 cm/s	35 us	4547 Hz
38 cm/s	35 us	4547 Hz
19 cm/s	70 us	2274 Hz

American NAB equalisation standards:

tape speed	high		low	
30 ips	17.5 us	4547 Hz	-	-
15 ips	50 us	3183 Hz	3180 us	50 Hz

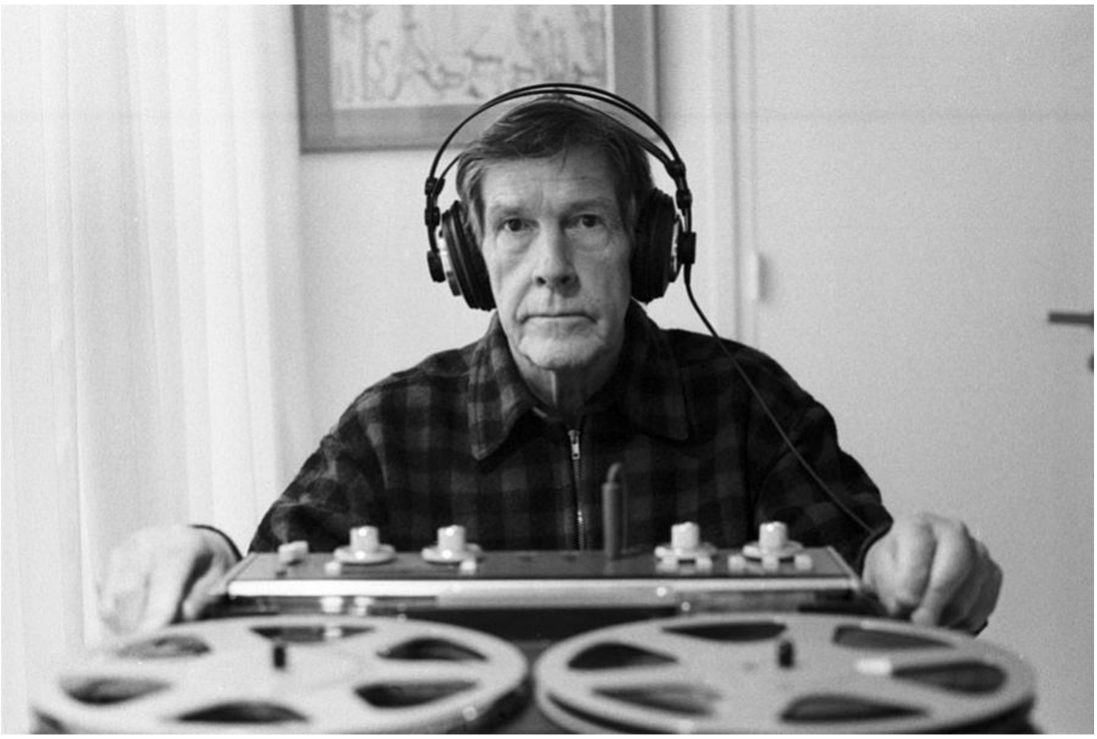
7 1/2 ips	50 us	3183 Hz	3180 us	50 Hz

Reel to reel analogue tape recorders on stage...

Without any doubt the Revox series of reel to reel tape recorders have once been the workhorses for all contemporary music production involving electronic technology in the timespan 1956 to 1990. In the sixties model G36 (model A was introduced in 1956!) was dominant (a very heavy, 22kg vacuum valve machine), in the seventies the A77 (produced from 1967 to 1977) model was predominant and in the eighties superseded with the B77 (introduced in 1978) and PR99 models.

On the picture below, dated 1970, we see Steve Reich experimenting for 'Drumming'. In the left of the picture we clearly see a Revox G36 machine.

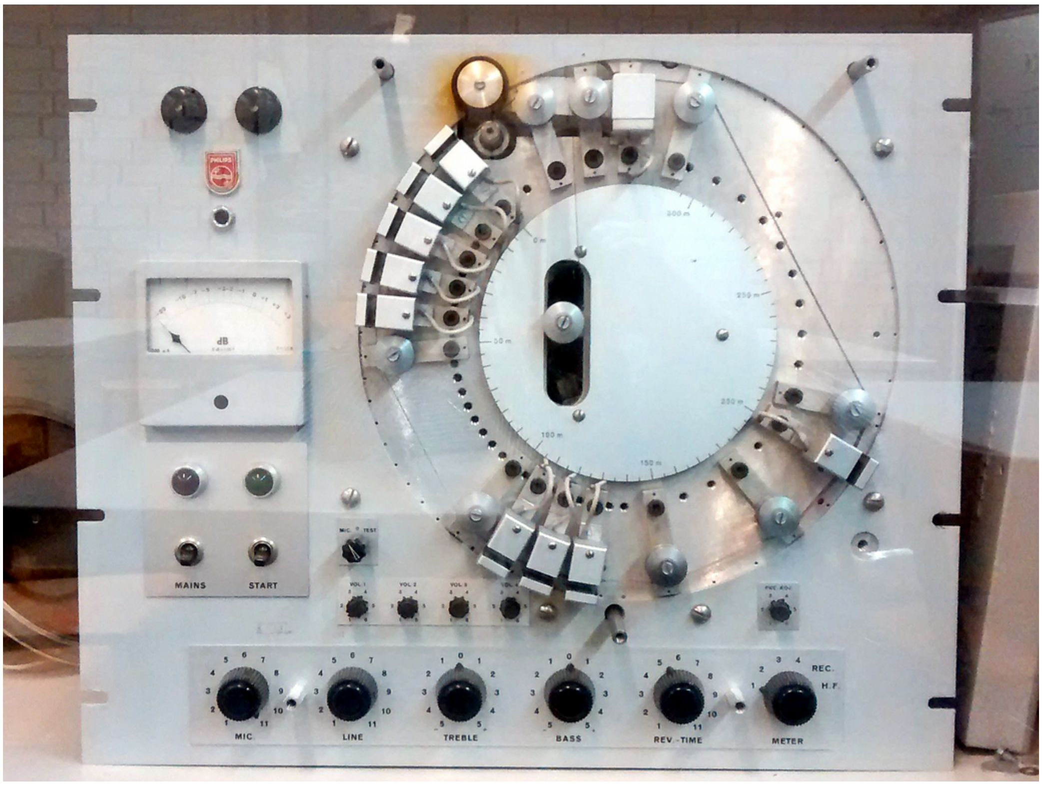
John Cage, using a Revox A77 is depicted here:



For all concert uses only the 2-track type, preferably with speeds 7.5 / 15 ips (19 cm/s and 38 cm/s) were to be used. Types with 9.5 cm/s and 19 cm/s speeds were also widespread. Their price was the same, but obviously by using lower speeds, one could save on tape expenses at the detriment of frequency response and dynamic range..

Taperecorders on stage as well as in studios were used in a broad range of applications:

- - playback of pre-recorded tape material in combination with instrumental music making or performance. The term 'composition for instrument and tape' is today still used by quite many composers although the tape has long been replaced by an audio track on a computer... We cannot image any composer or performer today misses the often very loud 'click-clack' sound accompanying start/stop commands that used to plague so many concerts involving musician(s) and tape-machines.
- - use as an audio delay line. The minimum delay time is determined by the distance between recording and reproducing tape heads. With a minor preparation it was (and is...) possible to enlarge this distance by pulling the tape out between these two heads and feeding it through an external pulley.
- - use as echo and reverb machine: this was done by recording on one track and sending the (delayed because of the distance between recording and reproducing tape heads) reproduced signal back to the other channel, thus creating a delayed feedback system. Special machines have been produced for this purpose as well. These were equipped with multiple reproduce heads at slightly different distances. (Tape echo machines). Here is a picture of one such machines:



- \- use for tape loops, using one or two machines. Some examples:
    - Stockhausen's 'Solo mit Rueckkopplung' is a good and sophisticated example.
    - Brian Ferneybough's Time and Motion studies.
    - Pauline Oliveros: 'The Bath' (1966), 'I of IV' (1966), 'Beautifil Soop' (1967)
    -

\- use for inverting sound and music by playing the tape backwards

- \- use for pitch transposition by changing either the capstan wheels or the capstan motor speed.
- \- use to cause a phase shift, as no two machines run at exactly the same speed: used, amongst others, by Steve Reich to realize 'Come Out' in 1966. Possibly the first example of such use.
- \- use as a generator of artefacts by causing interference between the bias oscillator and the input-signal. This was used by Richard Maxfield in 'Night Music' (1960). Maxfield connected the sawtooth (timebase) output of an oscilloscope to the input of the tape recorder and thus obtained difference tones in the audio range. The resulting sounds in this piece closely resemble sounds of nature, insects, toads, bats and birds however, they are all pure electronic.

Although it is perfectly possible to replace these analogue machines nowadays with digital, mostly computer based, technology we are convinced that in many cases an essential part of the performance gets lost by doing so.

However, using the old technology raises problems, as well functioning reel to reel tape recorders are harder and harder to find. In almost all cases the machines to be used will be between 60 and 30 years old and will be in need for maintenance and repair. Undertaking such a task requires a good knowledge of analog electronics as well as servicing experience, skills that are becoming rare.

Before starting repair or maintenance work on just about any tape recorder, first thing should always be to check the heads for wear. First clean them thoroughly with a soft piece of felt soaked in isopropyl alcohol (do not use methyl alcohol as it contains water, nor acetone as it may dissolve the insulation as well...) until their faces are fully shiny. Check for wear by observing grooves left by the passing tape. On new heads there should be no visible grooves at all. If the grooves are deep (let's say deeper than 0.2 mm) the heads need to be replaced as the reproduction of higher frequencies will drop seriously. It makes no sense to repair a machine with worn out heads as you will never manage to get it sounding the way it ought to sound. It must be said that it is very hard to find new head assemblies.

An essential part of maintenance on all reel to reel tape recorders consists in demagnetising the heads as well as all other ferromagnetic parts that come in contact with the tape. A simple tool is required to perform this task: a tape head demagnetiser:

This device simply consists of a coil with a fixed ferromagnetic core protruding from the coil. It connects directly to the mains AC voltage. It generates a strong magnetic field at the mains frequency near the tip of the protruding core. However simple the device, it should be used with care as touching the head surfaces with the tip can damage them severely. It should be held at very close proximity to the head for a few minutes, with the tape recorder switched off. Slowly move the demagnetizer towards and afterwards slowly away from the parts to be demagnetised. All three heads have to be treated but the most critical one is the reproducing head. Always do this with the machine turned off, as the voltage induced in the heads otherwise might overload and damage the preamplifiers in the machine.

Without a proper set of schematics, don't even try to repair a Revox or any other professional reel to reel tape recorder. The schematics as well as the service manuals are all available for download from the Studer website.

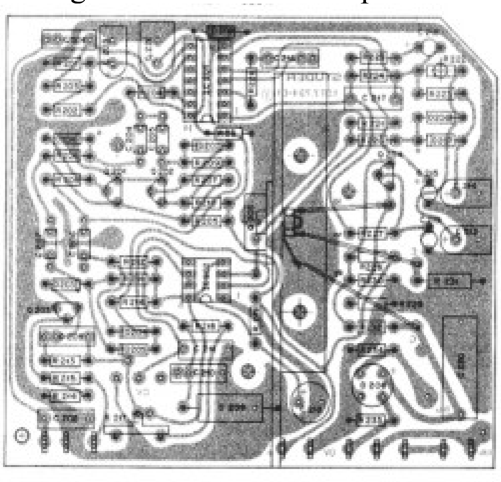
Here is a link to the circuits for the Revox A77 model on our own site.

Case reports on repairs performed by us on Revox tape recorders:

<u>Revox model A77, 1972 2-track</u>
This machine didn't do anything when switched on. First we discovered that the housing has a couple of pins that insert in a plastic cap covering the mains connection. They break the mains power as soon as the machine is removed from its cabinet. So first thing we did was inserting two bolts in these holes. This at least connected the machine to the mains power. But, this was not the real problem... We checked the 21V power supply to discover this voltage was absent. Not the 630mA fuse was blown, but the bridge rectifier appeared to be open circuit. We replaced it with a 4 A type mounted on the side of the chassis, as it was to big for the original PCB. Still it wasn't working, although the 21V power voltage was found O.K. now. Further checks revealed electrolytic capacitor C307 (250uF/25V) was shorted. Resistor R313 (39 Ohms) was found to be burned. We replaced both parts. The capacitor with one rated for a somewhat higher voltage. Therewith the job was done and the machine back in operational condition.

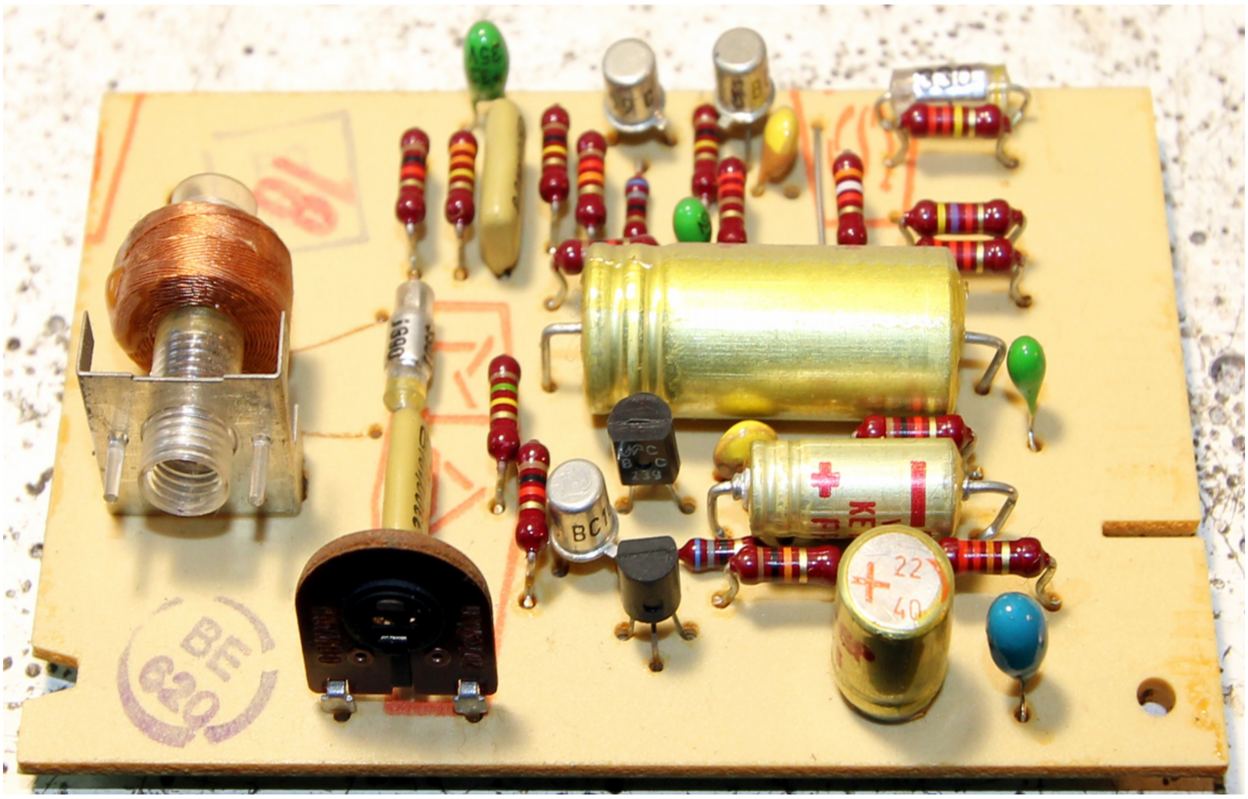
<u>Revox mode A77, 1975 2-track</u>
On this machines the lights came on, the take-up and rewind motors worked as normal but the capstan motor appeared dead. At the most, on the switch on surge, it would move a bit and then stop again. The 21V power supply was checked and found out to be O.K. and within specs. We

replaced the 3.5uF AC capacitor for the motor with the result that the motor started spinning, but way too slow and without any torque. We checked the tacho sensor with the oscilloscope and this worked O.K., however as soon as we reconnected it the motor started turning at a way to fast speed... This was going to be a tedious job, as this meant we had to remove the motor control board. Checking the board made us replace all electrolytic s and tantalum's with modern and fresh types.

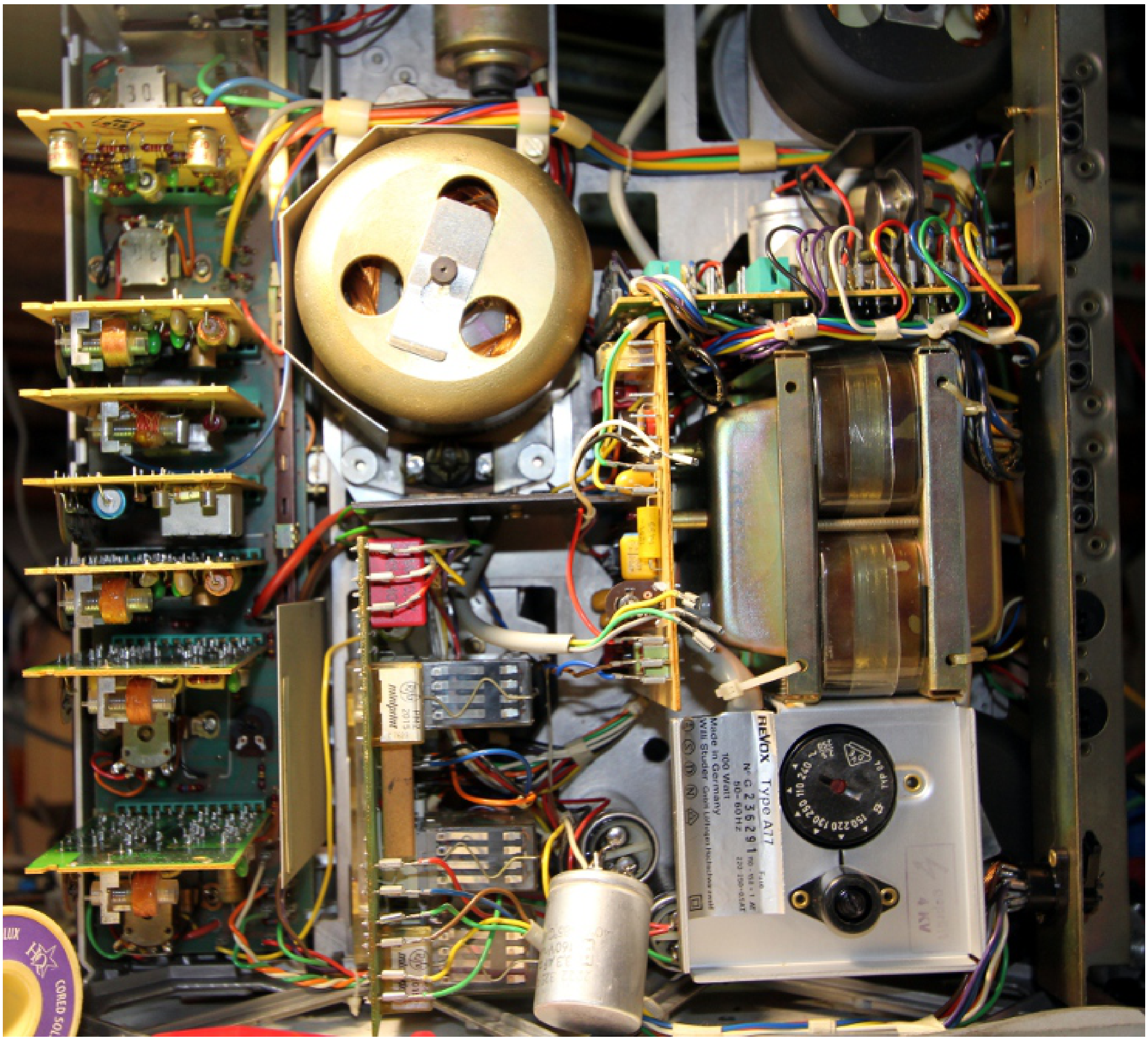


The power transistor driving the rectifier bridge in the motor control had a CE short. Its a 2N3583 in a cute TO66 package. Of course, this type is since long no more on the market. Looking up the specs, we decided to substitute it with an MJE13007 high voltage type. This one comes in a standard TO220 package and thus mounting and wiring had to undergo some changes. Still it wasn't working... We replaced the NE555 timer chip as well as the power transistor PNP driver (in the circuit specified as a BC178 but what we found soldered in was a BC308. This may have been an earlier repair... As we didn't have exact replacement parts in stock, checking the specs of the original transistor made us turn up a 2N3906. Before reassembling we checked the board's functionality by connecting a pulse generator (800Hz) to the tacho input, a lab power supply set to 21V. The board draws some 50mA. By connecting the speed switch wire (red) on and off the 21V we could verify the pin 1 output of the (obsolete and hard to replace) TBA931 reacting properly. The base drive for the power transistor changed accordingly. So far so good. We reassembled the board and fired up the machine again.
Success! It works again, the motor nicely spinning.

As we noticed a little bit of popcorn noise on the outputs of the playback amplifiers, we replaced the following capacitors: C803 (1600uF) -here we replaced it with a 2200uF type, as 1600uF is hard to get on the market; C811 (100uF), here we inserted a tantalum polymer type; C814 (10uF); C813 (25uF), replaced with 22uF. Here is a view on the board before the replacements:



Here is a view on the internal guts of the machine:



The motor control board can be seen on the picture: it's the vertical board to the left of the transformer. The 21V power supply is on the board seen in top of the transformer. A final problem with this machine was that due to unthoughtful treatment by someone not too familiar with fine equipment, the knob for the rotary turn-on switch was broken off. This obviously was not an electronic problem but nevertheless a pretty tedious one to solve. We had to make a new bus fitting over the 6 mm axle on the lathe in order to replace the knob.

<u>Revox model A77, 1974 2-track</u>
This machine was completely functional, except for the recording part. We checked the bias oscillator and found it working as it should. Then we suspected the board with the recording relay and indeed found R601 (10 Ohms) burned on the PC board. Capacitor C602 (470uF/6V) was measured and found to be very leaky. Here is a picture -showing clearly the burned resistor- of the board before the repair:

We replaced both and still it wasn't working. So we went on and checked the AC152 transistor. This one is no longer on the market anywhere. It's a Siemens product for which we digged up the data sheet: PNP Germanium transistor... The specs are pretty close to those of the AC128 (we had a box of these in stock) , so we gave it a try and performed the replacement. Replacing old germanium transistors is an always recurring problem in servicing equipment from the sixties and seventies, as these transistors are since long no longer in production. Never exchange a germanium transistor for a silicon one! If replacing with silicon is the only way to go, you have to redesign the entire circuit for operation within silicon boundaries. After also replacing the leaky capacitor C712 (250uF) on the oscillator board, everything was fine again.

A few general remarks.

If you Google the topic on Revox repair on the internet, very often you will encounter the practice of replacing all electrolytic and tantalum capacitors. We have serious doubts as to this practice. Modern electrolytic capacitors have a lifespan (MTBF) of only ca. 3000 hours. Moreover, in an experiment, we have removed some 30 electrolytic s from machines at least 40 years old and found that only 2% of them where really out of specs. We stick to the rule 'if it ain't broke, don't repair it'. If replacements are needed and the capacitor values are in the range 1uF to 100uF, it's a good -though very expensive- idea to replace them with tantalum polymer or aluminium polymer capacitors. Smaller caps really rarely fail and for the larger ones, electrolytic s are the only choice. If measuring capacitors they need to be unsoldered first and then measured both for capacitance and leakage. The Fluke 87 multimeters can handle this measurement very well and reliably. Generally leakage is the main problem with old electrolytic s.

Another issue is the 2-prong mains entry socket. This way of connecting equipment is not conform to modern rules with regard to electric safety. Many repairman have changed this connector for a modern 3-pole IEC model and connecting the earth pin to the chassis. If only a single machine is used, there are no objections against doing this. However, as in the case with on stage uses or professional studio set ups with lots of gear and equipment, we would rather not have this earth connection as it may introduce ground loops and thus inject hum in the system. If you are really concerned with safety (in fact the risk is quasi non existent, as the power transformer in these Revoxes is of outstanding quality and performs well as an insulation transformer) and/or rules, a better alternative is to use a hefty insulation transformer on stage and leave all audio equipment floating versus earth and mains power lines.

Not too many people nowadays seem to know that the mains voltage on the European continent has slowly been risen from 220V up to the late sixties to mostly 240V nowadays. So, we advice strongly to change the voltage selector switch (it can be seen in the picture above) accordingly. Doing so will reduce dissipation and heat production in the machine. This may very well be one of the main causes of common 21V power supply failures in the A77.

Nagra

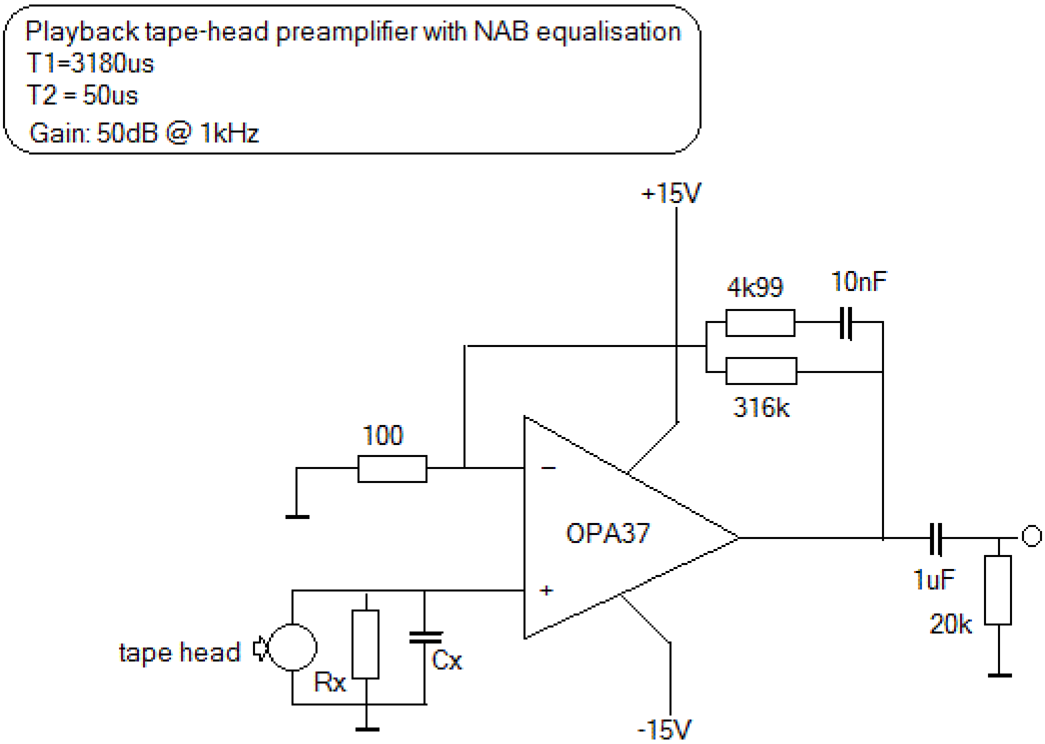
Revox model G36

This model uses vacuum tubes. The motors are much stronger than the ones used in the later (A77 and further) models. This made them very popular in applications involving long tape loops. Most often, the (mono) audio amplifier driving the internal speaker will be found to be defect. The best solution here is not to repair it at all, but to remove the two ECL86 vacuum tubes from their socket as well as the loudspeaker. All professional applications for such a machine use the line outputs exclusively. Moreover, by removing these components the machine draws substantially less current and also produces less heat.

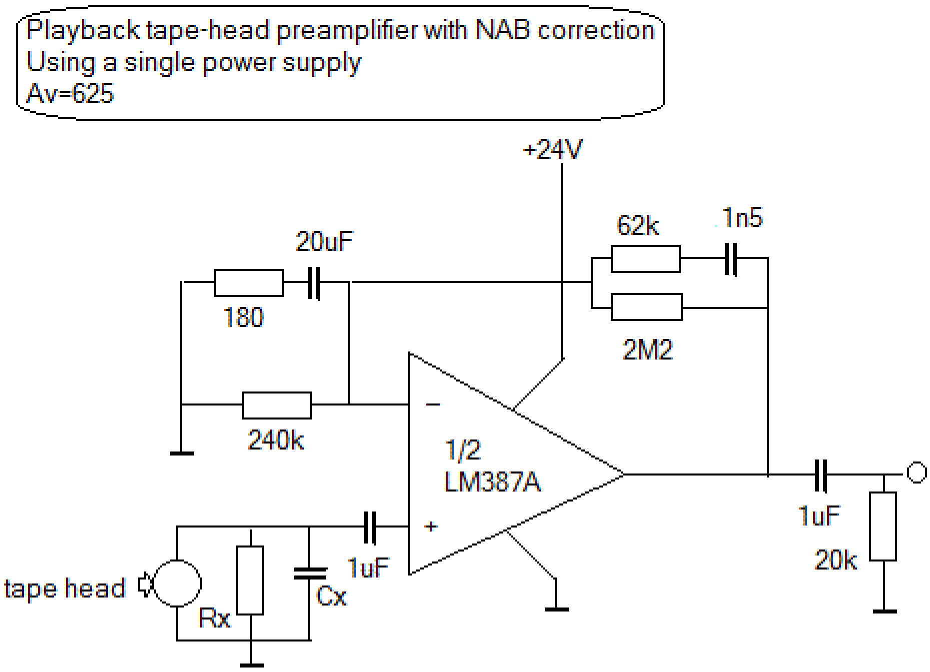
Here is a direct link to the service manual for all G36 models.

The main problem one may encounter when servicing these machines is finding replacement vacuum tubes. If you are lucky, you may find some new spares from old stock. However, often used vacuum tubes are offered for sale. Never buy nor use these as even if they still work, most certainly they do not work after the original specs. Moreover, vacuum tube testers have long vanished and even if you could dig one up, it would also be a vacuum tube instrument in need of servicing itself... Here is a picture of the vacuum tubes as found in the G36, a machine produced in 1964 (serial number 75027):

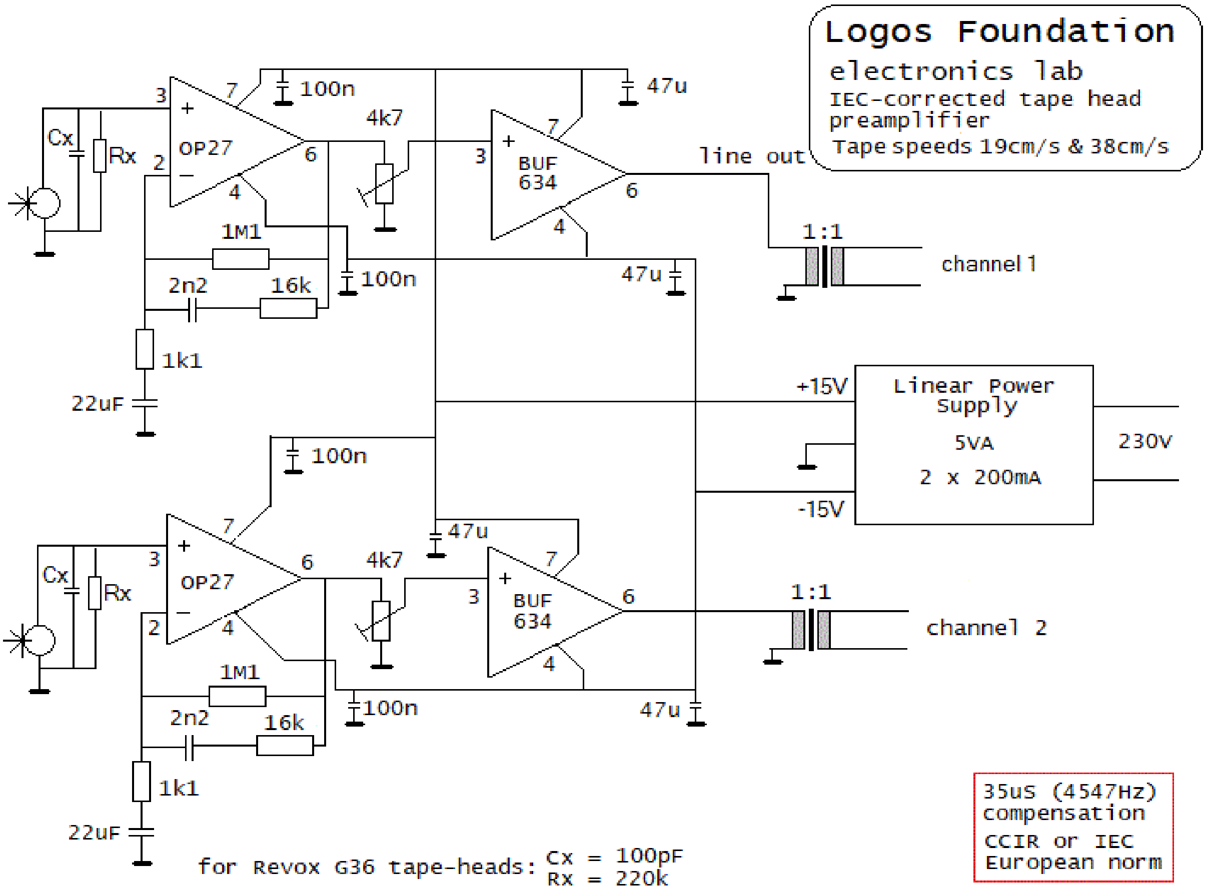
If the tape reproduction fails and no replacement components can be traced, you are left with the option to build a new playback amp. By using modern op-amps, the sound can be greatly improved over the original at the detriment of 'warm' vacuum tube sound. Here is a tested circuit, taken from an application note for the OPA27/37 operational amplifiers by Burr Brown:



If you find the use of a dual power supply objectionable or impractical, you could consider using the somewhat older LM387 dual audio op-amp in a circuit like this:



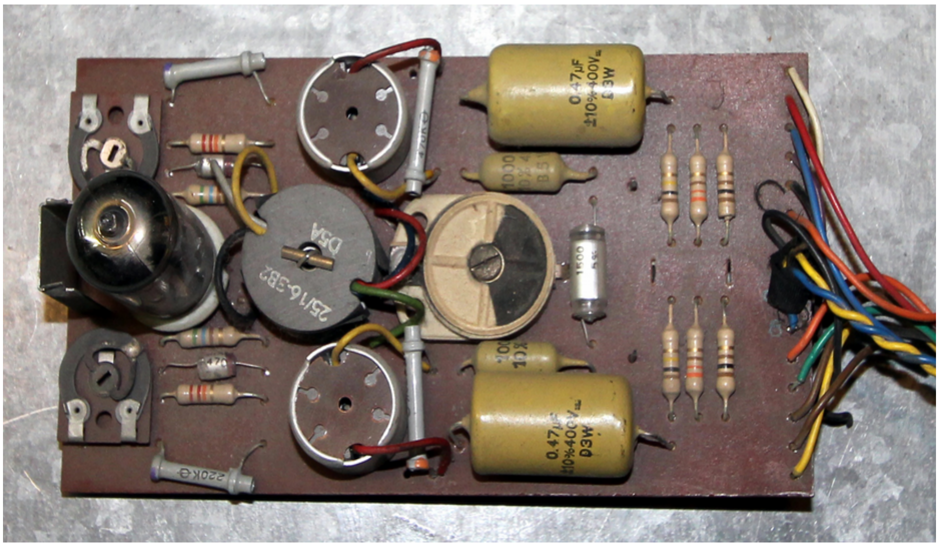
The values for Rx and Cx depend on the tape head characteristics. They should be copied from the original values found in the machine to be repaired. In magnitude order values for Rx -if present- in the range 10k to 47k and 100pF to 1.5nF for Cx are common. The result of this replacement -if performed with proper care given to wiring and component quality- will exceed by far that of the original circuit. If you want an IEC correction network (this correction was the standard for all European recording work on analogue tape), the following circuit can be used:



Note that in the IEC standard, there is no bass boost below 50Hz in the circuit and the roll off for the high frequencies is at 4500Hz instead of 3150Hz in NAB networks.

If a rectifier tube is found to be broken, replace it with a modern high voltage semiconductor double diode (BYV32 for instance). The same applies to selenium rectifiers, which we would always replace as all old types do leak now. This modification will not alter the sound at all.

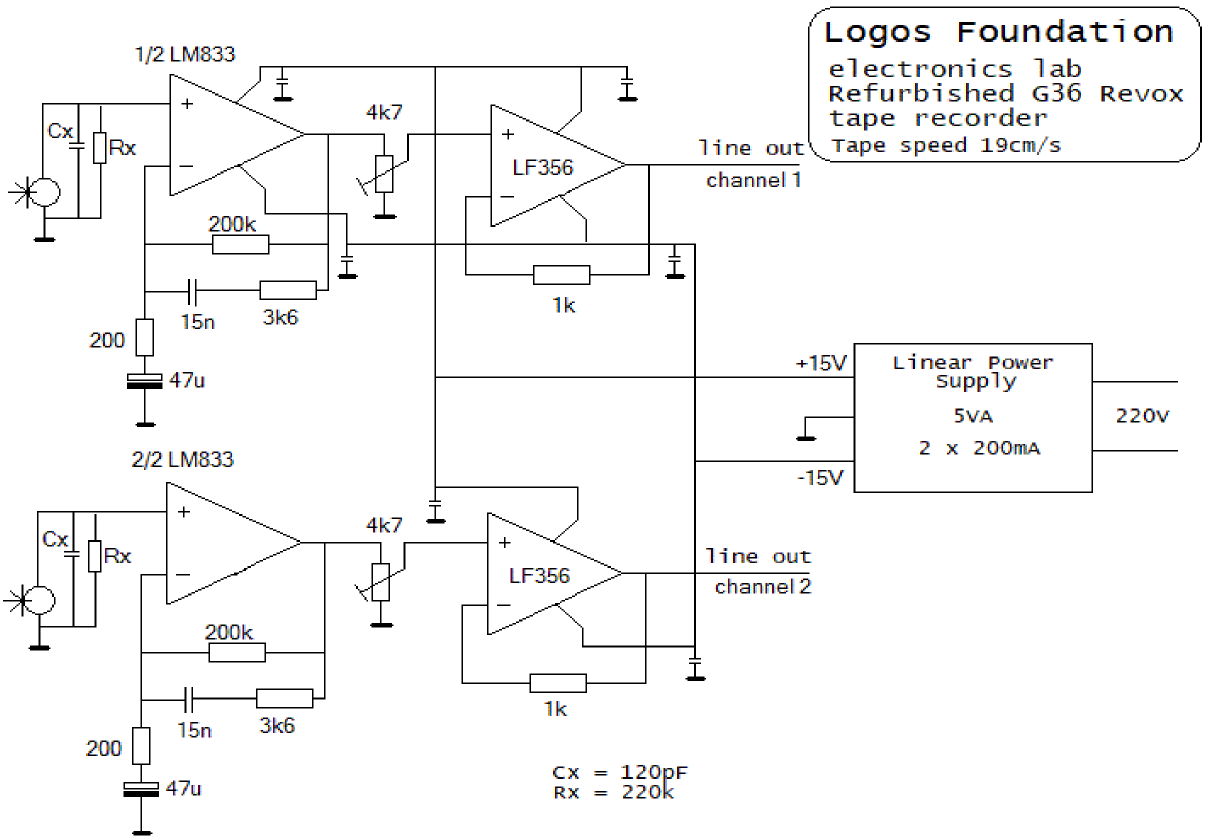
Here is a view on the bias oscillator from a G36, using an ECC83 double triode valve:



As the mechanical parts of this model are very sturdy and rarely fail, it's worth the effort to modify the entire machine for use as a high quality tape playback machine. Taking into account that it's unlikely anyone would want to use these machines in recording mode, limiting their functionality to playback only gives them a new life. Undertaking such a refurbishing involves removing all vacuum tubes. The output transformer for the monitoring amplifier (T853 in the circuit diagram) can be removed as well., We also advice to limit the recorder to a single speed only: the highest available (19 cm/s of 38 cm/s). This highly simplifies the wiring in the signal path and thus reduces hum and noise considerably. We performed the operation on a G36, series number 75027 dating from 1964. Measurement of the properties of the reproduction heads (Bogen) were:

- Inductance: nominal 540mH (left channel measurement: 519mH Q=6.74, , right channel: 569mH, Q= 6.38)
- DC-resistance: 268 Ohm
- Output signal at -3dB modulation: 4mV (taken from the data sheet and the circuit diagram)
- Load: 100pF across the terminals.

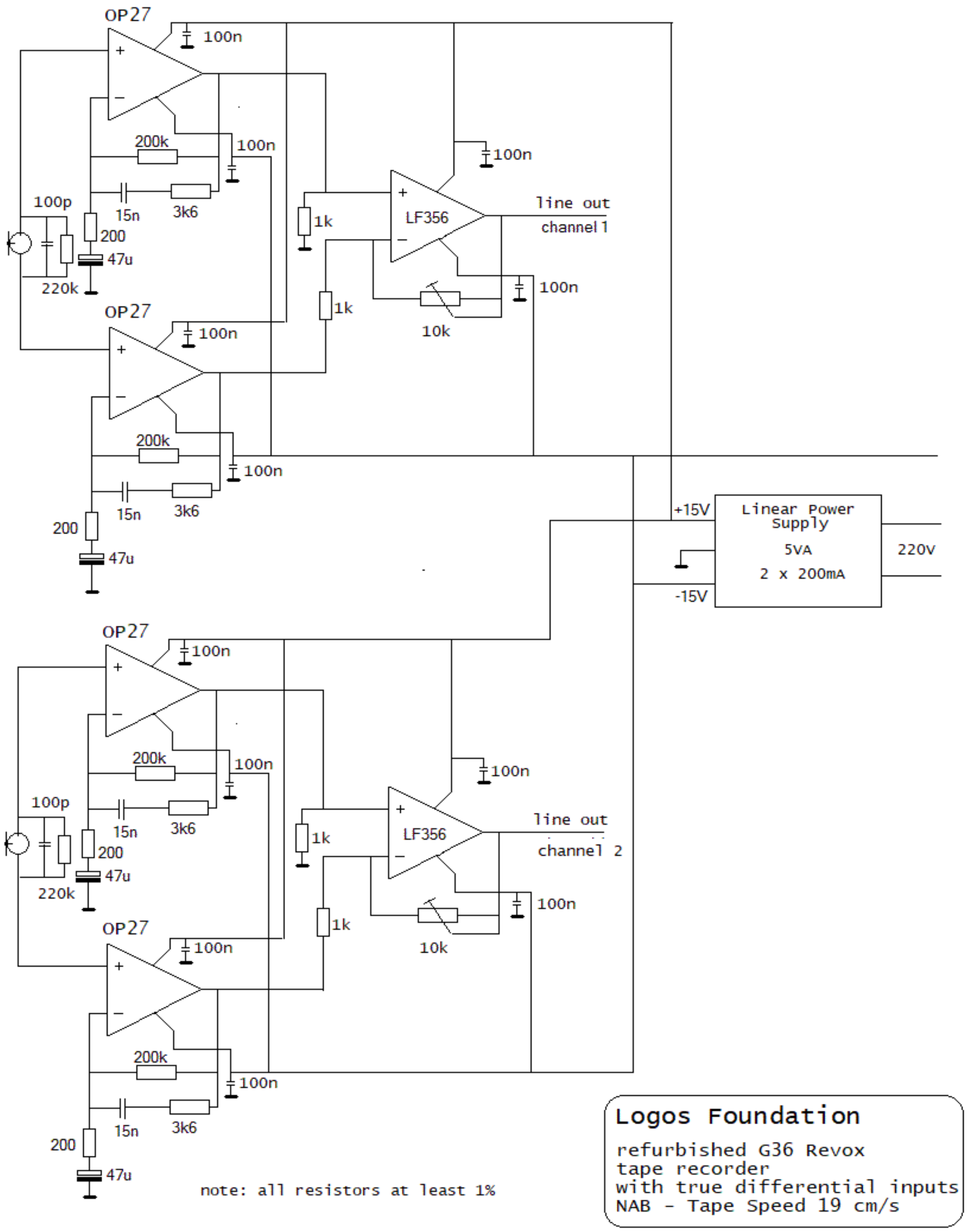
This became the new circuit:



2-channel NAB corrected tape head preamplifier

The 4k7 multiturn trimpot should be adjusted such that the output voltage is at 0dB (775mV) for a 100% modulated tape using a 1kHz sine wave signal. The outputs of the circuit can be connected to the 'cathode follower' RCA jacks on the backside of the machine..

A furher improvement is still possible: taking into account that tape heads are fundamentally differential devices (none of its terminals are internally grounded), we can take profit of a true differential amplifier, thus eliminating any induced hum and noise unrelated to the tape head signal. Here is a design example:



Obviously, hum picked up by the tape heads themselves, cannot be removed this way. Screening of the tapehead itself is the only remedy here. Note that if you only want to replace the reproduction part of the recorder, you can remove all valves numbered V7, V8, V9, V10 and V11. The saved space can be used for mounting the 15V power supply.

The dutch composer Dick Raaijmakers, certainly one of the first to consider the tape recorder as a tool for music production, early in his career, modified the predecessor of the Revox G36, the F36, with some extra heads, to make loops and delays
possible:

This machine is on display now in a museum in The Hague (Netherlands).

<u>Carad models R62,R53, R59</u>

Vacuum tube full track mono machines produced in Flanders (Kuurne) between 1955 and 1971. The later models using transistor technology and stereo. These later models were rarely seen used in live performance however. The older full track machines are worth to be preserved, as long as the tape heads are not worn. They were used by composers such as Henry Pousseur, Lucien Goethals, Louis De Meester and Norbert Rosseau. Here is one:

At the time of this writing, it is still in full working condition. Note that the bias frequency used on these machines is 56 kHz, so much lower than the 70 kHz used by the Revox G36 series.

<u>Truvox tape recorde</u>r

We ran into this type as it was in use in the private studio of my composition teacher, Norbert Rosseau. It's a mono half track machine with three heads and three motors with the odd property that it runs the tape from right to left... a British make, indeed. To make precise pitch transposition possible Norbert Rosseau had a set of alternative capstan wheels turned for him on a precision lathe. They run up in equal temperament chromatic steps. Here is a picture of the complete set:

When used to record a sound, on playback with the normal capstan, all sorts of chromatic pitch shifts became possible. Obviously, this way of transposing pitch, also affects duration. If pitch had to be preserved, a machine with rotary heads was required (Springer machine).

Hencot reel to reel tape recorder (1974):

Revox B77 and PR99 models:

These are improved versions over the A77 model. A feature of these machines was that it was possible to control motor speed continuously, thus giving rise to manifold new musical possibilities. Nevertheless they were considered to be less sturdy than the A77.

By the beginning of the nineties we saw a gradual disappearing of the reel to reel tape recorder on stage. Pieces for musicians and tape were more and more making use of the much smaller cassette recorder. However none of these compact cassette machines have ever been produced equalling the performance of the reel to reel machines. The signal noise ratio was at least 12dB worse and distortion was also much higher. To overcome this, the machines often were equipped with Dolby noise reduction systems and/or had to use special magnetic tapes (so called 'metal' tape). However, this caused quite some compatibility issues when using a different machine for playback of the cassettes. The favourite high quality brands and types used were:

Nakamichi 550:

Presumably the best cassette deck ever produced.

Sony TCD10

Sony WMD5 series

Although been widely used, their importance in live performance is minimal and there is no reason to call in cassette recorders if it comes to historical performance practice. Moreover, even if anyone wanted to use them again, the problems of maintenance are nearly unsurmountable. We have quite a collection of these machines, but all of their potentiometers (exception made for the Nakamichi) crackle and cause a lot of noise. Replacement parts are absolutely unobtainable and standard parts do not fit the housing. A replacement with computer audio or CD seems appropriate and legitimate here. After all, there is little 'creative' of 'performing' you could really do with these machines: no tape loops, no separate recording and playback heads, no tape speed control, tiny and fragile connectors, no way to cue the tape with any degree of precision...

Contraptions using tape recorder derived technology:

Loose tape heads with preamplifiers

In one of the first issues of the legendary 'Source' magazine, one page appeared consisting of a square layer of magnetic foil. This was to be used as score material by reading it with a spare tape recorder head. Instructions were given in the score. The composer was Jon Hassell. Here is a picture of the composer performing the piece:

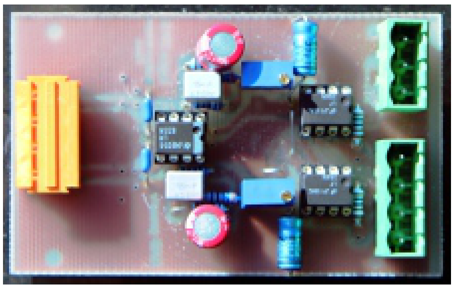
Mounted tape heads:

Dick Raaijmakers: 'Der Leiermann'

For this piece a machine is to be made allowing you to playback the pre-recorded tape (containing a recording of the famous Schubert lied with the same title) with a hand-crank driving the capstan. If any performer now would be willing to undertake a reconstruction of this piece, reconstruction of a copy of the original machine is a requirement.

Godfried-Willem Raes: 'Manipulofoon'

The NAB corrected circuit board looks like:



Two full-track tape heads are used and the device has two separate audio output channels. For playing it should be mounted on a camera tripod. Further documentation can be found here.

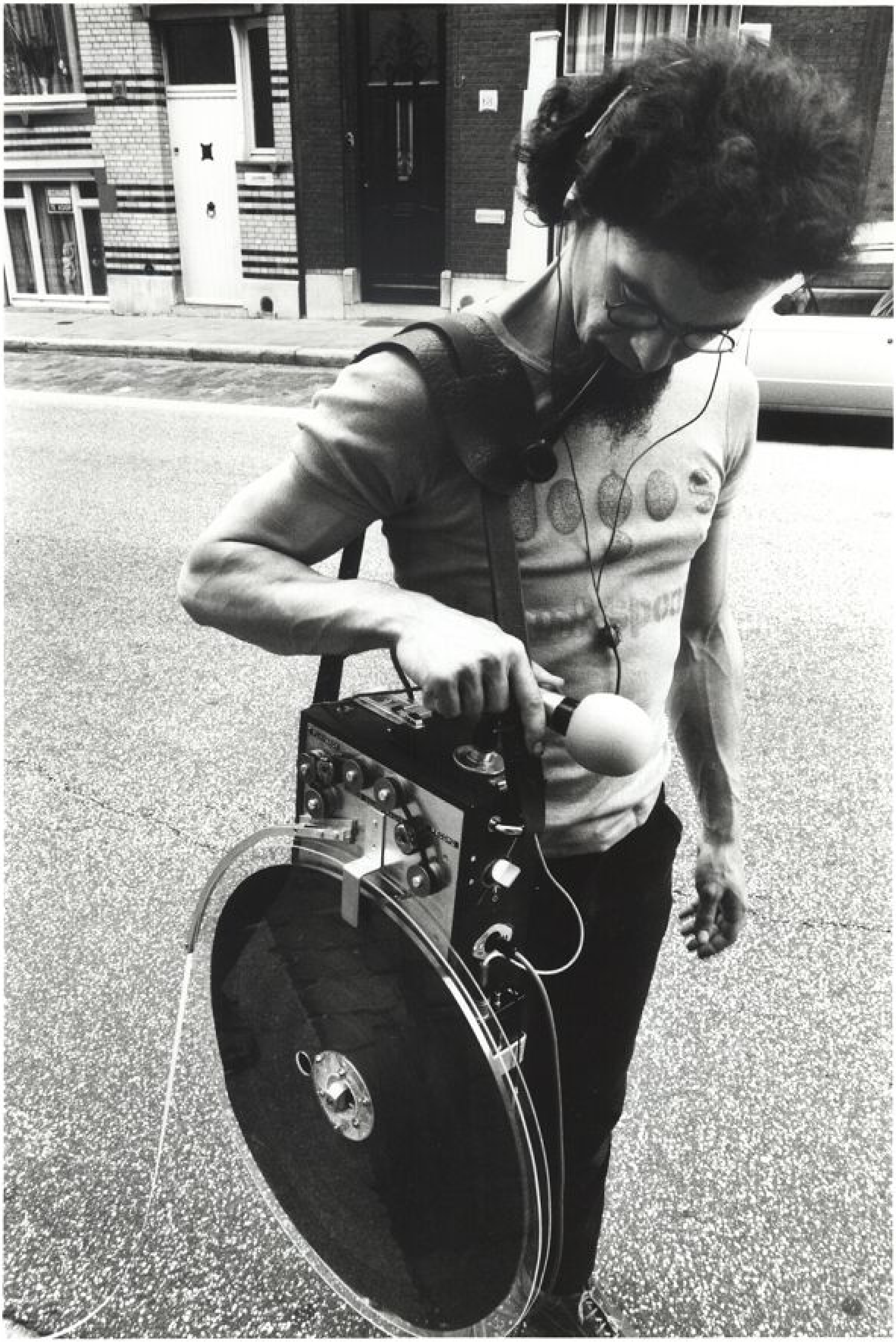
Michel Waisvisz

Michel Waisvisz' device is very similar, but he used a single tape head, megaphone loudspeakers and battery operated amplifiers.

Recorders without motor:

Godfried-Willem Raes: 'SoundTrack machine

SoundTracker
#1:

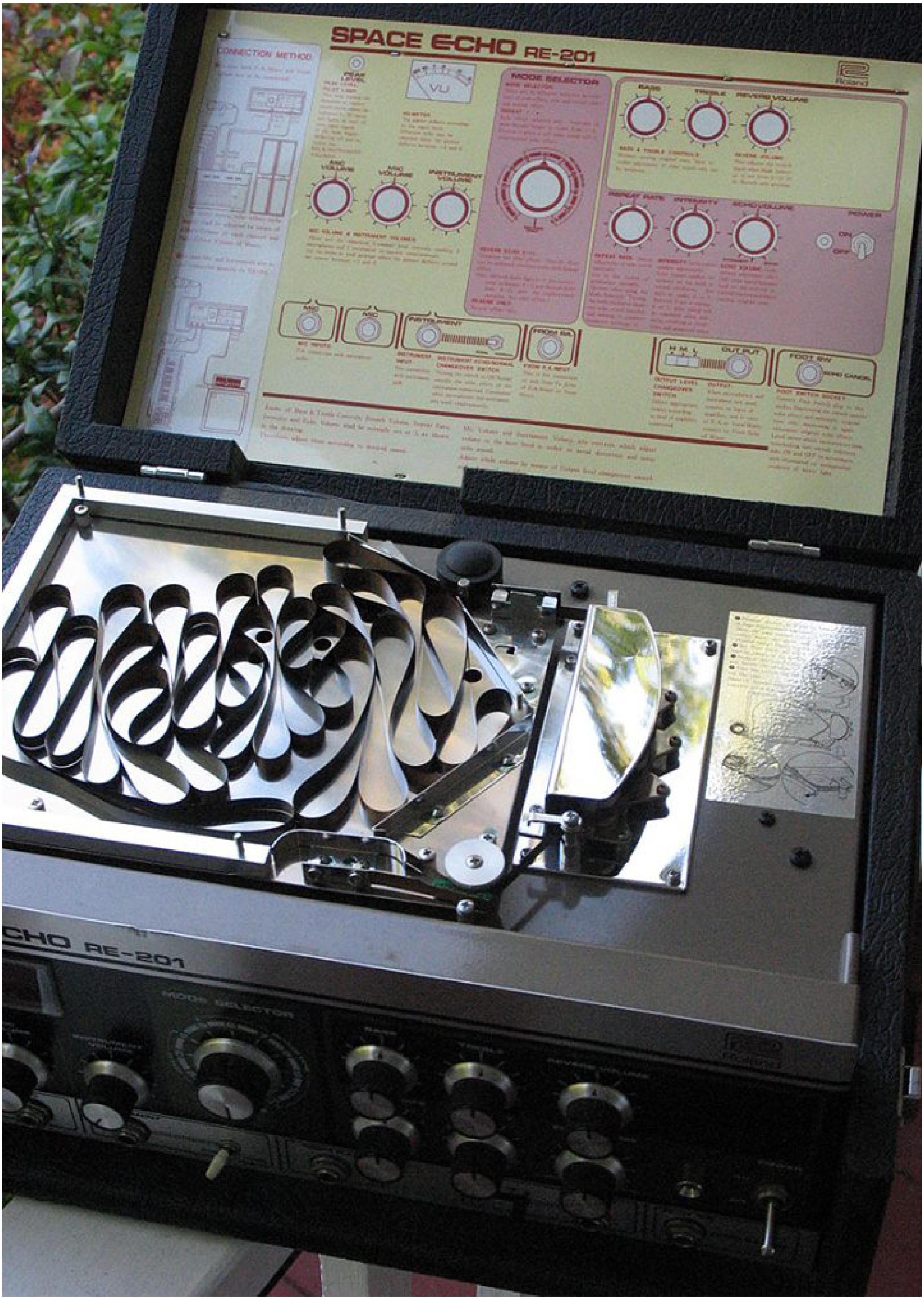


SoundTracker
#2:

More documentation on these projects can be found on the Logos Foundation website.

Tape echo and looping machines

These were amongst others produced by Roland around 1974. Here is a picture of their RE-201 model 'Space Echo':



Springer machine

This was a machine wherein a rotating playback head was used, a bit like in the much later developed video recorders. When the head was made to move against the movement of the tape, the pitch of the sound material could be transposed upwards without changing the duration of the recorded material. If the head was made to rotate in the direction of the tape, at the other hand, the pitch could be transposed downwards. No matter how beautiful a machine at first sight, it never worked without some flaws: glitches and dropouts. We donated our springer machine a few years ago to the MIM museum in Brussels.

As -as far as we know- Springer machines found no use in live electronics and performances, we see no reason for treating them in any depth here. Since the advent of computers powerful enough to handle audio in real-time, the function of these machines can very easily (and much better...) be replaced with software: pitch-shifting and time stretching are standard components in just about any modern audio editing program.

Case studies and reports with regard to real time performances using tape recorders:

Brian Ferneyhough, 'Time and Motion Study II' for cello and electronics.

A historical reconstruction of this composition was realised under our supervision and presented at De Bijloke, may 9th of 2015 with cellist Benjamin Glorieux. Tape operators were Maarten Craeynest, Pieter Matthynssens and Laura Maes.

For this piece four reel to reel tape recorders are required as well as ring modulators and volume controls. We provided all legacy equipment required for a historical performance.

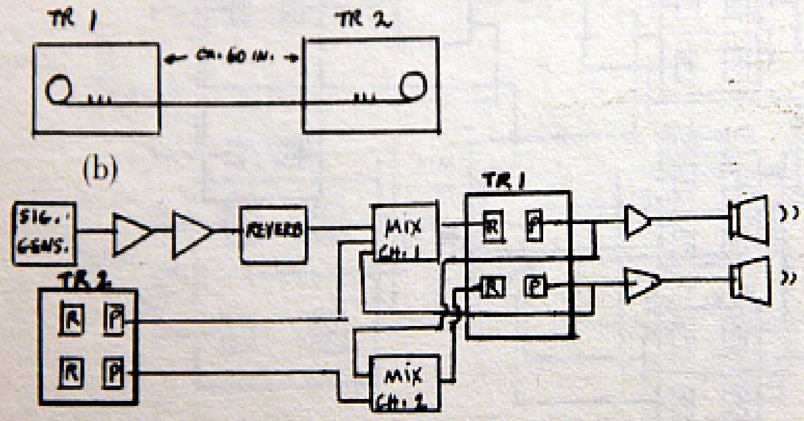
Karlheinz Stockhausen: 'Solo mit Rueckkopplung'

This is one of these pieces that nowadays is perfectly realizable using digital technology. Yet, if the tape loops required after the score are dismissed, the piece also looses all its charms. We have seen it performed my times since the late sixties -the Dutch oboist Evert Van Tright was a notorious performer of the piece- and every time it appeared to be quite a challenge to get the tape running smooth enough. Most of the time there was quite some jitter and a couple of times, we had to cope with tape fracture. Replaced with a simple computer patch, the piece becomes almost trivial and even seems to collapse.

Pauline Oliveros

Some of her compositions involving tape recorders, loops and delays are described in her collected writing 1963-1980, published under the title 'Software for People' (Smith Publications, Baltimore, 1984, ISBN 0-914162-59-4). Here are her descriptions of the different set-ups used:

I OF IV – 1966

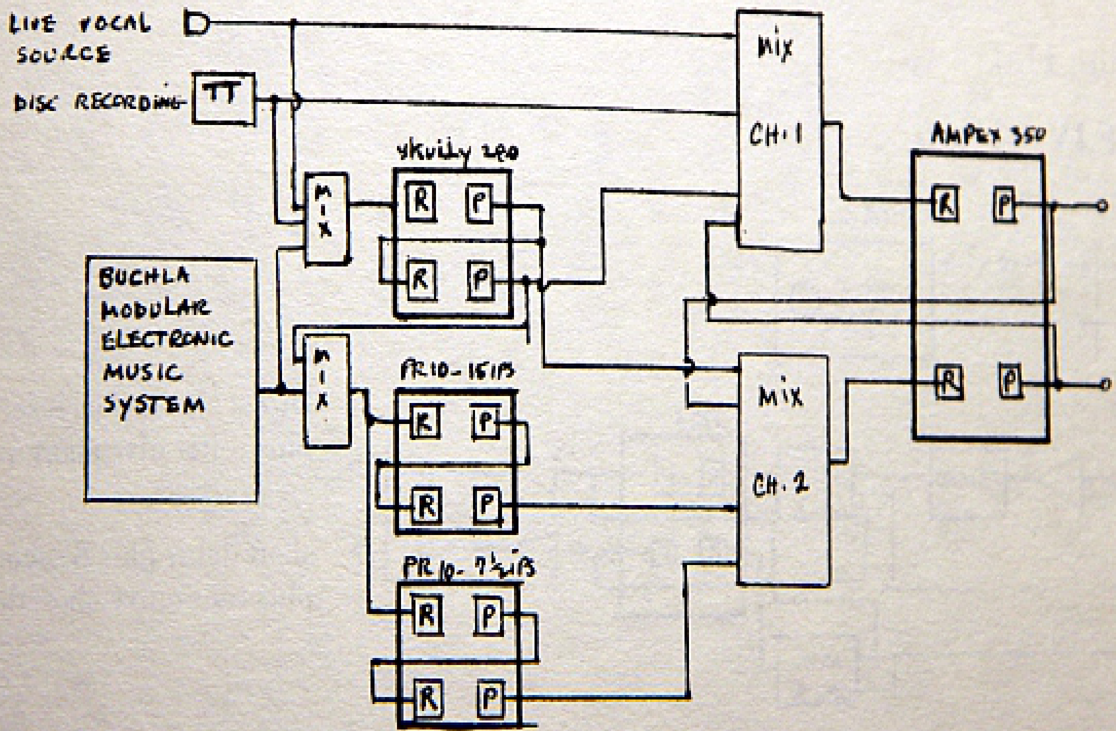


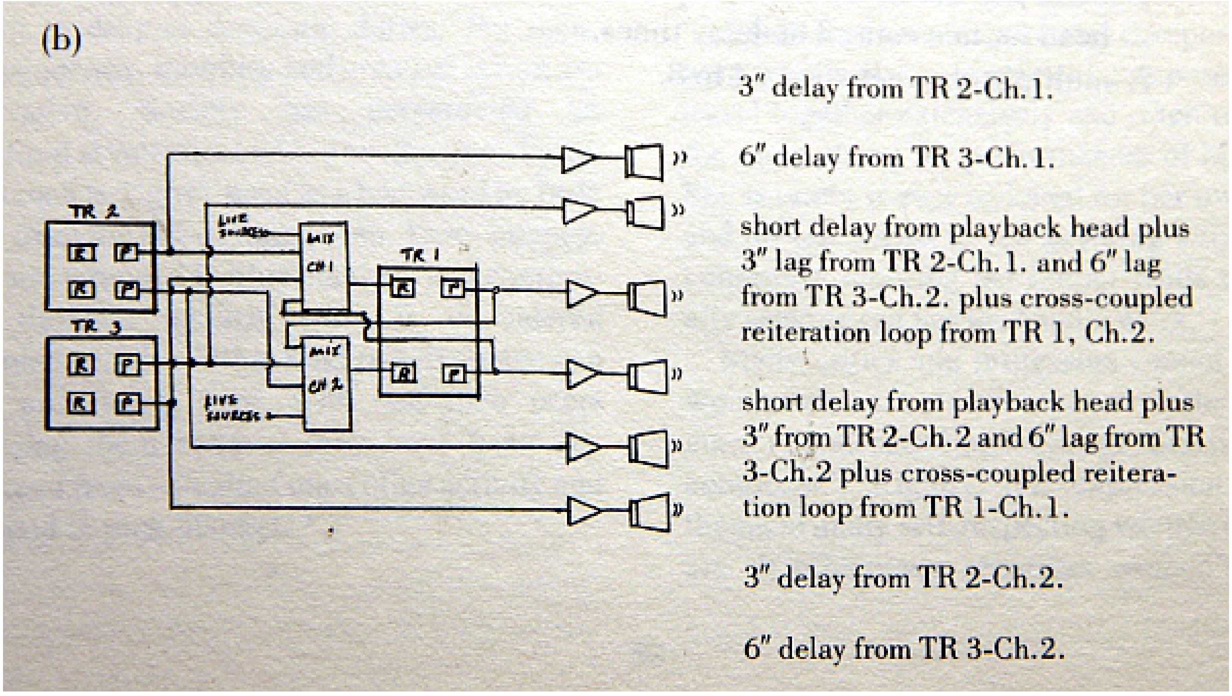
Direct signal plus 8 second delay plus reiterative short delay

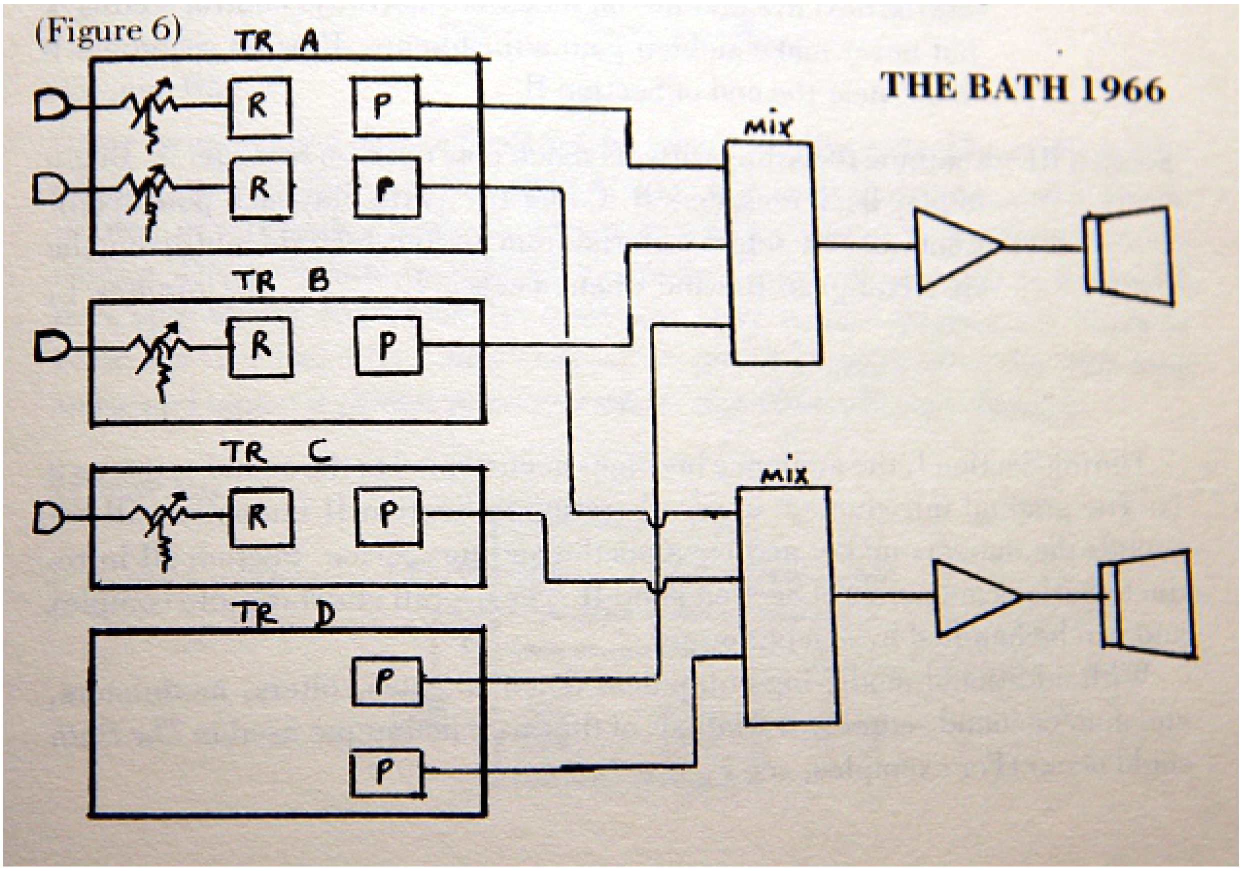
short delay plus 8 second delay plus reiterative short delay

(Figure 10)

BEAUTIFUL SOOP 1967







Here, as in the case of the Stockhausen piece, we think that the physical set up fundamentally adds to the rhetoric of the piece. It also makes the piece visually understandable. Only for recorded versions, a digital alternative sounds acceptable, but one could question the reason for doing this, as decent recordings made by the composer herself are available.

Godfried-Willem Raes 'FortePiano'

For this piece a single 15ips (38 cm/s) tape recorder is used as a delay machine. The short delay is caused by the physical distance between recording and reproducing heads.

Jerome Noetinger and Lionel Marchetti performing with Revox tape recorders (link to a Vimeo video from the STEIM archives). and loops:

Steve Montague 'Ambush'

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2. Electric and electronic sound: the tone generator as a musical instrument

Even long before the invention of the loudspeaker and the amplifier, scientists discovered that is was possible to generate sounds by pure electric means. These discoveries have inspired artists up to today, as very regularly we see contraptions based on these discoveries in performances and audio-art projects all over the world.

Likely the most important of such discoveries were:

1.- electromagnetic generation of sound: AC-current generators
2.- electrostatic generation of sound: the singing arc
3.- optomechanical and optoelectric generation of sound: August Lauste

Let's dig a little further into these technologies to see in how far they are still relevant for music practice today.

1.- Electromagnetic sound generation

Whenever a magnetic field changes, these changes can be converted to an alternating current by approaching the field with an inductor. This finding is at the base of technological realisations ranging from the motor, the alternator, the transformer, radio and TV broadcast and reception, radar technology... A large series of musical instruments based on electromagnetism has seen the light in the 20th century: the electric guitar, the clavinet, the Fender-Rhodes piano, the neo-Bechstein, the Hawai-guitar, the Chapman stick...

One of the very first electric instruments -the telharmonium, invented and build by Tadeusz Cahill, dates from even before the loudspeaker was invented. At the base of this instrument was the observation that an alternator (or dynamo) generates an alternating current whereby the frequency is a function of the number of magnetic poles wound on the alternator and on the speed of rotation of the rotor. By steering the speed of rotation it appeared possible to generate tones tuned to precise frequencies. By loading the so generated current with resistors, the amplitude could be controlled as well. As we all know, any alternating current in as far as its frequency falls in the audio range for human perception and its amplitude reaches out, can be considered the equivalent of a musical tone. In order to turn such a current into a tone, a device is needed to convert electric energy into vibration of air. Tadeusz Cahill had the smart, but still quite bizare, idea to offer (and sell...) the musical result of his telharmonium via the public telephone system! Listeners had to call a number and could then listen to a musical piece, generated in real time at the generator station. No matter how smart the idea, the company for the exploitation of the Telharmonium went bankrupt. The instrument did not survive in working condition.

2.- Another and completely different way for generating sound via electricity was the singing arc. A periodic discharge of a tuned LC circuit through an airgap between carbon electrodes caused a tone as a result of the fast heating and cooling of air particles in the gap and the compression/decompression as a result of this.

The ionic tweeter is another component in this category, using ionic wind caused by a very sharp point loaded with a very high voltage. By modulating the voltage, a sound wave can be generated.

High voltage sparks have been seen in quite a few audio art installations: Takehisa Kosugi (+), Rolf Julius (+), Ron Kuivila... Technically these installations often used the same technology that also underlies the electric lighter or the piezo-electric ignitor on water boilers running on combustible gas. The sound that accompanies the spark is a short and dry spike.

Our own 'Talking Flames' project can be considered to be the first pure digital loudspeaker, the digital to analog conversion taking place in the continuous spark between two slightly radio-active thorium-enriched tungsten electrodes.

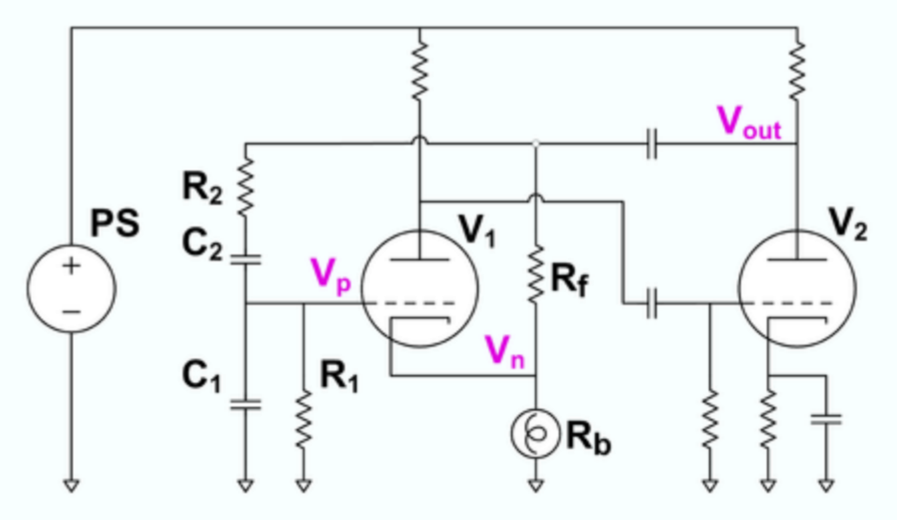
3.- Optical registration of sound vibrations is a technology invented by August Lauste quite some time before the electromechanical recording- and playback devices said to have been invented by Thomas Edison.
Only with the advent of the loudspeaker -an electromagnetic device using a light paper cone to convert vibrations of an electromagnet into vibrations of air - electric and electronic sound generation became practical. The 'Groves Dictionary of Musical Instruments' had a good article by the late Hugh Davies on the early beginnings of electronic sound.

Central in just about any electronic musical instrument capable of producing pitched sounds, is the oscillator. A few different types, each with a history of its own, have to be discriminated here:

1.- The sine wave generator

This type of generator was with certainty one of the first indispensible laboratory tools in any lab working in the realms electronics, acoustics and physics. It had a large dial to set and sweep the frequency and a callibrated attenuator to set the amplitude over a very wide range. The output on the professional lab types was floating with respect to earth. 'Balanced' in the electronics jargon. The impedance was low (generally in the order of 50 Ohms, but vacuum tube equipment often had 600 Ohm). When the first studios for electronic music started of, in the early fifties that was, we find them on the shelves. Not by accident nor chance, but very much inspired by a theory -one can also call it a belief- after which just about any sound can be thought of as being composed of a juxtaposition of an infinite series of sinusoidal components of varied (and decreasing) amplitude. In this theory the sinewave became the 'atom' in the world of sound and all sound imaginable could be generated by mixing nothing but sine wave components. This theory, mathematically underbuild with the Fourrier theorem on spectral analysis, was found to be very attractive amongst the early serial composers such as Karel Goeyvaerts, Karlheinz Stockhausen, Herbert Eimert, Gottfried-Michael Koenig... This paradigm promissed to offer the composers complete control over just about all aspects of sound. Additive synthesis was the keyword.
The sine wave oscillators used in the early days, were invariably laboratory types. The Hewlett-Packard 'Wide Range Oscillator' was very popular and reputed to be highly reliable. In fact we had two of these in the lab at Logos Foundation, they cary the production date -1954- and still -at the time of this writing- they do work fine... Here it is:

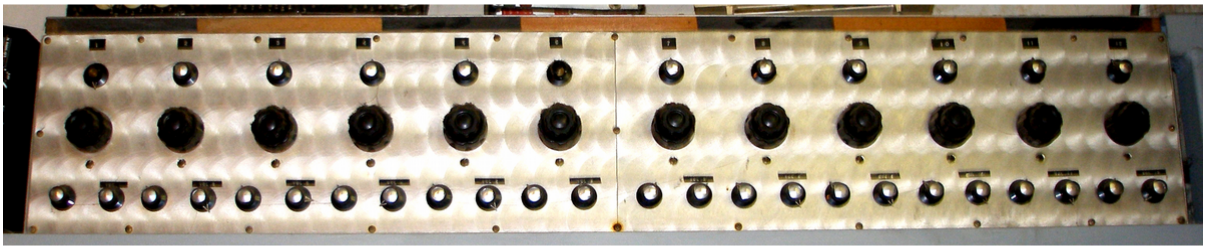
The first model of this generator was produced in 1939 (model HP200A) and the big innovation it brought was the amplitude stabilisation circuit using an incandescent bulb as voltage dependent resistor in the negative feedback loop. Here is the principle circuit as taken from the patent application:



The Wien-bridge components are C1, C2, R1, R2. The idea to use a bulb to stabilize amplitude can be attributed to Larned Meacham in 1937. Ever since -and even today!- this principle has found applications in numerous practical circuits, many of them with high relevance for audio equipment and electronic musical instruments.

Another, more recent and transistorized, popular lab model was this one by Leader:

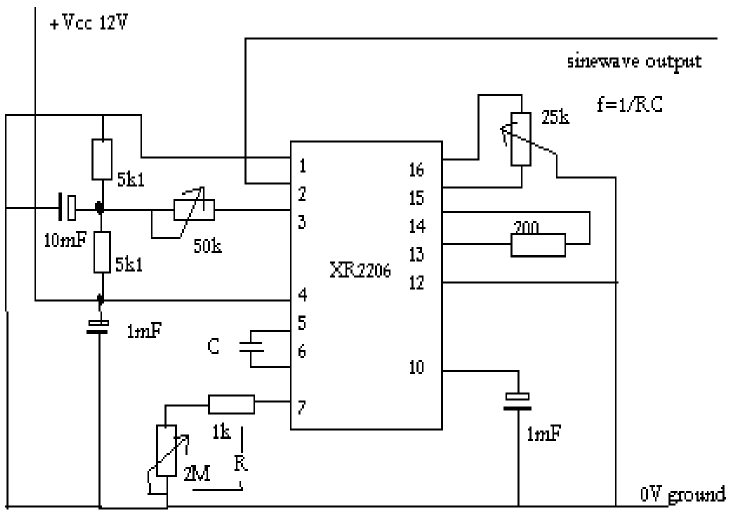
The circuit in use for variable frequency sine wave generators was very often a Wien-bridge, with variable capacitor. This was for instance the case for this assembly of 12 vacuum tube sine wave oscillators that were designed at the IPEM studio in the mid sixties:

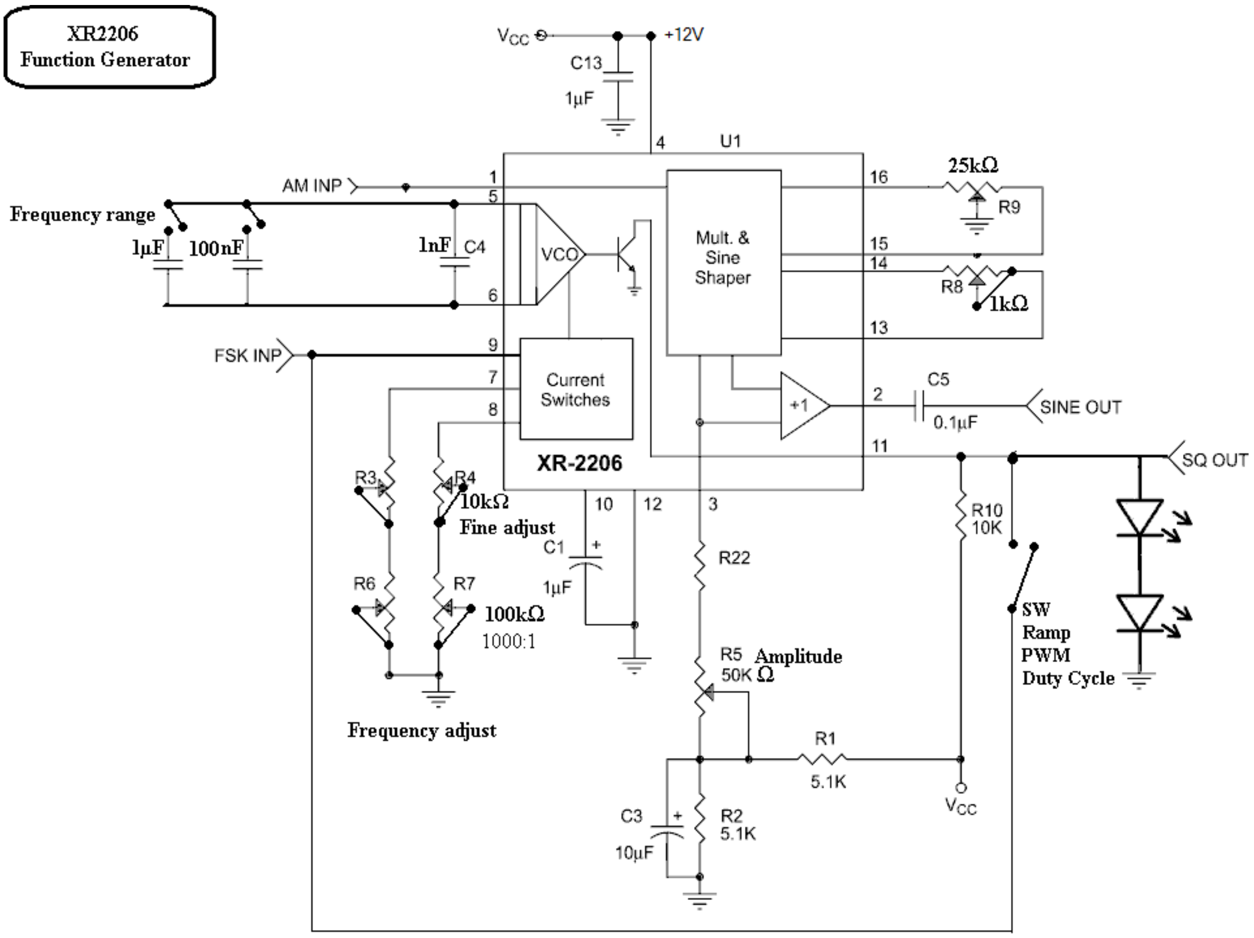


The upper row of knobs are rotary switches to set the frequency range, the second row -using multiturn knobs- are used the set the frequency, the bottom row has a switch to select the output channel and a potentiometer to set the amplitude. The mixer is part of the internal circuit.

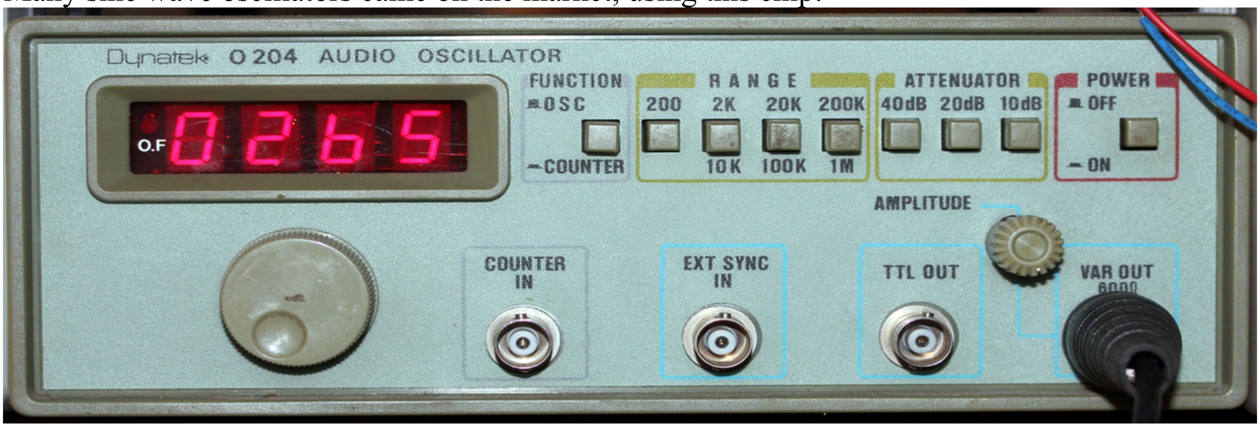
As said above, the Wien-bridge base oscillator was not invented and build until 1938. In early electronic instruments such as the Theremin as well as in the Ondes Martenot, the oscillators are invariably based on heterodyning. The advantage being that it was easy to sweep through the entire range of audible frequencies, without a need to change the range. This worked, but very much at the detriment of precision and stability. On 'normal', Wien-bridge based oscillators, the range on the big dial was a decade (1 :10, with a little extra, for overlap with the next/previous range).

Since the early seventies though, quite a selection of integrated circuits (chips) became available that made it pretty easy to make and design a sine wave generator. The most popular chip -still available at the time of this writing- being the Exar XR2206.



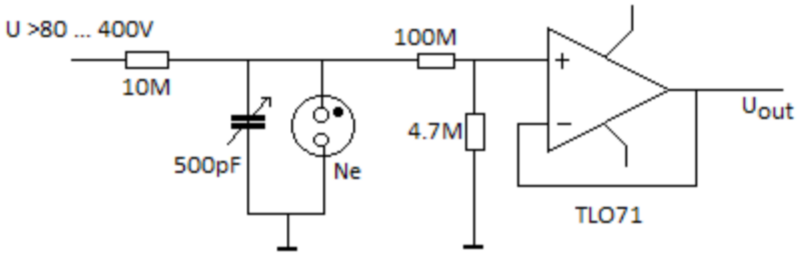


Many sine wave oscillators came on the market, using this chip:



2.- The relaxation oscillator (sawtooth)

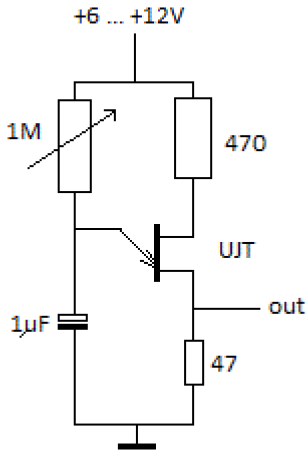
Presumably the simplest circuit one could think off, as it uses just a small neon-filled glowbulb, a resistor and a capacitor. By making the resistor variable, tuning is possible over a large range. The range can be selected, by allowing selection of different capacitors. If the operating voltage is high enough, a variable capacitor (500pF or so) can be used also to continuously vary the frequency. Here is a practical circuit, wherein a FET-input opamp is used to buffer the very high



impedance of the oscillator:

This oscillator type found application in the Philips 'Philicorda', the electronic Hohner accordions and many more brands of musical instruments. The dodekadent, developed by us for the Logos workshop in 1969, also used relaxation oscillators, in this case used as voltage controlled oscillators. The control voltages were between 65 V and 550 V for a range covering many octaves...

In the seventies the discharge-bulb design became less popular as a new component came up: the unijunction transistor. This component could easily be used to design a sawtooth oscillator very



similar to the circuit using a bulb. Here is a circuit:

3.- The multivibrator (square wave)

The multivibrator operates a bit like an oscillating on-off switch: if it switches off, it switches itself on again and vice versa. The speed wherewith it changes on-off or off-on states depends on the value of the two capacitors in the circuit. In the first half of the 20th century the circuit was build using double triode valves, but pure electromechanical versions were made as well. The latter used relays and were applied massively in telephone exchange stations as well as in power supplies for vacuum tube radios in cars (so called 'vibrators'). With the advent of semiconductors and the beginnings of digital technology, the term 'flip-flop' became the common name for the multivibrator circuit, basically a 1-bit memory element in a feedback configuration.

4.- The pulse- or spike generator

This type of generator found its main application as a triggering device for other circuitry. It can be used for generating spikes and bursts on its own, but also to synchronize different oscillators, sequencing devices, rhythm units etc. The circuit mostly used here is the one-shot. A very common integrated circuit to implement it, is based on the well known and ever popular 555 chip.

5.- The noise generator

This type of generator, an essential part of all acoustics labs since the first half of the 20th century, produces white noise. This white noise can be limited in bandwidth to create different kinds of colored noise. Pink noise is a special form of noise, obeying to the 1/f distribution. To change noise characteristics, filters are used and most often one will find these integrated in the generator. The type produced by Bruel-Kjoer, up to the mid seventies, could be encountered in all electronic music studios all over the world. For application in live-electronics, this type of generator was way too bulky (and expensive...) and thus a whole gamut of simple, often self-made, circuits came into existence to generate noise.

Noise generators in chip form became popular in the seventies, but these use a digital approach to generate pseudo-random noise. Indeed, this type of noise after low pass filtering, reveals its digital origin by producing a clearly recognizable always repeating pattern. Circuits based on these chips once were very popular in live electronics.

6.- Top octave generator.

This type of generator does not provide in a variable frequency, but instead generates a full octave of equal tempered very high pitches. These were used in the design and construction of all sorts of electronic organs and other keyboard instruments. Each of the twelve outputs of this generator was fed to a cascade of dividers, such that all pitches in all octaves below the pitch generated could be produced simultaneously. The chips used for this type of generator are since long out of production and there is no alternative for them, apart from redesigning the entire circuit using microprocessors.

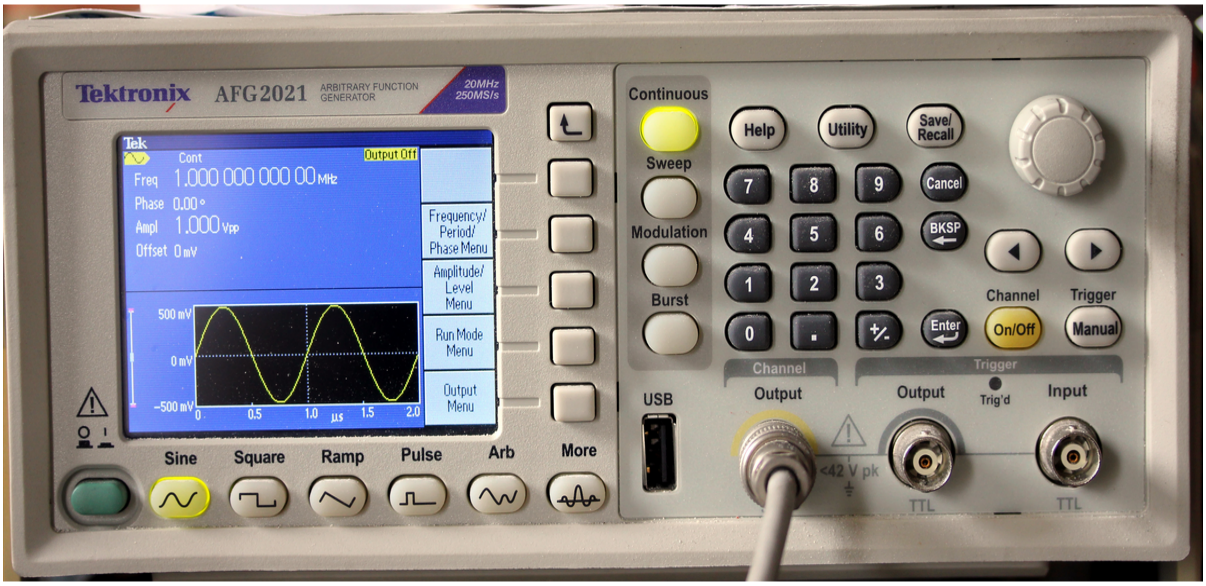
The top octave generator was a great improvement in electronic organ design, as it circumvented the problem of having to tune minimum twelve independent oscillators with regular intervals. Using this generator chip, the tuning of the entire instrument could be fine-tuned by adjusting just a single pot-meter or trim-cap.

Digital synthesis of arbitrary waveshapes

Sine waves, and for that matter in fact any periodic wave-shape, can be generated digitally. The principle is to implement a ring-counter and a DAC circuit. The ring counter sends a list of n values taken from a lookup table to the DAC at a given clock frequency, corresponding to n times the frequency of the signal to be generated. The precision is a function of the number of steps: with 256 steps we get 8-bit resolution.

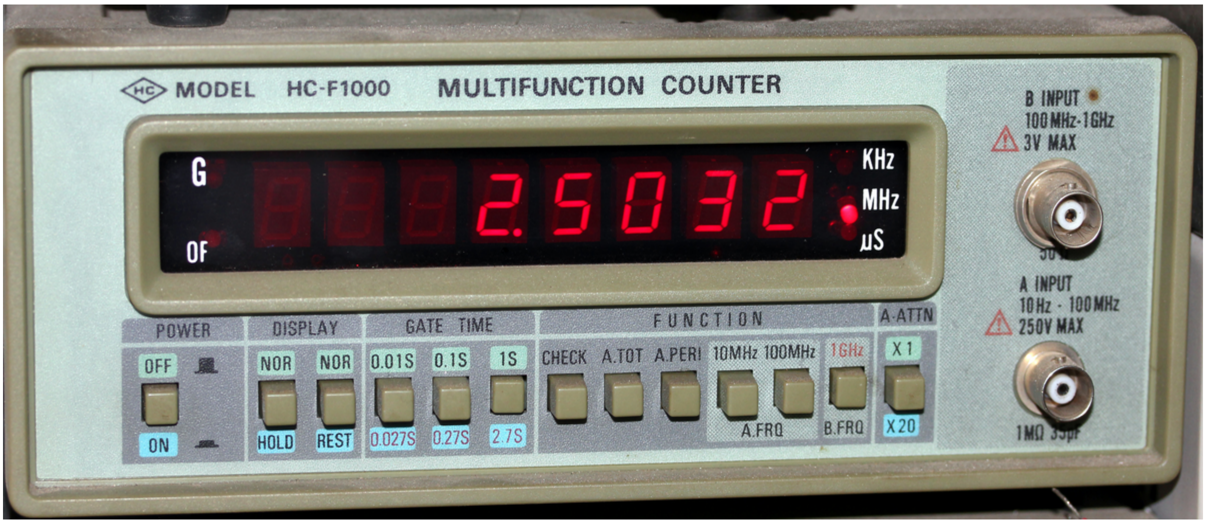
It is of interest here, to draw attention to the fact that this approach to sound synthesis is technically almost identical to the concept of a sequencer. The difference is only in the speed of the clock used. On a philosophical level, the idea that audible sounds can be brought down in frequency until they become rhythms , has been a source of inspiration for many composers. Karlheinz Stockhausen, used the concept both musically and philosophically in pieces such as 'Hymnen'.

Here is a picture of a laboratory arbitrary function generator made my Tektronix, our favorite of all times, although it cannot be used for live electronic performances: it does not have an easy to use dial knob for frequency and it has a fan, making noise...



Use as a musical instrument

The use of wave generators in electronic music studios in the fifties and sixties set aside, (sine)wave oscillators have been used very often in musical performances, on stage that is. The knobs and dials wherewith the frequency was to be controlled often didn't allow precise enough setting of the pitches as prescribed in the composers scores. The answer to that problem was the addition of a digital frequency counter to monitor the output frequency. With such a counter, the frequency became readable with high precision. However, the problem was that these counters had a rather slow response time. One second for a measurement precision of 1 Hz is normal.



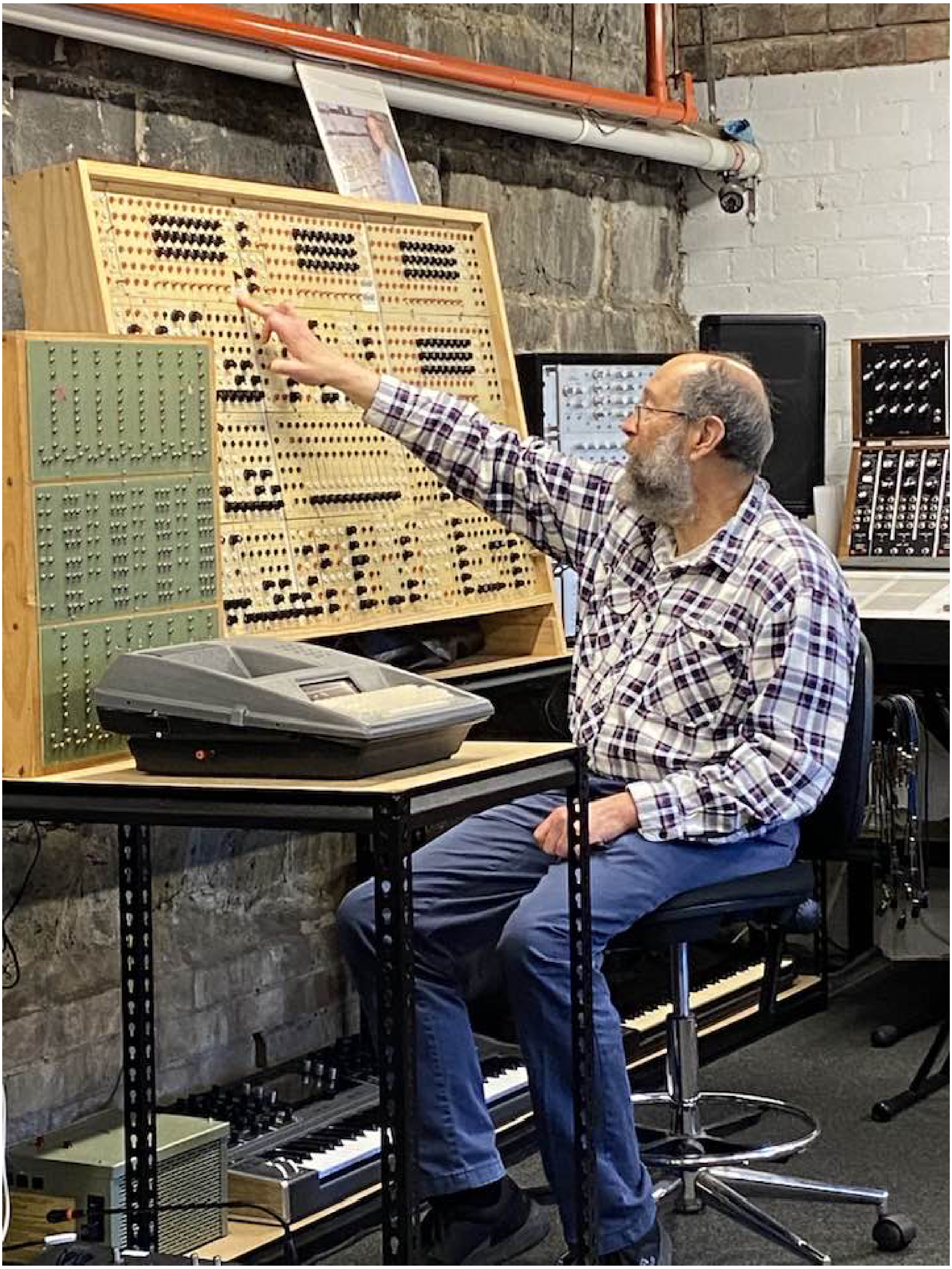
An alternative that became available only since the 21th century, is to use a digital tuner device such as made by Korg, Casio and other brands. There are now even a wide range of apps for smart phones available with just that function.

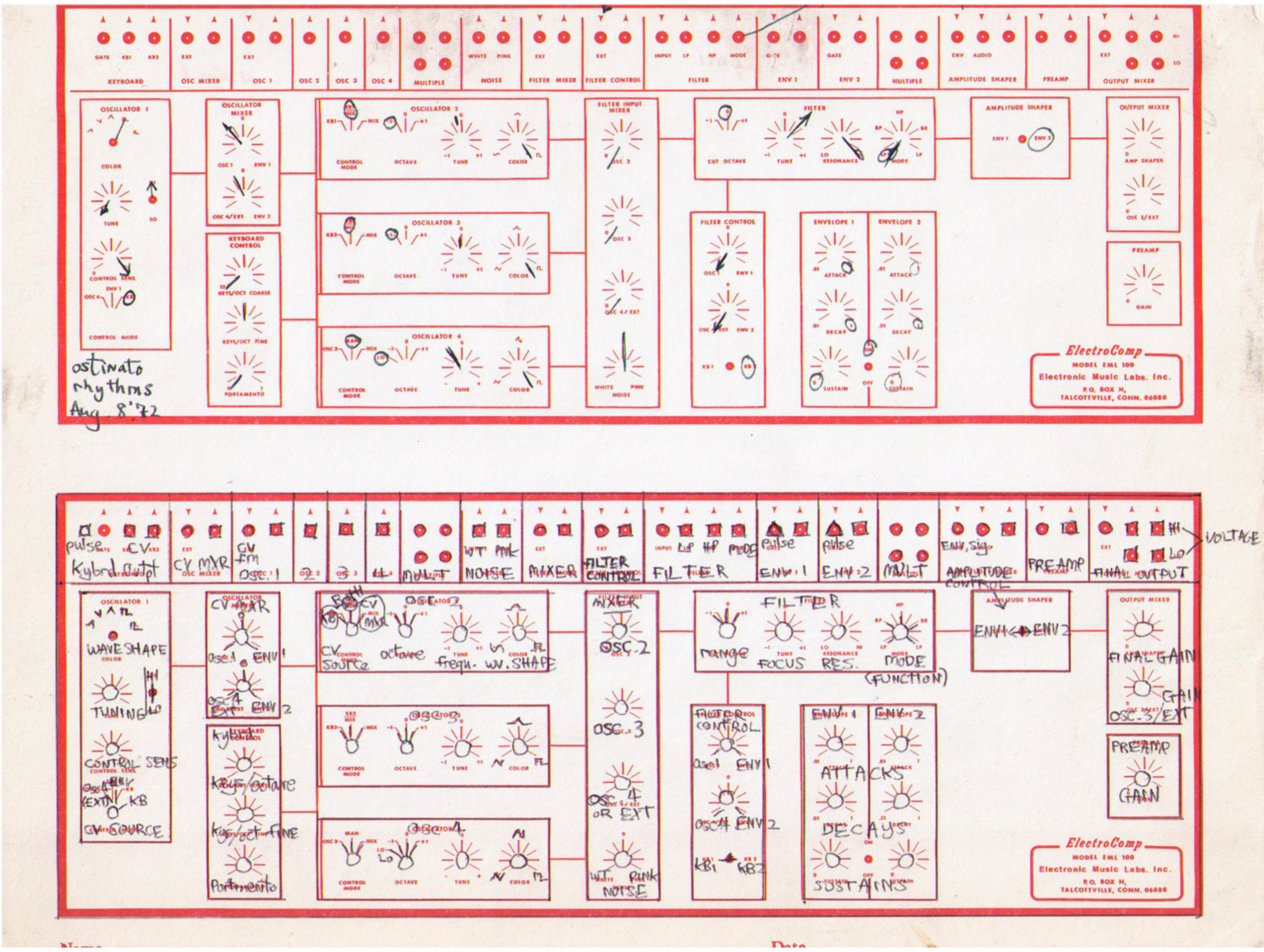
3. Synthesizers

In many respects, analog synthesizers can be considered as a specialized form of an analog computer. As a matter of fact, electronic synthesizers -unrelated to music- are in the first place analog computers used to synthesize arbitrary waveforms. They were and still are used in radio technology, astronomy, radar equipment and ballistic weaponry.
Basically analog computers just like music synthesizers consist of merely three basic electronic components or subsystems: operational amplifiers (used for integration, differentiation, summing, subtracting, amplification...), multipliers and exponential or other non-linear functions (comparators, limiters, exponential functions, lin-log converters...). As far as their musical applications are concerned, we treated the multipliers in our chapter on ring modulators. They are in fact technically identical. Operational amplifiers are very fundamental components in all analog electronic circuitry and there is plenty of literature available on them. The basics are actually pretty simple and not a lot of math is required to get them to work for you. Readers will notice that we encountered them also in our chapters on tape recorders, preamplifier's, contact microphones, filters etc...
Thus the only very specific issue we have to deal with here has to do with the logarithmic nature of our hearing apparatus. Both our hearing of loudness levels as our perception of pitch are fundamentally logarithmic. Hence the need for circuits with similar logarithmic characteristics if we want to design and use electronic synthesizers as musical instruments.
In the first generations of music synthesizers, the required log characteristic was often obtained by using components such as logarithmic potentiometers for volume and pitch control. With the advent of voltage controlled music synthesizers, that made forms of 'programming' possible, (since the late sixties that is), came the need to develop standards for steering the components. The universal standard here became the 1V/octave norm. Thus for a pitch span of 10 octaves, a control range of 0 to 10 Volts reached out. However, within every octave step of 1V, the curve had to be strictly logarithmic as well and so 'equal temperament' was required and implemented according to the twelfth root of two increments for chromatic steps. Achieving this was always a challenge to engineers and designers as the degree of precision and circuit stability required was very high leading to expensive components, precise temperature control, precision capacitors and resistors...
The first generations of synthesizers were modules without keyboards. They used patch cords and/or matrices for programming. Popular machines were the Synket, EMS VCS3 ('Putney'), the MiniMOOG, the Korg, ARP, Buchla, Serge instruments, Elektuur Formant.

Keyboard synthesizers were introduced in the seventies and caused their widespread use in pop- and commercial musics. The first types -such as the Korg model shown on the picture above- were monophonic. In the early eighties, MIDI-control became the universal way for controlling synthesizers and with it, the synthesizers turned into pure digital machines. The Yamaha DX7 (1983) was a great commercial success and used FM synthesis. By the end of the eigties, we saw sampling mechines taking over (Fairlight -a forerunner- in 1979, later Akai-SW (1985), EMU (1981) and many more). By the end of the millenium, there hardware started to vanish as the gradually were replaced with pure software based implementations. In the last years though, we notice a certain revival of old analog hardware with product series such as Doepfer a.o.

The term 'patch', still in general usage, is a leftover from the synthesizer epoch that actual wires with jacks on both ends were used. (Some brands used even banana plugs).
On the picture below, the Australian/American composer Warren Burt working on an old synthesizer:

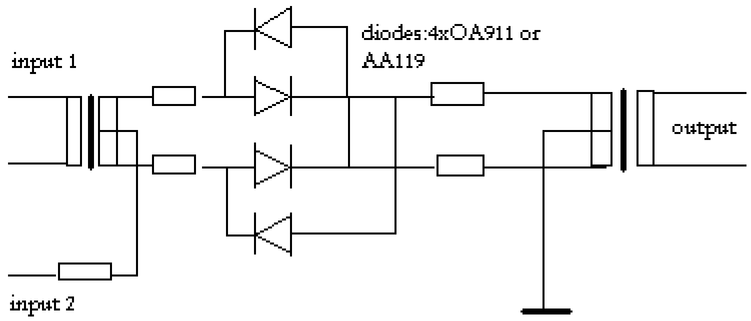




Music using synthesizers in general used 'patch-sheets' as score material. These sheets are extremely machine specific and -if the original machine is no longer available- rarely reproducible or translatable to another machine.

4. The Ringmodulator

Ring modulators are electronic circuits stemming from early telephony. They are at the essence of the possibility to send and receive many simultaneous telephone conversations through one and the same couple of wires. Soon enough the circuit has found abundant applications throughout 20th century music production and performance. They are fundamental also to the working of radio receivers and emitters, both for AM and FM. The historical passive ring modulator circuit looks like this:



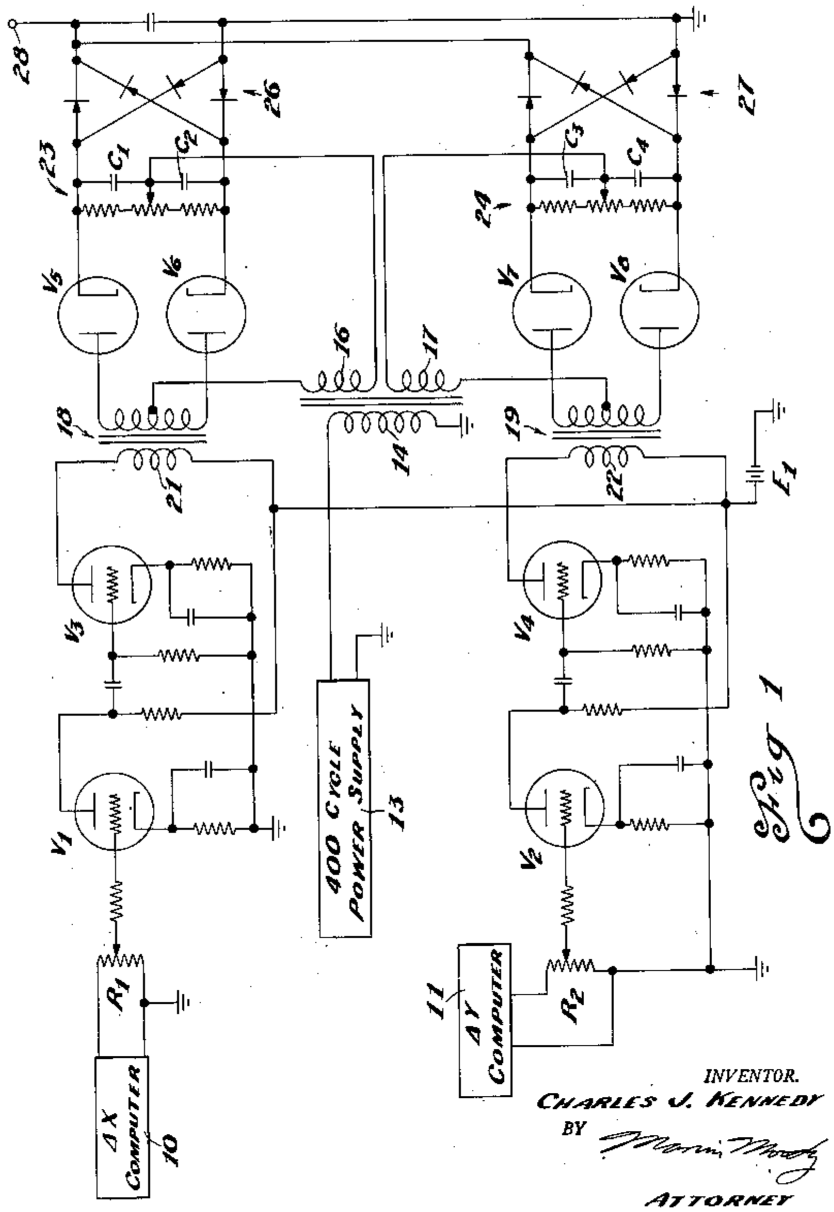
[ref.69.10]

There are two inputs and a single output. The two signals get multiplied in the germanium diode-ring configuration between the two transformers. Input signals for this circuit must be kept smaller than 25 mV. That's why there are four resistors in the circuit. Their value must be calculated such that they are larger then the conducting resistance of the diodes used under voltage conditions specified. The diodes must be matched within fractions of a percent. The ideal transformers for this circuit must be of the toroidal type and extremely well balanced. Hard to find for audio applications but if you are lucky you can every so often recycle them from old professional audio equipment stemming from physics labs, analog computers or electronic music studios. It should be noted, that the very first circuits, used vacuum tube diodes instead of solid state germanium of silicon diodes as drawn here. Amongst electronics and music afficionados on the internet, transformer based circuits as well as designs using vacuum tubes, still seem to have followers, generally synth players and e-guitarists with a popular music background. The ringmodulator still seems to have quite some popularity there. The number of forums we could find is pretty elevated. Things such as the kind of diodes to be used (germanium, vacuum tube, Shottky, XOR-circuits) lead to ongoing lengthy discussions. Here are some historical documents taken from the documentation available at the US patent office:

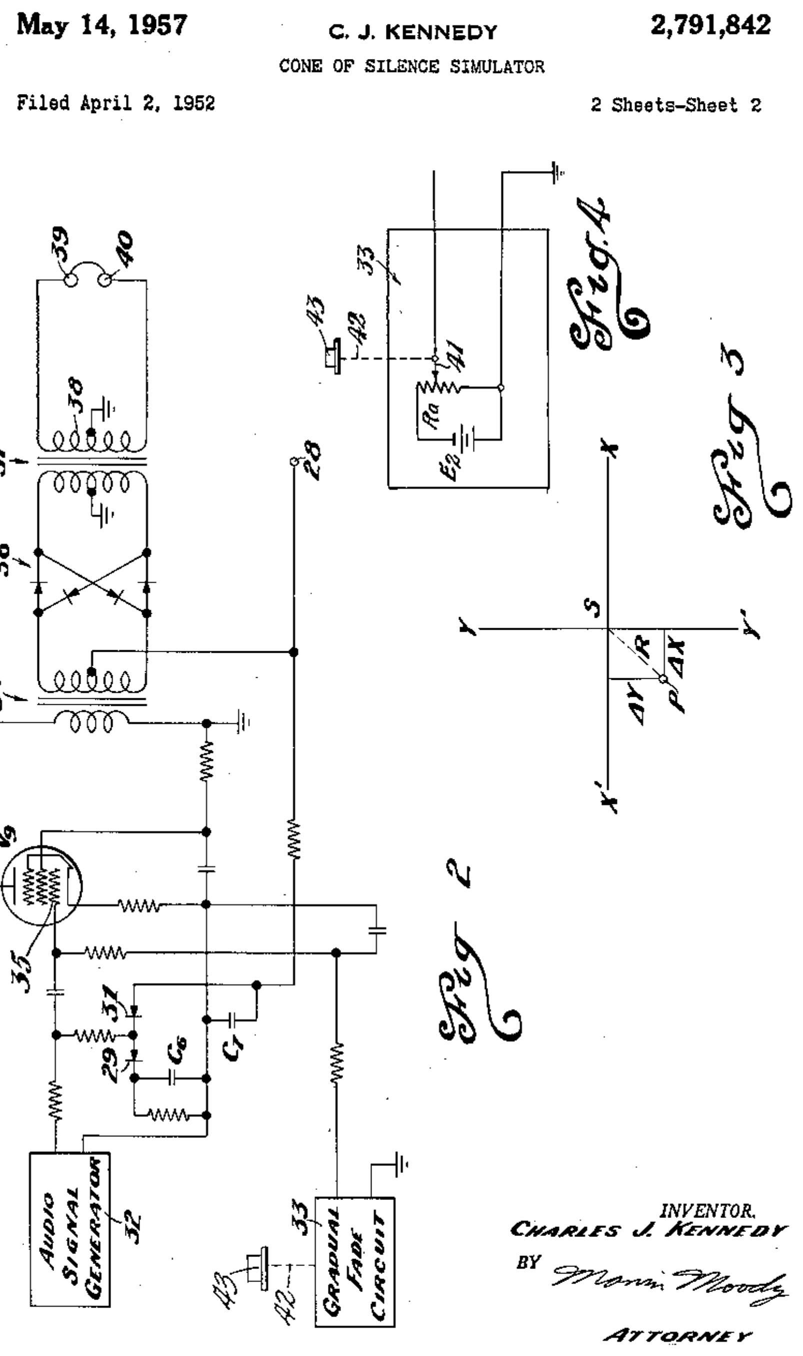
May 14, 1957 C. J. KENNEDY 2,791,842

CONE OF SILENCE SIMULATOR

Filed April 2, 1952 2 Sheets-Sheet 1



INVENTOR.
CHARLES J. KENNEDY
BY Morris Moody
ATTORNEY



Ringmodulators of all kinds can readily be found on the market. The Moog model has been quite popular for a long time:

It's usefullness for the performance of new music scores in general is not evident, as it is designed with a built-in LFO and carrier frequency generator. The foot-switch also, reveals the origin of the device as a guitar effect pedal. For most scored compositions by 'modernist' composers, this model will be not very usefull and getting or building a ringmodulator will be of much greater help.
On a technical level, the classic ringmodulator as shown is a purely analog circuit and its functioning can be analysed as follows.

- Two input signals are required. The output signal contains all sum and difference tones between both input signals, the input signals themselves disappearing from the output signal. From a mathematical point of view, the transfer function seen in the time-domain is:

$$\mathrm{Uout} = (\mathrm{Uin\_x} * \mathrm{Uin\_y}) / k$$

How this connects to the statement about resulting sum and difference tones is a bit more involved. The circuit behaves as a multiplier in the time-domain. Let's suppose the input signals are periodic functions such as:

$$\mathrm{Uin\_x} = \mathrm{Upx} * \mathrm{COS}((\mathrm{wx} + \mathrm{phi\_x}).t)$$

$$\mathrm{Uin\_y} = \mathrm{Upy} * \mathrm{COS}((\mathrm{wy} + \mathrm{phi\_y}).t)$$

wherein
$wx = 2 * Pi * fx$

and

wy = 2 * Pi * fy

Thus fx en fy represent the respective frequencies of the input signals and where phi_x, phi_y represent the phase-angles of these signals. Upx and Upy are the peak amplitudes of both signals.

Further development of this product leads to:

Uout = Upx*Upy * {COS[(wx-wy)* t - ph] + COS[(wx+wy)*t + ph] } * COS(wx+ph* t)

= (Upx*Upy/2) * {COS[wy*t + ph] + COS[-wy* t - ph] + COS[(2*wx +/- wy)* t +/- ph)

(suppose ph = phi_x - phi_y)

Expressed in words, this means that at the output we are getting nothing but the sum and difference tones between the input signals. The transfer function becomes even intuitive of we imagine both input signals being the same. In this case we would get:

Ui = Um * SIN(wx*t) input signal

[Um * SIN(wx*t)] * [Um * SIN(wx*t)] = (Um * SIN(wx*t))^2

If we now apply a standard rule in goniometry we derive that:

[Um * SIN(wz*t)]^2 = ((Um^2) / 2)) * [1- COS(2*wx*t)]

Or, expressed in words, the output of the circuit offers us a periodic signal with twice the frequency of the input signal. Simply said, we just made an octave doubler. For those amongst our readers wanting to dig further into the internal guts of such modulators and demodulators, we refer to the exhaustive technical literature available on the subject. The printed data books published by Analog Devices and Burr-Brown are invaluable sources of information. A thorough understanding of these devices is essential for a proper understanding of FM-synthesis as it has been fundamental for the development of electronic synthesizers since about 1970. FM-synthesis is entirely based on the multiplication of wave forms.

For pure sine wave input signals, the characteristics of the resulting output signal can be predicted easily, as shown in a few examples:

•suppose U1= 440 Hz (la) and U2= 660 Hz (mi)

gives: Uout+=1100Hz (do#)

as well as Uout-= 220Hz (la)

A pure fifth interval thus undergoes a transformation to the interval of an octave plus a natural major third.

•suppose:. U1 = 440 Hz (la)

U2 = 880 Hz (la')

gives: Uout+=1320Hz (mi)

as well as: Uout-= 440Hz (la)

In this example, the octave between the input signals, gives us an octave and a pure fifth at the output

•vb. U1 = 440Hz (la)

U2 = 466Hz (sib)

geeft: Uout+= 906Hz (la+)

+ Uout-= 26Hz (la-)

In this case we get at the output an A raised a quartertone together with a very low A, a quartertone down. The latter however will be perceived as a fast tremolo. As soon as we try to feed the ringmodulator with more complex waveforms such as those stemming from acoustic musical instruments, the results can get very complex. This is caused by the fact that the multiplier applies its transformation to the entire spectrum found in the input signals. If we feed at least one of the inputs with spikes or pulses, we can easily obtain sounds refering to metal percussion instruments such as gongs and bells. Needless to say that this has fascinated generations of composers and musicians using technology.
A special application of the ringmodulator is the octave doubler, obtained by shorting both inputs together and feeding it with a single periodic signal, as demonstrated earlier mathematically. Also, we can feed one of the inputs with a wide band noise signal and the other input with a sine wave. At the output colored noise centered around the frequency of the sine wave will be obtained. We can also use the ringmodulator as a VCA (voltage controlled amplifier). It is enough to use one of the inputs as very low frequency input or even feed it with ADSR signals. As a matter of fact, the VCA is nothing but a specialised application of the fundamental ringmodulator circuit. Technically a VCA is a one-quadrant multiplier.
Functionally, we encounter the circuit in many very old as well as new electronic instruments: the Theremin as well as the theremin derived Ondes Martenot

If we use the ringmodulator to multiply two signals in the ultrasonic audio range and if their difference tone falls in the audio band, then we can use it to detect just this difference tone as the co-existent sum tone a fortiori will be out of the audio range.

As soon as electronic music studios started rising up in the beginning of the fifties of the 20th century, the ringmodulator became a standard component for the production of electronic music. Early composers that made extensive use of them are Karlheinz Stockhausen (Telemusik, Mantra, Hymnen, Mixtur...), Wladimir Ussachewsky, Gordon Mumma, John Cage, Allan Strange, Alvin Curran... The analog electronic synthesizers as they were build since the seventies, almost all contained at least one patch-able ringmodulator (Synket, Putney of VCS3, EMS, Korg, R.Moog, ARP, D.Buchla, Synthelog, Serge etc).

Even in some orchestral compositions one may encounter requests for ringmodulators, for instance the Flemish composer Luc Brewaeys uses them in 'Trajet' and 'Due Cose'. The ringmodulator circuits for these performances were designed and made by us, together with some software to generate the signals to be used as modulators for the orchestra sounds on the inputs. At the time these pieces were written, it was already no longer common usage to use spare sinewave generators to this purpose. MIDI-controlled modular synths could perfectly replace the sinewave generators. Nowadays, the entire setup can be replaced with a simple computer program, a 'patch'.

Some pieces where the score calls for ring-modulators:

Toshi Ichianagi, 'Appearance', for 3 instruments, 2 oscillators and 2 ringmodulators.

Published in Source magazine, issue nr.2,

Although the original ringmodulator -or with its more technical name the double balanced modulator/demodulator- was build using transformers or coils, nowadays we would invariably make use of specialised analog chips. Of course, digital implementations are also possible and have become even commonplace as there is no material cost involved.

Usefull analog integrated circuits are:

LM1496 of LM1596 (Philips, National Semiconductor e.a.)

> HA2546, HA2547, ICL8013 (Harris)
>
> AD532, AD534, AD539, AD632, AD633, AD743 (Analog Devices)
>
> MPY534 (Burr Brown)

Good quality chips (with 0.5% precision) combined with a wide frequency bandwidth are not cheap. The MPY534 used to go for some 150 Euro. If you can live with 2% precision, you can go for the AD633 and expect to pay ca. 20 Euro.

Ringmodulators seen from the perspective of historical performance practice, make us think about the differences between period-circuits versus modern (software based) alternatives.

Other than in the case of tape-recorders, here the physical appearence on stage of the ringmodulator has no importance at all. It can take just about any shape. Most often it's just a metal box with some audio connectors and at the most, a few potentiometers to set levels. The musical result though may turn out very different, for one or more of the following reasons:

- analog ringmodulators 'suffer' a lot from leakage: that is, feed-through of the original inputs to the output. Most of the time, there is a noticable difference in leak-through, between the one input and the other. That's why inputs are often labeled 'signal' and 'carrier', or anything similar to make the distinction.

- analog ringmodulators have a very limited signal to noise ratio

- analog ringmodulators may distort the signal in quite some other ways than what is to be expected from the multiplication process on itself. The dynamic range of the device is inherently limited. The reason is simple: if we multiply two signals with their maximum allowable level, say 10, then the amplitude range of the output ought to cope with the product, say 100 in our example. This inherent -20dB reduction can be compensated for, but always at the detriment of signal-noise ration. This remark, by the way, also applies in full to the digital implementation.

However in a digital implementation, we can often anticipate and provide in a much higher bit-resolution at least for the input and modulator signals.

Wherever composers wanted to get rid of feed-through, they most often recursed to gating circuits, designed such as to oppress the output whenever no signal was present on both inputs together.

Bibliographical references:

- ORTON, Richard "Ring Modulator" in: Groves Dictionary of Musical Instruments, dl3.,p.249 (1984)

- BODE, Harald "The Multiplier type Ringmodulator in: Electronic Music Review,nr.1, 1967

- STRANGE, Allen "Electronic Music" ed.:New York, 1972

- RAES, Godfried-Willem "Een Onzichtbaar Muziekinstrument" , Doktorale dissertatie R.U.G., Gent 1993.

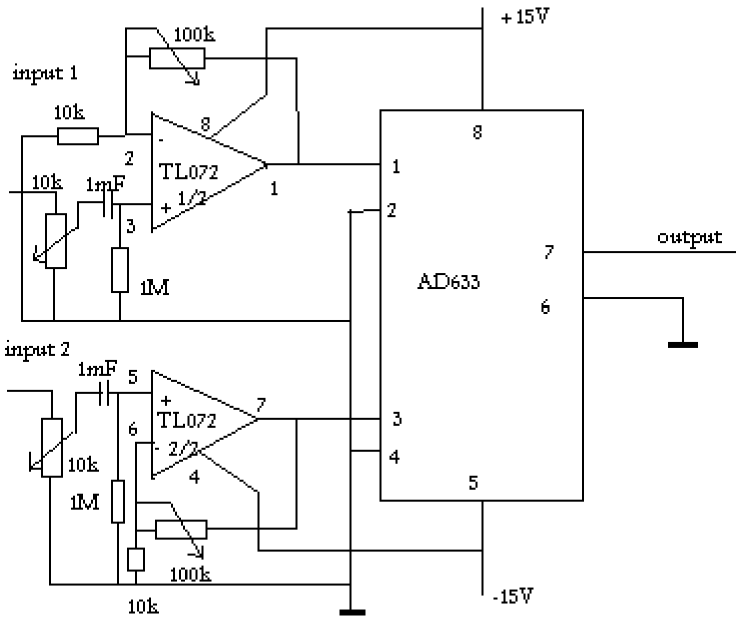
Modern practical circuits for analog ring modulators:

LM1496 circuit

zie: Elektuur, Formant-boek, deel 1 & 2


TABACOS PUROS
WILLEM II
CIGARS
WILLEM II
WILDE HAVANA
100% Tobacco
20 CIGARS MADE IN HOLLAND
MC1496P
L8623
WILDE HAVANA

AD633 circuit with balanced power supply:



[ref.:93-04]

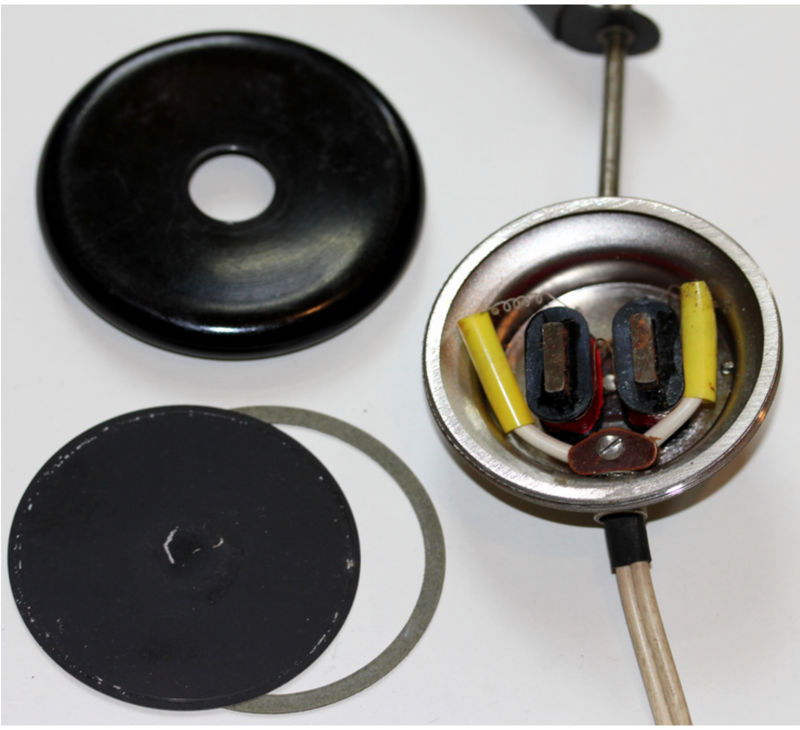
Here only two chips are used. The first one is a cheap dual operational amplifier. This serves to match impedances to the AD 633 multiplier chip. The 1uF blocking capacitor on the inputs is perfectly o.k. for audio use, but if the use as a VCA is anticipated, these capacitors should be replaced with wire bridges. This will of course lead to a potentially strong DC offset on the output. The input signals for this circuit should not go beyond 10Vpp. As the dynamic range of the output can be very high, it is often advisable to let the circuit be followed by a limiter or even a compressor circuit.

5. Electromagnetic pickups and transducers

Magnets and electromagnets are at the core of an awful lot of devices we encounter daily around us. Obviously they are essential in the construction of almost all kinds of motors, but also in audio devices they are indispensable. Loudspeakers and microphones are likely the first devices that will come up in our mind. But, also they are an essential part in musical instruments: the pickups in electric guitars for instance, but also instruments such as the Fender-Rhodes piano, the Hohner pianet, the e-bow, the mellotron, pipe organs, the neo-Bechstein piano as well as nearly all musical automatons build after world war 2.

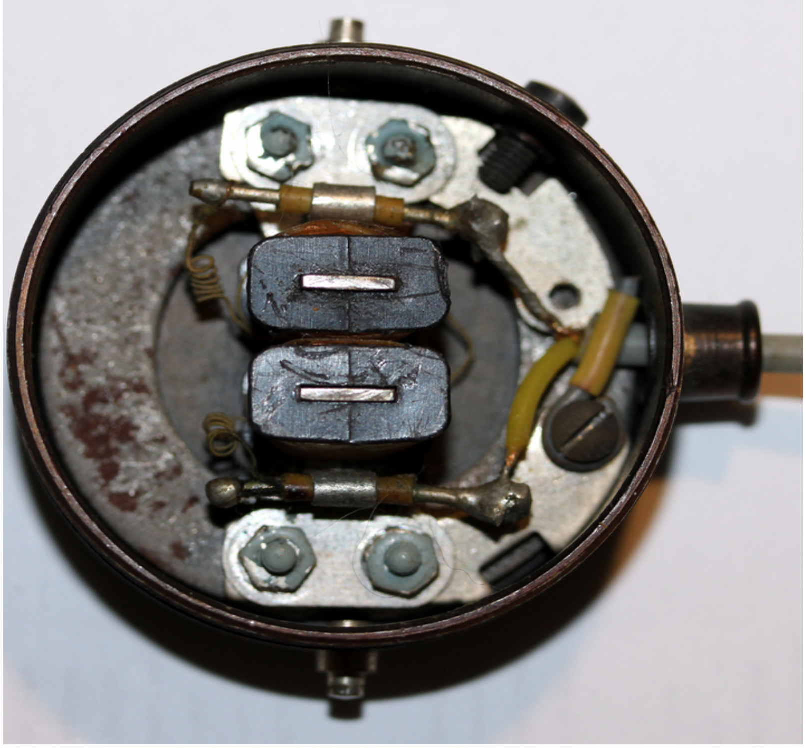
The invention of the horse shoe (electro)magnet is attributed to William Sturgeon (1825) . He found that the field strength of a magnet could be much improved by bringing the poles near to each other. The horse shoe magnet made telephony possible, not only for its use in the generator used to make a call, but also for the construction of the listening part of the telephone horn. In order to become an electromagnetic transducer, the only thing required was to wind a coil around the legs of the magnet. By sending a small electric current through the coil, the magnetic strength could be modulated and hence the force exserted to a ferromagnetic object such as a plate-membrane in the immediate neighbourhood. But also for the early development of radio, such transducers did prove to be indispensable. Here are legacy headphones made after this early recipe: 

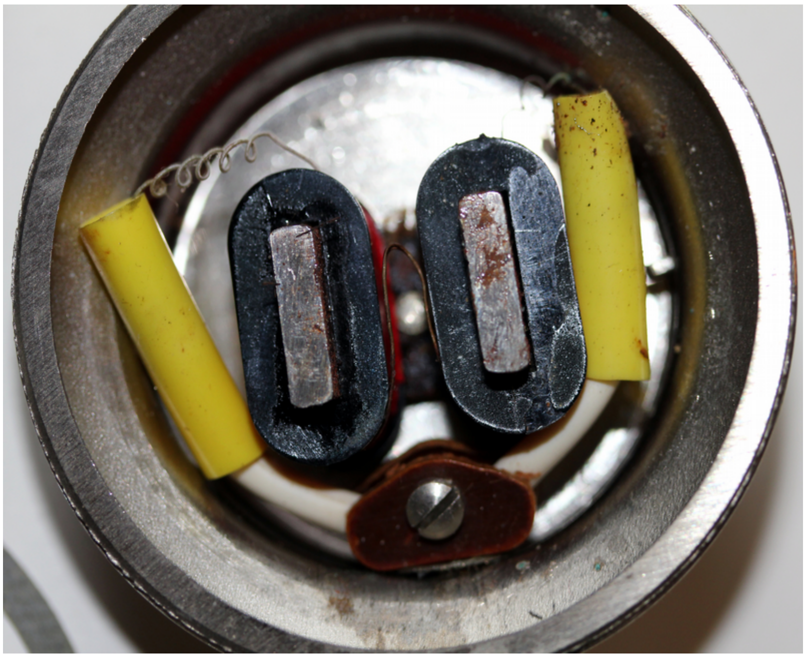
Disassembled they look like this:



The earpiece is made of Bakelite and screws on the holder. The membrane is made of thin carbon steel and sits in front of the U-shaped coil assembly with a gasket in thin cardboard. This gasket determines the dynamic range to quite a large extend. If it is too thin, the membrane may make contact with the coil and cause terrible distortion on high input signals. If it is too thick, sensitivity and loudness will suffer.

On the following pictures - three different specimens - we see the U-shaped magnet, the steel membrane that in normal operation is just in front of it, removed.

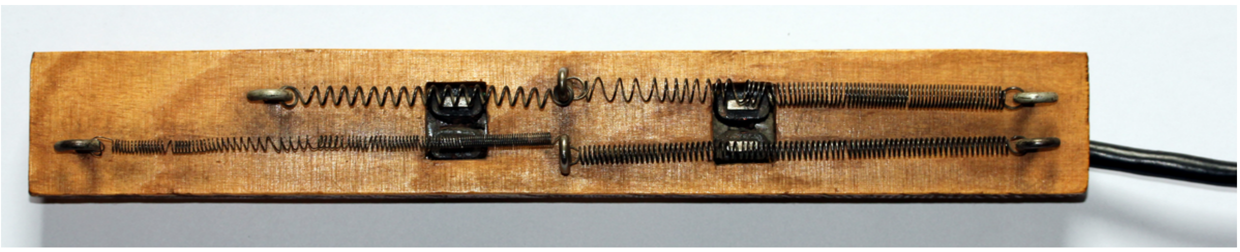




The first two types use 'external magnetisation', hereby the semicircular metal part is a permanent magnet coupled to the L-shaped iron cores holding the coils, The third type uses a permanent U-shaped magnet holding the coils directly. All types shown here have an impedance of 2000 Ohms. Nevertheless, the last shown type is a lot more sensitive than the first couple.

Although made and designed for the earpiece part, the device is perfectly reversible and thus works as well as a pickup. For such use, the membrane is usually removed and replaced with the ferromagnetic vibrating object one wants to pick up. In this shape we find them as transducers in many instruments and lots of composers and performers in the world of experimental musics, have used them. Most self-made instruments by Hugh Davies, make use of such pickups, invariably recycled from old military headsets. Also the German composer and performer Wolf Dieter Truestedt uses them for the construction of his wind harps.

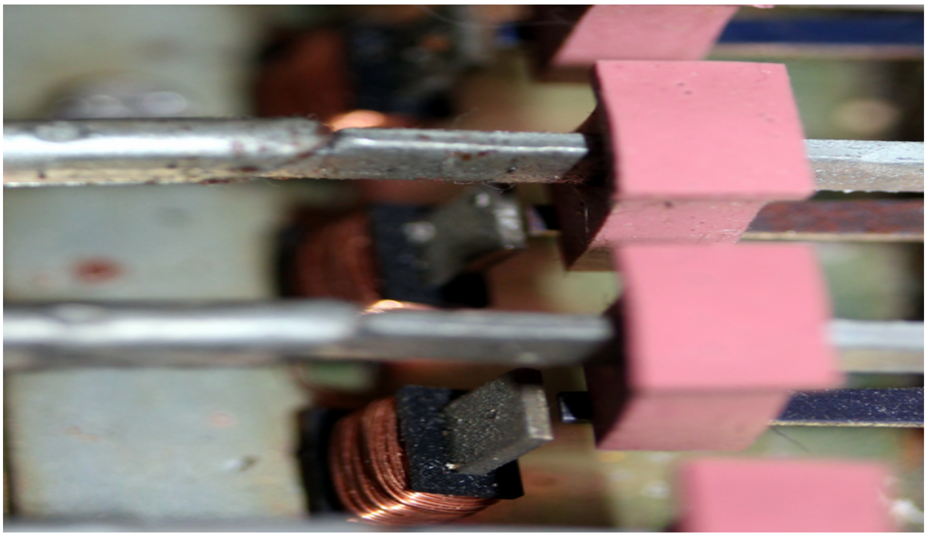
Here is a picture of one of Hugh Davies' original springboards, made in 1976:



If used as a pickup for vibrations in and of ferromagnetic materials (strings, springs, steel plates, tin cans...) it is mandatory that the U-shaped core of the pickup forms a permanent magnet. The problem that invariably shows up when using period transducers, is that their magnetic properties tend to be only a fraction in strength of what they were when new. In order to prevent demagnetization of horse shoe (and in fact this applies to all magnets) magnets, one must store them with a magnetic shunt bar between the poles. Of course none of the left over period transducers were stored that way and thus, you can be pretty certain that demagnetization has occurred. In theory it ought to be possible to repair the permanent magnet by applying a DC current of a polarity reinforcing magnetism for a substantial period of time. The problem is that often the wire gauge of the coils is too small to allow a substantial current to flow. The current to be applied ought to be such that any further increase does no longer lead to an increase in magnetic strength. This strength can easily be estimated using a magnetic compass at some distance from the pickup. If such a strategy can not be followed - generally because the coils get too hot - one can consider to apply a modern neodymium magnet to the existing core. This trick, by the way, also helps to restore old pickups that lost sensitivity on electric guitars.

Common impedance's for these transducers vary between ca. 4 kOhm and some 24 Ohms. The input impedance of the preamplifiers or mixer inputs to match these transducers must be ten to twenty times the impedance of the transducers. If the input of the preamp is too low, high frequency reproduction will suffer. This is because the transducers are inductive devices wherefore the impedance rises with frequency : Xl= 2.Pi.f.L. The exact opposite of piezoelectric devices. Hugh Davies (1943 - 2005) always used them in combination with his (battery powered) Uher mixer, having 47kOhm input impedance.

Other mixers that were very popular (and cheap...) in the early seventies were the Eagle mixing boards using slider faders instead of the rotary potentiometers in the oldest models.

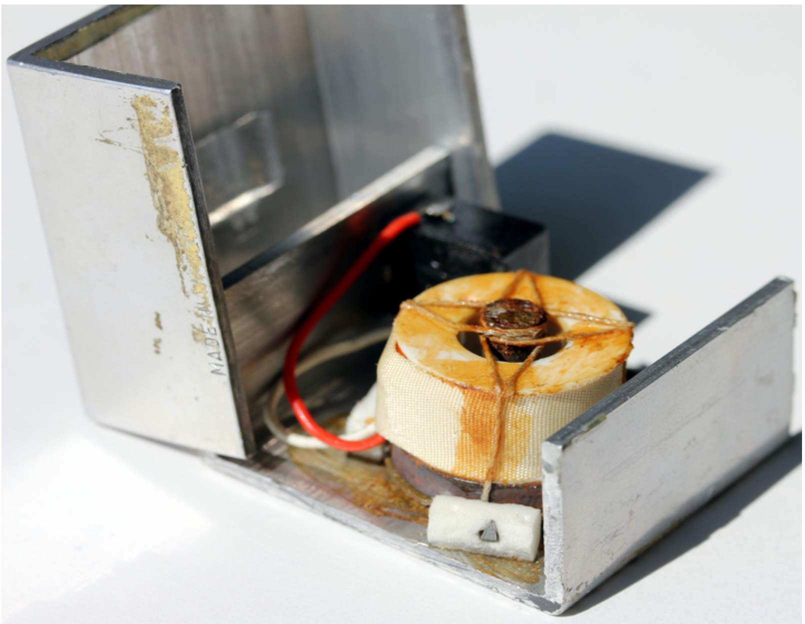
The electromagnetic pick-ups used in commercial instruments such as the Fender Rhodes piano and the Hohner pianet where obviously not recycled parts, but custom made transducers. Their construction nevertheless is quite similar to the devices described above. Here is a picture of the transducers used in the Hohner pianet: 

It is worth noting that all 55 transducer coils are simply connected in series. The use of the term 'connected in series' may even be an overstatement in this case, as in fact the entire transducer assembly is wound using a single uninterrupted copper wire. The mounting plate under all the coils, is a strong permanent magnet.

Moving coil transducers

The basic construction of such contact microphones is a fixed permanent magnet and a coil that is suspended around the magnet and free to move. Such constructions were on the market between 1975 and 1985 and advocated for use on the double bass, the cello and the piano. The moving part being light, the transducers works on the principle of inertia. They are very sensitive to the direction of the vibration and their sensitivity is at maximum for movement and vibration perpendicular to the stave magnet. Here is a popular type - we bought it in 1974 - made by FreeTone:

Opening the aluminum case reveals the utmost simple construction and the moving coil:



The screening here is very good and hence hum is reduced to a bare minimum. A particular advantage of this type of transducer is their behaviour at overload: they do not clip nor saturate but rather output a compressed signal on exposure to very high excitation levels. This property makes them very suitable as pickups for heavily vibrating objects. We have used them on carillon bells, where the use of piezo disks was impossible because the vibrations were way too strong. In fact, its pretty easy to make them: one can use small (<50mm) loudspeakers with impedance's between 16 and 50 Ohms, on which the entire paper cone should be cut away such as to only leave the moving voice coil. A sharp razor blade must be used and one has to be very careful not to cut through the tiny copper wires leading to the voice coil. The paper cone must be cut away to make the transducer insensitive to acoustic sound. They can be used just like normal dynamic low impedance microphones and connected as balanced sources to standard mixer inputs with XLR connectors. A disadvantage of this type of transducer is that their mass functions as a damper on the source of vibration. Thus their use will be limited to those cases where the proportions of the vibrating body to the mass of the transducer is large. Hence their indication for the double bass and the piano.

It is important to distinguish two distinct electromagnetic transducer types: first, those using a permanent magnet whereupon a coil is wound (this is the case for all transducers described above but also for loudspeakers) and those designed to pick up changes in magnetic fields. In this category we meet the tape recorder heads as well as the telephone pickup coils that were popular with old telephone sets, to pick up or amplify telephone conversations. Remind you it used to be strictly illegal to make any galvanic connection to the public telephone network! Also the earliest telephone modems -acoustic modems they were often called- made use of such technology for the same reason.

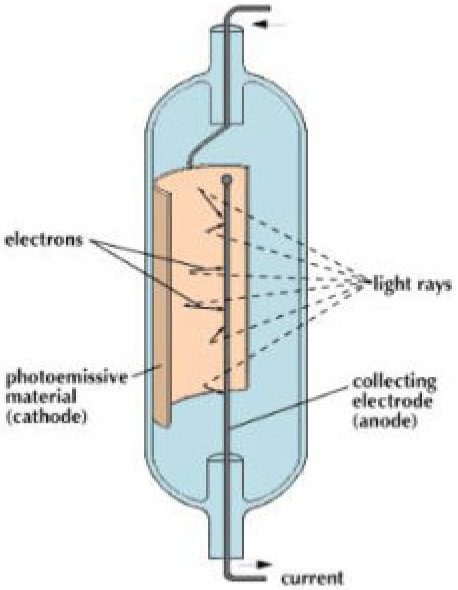
Pick-up coils have been used by sound artists and performers to listen to the internal guts of computers and other digital equipment. There is quite an interesting sonic world to be explored in there...

note: buzzers and vibrators exist in both flavours. Comment on octave doubling!

Photocells, light sensors and optical sensors

Vacuum tube photo cells

Devices to convert light into electric signals were already discovered and available in the 19th century. In fact, the discovery of the photo-electric effect was at the base of quantum physics and Einsteins relativity theory. Photons are freeing electrons from a suitable cathode, thus causing an electric current to flow.



Here are a few historic photo cells:

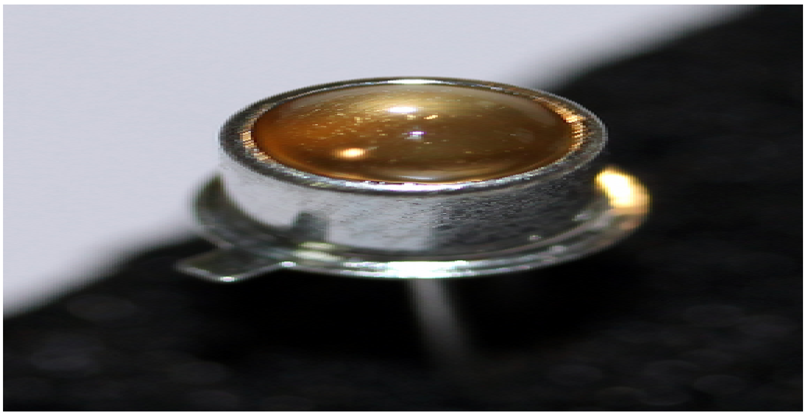
Here is an old vacuum tube photo cell. intact but the tube had become loose from its socket: :

Gas-filled photo cells, as used a.o. in fimprojectors with optical sound track:

This one, removed from a Siemens 16mm film projector because its signal had become way too weak. Although one may encounter such original photocells from the fifties, it will be of little use to try them out: photocells , as used for picking up optical sound in film projectors with an optical soundtrack, are gas-filled vacuum tubes. They are severely sensitive to ageing and the sensitivity of them now, is -after our own findings and measurements- worse than 32dB down. The cathode most often uses a layer of cesium deteriorating over time and exposure to radiation. Also, tube photodiodes are very prone to mishandling: if ever the voltage over them reaches the ignition voltage threshold, they glow up and are thus destroyed right away... Normal working voltages are in the range of 50V to 90V and the cells have to be used with a very high series resistor (> 1MOhm). The dark current for a good cell ought to be in the range of 0.1uA, but of course exact values ought to be read from the datasheets for the individual photo cells. In the industry gas filled photocells have since long largely been replaced with by far superior semiconductor types. For very specialized applications (radioactivity measurement with Geiger-Mueller tubes is an example) some types are still produced.

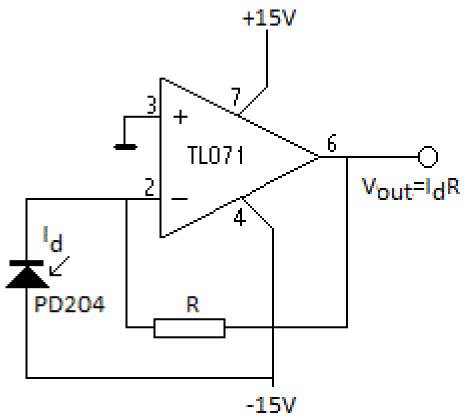
Solid state semiconductor photodiodes and transistors

Semiconductor photodiodes and transistors have been available since at least the early sixties. That was not such an invention as the photosensitivity of PN junctions was already observed long before and in fact many early applications using phototransistors, make use of ordinary transistors where from the black lacquer was removed. Types that were often used as phototransistors where the Germanium transistors OC44, OC70, OC71, OC72. The OP-types where specialy made for their photoelectric properties. Semiconductor diodes in a glass packages are also inherently light sensitive and here again, special photodiodes are also produced.



A photodiode can be run either in photovoltaic mode (when run in forward bias) or in photodiode mode (when run in reverse bias). Photodiodes are normally run in reverse bias as this provides a linear response, and the responsivity range can be quite large. Avalanche photodiodes are also available and are always intended to operate with bias very near the reverse breakdown voltage. Once light is incident on the device, the number of photo-generated carriers is multiplied by the external bias as the device runs beyond the breakdown voltage. This produces gain during illumination. These photodiodes are designed to run in breakdown and are useful for detecting weak optical signals. They are less responsive and thus not used for audio applications. Photodiodes alongside an amplifier and analog-to-digital converter can also be used to receive digital data encoded in amplitude-modulated light or in PWM optical pulses. In the case of PWM, the bandwidth of the photodiode and amplifier have to be high enough as this limits the maximum data rate. The response time of a photodiode is related to its terminal capacitance. The higher the capacitance, the slower the device. The maximum response frequency is typically taken as the knee frequency for a digital pulse with a particular rise time, which is equal to 0.35/ (response time).

Photo diodes require a current preamplifier in order to be of practical use. Typical circuits look like this:



Photodiode amplifier

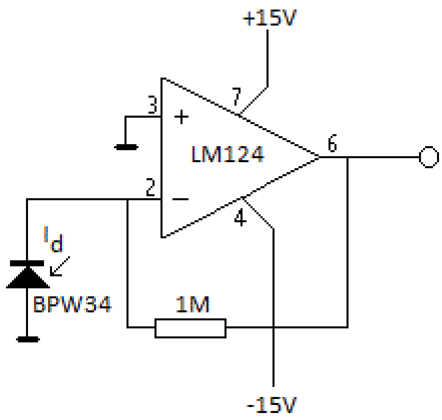
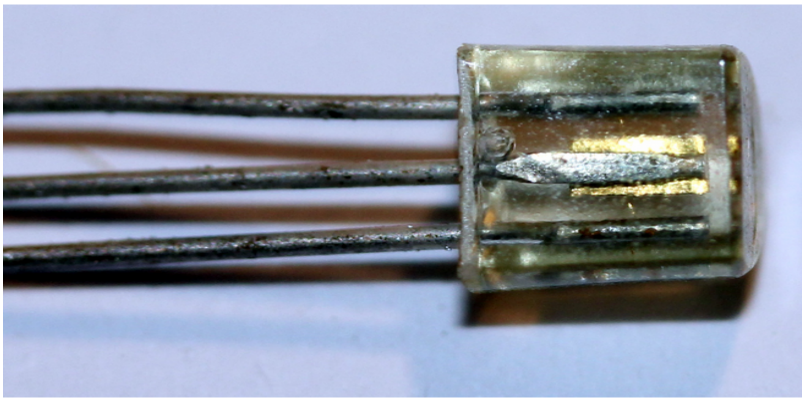


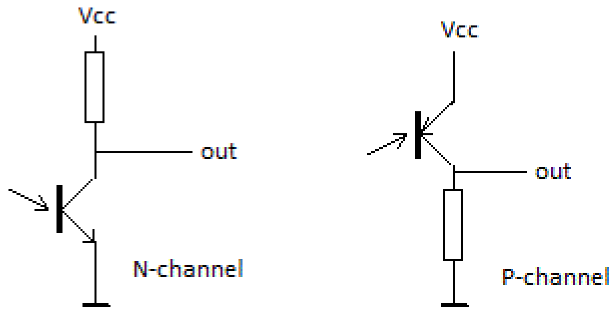
Photo voltaic cell amplifier

We see no reason for not using semiconductor or photo resistive sensors if it comes to performing practice of pieces and installations originally using vacuum photodiodes. However, it still seems important to understand the working and specifications of the original components in order to replace them with modern components. In particular the spectral sensitivity is an important parameter as not all photodiodes or transistors work on visible light. Many types are only sensitive to infrared radiation.

Phototransistors are available as well and their use is pretty much the same as photodiodes. However, they tend to be somewhat slower. Their sensitivity can be set to a particular range, by applying some base current. Most often though, they are used with the base left unconnected. Here are some specimens:

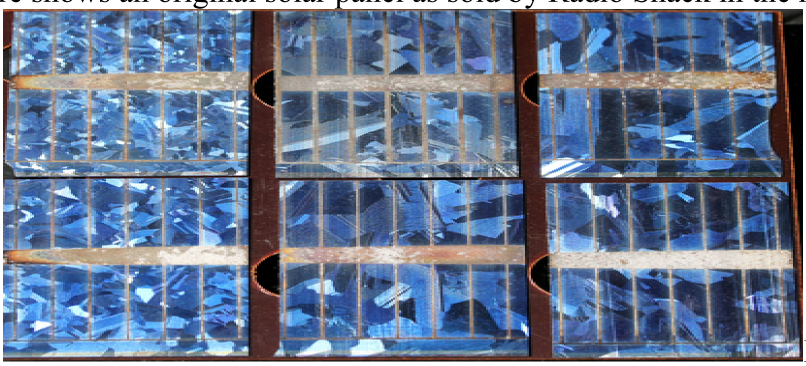
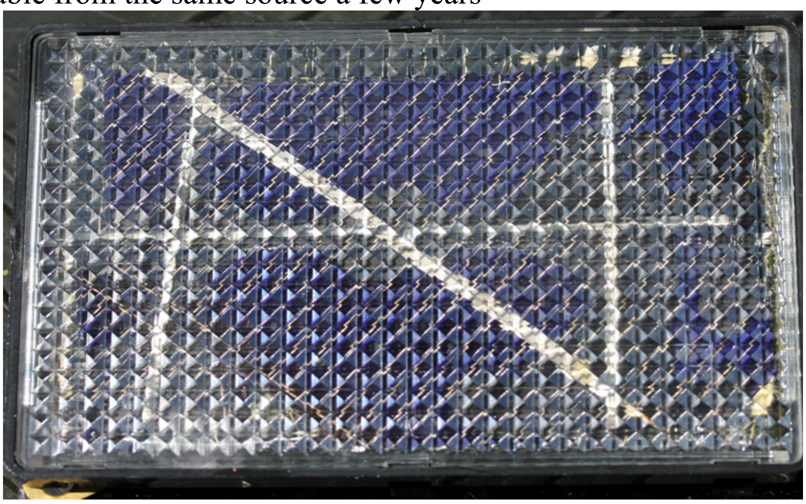


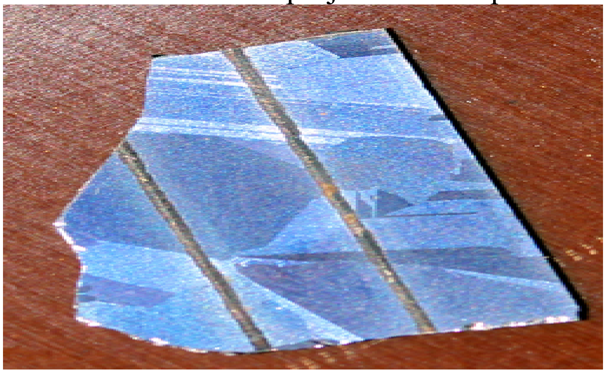
If you are shopping for these components, look at the data sheet to get information on the spectral sensitivity. Many types are specified for the infrared range rather than for visible light. If you are replacing an old and obsolete component, often it will be a P-channel Germanium type. P-channel phototransistors are rare now, but redesigning the circuit to operate with N-channel components is not too difficult.



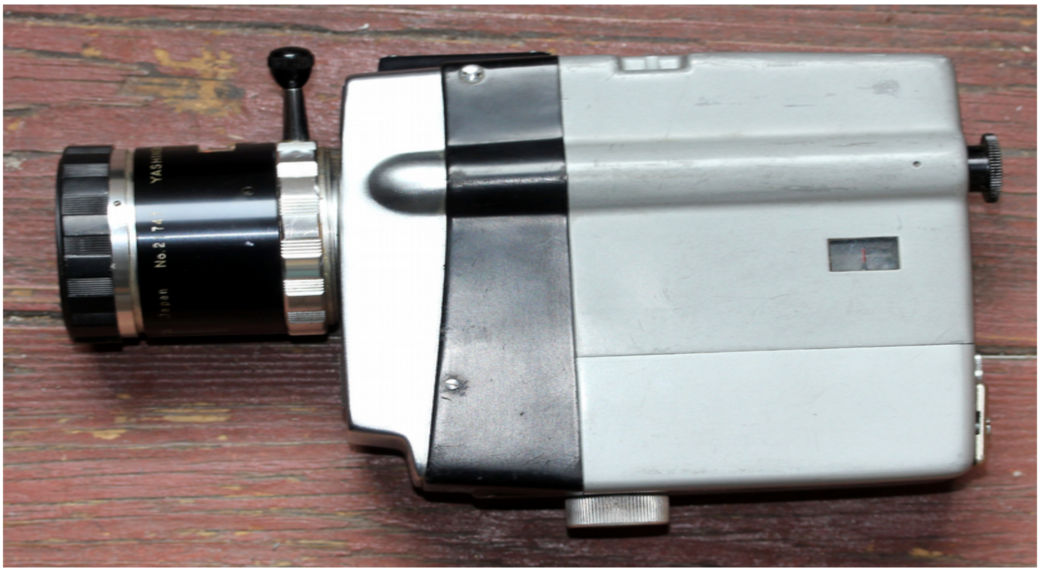
Phototransistor circuits

Photovoltaic cells - solar panels

This picture shows an original solar panel as sold by Radio Shack in the mid seventies:  It is fully unprotected and the six elements that make the panel glued to a piece of insulating impregnated hardboard, are connected in series on the perforated back side. Plastic protected solar panels became available from the same source a few years later: 

As a replacement for a vacuum photocell in an old filmprojector with optical sound, a small piece of solar panel can be used:  In general, such pieces of polycrystalline photovoltaic cells can be used to convert modulated light to sound. The voltage output range is almost independent from the crystal size, but the delivered current certainly is. For the specimen shown here, the maximum output voltage was ca. 500mV. Pre amplification poses no specific problems and even direct connection to mixing desks is possible. The only problem one will encounter is to solder connections to the material. Note that a special type of solder must be used to do this! Regular Sn-Pb solder definitely does not work here. The French sound artist Jacques Dudon uses them as the core of his optophonic concerts and installations.

We rebuild a bunch of old 35 mm photo and 8 mm film cameras by turning then into 'optosound' cameras, just replacing the spot for the film with an equivalent surface of a photovoltaic cell. The optics remain what they are and can be used for diaphragm (this becomes signal output) and focusing. This was done for a composition wherein we needed to turn the movement of very large steam engines and air plane motors into sound. Thus we could 'amplify' the sound of the flywheels, the Watts regulator, propellers and cylinders of these huge machines. They can also be used to make solar eruptions audible. This was done in performances by Pauline Oliveros and others. If you want to amplify and listen to the stars, we can tell you this is barely possible as the sensitivity is too low...

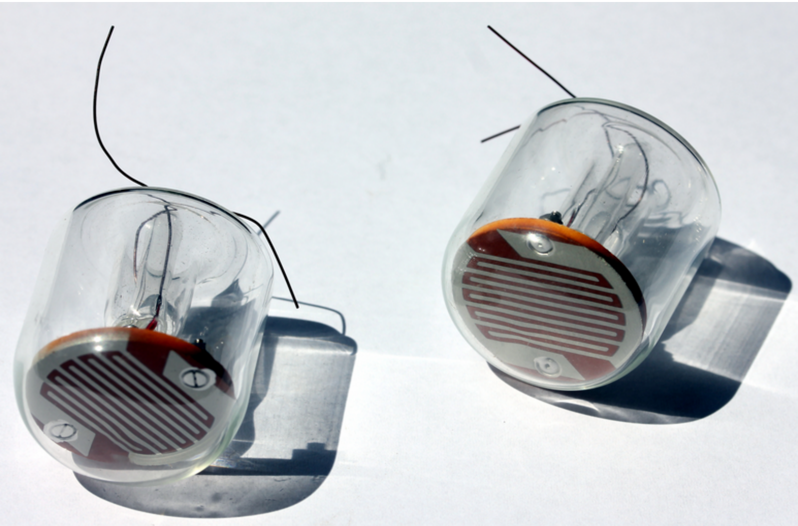


Both 'optoson'

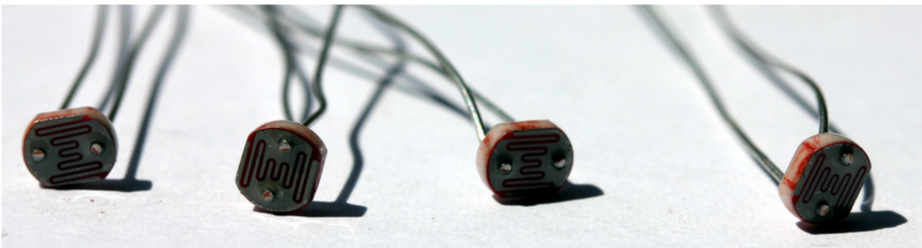
cameras in the picture, have a balanced XLR output and make use of phantom powering just like regular capacitor microphones. In the camera on the first picture, we preserved the camera housing and used its internal space to hold the preamplifier. In the second picture, we disposed of the original camera body altogether and just kept the optics, mounted on a standard BIM box.

Light dependent resistors (LDR's)

For many applications, LDR's can be used although it must be said that in a short time from now, they will also become obsolete and their production even outlawed as they contain a substance toxic to the environment: cadmium sulphide. These components have been around since the sixtees and the first types were quite large, some even encapsulated in glass just like a vacuum tube:



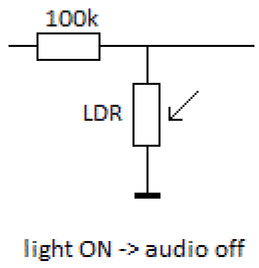
Modern ones look like this:

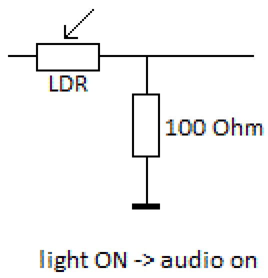


. They found astonishing many applications in audio devices such as guitar or organ volume pedals, amplitude modulators, audio choppers... They always have been very popular amongst electronics enthusiasts as they are very easy to use and calculate. Their dark resistance is ca. 1 MOhm but can reach 200 MOhm for certain types. The smallest resistance values are around 100 Ohms. The light versus resistance curve is far from linear and their application in audio circuits may introduce some non-linear distortion. Furthermore, their response to changing light conditions is highly asymmetric. The light to dark transition may take up to 10 seconds, whereas the dark to light transition happens within some tens of milliseconds. For many designs this has been considered an advantage as it kills all risks for glitches that invariably accompany fast changes of audio signal levels. Hence their use also in compressor and limiter circuits.

Some very simple, yet reliable circuits and circuit ideas as we encountered them through our performance experiences:

Circuit to enable or disable audio when someone of something obscures the LDR sensor:





Of course these or similar simple circuits can also be used to steer control voltages for synth-like equipment modules such as VCO's, VCA's, VCF's...

Properties:

speed: diodes are fastest

signal: phototransistors

photovoltaic cells

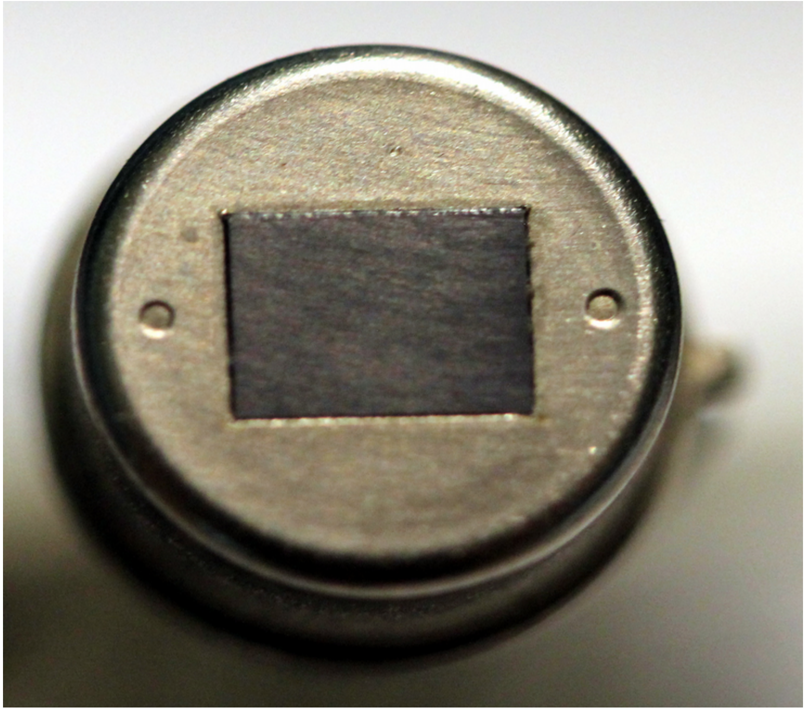
slow: photo resistors (LDR)

Musicians and composers having used photo cells and optical sensors before 1980:

David Tudor, Jacques Dudon, John Cage, Richard Lerman, Nam Yun Paik, David Behrman, Wolf Dieter Truestedt, Hans Otte, Walter Giers, Jerry Hunt, Michel Waisvisz, Jozef Anton Riedl, Dick Raaijmakers, Lucien Goethals, Takehisa Kosugi

Pyrodetectors

These semiconductors form a category of their own in the area of optical sensing. They are sensitive to infrared radiation with frequencies corresponding to the ones emitted by living bodies of humans and warm-blooded animals. Although invented by Herbert Berman in 1970, the first working commercial specimens were introduced on the market in the early 80ties. As early as their availability on the market, they have been used as triggers in live-electronic interactive setups. They are not very suitable for gesture sensing as to that purpose they are way too slow. The fastest signals they can possibly generate are below 4 Hz. However they are a reliable solution for the detection of moving human bodies in a given area and within distances up to 10 meters.

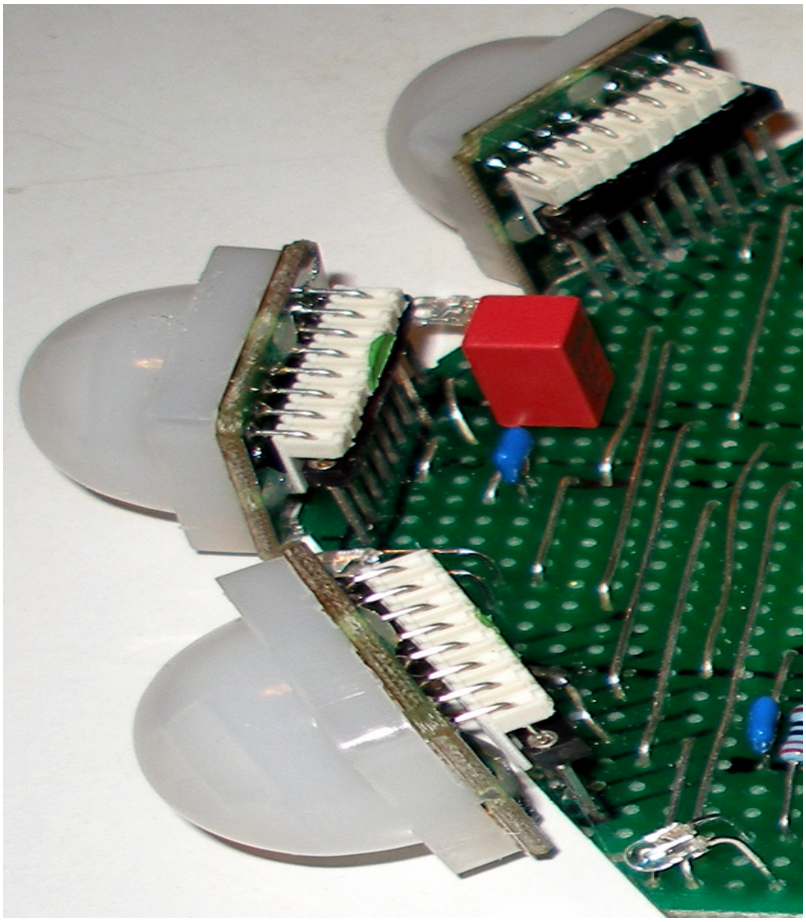


In itself, infrared technology seems very promising if it comes to detecting moving live bodies since here we work with the radiation emitted by these bodies directly. Human bodies normally are at a temperature of 37 degrees Celsius (310.15 °K) and hence must radiate electromagnetic waves with a frequency spectrum showing a maximum at fm= w.T , according to Wien's law. In this relation w, derived from Wien's constant, equals 1.035E11 Hz/°K and T the absolute temperature of the radiating body. Thus, for a human body we come at a frequency maximum of 32THz corresponding to a wavelength of 9.368µm. The emission power, a normalized factor ranging from 0 to 1, equals unity only for a perfectly black body. This leads to the funny conclusion, that a sensing system based on body radiation works about twice as well for black people as compared to whites...(note 6). Obviously, this only applies when their bodies are not insulated from the environment with clothing. This frequency is in the very lowest part of the infrared spectrum. These frequencies are too low for detection by common infrared diodes, designed to operate somewhat below the visible spectrum of light, generally between 620nm and 950nm. Pyro-detectors, using a completely different technology mostly based on ferroelectric properties of triglycine sulfate (TGS), can detect this radiation range very well, showing sensitivities ranging from 7µm to 14 µm. The pyro-electric elements themselves are capacitive charge displacement devices and show off an extremely high impedance (ca. 100GOhm) and for this reason they are always produced directly coupled to the gate of a MOSFET. Due to this extremely high impedance, the sensing elements must show off a sensitivity inverse proportional to frequency. This means that it is inherently pretty slow, limiting frequency of input amplitude changes to below ca. 50Hz. In practice, the output signal is invariably low pass filtered to below this frequency, mostly to suppress the 50Hz mains frequency component. Note that typical output voltages for these sensors are below 450µV. (note 7) The problem in using these detectors for body sensing, is that the spectrum is obscured by many other low infrared sources in the environment, since emitted spectra are continuous and therefore objects at higher temperatures will also emit waves in the band we are interested in. Thus the received signal gives only a measure for the overall intensity of the radiation. For this reason, it would not suffice to place a very selective filter in front of the sensor. By using focusing optics, it is possible to measure temperatures on a small spot, as used in remote thermometers, but this does not help us for developing movement sensors. Rescue comes from another consideration: if the radiating bodies move, we can discriminate the emitted signals by placing a fine Fresnel lens in front of a sensor composed of at least two sensing elements. This is the fundament behind the design of PIR sensors, reacting only to changes in the frequency range of interest. This also seems to entail that the differential signal amplitude should correspond to the distance between body and sensor, since the surface of a body remains pretty constant. The relation, in theory, follows a square law. It also appears that it should be easy to determine the precise angular position in space, given a suitable lens system. For the latter, in regular PIR devices, Fresnel lenses placed in front of the pyro detector are invariably used. For good hemispherical sensing, the sensor should be internally composed of four elements. Angular speed seems derivable from the frequency of the output differential signal. The newest PIR sensor we examined was the PIR STD (Order number CON-PIR-STD-172526), claiming a sensitivity up to 12 meter. After the data sheet, we get from the analogue outputs a signal with a frequency between 0.2Hz and 10Hz, proportional to detected angular speed. The resolution is pretty low as it is determined by the optical characteristics -the number of bands- of the Fresnel lens used. In this case, the data sheet gives 11 bands horizontal with a spacing of 10 degrees and 5 bands vertical, with an unevenly distributed spacing of -30, -20, 0,+20 and +30 degrees. This signal is symmetrical around the halved power supply voltage, but the circuit provides in a second analogue reference output used internally as Vc/2. This provision makes interfacing to a micro controller pretty simple. The sensitivity angle is ca. 100 degrees, horizontal. In the vertical plane it is 60 degrees. A digital output is available as well, derived from a simple window comparator circuit in the sensor module.

Better sensors using this technology may become available in the near future, the range of potential applications being quite large. In our wildest fantasies, it seems possible to design and fabricate a (sort of) CCD chip composed of say 1000x1000 round pixels, each a little smaller than 10µm in size such that each pixel could be tuned to resonate at the wavelength of 32THz. Such a chip substrate, preferably in a triangular 60 degree arrangement, would measure 1 square centimeter, and due to the small size of the pixels, respond quickly. The required mosfets can, with available technology already be fabricated on chip. The read-out logic could profit from already existing designs for CCD cameras. After we wrote down this dream, we discovered that devices implementing this approach actually do exist, but not only in as far as they can be bought by private persons or institutions, are extremely expensive but also, they all fall in the domain of military technology. Some relevant links: http://en.wikipedia.org/wiki/Thermographic_camera as well as http://en.wikipedia.org/wiki/Night_vision_goggles. If you delve any further you enter into military domains. Who takes up the challenge to produce something more constructive?

(6) This only under the assumption that black people can effectively be considered more black than white people for the wavelengths we are talking about here, an assumption that needs to be proved by experiment. So, don't take it for granted.

(7) Cf.. data sheets for the RPW100, RPW101, RPW102, RPY100, RPY102, RPY107, RPY109, RPY222 devices in the Philips Semiconductor Sensors data book, volume SC17, 1989 edition.



Musicians and composers known to have used pyro-detectors in their compositions and/or performance practice:

Jerry Hunt

Video cameras, Vidicon tubes and CCD sensors

We mention these clearly optical devices here for completeness sake, without discussing them for we know no composers but one -Jerry Hunt (1943-1993)- that used them in their performing practice. The only other exception could be Nam Yun Paik, but in his case, it would much rather apply to his video and installation work, than to musical work he made. Of course we are not referring to composers having used video as such in their composition or performances, as such 'normal' use poses not the slightest problem with regard to historic performance practice.

As to Jerry Hunt, we have known him pretty well and have discussed technical stuff a lot with him, we must state that he didn't leave any technical details and even the scores he left are so obscure that it barely makes sense to try to perform them. The only thing one could do is, based on the video's and recordings of his performances, attempt to reconstruct them. We doubt such reconstructions would be convincing given the strong scenic personality of Jerry Hunt himself.

Microwave and radar sensors

Microwave based doppler radar sensors have been available since... [military restriction...]

Used as alarms and for home automation.

Used a lot in the shamanic performances between 1978 and his death, by the American composer Jerry Hunt (1943-1993)

Ultrasonics

Ultrasound in performance : bat's, holosound bells, Tinti, Chi

Ultrasound based gesture recognition technology

parts of this paragraph also in the case-studies article under John Cage. It may be better placed here...

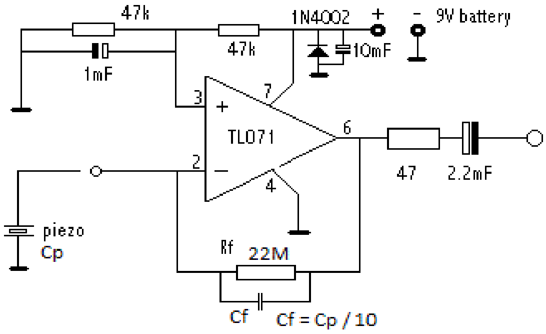
For interactive applications -more specific, for productions with Merce Cunningham and his dance company- Cage has been using photocells. He nowhere gives more specifications about them, but we are sure he confined the technical details to his collaborators some of which were very knowledgeable in electronics: David Behrman, Takehisa Kosugi, Gordon Mumma to name just a few.

Installations and scores mentioning photo cells should be re-engineered with a good understanding of the function they fulfilled in the period-equipment or setup. Phototransistors can be used, but their use is a little more involved: they need some kind of a circuit and their output is a voltage or current. Often, for instance if the photo cell is used as a sound generator, picking up amplitude modulated light, an excellent solution consists of using a solar cell. These are in their common appearance in general way too large, but it's quite easy a break off a small part,

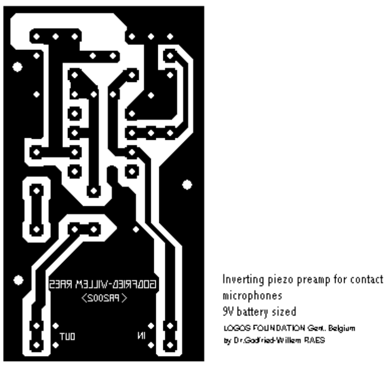
6. Piezo disk contact microphones: problems and solutions

Piezo disks are used almost universally by experimenting musicians all over the world, not in the last place because they are cheap. They have been around since the mid seventies of last century. (1). Generally people connect them to unbalanced microphone inputs of power amplifiers. The resulting sound is more often than not, harsh and metallic. Here we propose a much better solution, based on a good understanding of the nature of piezoelectric transducers. A piezo disk basically is and behaves as a capacitor, generating a voltage when exposed to vibration. As a consequence, the impedance is a function of excitation frequency and can be considered infinite when no signal is generated. Such a generator makes a very bad source for a regular microphone input with constant input impedance.

Here is a much better solution:

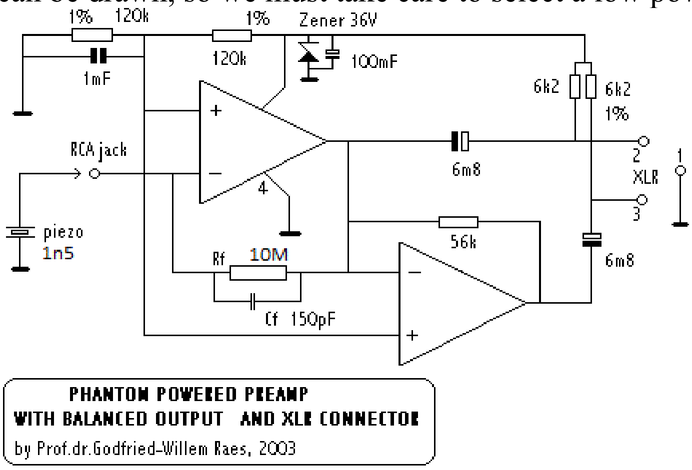


The op-amp in this circuit is configured as a current amplifier. Basically the piezo disk connects to the inverting input and thus sees a zero impedance. It is virtually shorted. The current produced by the disk is amplified by a factor Av = Xf / Xi, wherein Xf = 1 / (2.Pi,f.Cf) and Xi = 1 / (2.Pi.f. Ci). Substitution then leads to the conclusion Av = Cp /Cf , where Cp corresponds to the capacitance of the disk, generally in the order of 10 nF to 24 nF. (see table below). Note that Av is inherently independent of frequency here, thus we avoid the sharp and metallic sound obtained when using a non-inverting op-amp configuration. The 22 MΩ resistor limits DC gain and prevents oscillation. It determines the low frequency roll-off. The higher this resistor, the lower the cutoff frequency. Resistors with such high values - values up to 100 MΩ can be used- are not common. (2) It you are in trouble tracing them, you can also use common 10 MΩ resistors. Note that when using very high resistor values here, the op-amp used must be a type with very low drift. The 1N4002 diode (about any type will do) serves as polarity reversal protection, no luxury as we noticed many users try connecting a battery in its holder until it fits... The TLO71 op-amp specified here, operates at 1V below its minimum Vcc-Vss range, yet it works. It can be exchanged for any low noise low voltage type. We designed this circuit some 35 years ago, at a time low voltage op amps were not readily available, the standard being -15V and +15V. Here is a simple single sided PCB for the circuit (here at 200%, so for etching you need to reduce to 50%). The PCB has the size of a 9V battery and thus can be mounted on a battery holder.

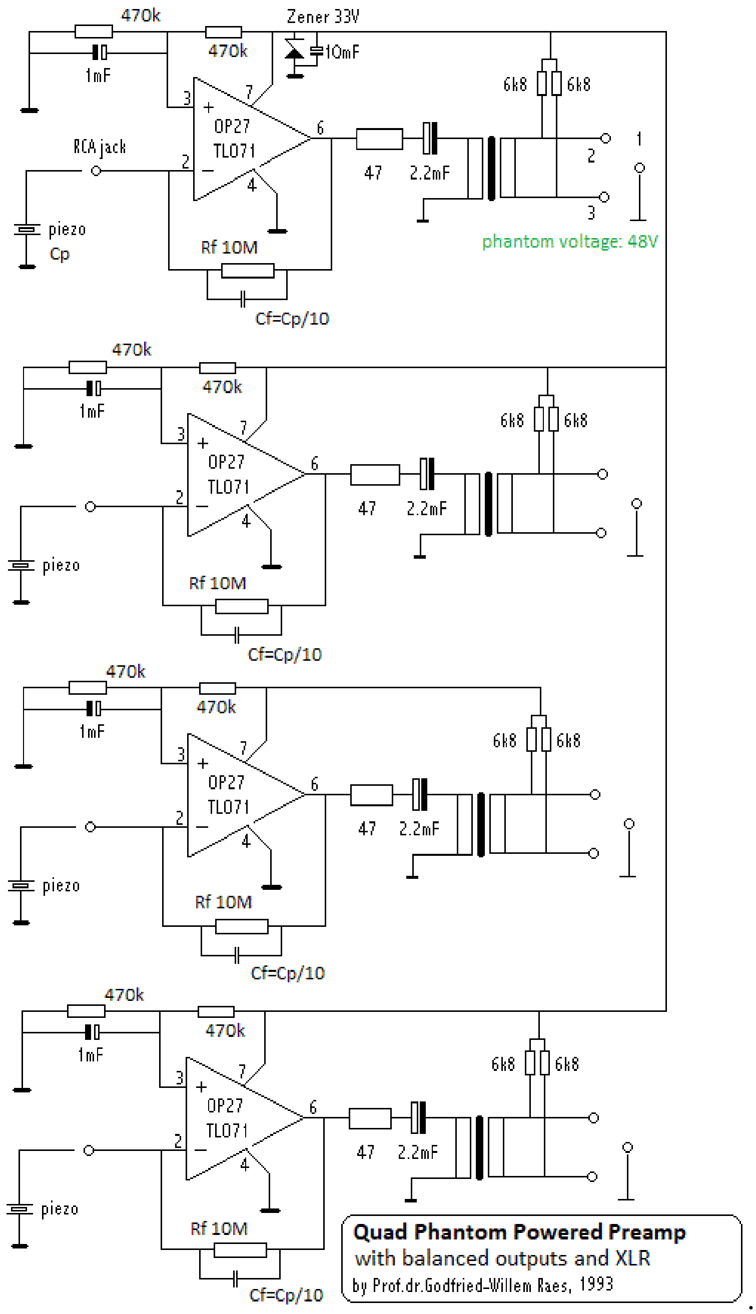


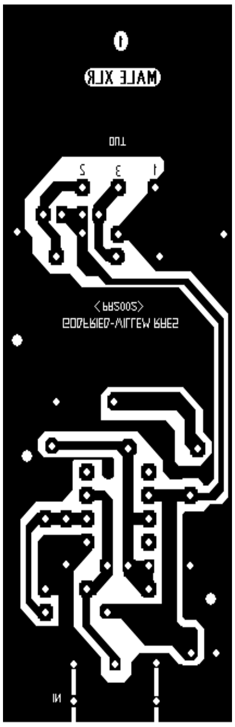
A typical application of this circuit was for an amplifier for a contact microphone picking up the vibrations of a cello bow. For this project, a small piezo disk was clamped via a small wooden bridge between the bow hair and the wood of the bow. Here is

a picture:  A shielded highly flexible silicon wire should be used, as stiff wires will cause unwanted noises.
A variation of this circuit was worked out to facilitate connection to standard mixing boards with provisions for phantom powered microphones. The phantom voltage is 48V but only a few mA can be drawn, so we must take care to select a low power op-amp in the circuit:



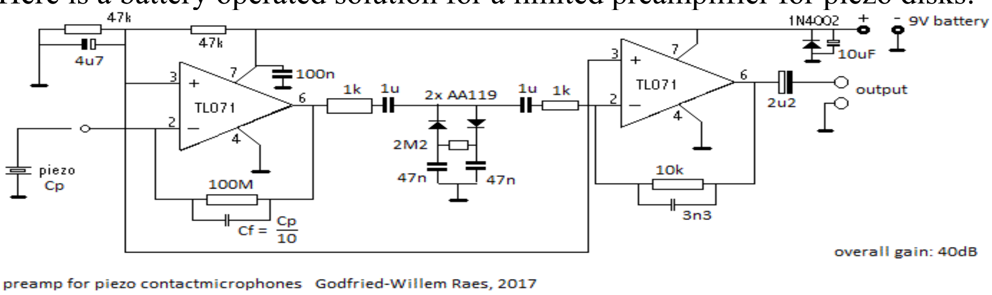
. This is a transformer-less design, but audio transformer designs are possible as well. Here is a circuit for a PCB accommodating 4 separate piezo disk inputs and four phantom powered transformer balanced outputs:


470k
470k
Zener 33V
10mF
1mF
6k8
6k8
3
7
OP27
TL071
6
RCA jack
2
4
47
2.2mF
1
2
3
piezo
Cp
phantom voltage: 48V
Rf 10M
Cf=Cp/10
Quad Phantom Powered Preamp
with balanced outputs and XLR
by Prof.dr.Godfried-Willem Raes, 1993

The PCB for a single channel looks like this:  This PCB was designed to use Amplimo audio transformers. If other transformers are used, the foot print may need to be changed accordingly. Here is a picture:

One serious problem often met when using piezo disks as pickups in live electronic projects is that very high voltages spikes can be produced on strong excitation of the disks. This can eventually lead to destruction of either the power amplifier or the speaker system driven by it. Also, it's not too healthy to the ears of the audience... Hence the need for some kind of limiting or even compression of the signals.

Here is a battery operated solution for a limited preamplifier for piezo disks:

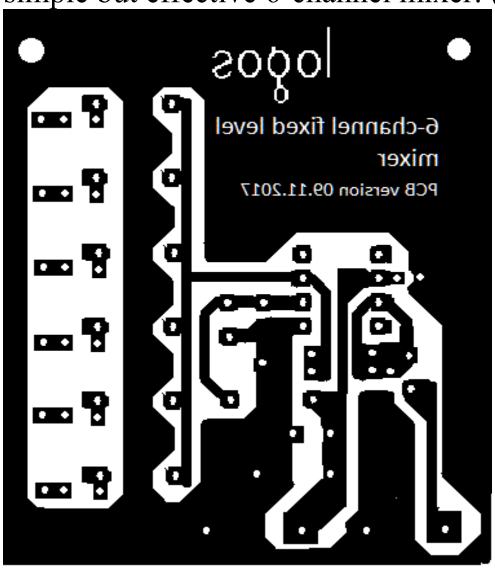


The diode circuit following the first amplifier stage, makes a limiter. If the signal on the input exceeds the diode forward voltage drop (ca. 250 mV for germanium diodes, 600 mV for silicon types), the excess signal flows through the capacitor and high frequencies get attenuated more than low frequencies. With the 1k series resistor and the 47nF capacitor as drawn, the -6 dB point is at 3.4 kHz. Thus it makes a simple soft clipper/compressor. The response time is

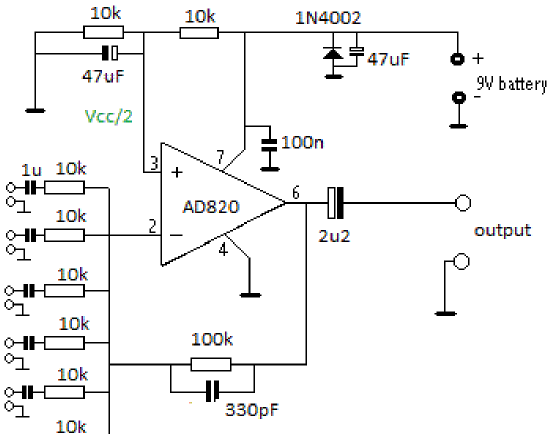
determined by the RC time for the 47 nF and 2.2 M Ohm combination, so in this case 100 ms. It is the kind of circuit that once was very popular amongst short-wave radio amateurs for listening to Morse code broadcasts. The germanium diodes (AA119, OA91 etc.) nowadays may be hard to find. (3) They can be replaced with BAT86 Schottky diodes (Vf = 260mV). Do not use regular silicon diodes, as at clipping, they will cause the second op amp to saturate. This is because the peak voltage of the signal with regular silicon diodes equals 1.2 Vpp. As the second op amp has a gain of 10, the output should become 12 Vpp, a value that could only be reached when the power supply is higher than 12 V. If you find the operation of the soft limiter still sounding too harsh for your application, increase the value of the 47nF capacitors. If you want to increase the attenuation, increase the value of the 1k resistor connected to the output pin 6 of the first op-amp.
The final op-amp, just mixes and amplifies the signal further to a strong and hefty line level. If long cables are to be used, select an AD820 op-amp rather than a TLO71.

A typical application for such a circuit was in the construction of a pick up to be used to amplify human heart beats. For this project a quite large piezo disk should be used (30 mm diameter or so) and a weight should be glued with silicon compound to the upper-side of the disk. This kills unwanted higher frequency signals right at the source. Of course, components have to be selected for optimum very low frequency performance. The limiter circuit adequately protects against spikes here. Similar applications are: larynx microphones and monitoring devices for all sorts of body sounds. For my own 'Woman's Quartet', I even made a set of four vagina microphones...

As in some applications such as sound sculptures and audio art, often the signals of more than a single piezodisk are to be combined, a mixer circuit comes in handy. Here is a small PCB for a simple but effective 6-channel mixer: (reduce to 50% for printing the film).

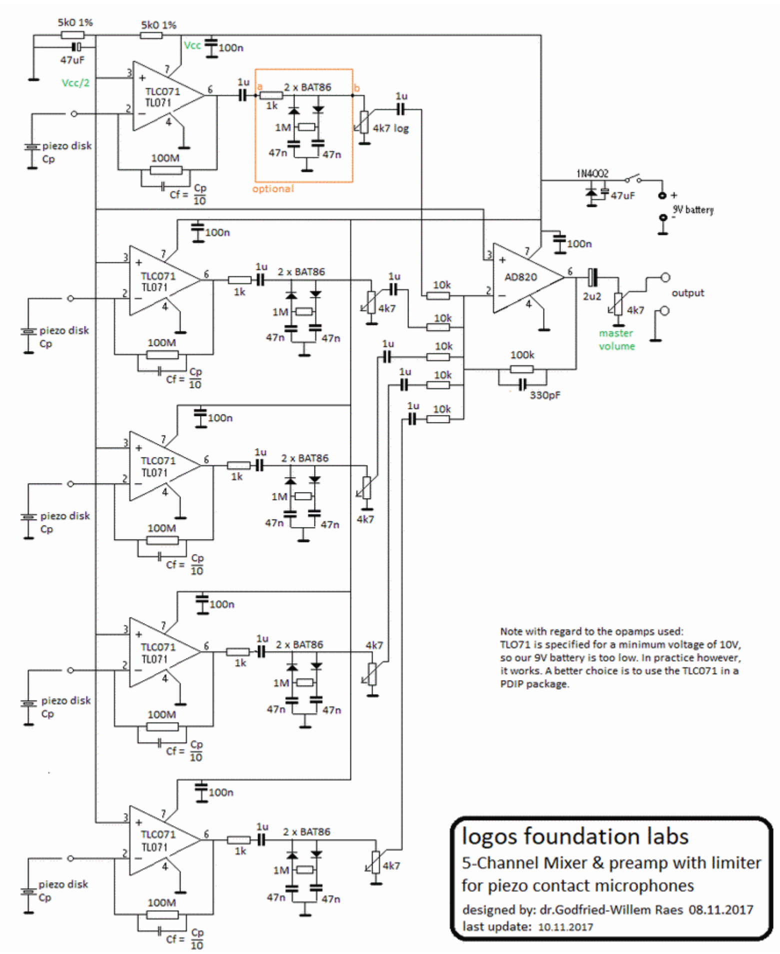


Here is the corresponding circuit:



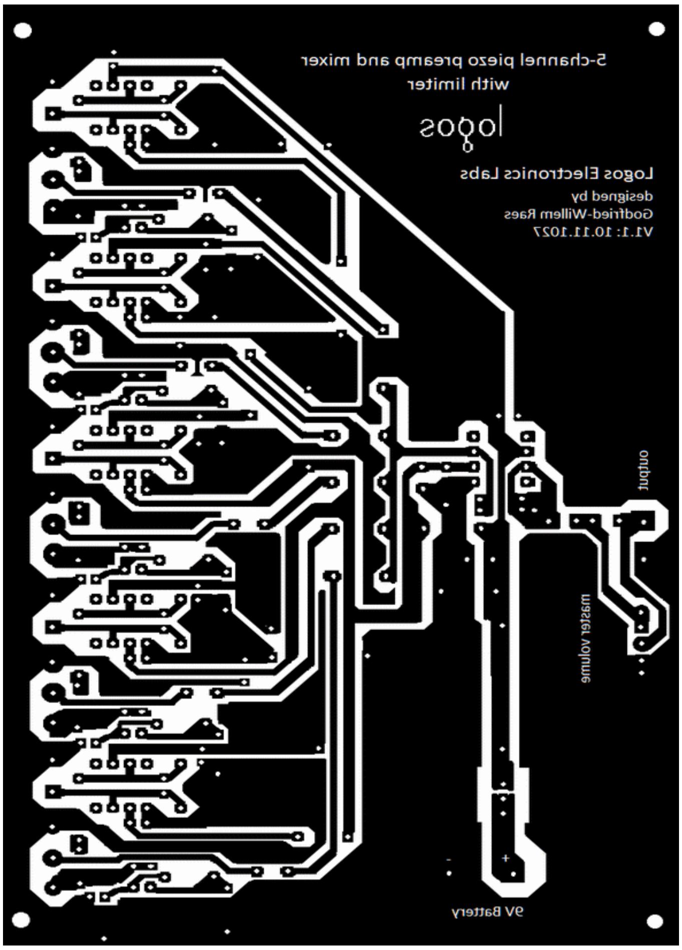
6-channel mixer with 20dB gain

In the following and more elaborate project design, we combined these ideas to realize a 5 channel piezo disk mixer, with individual channel volume controls as well as a master volume control. Here is the circuit:



Note that metal film resistors should be used throughout for lowest possible noise. The mixer stage can make use of an TLO71. AD820 (data sheet) or TLC071 (data sheet) op-amp, but if the circuit is to drive long cables, the AD820 makes a better choice, as it can drive quite large capacitive loads without stability problems. The headroom and maximum output voltage swing of this circuit can be improved by raising the power supply voltage. Make sure not to exceed the absolute maximum ratings for the op-amps used. If different types of op-amps are used in this circuit it is mandatory to go for JFET input low noise types, preferably with output swings up to the supply voltage. Also, make sure to check the minimum Vcc voltage for the part. If specified

at higher than 9V, consider using a 12V or higher voltage battery pack. Our favorite long line driver op amp, the rock stable LF356, unfortunately requires a 30V power supply and cannot be used, unless we redesign the circuit with a proper -15V - 0 - +15V power supply. The AD820 performs very well here, but it's a rather expensive component.
If you want to check it out and build it yourself, here is a single sided PCB design (at 200%):



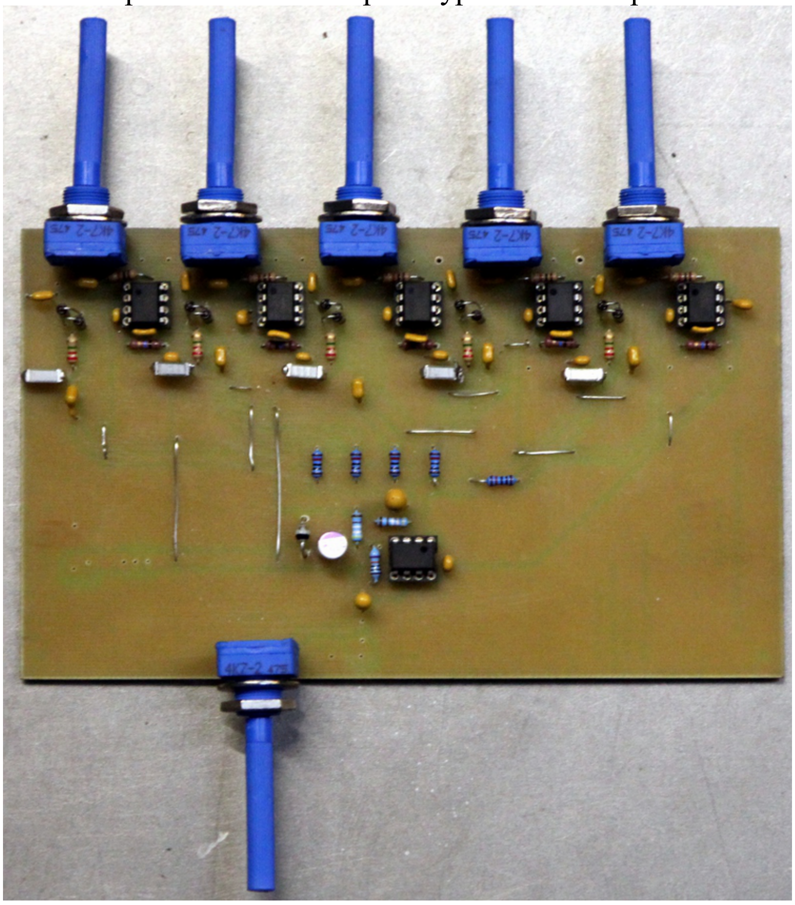
In quite many experimental instrument building and sound-sculpture applications, it is not required to build the circuit with input volume control potentiometers. If the circuit is to be build into such a project, the pot-meters can be replaced by fixed resistors. Determine their value by making a good sounding balance, measure the resistance value of the pot-meters at that setting and replace them with fixed values. Not only this improves lifetime of the circuit, but also it entails quite a saving on building costs. Renounce the tempting idea to replace the pot-meters with trim-pots, as these components are amongst the least reliable of all. Of course this also applies to the output volume control. Generally the only thing you have to make sure, is that the output is a nominal 0dB level, or whatever level you decide to stick to. Remind you: 0dB means 0.775 Vrms, or 2.2 Vpp.
If you want to etch the PCB, save the gif image on your computer and laser-print it on transparent foil reduced to 50% in size. The circuit fits a standard Eurocard PCB (100 mm x 160 mm).
As the value of the feedback capacitor in the preamp circuit depends on the capacity of the piezo disks connected, we measured some different types. This may be useful for users not equipped with a capacitance meter or a detailed data sheet for the product. As you will notice, the capacity is not always a function of size, as it also depends on the thickness of the piezo ceramic layer. Here are the results of our measurements:

diameter	Disk thickness	measured capacity	practical value for Cf	source	remarks
12 mm	0.22 mm	7.8 nF	820 pF	Farnell nr.2433032	fres= 9.5kHz
15 mm	0.22 mm	9.35 nF	1000 pF	Murata Farnell nr.2443195	fres = 6 kHz
15 mm		14.6 nF	1500 pF		
15 mm	0.13 mm	34 nF	3300 pF	Multicomp Farnell nr. 2667639	fres = 4 kHz
20 mm	0.2 mm	10 nF	1000 pF	Murata Farnell nr. 1192520	fres=6.3 kHz
20 mm	0.22 mm	20 nF	2200 pF	Murata Farnell nr. 2443197	
20 mm	0.4 mm	6 nF	680 pF	Multicomp Farnell nr. 2433033	fres = 6.4 kHz
27 mm	0.52	20 nF	2200 pF	Multicomp Farnell nr. 1675548 (wired) Farnell nr. 2443198 (unwired)	fres = 4.2 kHz
28 mm		23.8 nF	2200 pF	Hungary	
30 mm		26.1 nF	2700 pF	Hungary	
35 mm	0.3 mm	30 nF	3300 pF	Murata Farnell nr. 1192552	fres = 2.8 kHz
35 mm	0.53 mm	22.9 nF	2200 pF	Murata Farnell nr.2443199	fres = 2.8 kHz
35 mm	0.51 mm	37 nF	3300 pF	Multicomp Farnell nr.2433034 (unwired) Farnell nr.2433035 (wired)	fres = 2.8kHz

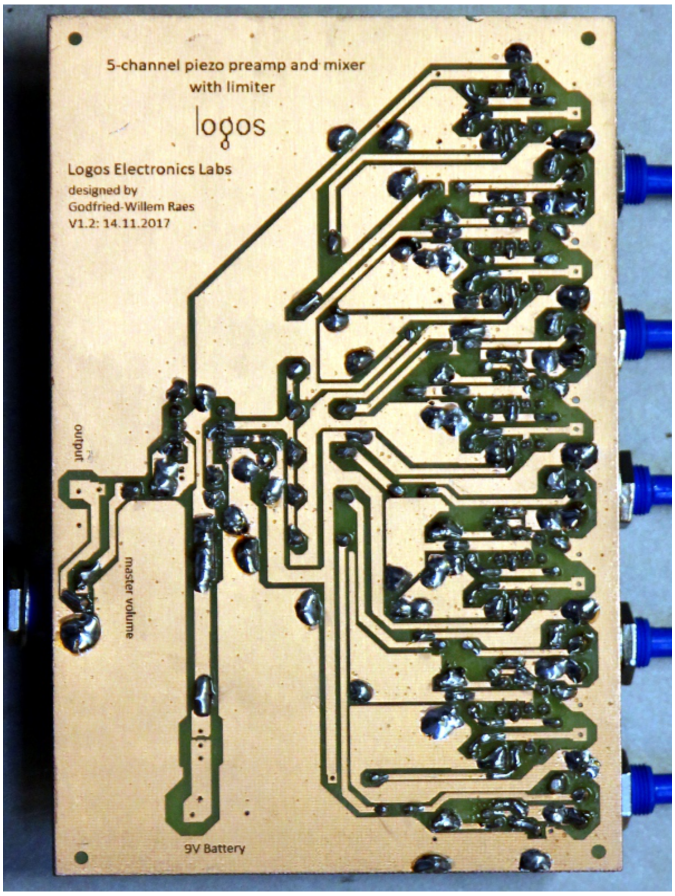
50 mm	24.8 nF	2700 pF	Conrad order nr. 541-022-08	type FT-50T
6 mm x 14 mm square	4.7 nF	470 pF	Hungary	

If you have no idea what kind of piezo will be connected to the circuit, it is a good compromise to go for a 1.5 nF value.

Here is a picture of the first prototype of the complete circuit:



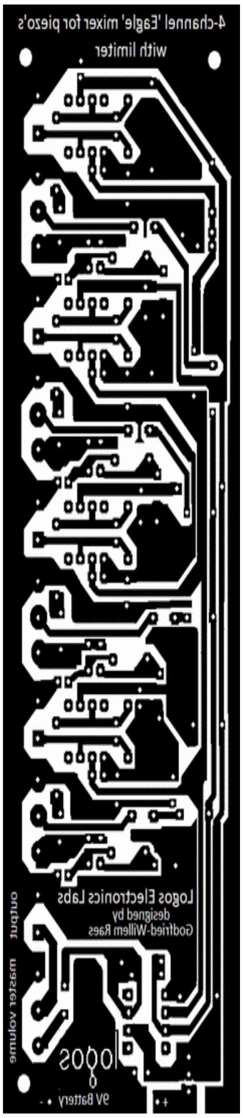
It may help potential builders to figure out correct component placement on the board. The battery holder and the piezo disks are not yet connected on the picture. Here is a view on the copper- and solder side:



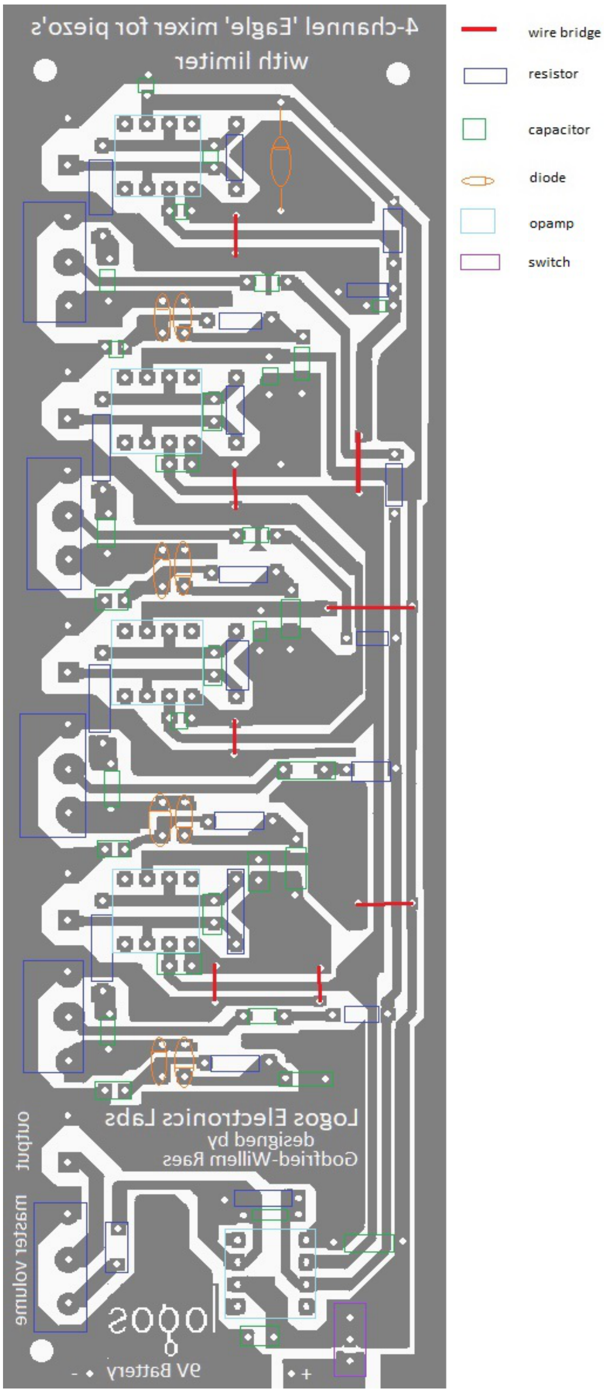
For people preferring a version using a dual bipolar power supply (or two 9V batteries), we worked out a circuit as well. Note that this version shows a clear saving on relatively expensive capacitors. Another advantage of this version is that the inputs for the piezo disk are at ground level.

Current consumption of the circuit is ca. 10 mA, using six AD820 op-amps, so a regular 9 V alkaline battery (the capacity is ca. 550 mAh) can be used with an expected lifetime of ca. 40 hours. For the dual supply version, this becomes 80 hours. Note that TLO71 op-amps draw about twice as much current as their AD820 counterparts.

In the early days of live electronics, between 1965 and 1973, one of the devices seen all over setups was the infamous Eagle mixer, a very small 4 input line level mono mixer. The internal circuitry was of an utmost simplicity: a passive mixer followed by a 1 transistor (a Japanese PNP germanium type...) emitter follower. This mixer was very cheap and its performance pretty poor as well. It produced a lot of noise and caused distortion on loud signals. Nevertheless it was in use amongst hundredths of musicians and has populated just as many stages. With some nostalgia, we remade such a four-input mixer, here in a version for four piezo disks, in the same form factor. The PCB would even fit in the old housing, but if anyone considers reusing an old cabinet, please replace all potentiometers as the original types are of an extremely lousy quality.

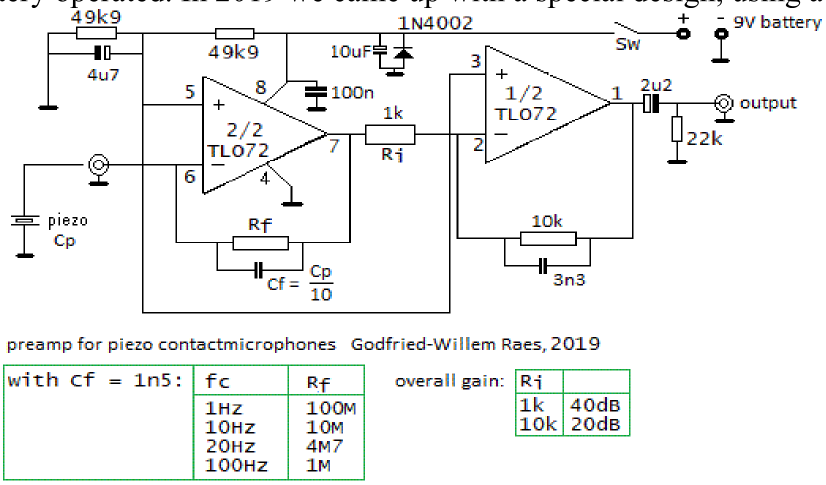


To facilitate component placement on this PCB , here is a sketch:

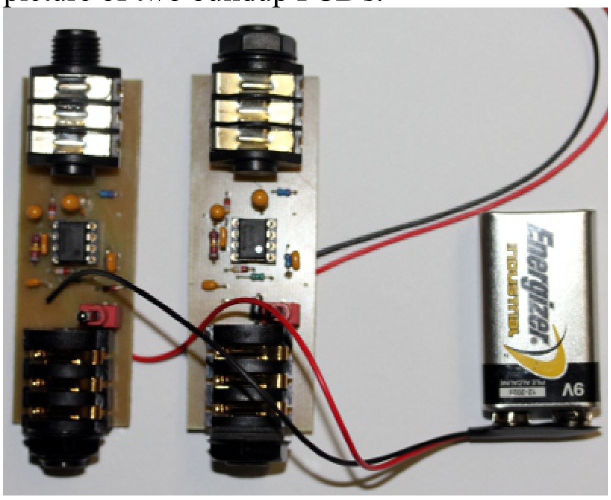


A single Eurocard (100 x 160) PCB will accommodate two of these circuits. Here is the picture of the housing, internally completely rebuild:

All of the circuits above have been copied and used by quite many musicians and quite a few of them we did build on demand. A request that regularly came back, was for a design using 1/4" jack input and output plugs, battery operated. In 2019 we came up with a special design, using a



dual op-amp This is the circuit:
As to the op-amp used here, a low voltage type such as the TLC277, is even a better choice (although a lot more expensive) than the TLO72 specified in the circuit drawing. Here is is picture of two buildup PCB's:

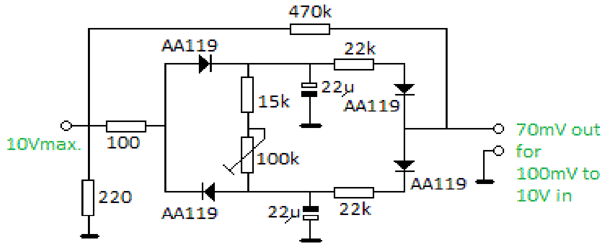


If the vibrations on the piezo disc are too strong, the amplifier may go into lockup. In such cases, we advise users to rather use damping material on the piezo rather then changing the circuit component values. The more you mechanically damp the piezo disk, the more you also damp its resonance peak. If the saturation persists, change Ri to 100kΩ or so. A Eurocard PCB design, accommodating six preamps is available.

Notes:

(1) A historical source on piezo-electrics is: Ed. Palmans, 'Piezo-electriciteit'. ed. P.H.Brans, Antwerpen,1942. A bit more recent: J.W.Waanders 'Piezoelectric Ceramics', ed. Philips Components, Eindhoven 1991. Early adopters of piezo disks in experimental musics were Richard Lerman, Alvin Lucier, Takehisa Kosugi, David Behrman, Hugh Davies, Hans Karsten Raecke, John Hudak, Maurizio Kagel, Karlheinz Stockhausen, Daniel Senn, Michel Waisvisz, Paul Panhuysen, Dick Raaijmakers, Wolf Dieter Truestedt and many others.
(2) 100 MOhm resistors are a/o. made by Ohmite and can be ordered from Farnell (ord. nr.2664957). They are a lot more expensive than regular types.
(3) Not a single germanium diode can be found in Farnell's otherwise extensive catalog. AA119 diodes are popular in music circles and go for prices in the order of 4 Euro's a piece on Ebay. Matched pairs go even higher prices. The reason why these diodes are so much looked after is that their forward voltage drop is very dependent on the current flowing through the diode. (cfr.

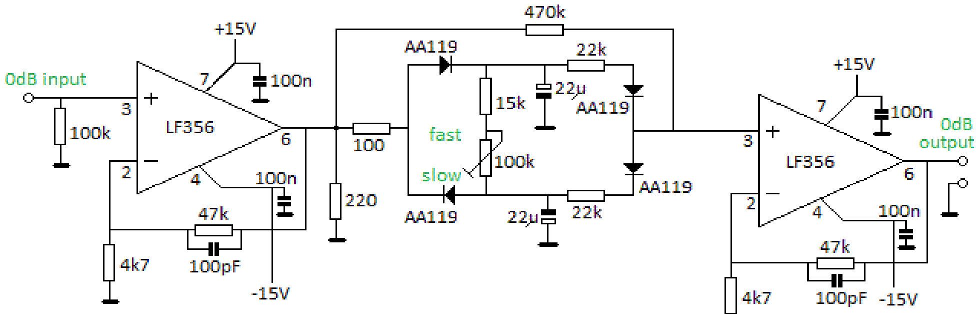
[data sheet](). Hence its application in quite many simple audio compressor circuits. This particular circuit has been around at least since the early seventies:



logos foundation

legacy diode audio compressor

We have no clue as to the original author. The trim potentiometer is used to adjust the decay time of the limiter-compressor. Note that the input impedance is very low and thus the circuit must be driven by an amplifier output. The output also requires buffering, as it has a high impedance. Worked out with this in mind, the practical



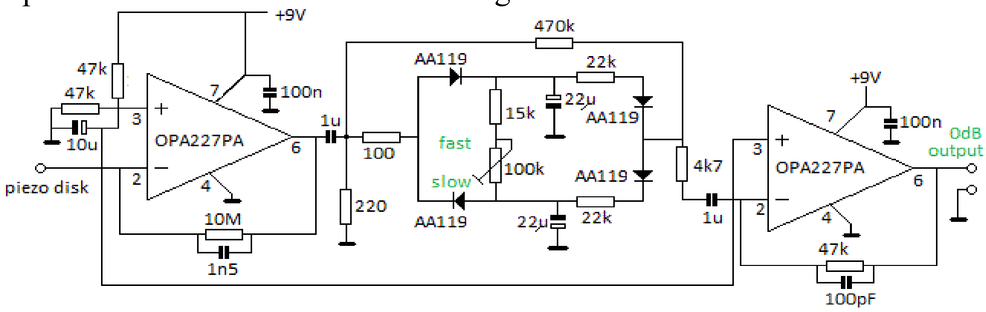
logos foundation

Buffered line level in & out audio compressor / limiter

circuit becomes:

At Logos Foundation we used to have quite some stock of new AA119 matched paired diodes, such that we always could come to rescue for musicians in need of them. They always went in multiples of two.

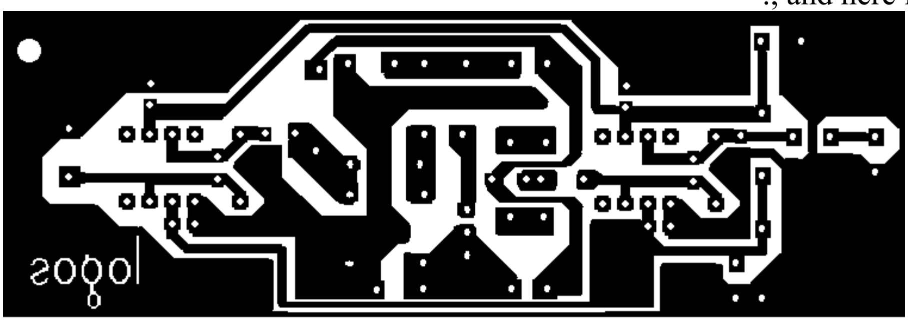
We also designed a PCB for a single supply version, operated with a 9V battery and an input for a piezo disk. Here is the circuit drawing:



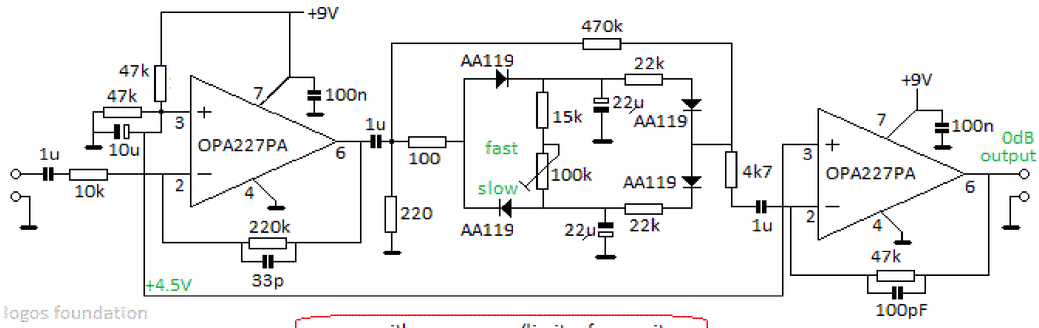
logos foundation

Buffered in & out audio compressor / limiter

:, and here is a



PCB to go with it: PCB for AA119 compressor with buffered in and output. Operation on 9V battery. Reduce to 50% for etching. Inverting This version can easily be built into a regular audio (versus midi) volume control pedal. With some slight modifications in component values, at can also be used for electric guitars and similar



preamp with compressor/limiter for e-guitars

sound sources:

7. Effects

Circuits to modify sounds are about as old as the electrification of audio itself. Under the chapter ring modulators we treated already a great number of very commonly used affects, as in fact a lot of them can be implemented with ring modulators. Examples are tremolo-circuits, choppers, voltage controlled amplifiers (VCA's), vibrato circuits...
A category on its own are the audio compressors.

Compressors

The dynamic range of the human ear is in the order of 100dB, that means that the difference between the softest audible sounds and the loudest non-damaging sound is 1 to 100000. Technical apparatus capable of handling such a wide dynamic range of signals only pretty recently became available. Tape recorders had a dynamic range of ca. 46 dB, AM-radio worse than 40 dB, power amplifiers 60 dB or better... Thus compression was required whenever audio had to be transmitted or recorded electronically.

Here is a compressor as designed and used by radio amateurs. The device is inserted between the microphone and the modulation input of the broadcast station. This and similar pieces of equipment were also often used in live electronic performances. In case where a (vocal) signal of a microphone needed to be fed into a ring modulator, it was of great help in limiting the dynamic range to the possibilities of any subsequent equipment.

Companders:

These circuits became quite popular in the seventies and several analog chips to implement their construction were available. They were often used in concerts were electronic music recorded on tape was presented. The tape was considered 'compressed' and the expander part of the circuit greatly improved the dynamic variations in the music. The once very popular 'Dolby' system for noise reduction was in fact nothing else than a patented frequency dependent compander system. It became fully obsolete with the advent of digital audio and its much larger dynamic range. Once in a while, one may run into old analog tape recordings -or even more likely, audio cassettes- that are Dolby encoded. To playback or re-master such recordings,a Dolby decoder is required. In the seventies and eighties, special analog chips were available to fulfill this function, but they where never widespread nor cheap. The Dolby encoding was patented and hence there was a charge on every single chip implementing this encoding/decoding system. A solution we ran into is the existence of software plug-ins to use with audio-editing programs that implement the decoding pretty well.

A second and quite different series of commonly used effects resorts under the general category of filters.

Filters.

lowpass

highpass

bandpass

These three types, before the mid sixties, most often were built using only passive components: resistors, capacitors and inductors. If one runs into them nowadays, chances are that they still work flawlessly. Only with capacitors of somewhat larger values (say, from 0.5uF on), every so often there may be a problem. Professional audio-filters usually involved hand-wound inductors and precision components, certainly when higher order filters are involved.

parametric

equalisers

VCF's

resonance filters

formant filters

Vocoders

Harmonisers

An important source of inspiration for all musicians involved in live electronics has been the literature on the design of electronic filters used in analog electronic organs.

Distortion

Echo and Reverberation effects

plate reverbs

spring reverbs

tape reverbs

bucket brigade delay line reverbs

digital reverbs

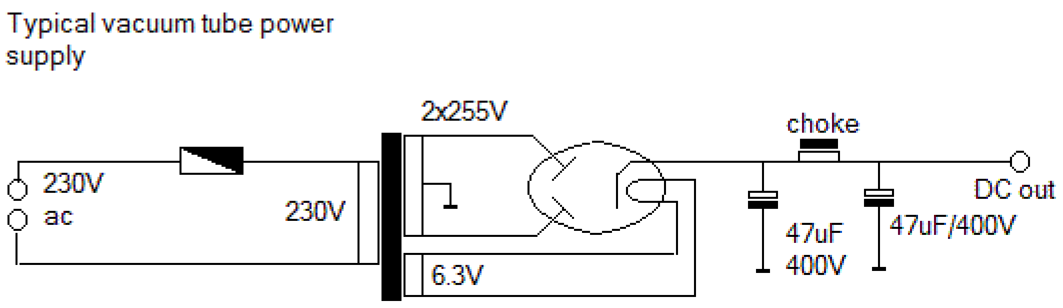
software reverbs

8. Power supplies

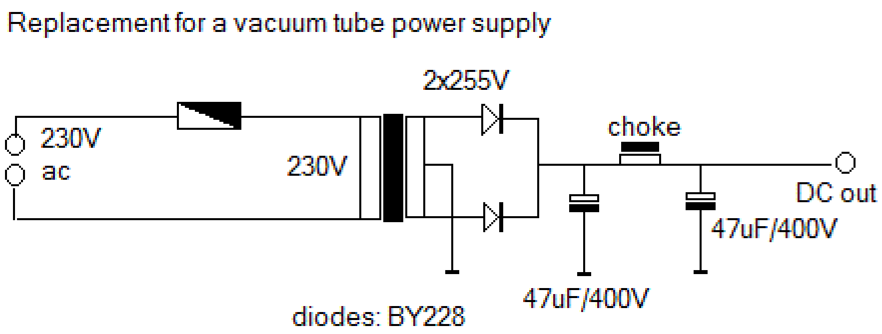
No matter what kind of electronic device, unless powered from batteries, it invariably will have a power supply. The vast majority off all circuits used in audio internally work on DC voltages. Today these power supply components have long become commodity items that you just buy off the shelves, but in equipment build before the mid seventies more often than not, you will encounter discrete circuitry. When servicing legacy equipment, the very first thing to be checked always will be the power supply. Very often, if found defective, it will also be the last thing to do to get the repair done. Hence, this little chapter on power supply issues despite the fact that this topic might appear to be irrelevant for the performance practice of music using legacy equipment.

Long before semiconductors and vacuum tubes came into common use, stabilized power supplies did already exist. [Here is a description](), for those interested in what could be called the prehistory of electronics.

Power supplies in vacuum tube equipment in general look like this:



Sometimes you will find rectifier tubes labeled GZ34, working with a filament voltage of 5V, but most commonly 6.3V filaments were used. Typical vacuum tubes have been EZ41, EZ9x, EYx etc... The first character of the type number -at least for European vacuum tubes- gives the filament voltage: A= 4V, G = 5V, E=6.3V, D=1.5V. It's not a good idea to keep the original tubes nor to replace them. Nothing will go wrong if you replace them with the alternative semiconductor circuit given below:



If you have an old plug, fitting in the vacuum tube socket, the diodes can be soldered on the plug, replacing the tube. If no plug is available, you might consider to use the vacuum tube base itself as such, in which case you need to break the tube and remove it fully from the socket. The filament wires remain unconnected, unless in the rare cases you have to do with directly heated cathodes. In that case, connect one of the filament/cathode wires and leave the other one unconnected. Obviously, high voltage diodes are to be used. Check the electrolytic capacitors, as very often they will be leaky. They can even explode if the equipment has not been used for a very long time... In many implementations, the choke will be replaced with a simple power resistor (in the 100 Ohm range). If the equipment suffers from hum, the value of the electrolytic capacitors can be raised. However, never do this if you want to leave the vacuum diodes in place as the tubes cannot withstand the current surge with larger capacitance values. If, after the replacement, you find the output voltage to be somewhat higher than it ought to be, this is due to the smaller voltage drop in semiconductor diodes. You can lower the voltage a bit by inserting a small series resistor or -if possible- by playing around with the tabs on the transformer. Most of the time, a slightly higher voltage does not have adverse effects on the functioning of the device. In vacuum tube equipment very often double or even triple electrolytic capacitors were used. Nowadays these type are obsolete and no longer in production. They are in fact nothing but two or three capacitors (generally between 33 µF and 68 µF in value) in a single aluminum can with a shared negative electrode. They can be replaced with two or three separate capacitors. Generally, there is enough place for the replacement as modern types tend to be quite a bit smaller.

In general, voltage stabilization was rarely used. If you run into a circuit with vacuum tube stabilization (commonly using a gas filled tube), you should consider replacing it as well with a semiconductor alternative. Stabilizer vacuum tubes have become impossible to find now taking into account that they have become completely obsolete. Linear high voltage regulators can be found on the market and can easily replace them. Often Zener diodes of appropriate specifications can come to rescue here.

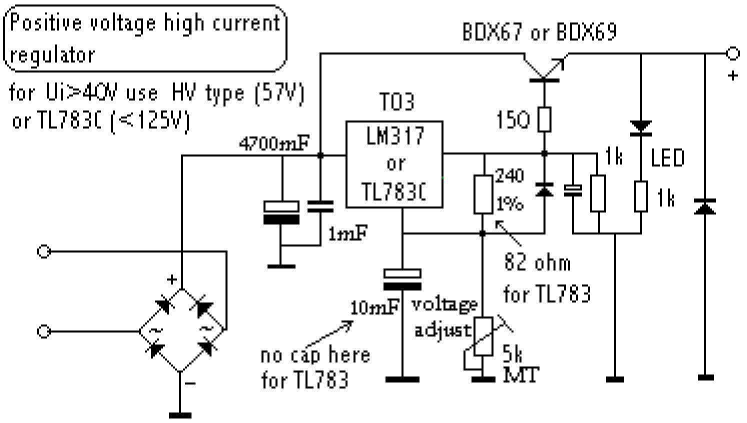
Power supplies in discrete analog semiconductor equipment:

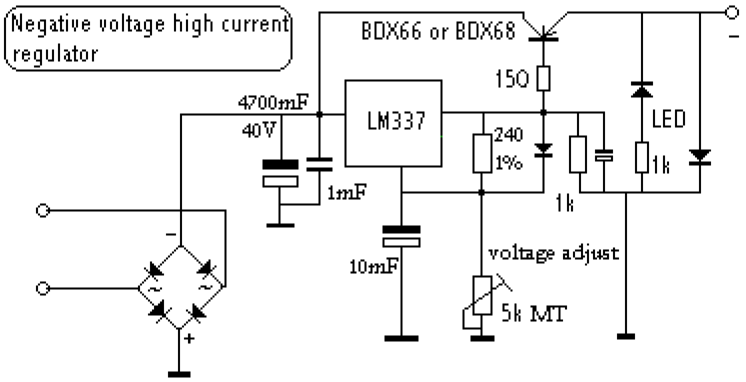
Before the advent of linear stabilizer IC's, power supplies were build from discrete components. Some of them even with great circuit complexity. If in such circuitry a non obvious fault is detected, attempts to repair should be limited to checking the large pass transistor, mounted on a heat sink. Only if this transistor is found to be defective and only if it's a silicon type, you can replace it and see whether it works again. If this fails, don't loose time but rather design a new power supply using linear regulators. These exist with fixed output voltages such as 3.3V, 5V, 6V, 8V, 9V, 10V, 12V, 15V, 18V, 24V as well as with adjustable output voltages up to 80V. Output current is generally limited to 1 or 1.5A (for common voltages such as 5V and 12V 10A types do exist), but if you need higher currents, these regulators can be paralleled. Don't forget cooling!

Analog equipment using chips instead of discrete transistors almost always make use of an integrated analog regulator in the power supply. If found defective, search for a possible cause. To do this a lab power supply is required. It allows you to monitor current and voltage at the same time. If no cause is found, it could have been caused by external equipment, wrongly connected to the device. Replace the linear regulator and check the capacitor.

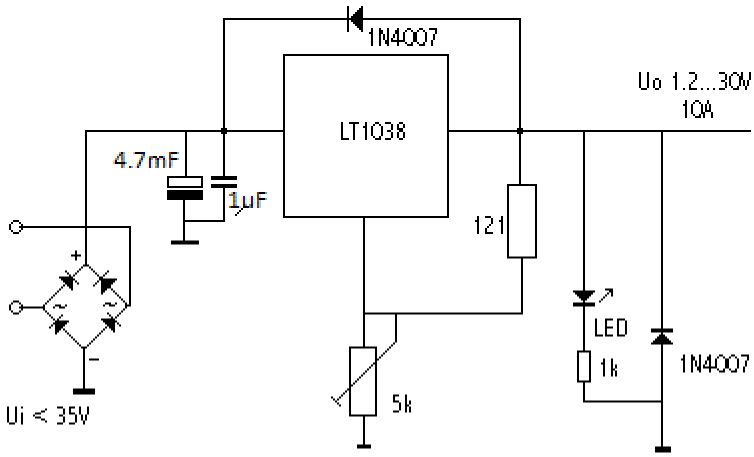
Note that transistorized equipment from the sixties and early seventies may make use of germanium transistors. Most often such circuits have the positive power supply pole at ground level! If this is found to be the case, replace with a negative power supply circuit.

Here are some circuits, with adjustable output voltage, that can be used as replacements in all cases where the current exceeds 1 Amp. If less is required, just use a single linear regulator.:

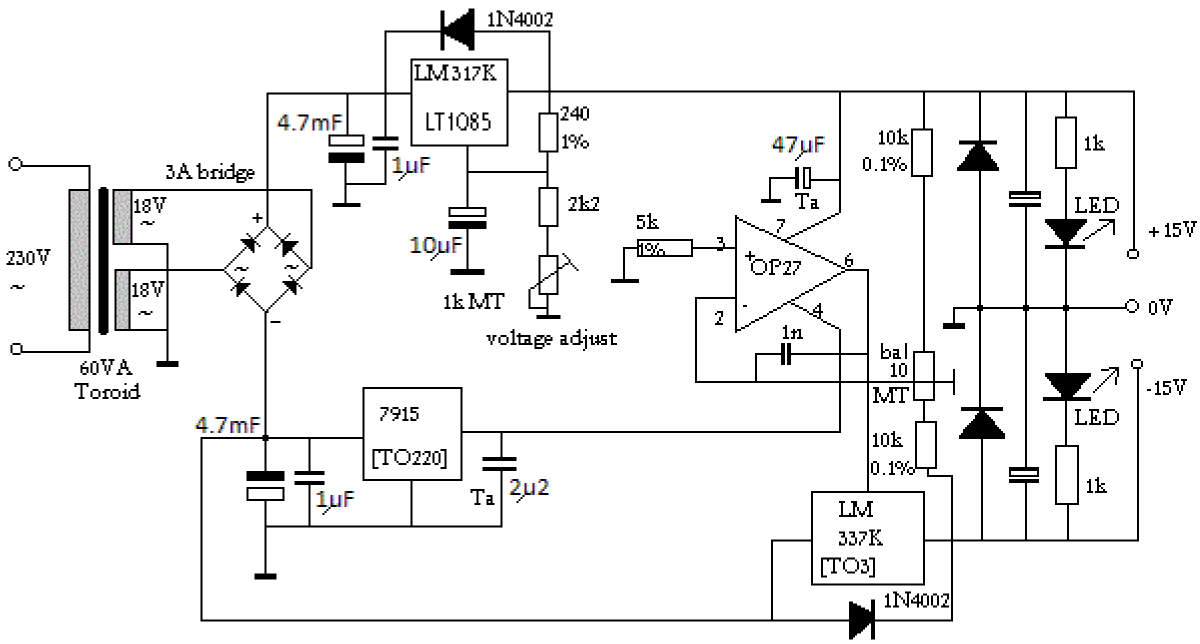




Or, using an expensive 10A linear regulator:



In analog audio equipment, dual power supplies are the standard. Most operational amplifiers work best with dual supplies. Common voltages are +15V / -15V. Precision circuits, such as found in synthesizers, sensor interfacing circuitry and analog computers require tracking dual power supplies. In these, the absolute value of the positive and negative output voltages are kept equal in absolute magnitude within 0.1% or much better. Here is a well tested high performance circuit example:



High precision tracking power supply

This circuit differs from most textbook circuits and designs in that it lets the negative voltage track the positive one. This design was developed by us based on the practical experience that in general the positive side tends to get more load then the negative one. Loads are rarely balanced in practical circuitry.

In audio power amplifiers these dual voltages can be much higher. The voltage is directly proportional to the output power of the amplifier and stabilization of this voltage is normally not applied, exception made for the preamp sections. If these amplifiers fail it's quite pointless to repair them. Unless the repair is limited to replacing a fuse, just get a new amp and save your time for something more creative or artistic... A good amplifier is a good amplifier and can easily be replaced without any effect on the audio. A bad amplifier -some types are popular in pop music- isn't worth being repaired.

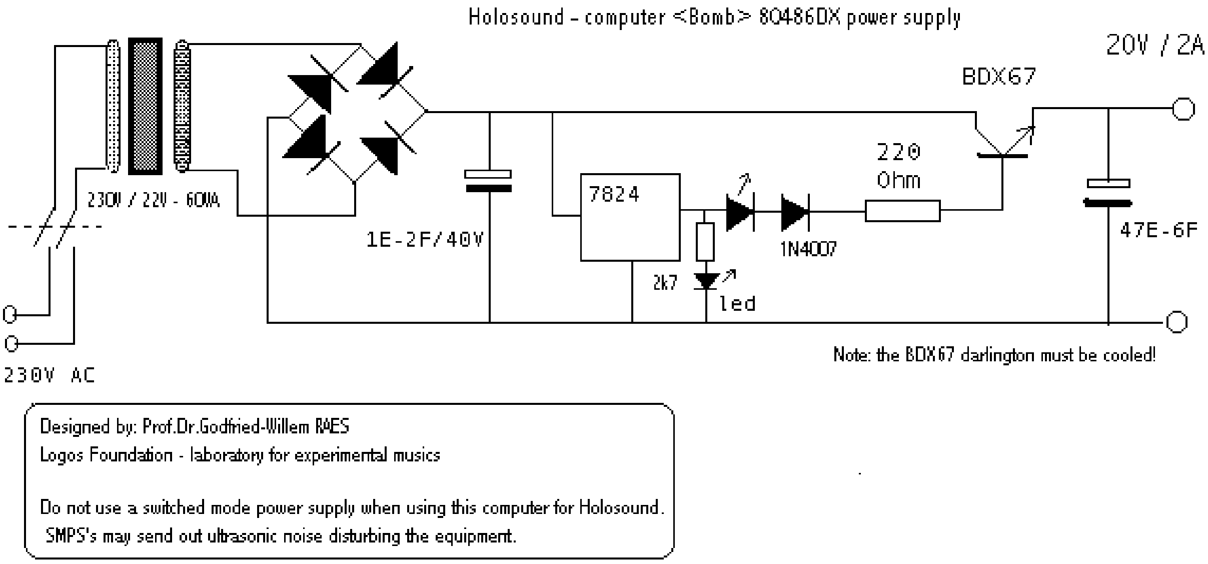
Never exchange a power supply in analog equipment with a switched mode power supply module. Doing so causes audio interference as well as malfunctions on the analog inputs, not being designed to cope with high frequency ripple on the supply lines.

Power supplies in early digital equipment.

From the early days on, almost all digital equipment is powered with an SMPS (Switched mode power supply). For the earliest devices, build up using discrete components. If these fail, do not attempt to repair them. Replace with a suitable off the shelf unit within the specs.

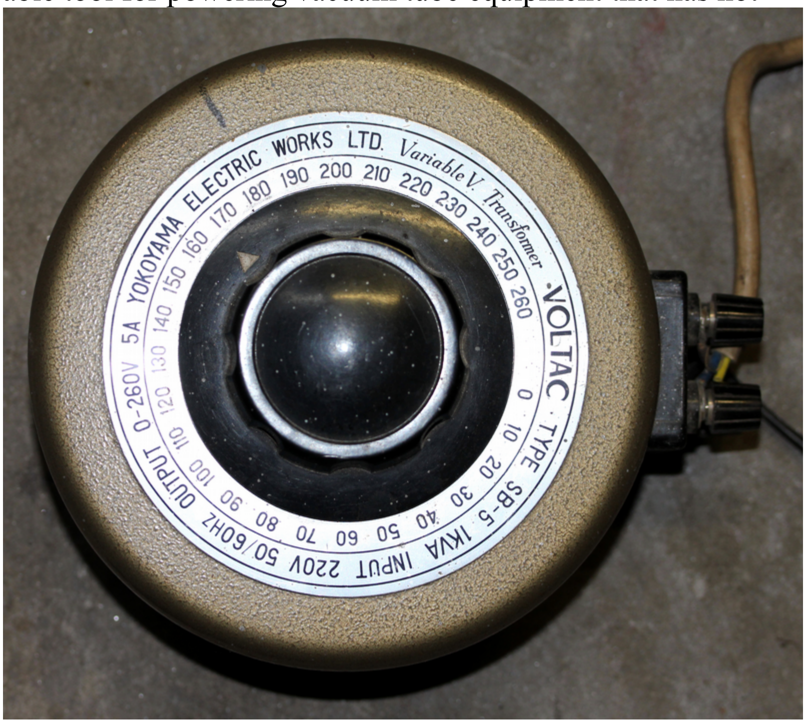
Sometimes it is a good idea to even replace an SMPS power supply with a linear stabilized alternative. The latter will certainly turn out to be much more bulky and expensive, but the advantages in audio performance of the equipment can be tremendous. In numerous cases, we saved musicians concerts by replacing their laptop power supply with a linear one. For the first generations of Apple laptop computers, this was a very common problem and the owners could only produce decent audio from their laptop when fed from their internal batteries... as long as they last.

Here is an example for a replacement for an old laptop power supply requiring 20V at 2A:



Lab power supplies:

The most common types offer single or dual voltage outputs ranging from 0 to 30V and can deliver currents up to 3 Amps. For most repair work they perform quite O.K. However if you have to service vacuum tube circuitry, the voltage range will be way too low. Confronted with this problem, and also because of our needs in musical robotics, we designed a high voltage variable bench power supply ourselves. Here are the [details](). Next to this, a large rotary transformer (a variac) is an invaluable tool for powering vacuum tube equipment that has not



been switched on for a long time.
The procedure consists of setting the transformer to about half the normal operating voltage, switching the device on and over a timespan of an hour or more, gradually bring up the voltage whilst monitoring what happens in the circuitry in the meanwhile. This procedure helps the capacitors self-healing a bit, although one shouldn't count on it. They may just as well cause a

short or explode...

A noted of warning here: Variable voltage transformers are so called 'auto-transformers'. They have no split primary and secondary windings, hence they do not isolate the user from the main power lines. This is an inherently unsafe situation and the only way to remedy this health hazard is to feed the variac from a real safety transformer, thus floating its output voltage.

Hum

The main reason why we pay so much attention to power supplies here, is the simple fact the one of the most common nuisances in early electronic music (live as well as in studios) was the phenomenon of hum. In Europe, 50 Hz and in the US 60 Hz. Searching for the source(s) of hum plagued almost all lineups and suppered up many hours of investigations. This nasty noise was extremely hard to get rid of. It had three quite distinct causes: the power supplies of the equipment used, ground loops, and mains voltage induced noise from low voltage, high impedance lines and transducers. Power supply caused hum is what we treated here together with some possible suggestions for solutions. One important solution- we left out here -because it is not generally thought of as being part of the equipment- is that power supply caused noise can be completely canceled, by using batteries as power sources. Of course, this is not suitable for vacuum tube equipment. For transistorized gear, very often in live electronic performances, batteries or even car batteries have been used. The British composer/performer Hugh Davies, almost always was feeding his equipment from batteries for that reason. Ground loop noise is very hard to find a simple remedy for. Our own solution has always been to use a separation transformer to feed all equipment in the setup, such that it can work 'floating', that is, free from any connection to mains ground. The last source however, can only be eliminated by using balanced lines, lowering impedance's and avoiding inductive pickups. The latter is often just impossible: tape heads are inductance's just as well as electric guitar pickups and many common types of microphones.

In top-quality audio equipment using vacuum tubes, often one will find that the filaments of the vacuum tubes are being powered with pure DC instead of the standard 6.3 V ac. This was done as yet another attempt to minimize hum in the audio signals.

Bibliography and references:

De Muiderkring B.V., 'Electronic Tube Handbook', 19th edition, Bussum, 1981.

Jansen, J.H., 'Transistoren, theorie en praktijk', Kluwer N.V., Deventer, 1964

Schultz, Reinhard 'Netzgeraete', M.Frech Verlag, Stuttgart-Botnang, 1969

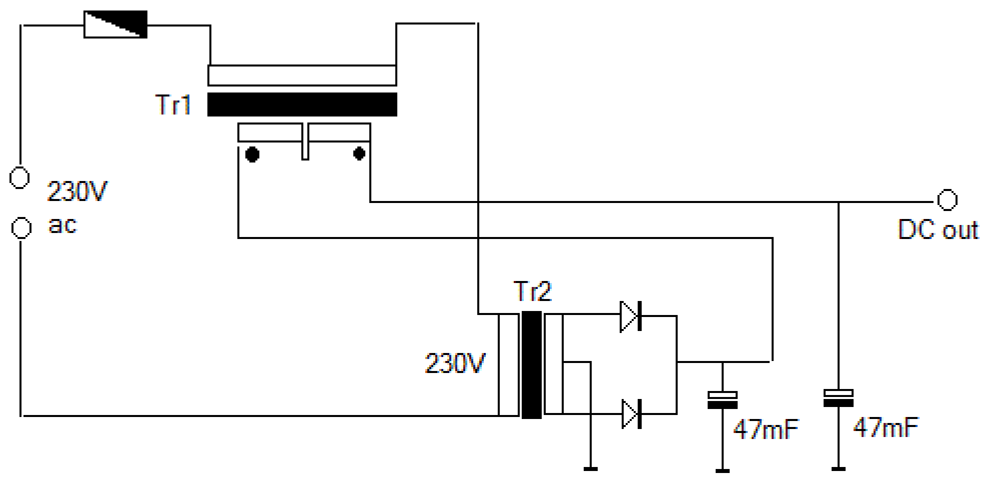
An old recipe for a stabilized power supply without using semiconductors nor vacuum tubes: the saturated core transformer.

Long before vacuum tubes, let alone semiconductors, came to join the world of electric design, it was already possible to design and build regulated power supplies. Quite a few techniques were used:

- - stabilization by using the voltage dependent resistance characteristics of tungsten filament lamps. The lamps are connected in parallel with the load. This technique for voltage stabilization can be found in very many designs of sine wave generators as well (Hewlett Packard, for instance).
- - motorized regulation using a variac or a rheostat, pretty similar to the Watts-regulator as used in steam engines. Such designs mostly apply to very high power supplies from 500 W on.
- - stabilization by using current controlled inductors

The device required to achieve this last option is a saturated core transformer. Such a transformer has three windings, often wound on three separate legs of a closed E-core. The main winding is switched in series with the regular power supply transformer and it is the inductance of this winding that will be modulated by the current taken from the power supply.

Stabilized Power Supply using a saturated core transformer

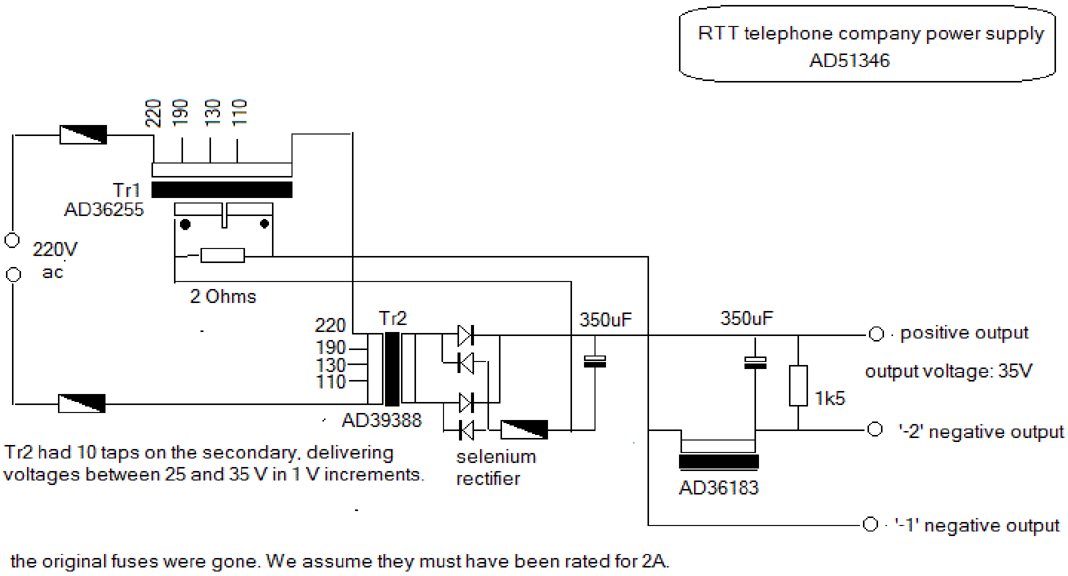


If you look at the circuit drawing above -the saturated core transformer is labeled Tr1- you should notice that the two secondary windings are connected in series but out of phase. Thus no AC voltage will be generated over these windings, provided those windings are closely identical. The DC current through the load runs through this winding. As the inductance of the transformers primary goes down as the saturation is approached, the impedance of the coil will go down as well, and as this coil is in series with the power supply transformer, this one will get a higher voltage. Thus the power supply will regulate load variations by modifying the primary voltage of the main transformer. Here is a picture of such a transformer:


Note also that the core ought to be under dimensioned in comparison to a normal transformer, to guarantee saturation to occur.

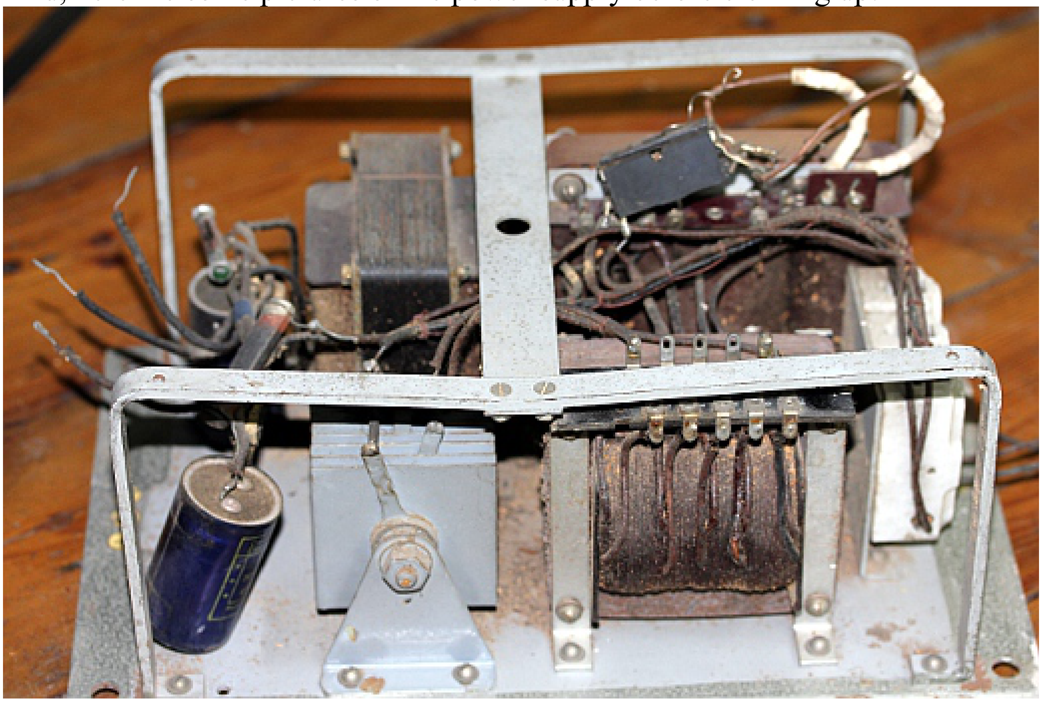
Power supplies build along this principle tend to turn out very bulky. However, they have an inherent sturdiness and as long as their current rating is not exceeded for a long time (the fuse is an essential component...) they have eternal life and require no maintenance.

We never encountered such circuits in high voltage power supplies for vacuum tube circuits -probably because the current these tend to draw is relatively low- but at the other end, very often in power supplies serving telephone exchange equipment. These power supplies offer voltages in the range 24 to 48V at current rating between 2 and 10 A. One day, we took such a circuit (it must date from the period between 1935 and 1950) apart and reverse engineered it completely. We got it from the equipment that was to be disposed off at the move of the former IPEM studio in Ghent, from its premises at Muinkkaai to the much smaller university rooms at Rozier. It was used in that studio as a power supply for one of the sound generators they had in use in the sixties. The keyboard part of that generator is now in the collection of the MIM in Brussels, as we donated it to them. Here is the reconstructed circuit drawing of that power

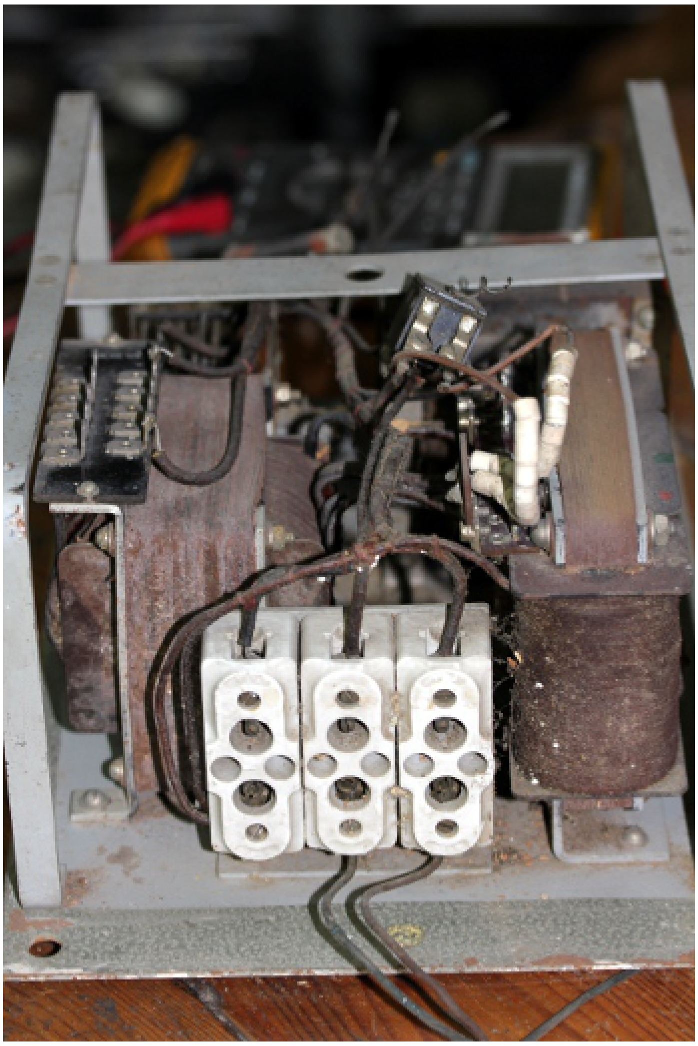


supply:

And, here are some pictures of the power supply before cleaning up:

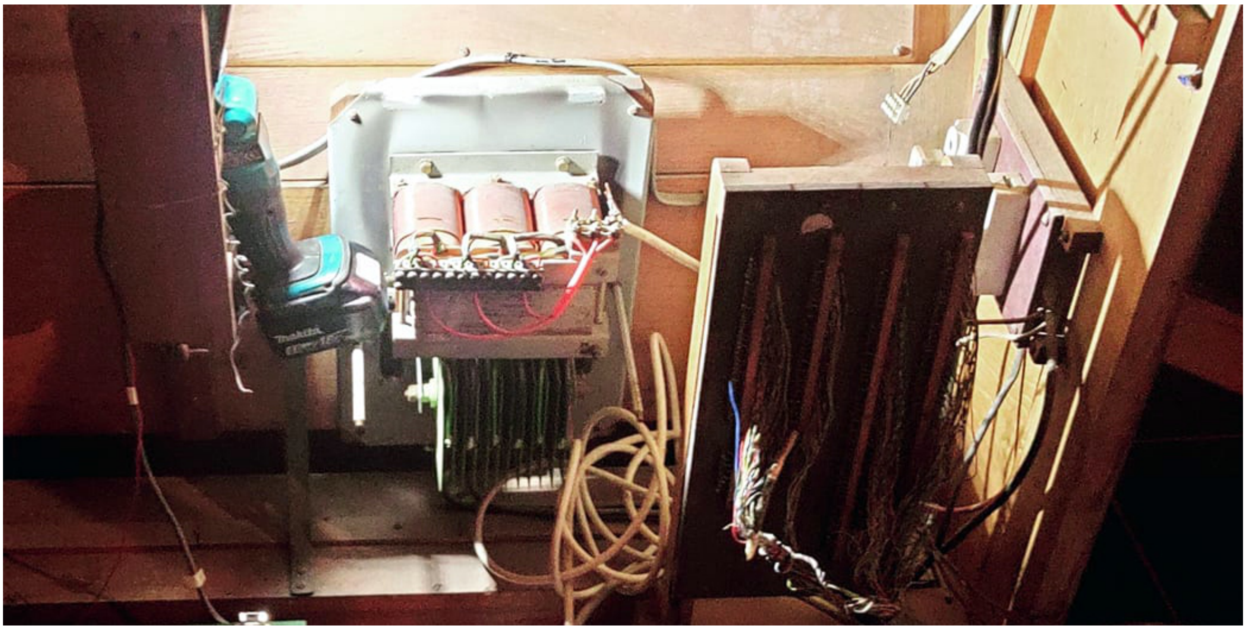



-1
-2
+



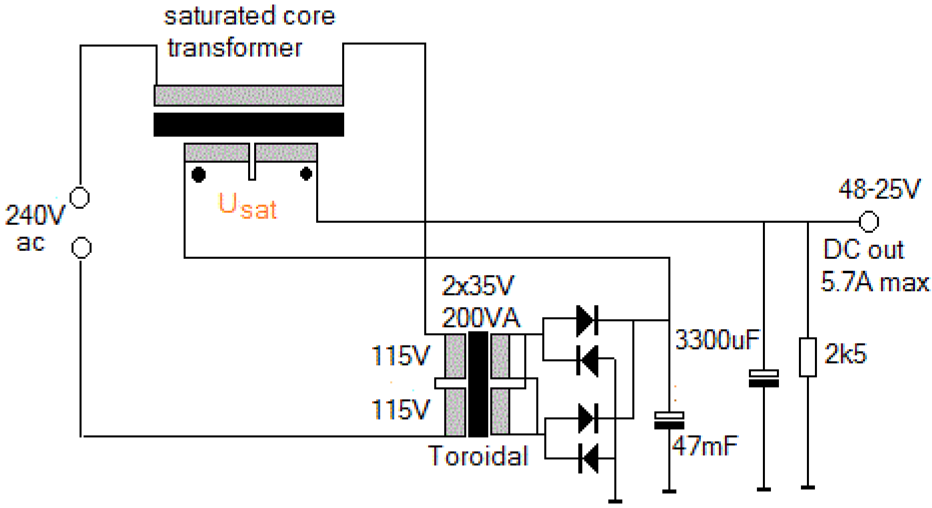
We measured the wire gage on the saturation transformer Tr1 secondaries as 1 mm diameter. The DC resistance of both windings in series equals 0.8 Ohm. If we assume a current density in the copper wire of 3 A/mm2, this would indicate a current rating of 2.5 A. This seems consistent with the size and weight of the transformers, indicating a power between 65VA and 100VA. None of the output terminals are grounded, thus the voltage is floating versus the chassis. There is no mains grounding provision neither, but presumably the entire unit was built into a steel cabinet that itself was connected to mains ground. Note that the device still works today! If we would change anything, it would be the leaky selenium rectifier by replacing it with a silicon type, and the two electrolytic capacitors. We would at the same time increase their value to at least 10mF, thus improving the ripple reduction achieved with the Pi low pass filter using a choke considerably. Capacitors with such high values were simply not available at the time this supply was designed and build. By selecting one of the ten taps provided on Tr2's secondary, the voltage can be set to voltages in the range 25 to 35V. The mains voltage can be 220, 190, 130 or 110Volts. All measurement data come from us, as no documentation could be traced.

Power supplies following this principle can also be found in electronic music equipment in use up to the mid sixties of the 20th century. Examples are electronic organs (Hammond and derived designs), electromechanical sequencing equipment, tape delays and old Springer-machines (used for changing pitch without changing duration with analog tape recordings) with rotating heads and such more. Also, we regurlarly encountered such power supplies in pipe organs using electromagnetic valves in their windchests, as can be seen in this picture:

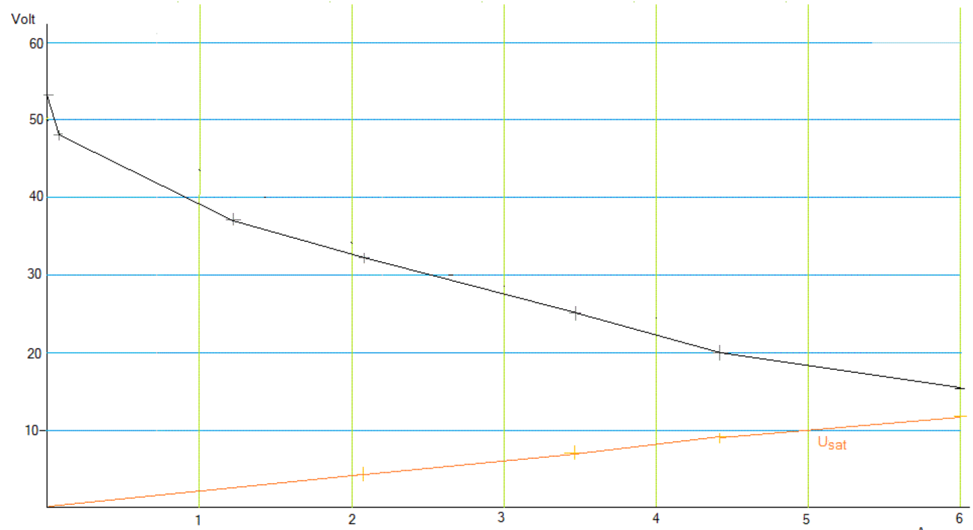


If it comes to restoration, our advice would be not to replace them if they still work properly. At the most, the selenium rectifier should be replaced with a modern silicium type. If a repair is practically unfeasible, it should be replaced with an alternative pure analog and linear design. Stay away from SMPS type power supplies as these will introduce a lot of high frequency noise in the audio circuitry driven by them. Moreover, if they were used in the original piece of equipment, to drive inductive components of any kind, only linear alternatives are possible as SMPS power supplies rarely function well in circumstance where they have to feed fast switching heavy inductive loads. In fact the saturated core power supply is still the most sturdy design if it comes to regulated power supplies for inductive loads such as solenoids and stepping motors.

For this reason, in part also as an experiment, we used one of the spare saturated core transformers we had in stock (type number AD36255) , to construct a new power supply using otherwise modern components in 2015. The circuit diagram follows closely the original design:



Ripple on the output is <10mV with a 1A load. The AC voltage over the primary of the saturated core transformer varies between 30 and 105V, depending on the load. The measured regulation results are -for nowadays standards- far from ideal:



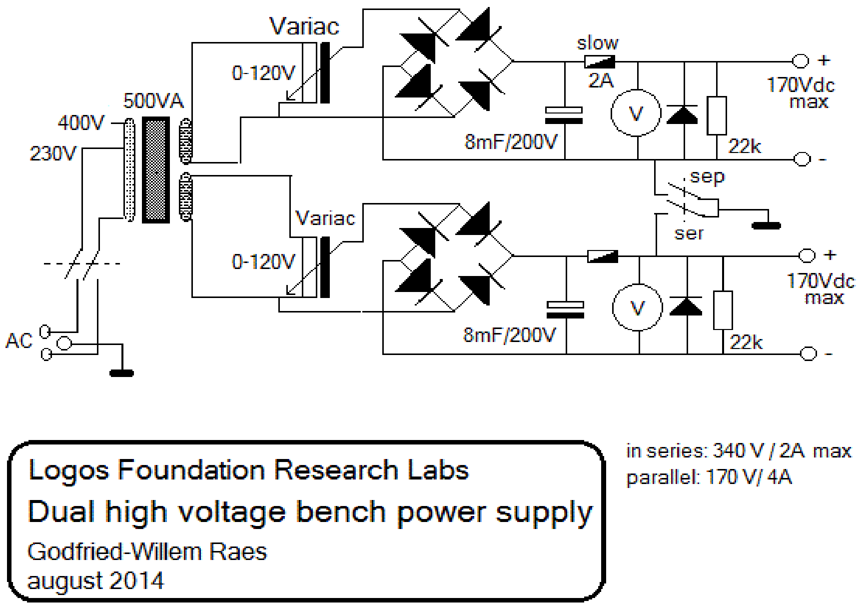
Output voltage versus output current

This curve was obtained with a 50 Ohm / 200W variable rheostat as a load. A noticable mechanical hum -proportional to load current- was noticed from the saturated core transformer. This hum is clearly not caused by our chassis, as we made it from stainless steel. Only impregnation of the windings with epoxy rosin could improve this. It will be clear that regulation gets better as the current is increased. For the components used here, the maximum output current should be limited to 5.7 A. The 2.5 k wire wound resistor on the output serves as a 'preload' limiting the open voltage output to 48V as well as for decharging the capacitors when the power supply is switched off. An incandescent bulb of a suitable value at this place could further improve regulation. The curve drawn in orange gives the DC voltage measured over the saturation winding of the transformer. DC resistance of this winding equals 0.8 Ohm. As predicted by Ohm's law, this is nearly a straight line. The slightly downward curve near the right is caused by the increasing resistance of the coil as it starts warming up.

A simple solution for a high voltage 500 Watt bench power supply

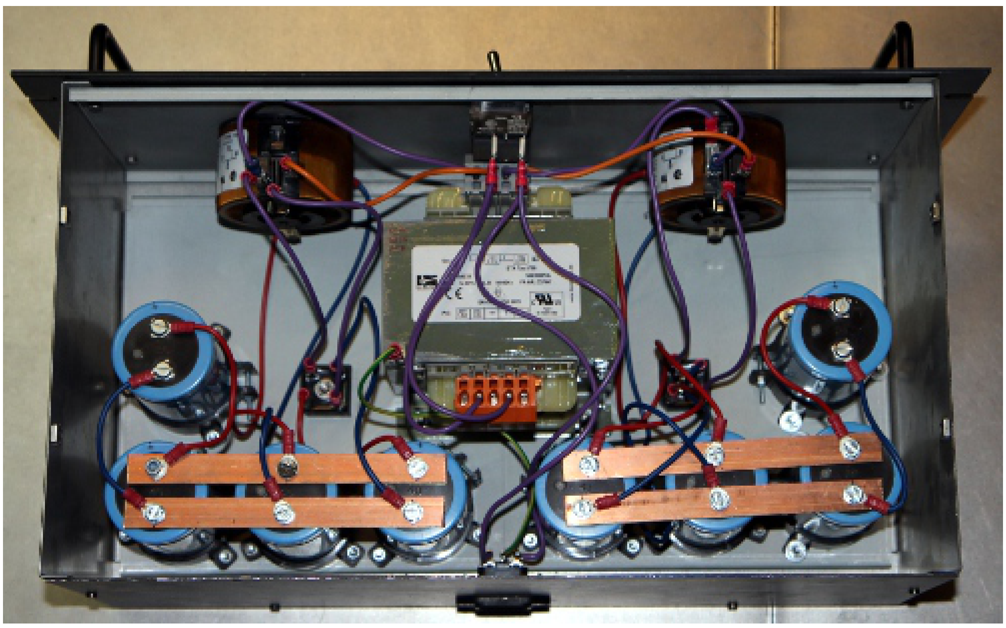
In the many years that we have been involved in the design and construction of musical robots, we must have blown up at least some twenty factory made laboratory power supplies. Not that we exceeded their ratings (in theory they ought to be well protected against that...), but because none of them were capable of dealing with fast switching load conditions or driving heavily inductive loads. Such loads and operating conditions however are normal modes of operation in robots, where all mechanical action is achieved by switching solenoids and/or motors. The culprit in commercial designs seems to be the way the current protection circuitry works. Hence we felt an urgent need to design a variable voltage power supply capable to withstand high surges and with a voltage range quite a bit higher than what's commercially available (mostly 0-30V). The most evident solution seemed to be to make use of a variable rotary transformer (a variac) rather than a conventional linear analog circuit to control the output voltage. As such that would be a very dangerous solution, taking into account that variable transformers are basically auto-transformers providing no insulation at all from the mains power. Hence the use of an isolation transformer to drive the variable transformers in our simple design for a dual voltage bench supply. We used two variacs rated for 120V. The data sheet for these transformers can be retrieved via this link. These transformers can be obtained from Farnell. The outputs can be switched in series with the midpoint grounded with a panel mount toggle switch. There is no provision for voltage stabilization. An extra advantage of this utmost simple design is that is has -in contrast to most commercial designs - no noisy fans, a major source of nuisance

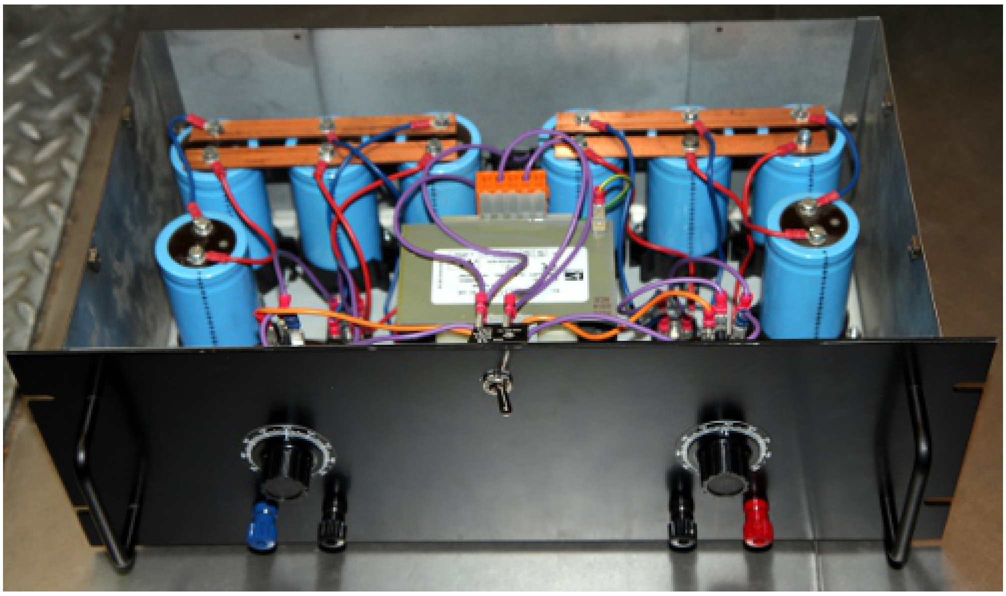
when working on musical instruments.
We mounted two analog voltmeters on the front panel but left out the usual current meters. The fuses (automatic resetable types) are mounted on the front panel as well and can be taken out for ease of current measurement using just about any external meter with a suitable range.



If parallel operation is required, make sure to first adjust both output voltages to the same value before paralleling the outputs. Also, the ser/sep switch must be in the sep position.

This is how the lab supply looks mounted in a 19" rack enclosure:





An easy way to implement voltage regulation in this circuit would be to have potentiometers mounted on the shafts of the variacs and connect these such as to set the base current of a couple of sturdy pass transistors or power MOSFETS. However, for the purpose of switching heavily inductive loads, including stepping motors, stabilization of output voltage is rarely required.

This bench tool has helped us also a lot in servicing and testing old and legacy vacuum tube equipment, as it easily meets the anode voltage requirements for such circuitry.

Case-studies

Electric and electronic sound in the scores of John Cage

Electronics and technology appear in a great manifold of John Cage's scores. It would constitute a study on its own to cover the composers complete works from this perspective, and therewith to pay attention to all problems one may encounter in attempts to perform it 'historically correct' now.

Here we will discuss just a few of his scores, where problems related to the electronic legacy can rise.

Radio Music (1956)

To be performed as a solo or ensemble for 1 to 8 performers, each at one radio.

Problem:

The numbers in the score seem to represent 'tunings' for the radio sets. No units are given however. Numbers given range from 55 to 153 and -as they are not continuous- are likely derived from a tuning scale found on a given radio set from the fifties. The numbers written in the score as tuning instructions are determined by chance operations. They neither follow the official standard for channel separation used in Europe (10 kHz), nor in the US (9 kHz). To interpret this score, first thing we need to do, is to find a correct interpretation for the numbers. Usually period radio sets had a tuning scale with tuning expressed either in wavelength (m) or in frequency units (kHz or MHz). The numbers in Cage's score suggest a scale encountered on a radio receiver like this one:

Thus it is suggesting the radio broadcast AM band, with frequencies expressed in 10kHz units. However, therewith the problem is not solved at all: recordings from period performances, reveal that short wave radio was used: we hear Morse signals and all audible artifacts so typical for short-wave bands. Further evidence for this is in the fact that the radio AM band was reserved for national radio stations and not legally allowed for use by amateur Morse broadcasts. But, if we have a close look at period radio receivers, we will notice that the tuning scale often has many numbers printed on it. The radio on the picture, has a 0 to 10 scale, below the large-number scale. The largest set of numbers, standing for the wavelengths of common radio stations in the AM band. A rotary switch (most often) allowed the user to switch the radio receiver to different frequency bands: The long wave range as well as one or more short-wave ranges.

Cage does not give even the slightest hint as to the frequency range to be used. The only other knob he has instructions for, is the volume controller. This fact in any case, already entails a hint for the equipment to be used: portable self-contained receivers and certainly not tuner's fed into a central mixing board. So there should not be any external amplification.

If attempting to perform this piece first thing to be done would be to recalculate all numbers/tuning in the score such as to correspond to the scale used by the radio receivers at hand. Dividing the given scale proportionally to the number range given in the score is one solution, a better idea might be to use a length of white sticky tape, cut to the length of the physical scale, and copy Cage's range with marks on it. Then, just stick the tape on the receiver scale and perform the piece from the original parts.

A technical detail, however with high relevance to performance practices for this piece (and other pieces using radio receivers) is that old radio receivers rarely had a squelch circuit. This is a circuit that suppresses automatically all noises that normally are received when the radio set is not exactly tuned to the carrier frequency of the broadcasting station. On almost all multi-band receivers produced after the vacuum tube era, such circuits became universal. Of course, on sets with such circuits, most likely a performance of 'radio music' could likely lead to nothing but silence, as the composer doesn't want the performer to adjust the tuning for good reception. The only thing he is supposed to do is setting the dial for the frequency and turning the volume all the way up, regardless what the receiver spits out.

In our opinion it is and remains mandatory to use, if not period radio receivers if still in working condition, than at least equivalent modern ones. It is not very hard nor expensive to build simple short wave receivers and one can find literally thousands of proven designs on the Internet. They are even available as DIY kits. At Aliexpress, we easily could even find kits using vacuum tubes. We have yet to try it though. As there are much more sources of aether-smog nowadays than in the fifties of last century, we would advice to use reasonably long antennas. Modern world-receivers are to be avoided, as more often than not they are way too sophisticated: they have autotune facilities and digital frequency synthesizers, automatic volume control, squelch circuits and automatic interference suppression. Moreover, they rarely have a tuning dial and instead use a numeric keyboard to enter the required tuning frequency.

So far, we have been talking about the technical aspects of performing 'Radio Music' from the performers perspective. But, there is a second maybe even more prohibitive problem, unrelated to what a performer can or could do: the aether and the radio signals occupying it itself. The short-wave bands as well as the AM broadcast MW band, are more and more abandoned or, used for all sorts of radio beacons, digital and encrypted transmissions. In other words: the entire radio environment has changed tremendously. It is pointless to attempt to re-create the aether as it was in the fifties, filled with Morse signals and short-wave stations with national news from all countries in the world. There really wasn't much music to be found there, as the possible quality that could be obtained in those radio bands was way too low for even recreational pleasure.

One may get seduced into trying to perform this piece, using modern portable radios using the FM band. We are absolutely certain that this entails a complete misunderstanding of Cagean aesthetics. The piece would sound more or less like a John Zorn collage, because on a modern FM receiver, stations always come out right and thus you would get mostly commercial and entertainment musics, as that is what occupies this band. Also, as FM radio covers only an area of at most 50 km around, the adventurous character of tuning into remote radio stations, evaporated. Alvin Lucier in his book 'Music 109', reports about Cage at a performance with a radio set, happening to hit Radio Vatican, and got the pope speeching for a while...

As a conclusion, we are convinced that a historical performance in the sense of a recreation of this piece is impossible. It would never sound the same as in the time of its conception. At the other end one may argue, that the changes the piece undergoes by historically informed performance of it, are an intrinsic part of the Cagean aesthetic. It we extrapolate it into a further future, at the end, we may end up with nothing but silence. We are quite certain Cage would have loved that consequence.

There are other pieces by Cage where radio receivers are required in the score: In 'Water Walk' (1959), five portable radios of inferior quality are prescribed. No further specs nor details given. The score just specifies them to be switched on and off at given times.

The oldest Cage piece involving radios is 'Imaginary Landscape #4' (1951), scored for 12 radios and 24 performers with one conductor. Each radio requires two performers to use it: one for tuning and the other for the amplitude and timbre changes. This is the first composition wherein Cage radically tries to detach from his own work and wherein he attempted not to have any control on the composition. (Cf.. his book 'Silence', p.57-60). In reference to this, Cage commented: "It is thus possible to make a musical composition the continuity of which is free of individual taste and memory (psychology) and also of the literature and 'traditions' of the art. The sounds enter the time-space ... centered within themselves, unimpeded by service to abstraction". Performances of this piece we listened to, -all recordings we could find data from the 21st century- reveal that the AM radio band was used. In-between-stations noise was anticipated and is an essential part of the piece. In all too many recordings this is missing or way to damped. Hence, here again, the use of modern radios with automatic tune-adjust and squelch circuitry is inappropriate. The 'six transistor' small radio sets cheaply produced and omnipresent in the sixties, would be a good choice. Original portable radios from around 1951 would all be vacuum tube types. If they would still work at all, for sure the required batteries to feed them are since long no longer produced. They needed a hefty '90V anode battery' as well as a 1.5V battery for the filaments. The 90V battery is obviously the problem here.

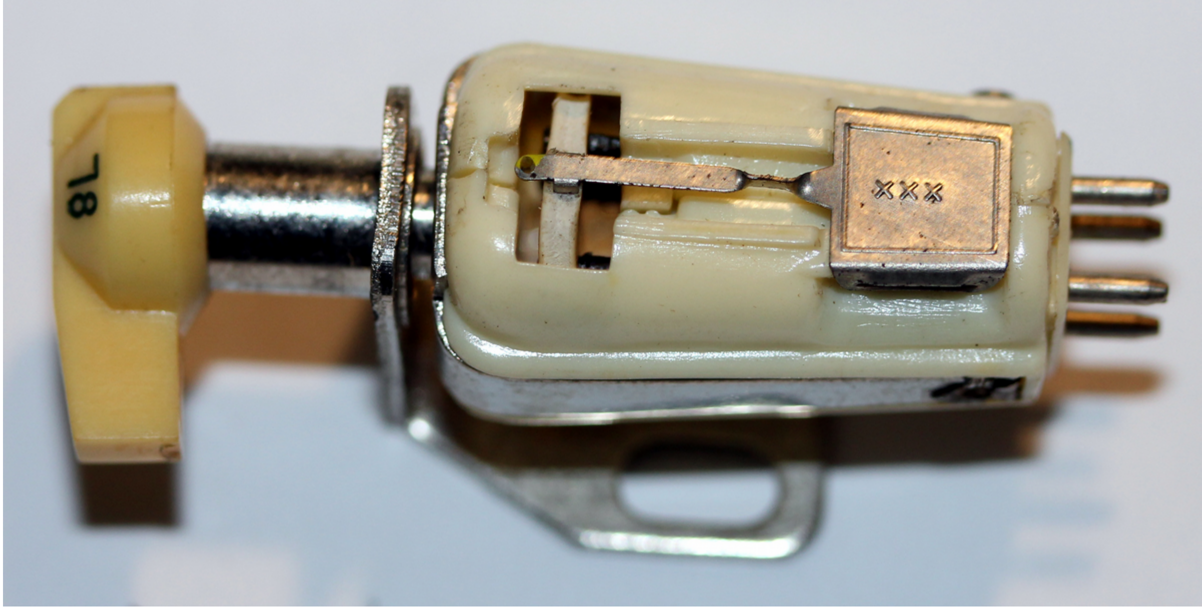
The score is prefaced by an extensive explanation on the indication of duration's, station tunings, dynamics (numbers ranging from 3 to 15, 3 being turned on but inaudible, 15 being maximum volume).

Cartridge Music (1960)

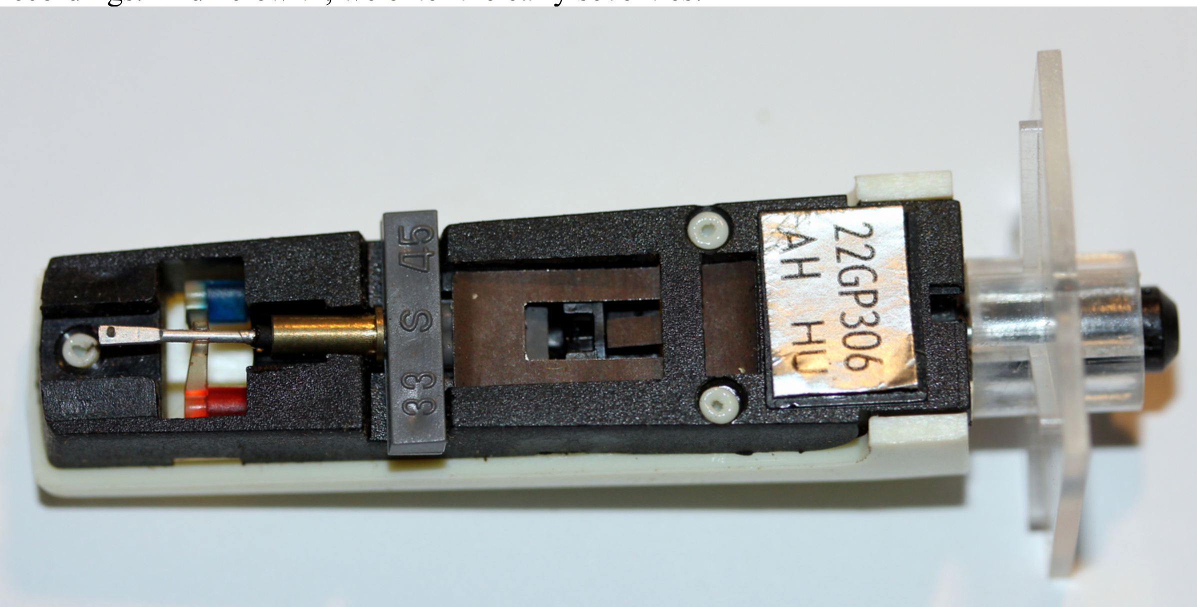
In a footnote in the score, Cage writes: *'A cartridge is an ordinary phonograph pickup in which customarily a playing needle is inserted. Instead of a playing needle, any object that will fit into a cartridge may be inserted (e.g., a coil of wire, a toothpick, a pipe-cleaner, a twig etc.).*

Turntables, certainly those commonly used in the sixties, are harder and harder to find. The cartridge in an old turntable is about the very first component to fail... The main reason for failure as we noticed, is moisture as the crystals used for the transducer are extremely sensitive to humidity. So hunting for well functioning old phono cartridges may have become quite an undertaking by now. Here are some pictures of period-cartridges of the kind used by Cage:

Notice that these cartridges even have a small screw wherewith the needle is to be secured in the cell. This makes attaching other objects as Cage suggests for this piece, an almost trivial undertaking. Also note that all such cartridges are mono, as they date from before the time stereo records were introduced. Cartridges from the mid-sixties look like this:

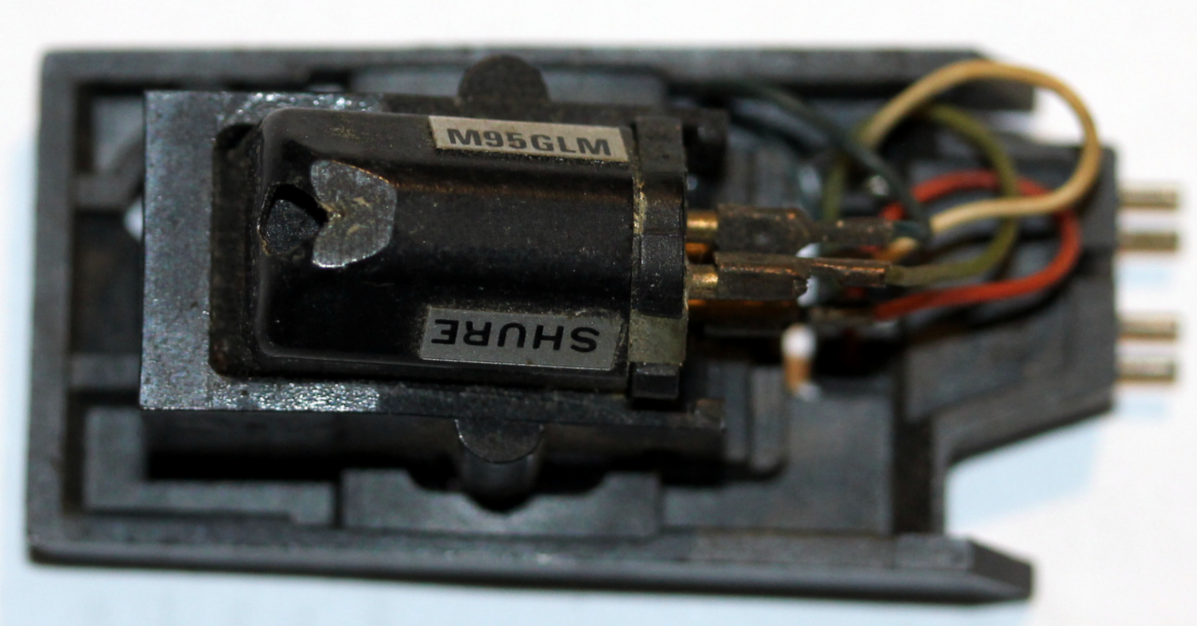


Mono cartridges became very rare and only made for the reproduction of older gramophone recordings. And herewith, we enter the early seventies:

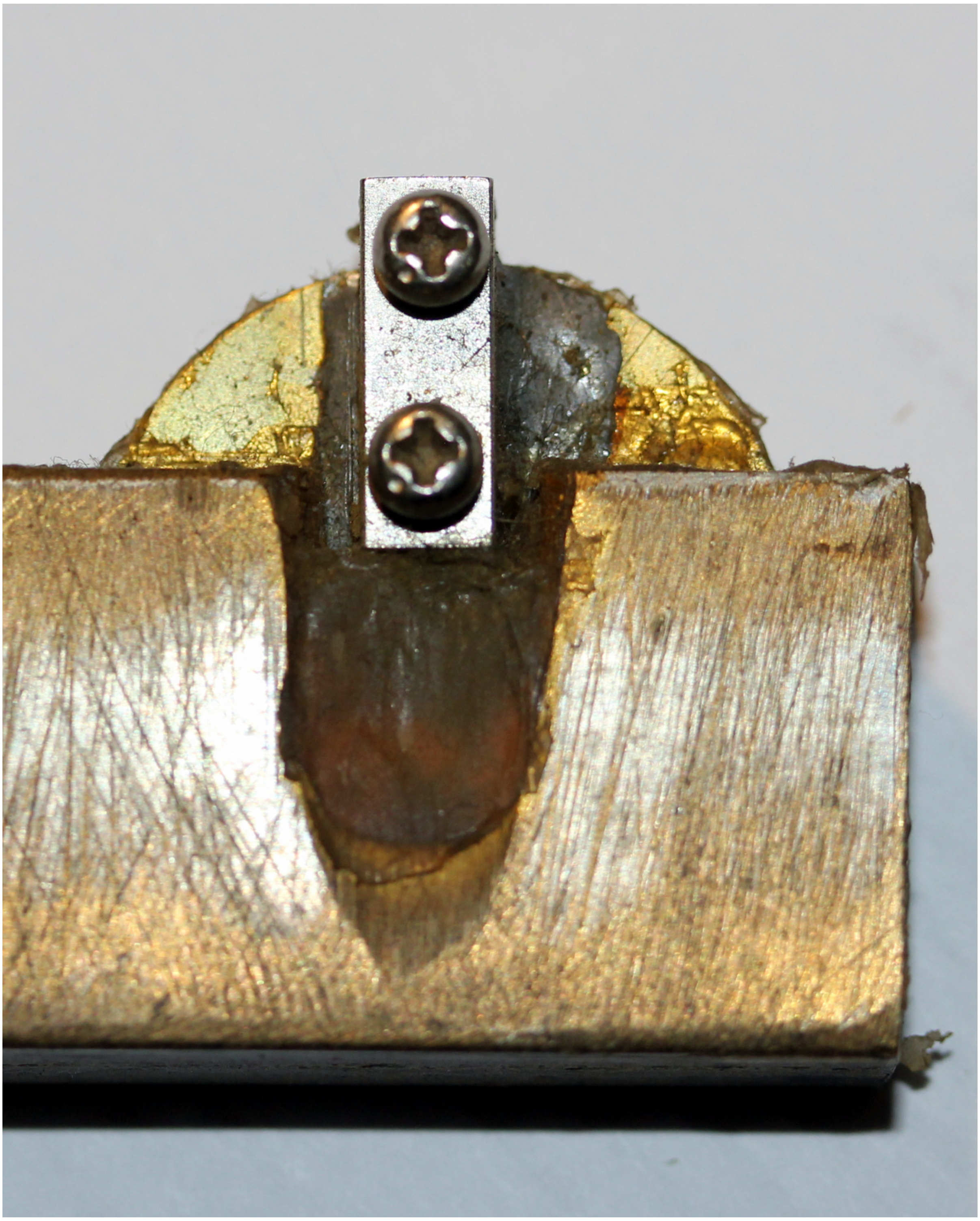


As one sees right away, its already a bit more tricky to secure other objects. Still it's feasible with a bit of handiness.

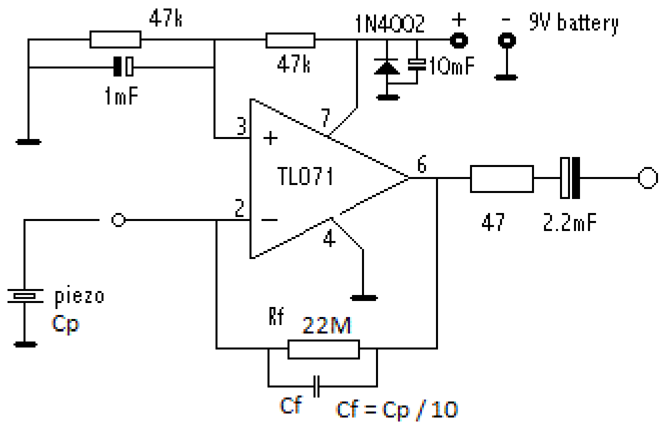
Good quality modern cartridges as made by Shure, Pickering, Orthophon and the like, are still made today in answer to an unexpected popularity of vinyl in some circles of audio afficionados. These high quality and expensive types -always stereo- are very hard to use for performing this Cage piece because they are very fragile and too light to be handled in a performance situation. It's difficult to replace the stylus / needle assembly for inserting other objects. In no time, the cartridge is ruined by doing so. Here a picture of a Shure cartridge, the needle removed:



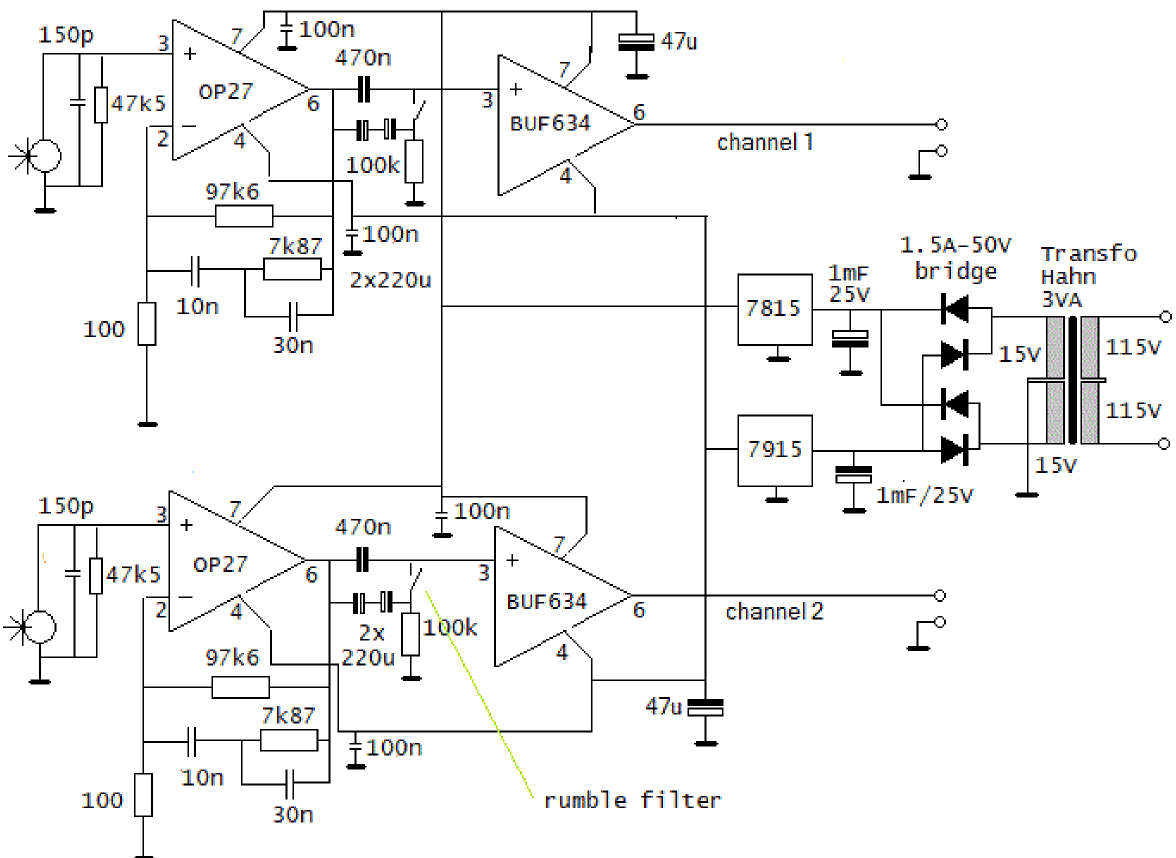
If it appears impossible to obtain original cartridges, there are acceptable alternatives, requiring some construction and simple circuit building. Piezo electric material can be used, in combination with a suitable preamp. Here is a picture of a very simple yet sturdy construction, using a Piezo disc, clamped in silicone rubber and provided with a screw mechanism to attach all sorts of objects:



And this is a circuit for the buffer preamp to go with it:



If original cartridges are used, of the magnetic type, one should use a circuit like this one:



Precision RIAA-preamp for pick-up cartridges rev.03.2019

This is designed and drawn for a stereo magnetic cartridge, providing in proper equalization as specified in the RIAA standard for phonograph recordings. Modules with spare preamps of this type are still available on the market, one of the reasons being that many modern amplifiers do not have the required inputs anymore for connecting turntables.

In the score instructions Cage is very clear about the idea that each cartridge and player should have its own amplifier and loudspeaker in close proximity. The amplifiers used should have individual volume as well as a single sound control. The latter used to be a simple low pass filter. The control knobs must be of the rotary potentiometer type, as a scale for them to use is included in the score. The amplifiers are preferably to be setup around the audience. So, the use of mixing desks is certainly a wrong practice for this piece. It kills the sonic transparency of the piece.

A good source for information with regard to old phono-cartridges is: James Moir, High Quality

Sound Reproduction, ed. Chapman & Hall Ltd., London 1958.

Bird Cage (1972)

Twelve tapes to be distributed by a single performer in a space in which people are free to move and birds to fly

Apparently, the tape recorders Cage envisaged for this piece are mono machines, nominal speed 7 1/2 ips (19 cm/s), each with an amplifier of its own. Cage explicitly states that other machine speeds may be used. The performance, for and by a single person, consists of manipulating the tape recorders: switching on and off, swapping tapes, rewinding... Also it is clearly not meant to be a 'concert' piece but much rather what is called nowadays an 'audio art installation' or a soundscape.

Alternative technologies that can be used here: analog cassettes, DAT tapes. Digital computer based technology is not appropriate for performances here.

Telephones and Birds (1977)

For three to perform.

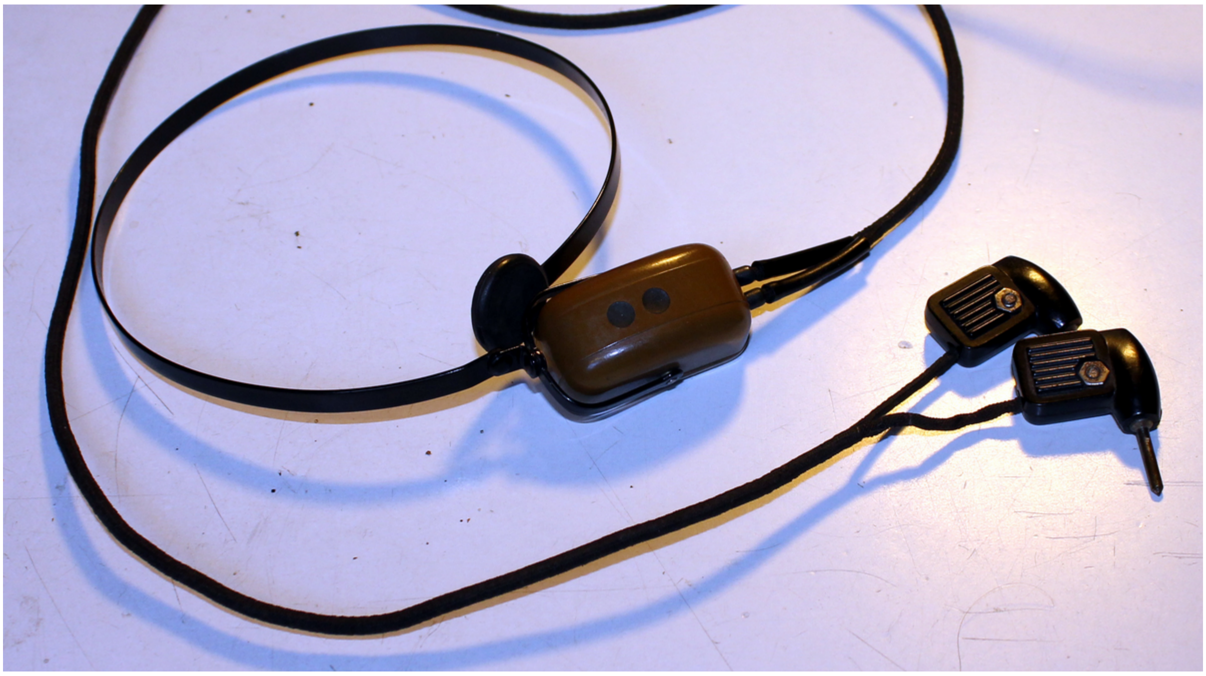
Next to calling for tape recorders -their noises are declared explicitly part of the performance- this piece calls for telephones, yet another technology that has changed substantially over time: dialing a number, automated answering machines, ring-tones, taking up the horn... all these once so typical sounds for telephony have changed. Also the list of phone numbers, 17 USA numbers for the National Rare Birds Alert Network, specified in the score has not a single number that is still connected now.

Trying to perform this piece with period telephones is vain, as technology made old phones obsolete. They can no longer be used to connect to the phone network. This piece, if one judges it worth the effort, ought to be recomposed. Using chat sessions and the Internet may be a valid way to go. However, in performing, it should be made clear to the public that it is an adaptation and that the performance as presented may deviate heavily from any period performance.

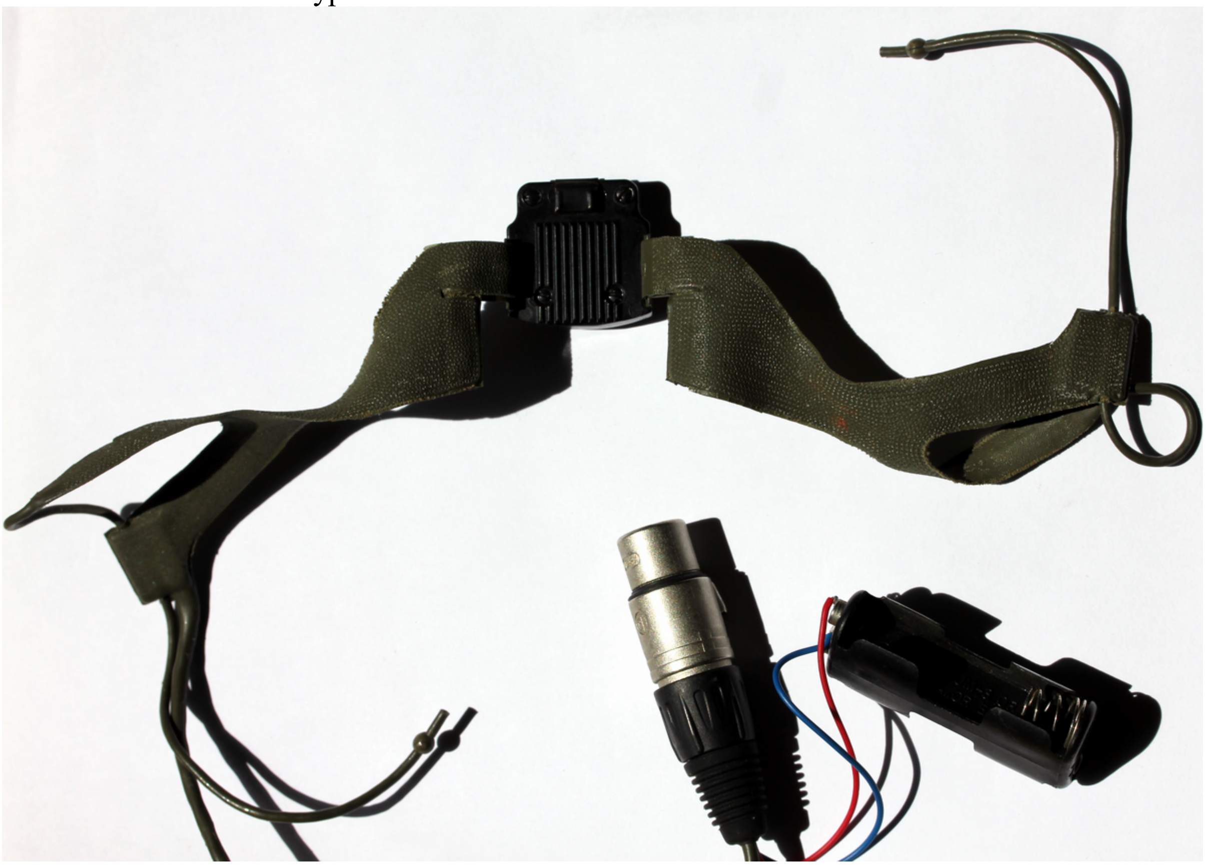
Next to the scores explicitly calling for electronic and technical devices, Cage has published a whole lot of scores and score materials, wherefore electronics of some kinds seem particularly indicated. The series of 'Variations' is a good example. In his own performing practice, simple electronics have more often than not, been present. The many concerts he gave as a composer/performer, very often were collaborative projects with soulmates such as David Behrman, Robert Ashley, Alvin Lucier, Takehisa Kosugi, Frederic Rzewski, David Tudor, Gordon Mumma and a few more. The staging for most of them consisted of one or more tables filled with electronic devices. We remember assisting to one of his concerts in Los Angeles in the eighties, a production with Merce Cunningham, where we saw Takehisa Kosugi handling a soldering iron on stage as part of the performance... Contact microphones, small mixers, modulating devices, gating devices, photo cells triggering oscillators, tape recorders, oscillators, radio circuitry, amplifiers as well as toys and gadgets formed the gamut of sound sources for the performances. We do not remember where the picture comes from, but it gives a pretty good impression of the stuff used. We see Frederic Rzewski and John Cage performing one or another Cage piece:

A device of particular interest Cage used (a/o. reported by Alvin Lucier) is the throat microphone. We state it as of interest, because some other composers also used and prescribed these special microphones in their scores. The most wellknown piece here could very well be 'Maulwerke' by the German composer Dieter Schnebel. But we have also seen it used on stage by artists from the sound poetry scene, such as Henri Chopin, Lilly Greenham and Gust Gils. Throat microphones in origin are strictly military devices. Their invention and application was an answer to the problem of pilots in military aircraft (bombers and suchlike) in communicating over their radio system with commanding ground. Normal microphones, even placed extremely close to the mouth of the pilot, were pretty useless as the sound level of the engine in the cockpit made it impossible for a voice to get over it. Thus the idea of picking up the vibrations of the larynx directly arose. In fact it can be considered to be a contactmicrophone, designed to pick up larynx vibration as conducted through the bony structures in the human neck. Many models and types were made throughout world war two. Carbon mikes, electromagnetic types, crystal type etc... Do not expect a throat microphone to deliver a clear and nice vocal signal! Even understanding speach with it, requires a lot of practice from the side of the listener. This is because the articulation through the mouth cavity is largely absent in the signal. Throat microphones can still be found in the catalogues of professional microphone manufacturers such as AKG. Here are some pictures of the historic types as used by composers and performers:

This is an electromagnetic type:



Here we have a carbon type:



In order to give out a signal, it needs a 1.5V battery. Carbon mikes, being vibration-dependent resistors, do not give out a signal unless fed with a current. The output of both these types is balanced and low impedance, thus they can be connected straigth into any microphone input on an amplifier or mixing desk.

Throat microphones can still be found on flea markets and in army-stock warehouses.

For interactive applications -more specific, for productions with Merce Cunningham and his dance company- Cage has been using photocells. He nowhere gives more specifications about them, but we are sure he confined the technical details to his collaborators some of which were very knowledgeable in electronics: David Behrman, Takehisa Kosugi, Gordon Mumma to name just a few. Although one may encounter original photocells from the fifties, it will be of little use to try them out: photocells , as used for picking up optical sound in film projectors with an optical soundtrack, are gas-filled vacuum tubes. They are severely sensitive to aging and the sensitivity of them now, is -after our own findings and measurements- worse than 40dB down. In the industry gas filled photocells have since long been replaced with by far superior semiconductor types. Installations and scores mentioning photo cells should be re-engineered with a good understanding of the function they fulfilled in the period-equipment or setup. For many applications, LDR's can be used although it must be said that in as short time from now, they will also become obsolete and their production even outlawed as they contain a substance toxic to the environment: cadmium sulfide. Phototransistors can be used, but their use is a little more involved: they need some kind of a circuit and their output is a voltage or current. Often, for instance if the photo cell is used as a sound generator, picking up amplitude modulated light, an excellent solution consists of using a solar cell. These are in their common appearance in general way too large, but it's quite easy a break off a small part, and solder connections to it. Note that a special solder must be used to do this! Solar cells thus prepared, can be connected straight into an audio mixer. The French artist Jacques Dudon uses them as the core of his optophonic concerts and installations.

Electronics in the scores of Alvin Lucier

'Vespers'

For this piece, Alvin Lucier used ultrasonic transducers of the kind that emit a strong and short burst of ultrasonic sound. After the end of the burst, a timer is started in the circuit. That timer is stopped the moment that an echo is received. The time interval is thus proportional to the distance from transducer to object that caused the echo. Lucier, for this piece, was not interested in the function of the device as a distance measurement thing, but much rather by the fact that, if the bursts were strong enough, the echo could even be heard by human ears. This is of course caused by the fact that at the onset of the burst -normally a finite number of complete sinewave periods- is a spike and thus a broad spectrum signal. Even though the frequency is ultrasonic and as such inaudible for humans, the start of the series of pulses can be heard as a click or a spike. Such spikes can be used to get echoes from objects not too far away, as long as they are reasonably reflective. That's exactly what Alvin Lucier is exploiting in this piece. In his own descriptions for 'Vespers', he mentions the device as a Sonplus [check] , experimental prototype one of a kind. He also affirms he didn't want to let other people use it as he considered his device unique and irreplaceable. As we did quite extensive research into the area of ultrasonics, we found that in fact there are quite a few such devices on the market (Pepperl+Fuchs, being one of the best known brands) and, moreover, it's not extremely difficult to build such devices, if only the burst property is of interest.

'I'm sitting in a room'

This piece could also work with other technology, as the gradual deterioration of the signal is only to a certain extent caused by the lower S/N-ratio of tape recorders as compared to pure digital recording devices.

'Brainwave pieces'

Gordon Mumma

mesa

David Tudor

Although David Tudor as he performed, had almost invariably a stage full with a diversity of electronics devices. Much of it, he designed and made himself. He used to house his circuits in plastic household boxes. However, he kept his circuits secret and thus it becomes pretty difficult to reconstruct. He clearly didn't want others to perform as he did. It's not that he wasn't communicative, but he simply didn't disclose nor document his musical tools. Even his close friend Alvin Lucier witnessed this. (Lucier, Music 109, p.62-63). We know that ring modulators were very often present in his set-up and that many of his modules were very similar to what can be found in a lot of modular synths from that period: VCO's, VCA's VCF's, oscillators of different kinds and waveforms...
One of the best known pieces for his certainly is 'Rainforest'. In our opinion -we have assisted to quite a few presentations, both with him as without him- this piece should be considered more or less 'conceptual', and every realization is in fact a reconstruction, if not even a recomposition.

David Behrman

'Runthrough'

As described by Alvin Lucier (Music 109, p.77) , this improvisational piece makes use of photo resistors placed into empty tin cans of Campbell Soup. The players use flashlights to cause sounds to spin around on a four-channel loudspeaker system. Voltage control is used to steer modular synth components of his own invention and making.

wave train

Robert Ashley

Georges Antheil

'Ballet Mecanique'

To meet the requirements for this composition, we decided to go for a fully automated robotic solution: the <Balmec> project. Problems connected to performances of this composition and that we tried to solve by designing this <Balmec> project are:

1- the required player piano's
2- the three required airplane propellers
3- the sirens
4- the electric bells

Balmec project: <Bello>

Common electric bells use an interruptor mechanism to get fast repeating strokes against the bell. As this interruptor breaks the current through the coils, a back EMF is generated. If you look closely at the interruptor contacts you can easily see the sparks this EMF generates. The voltage generated when breaking the current through a coil can easily become larger then ten times the voltage applied over the bell. This explains why, even when feeding a bell with an ordinary 4.5 or 6 V battery, one can get an electric shock when touching the bell wires.

The effect of the spark is not only that eventually it burns the switch contacts but also that it generates quite a lot of HF radiation. Radio receivers are very sensitive to this, particularly in the AM bands. The explanation for this is that a sudden current change through the coil causes a spike voltage over the switch. The sharper the edges of the spike, the more HF radiation (and the broader its spectrum) is produced. (Fourier analysis of a spike...)

In order to reduce and damp these sparks, the better bells have a capacitor in parallel with the interruptor contacts. The value is in the range of 47 nF to 150 nF but the voltage this component will have to withstand must be twenty times the working voltage of the bell. To save on capacitor expenses one often sees a capacitor in series with a small resistor (around 470 Ohms). This reduces the voltage over the capacitor by limiting the charge current. The time constant of the circuit (the RC product) must be larger than the period of the interruptor if all sparking needs to be damped. A combination of 100 nF and a 470 Ohm resistor gives RC= 47 us and will only damp HF radiation higher than 21 kHz. With an AC capacitor 33 uF and a 680 Ohms resistor, one can obtain full damping at 44 Hz. However, it comes with a price as bipolar capacitors in this range are pretty expensive and large as well.

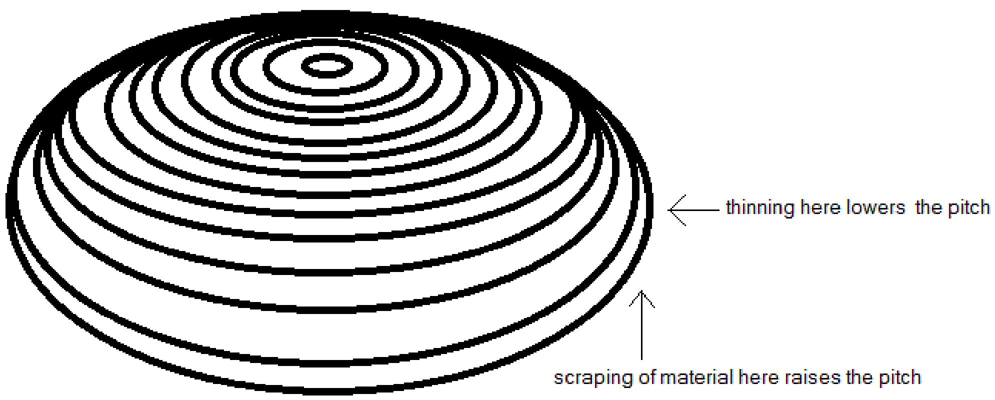
Bells that work using an interruptor will have a repetition frequency determined by the mechanical properties of the contact assembly. The larger the trajectory of the interruptor, the lower the repetition frequency. The size of the trajectory is -at least on older electric bells- adjustable with a small screw. Other parameters are the mass of the beater and the strength of the return spring. The mass of the beater shouldn't be played around with by the user, as this is optimized at a certain percentage of the mass of the bell to be struck. Rule of thumb here is 10% of that mass. As to the return spring: most often its a blade spring. On old bells very often this spring will be rusted and it may have become too weak. It is very difficult to find a replacement spring, also because working on spring steel is problematic.

Acoustic alarms such as bells, car horns, sirens, ship horns etc. Have been used every so often by composers ever since these devices existed. One will encounter electric bells and sirens not only in George Antheill's 'Ballet Mechanique', but also in Erik Satie's 'Relache', in pieces by Mauricio Kagel, Alvin Curran, Llorenc Barber, Anton Riedl, Davide Mosconi, Dick Raaijmakers, Gyorgy Ligeti, to name just a few that come into our mind.

Other than what the Antheil score publisher states (7 different sized bells required) the score prescribes pitches, notated in the treble staff, and more than 7 bells. Occuring pitches in the score are: 69, 73,76, 77,78,79,80, 81, 82, 83. This makes 10 bells, not seven! For the construction of an automaton to solve this problem once and forever, we generalized the concept such that we could offer a more continuous range of pitches. Finding suitable industrial electric bells -in construction and sound as close as possible to what Antheil could have had in mind- was not a trivial matter. Moreover, the mechanism of the bells we got from Infrabel -the local railway company- (Funke and Friedland) have a mechanism using a spring such that the actual hitting of the bell happens at the release of the electric pulse driving it. This has quite some implications for the firmware to drive these mechanisms.

Obviously if the pulse lengths get longer than 10ms, there would be a noticeable latency. The tuning of the bells is possible using a regular column drill and a clamped file. The ideal tool for doing this is a vertical lathe, but not too many people do have such equipment available... The rules are clarified in this little drawing:

Tuning procedure for shell bells

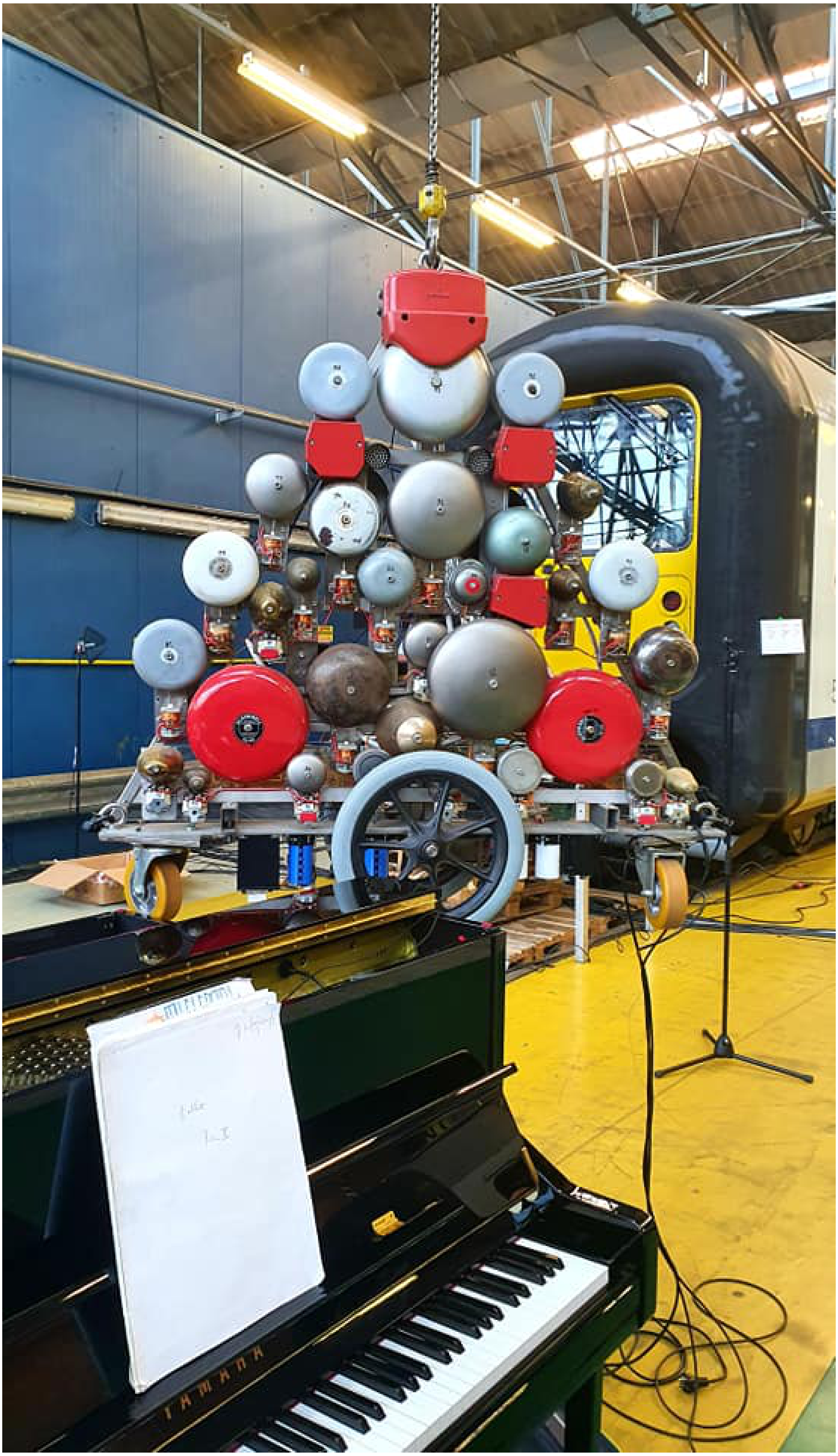


The range for tuning is pretty limited. Lowering the pitch can be done up to a semitone. If you go lower, the sound volume will suffer as the material gets too thin. Raising the pitch can be done up to a minor third. Welding on the rim of shell bells made of steel or stainless steel never gives good results. The bell looses all resonance because of the unavoidable deformation of the shape.

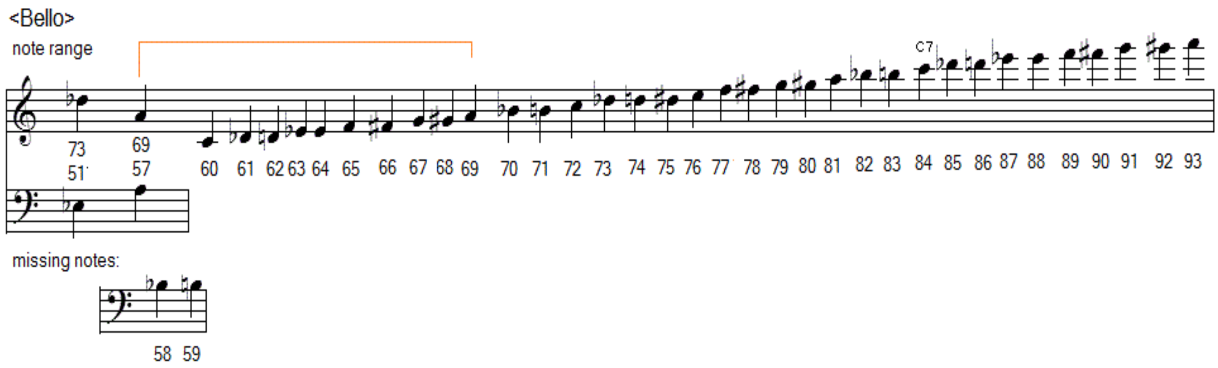
The note mapping for <Bello> is given under 'midi implementation' further below. The notes indicated as missing may be added at a later stage, when we can find suitable dome bells. For the notes 62, 63 and 66 we used U-shaped pieces of steel tuned to the right pitches. These bells have different sonic qualities as they are not real dome bells. Users of our robot orchestra that are really in need of the missing lower bells, can use the bells on our <Llor> and/or <Belly> robots. For an alternative F# (note 66) the <Harma> robot includes this one as well. Note that this robot uses mostly bells from very different origins and of very different composition, hence their sonic qualities are very different. It was not our intention to create a homogenous instrument in terms of sound color.

Note that it is very important to decrease the velocity values as the note repeat frequencies are increased. Not only will the bells not sound properly, but moreover, there is a high risk of burning out the coils. The safe maximum value for key-pressure is 106. This value guarantees the duty cycle to be lower than 50%. However, users should not consider this to be the optimum value. At high repetition rates, the lowest possible velocity values generally sound best. The highest possible repetition rate corresponds to that of a 60 Hz American AC driven alarm bell. European AC driven alarm bells sound at 50Hz. Ordinary AC/DC bells using an interruptor mechanism have much lower repetition rates. Modern bells have faster repetition rates (due to their lighter construction) than antique ones. For quite a few 19th century electric bells, we measured repetition rates as low as 4 Hz under normal operating conditions.

Here is a picture of the <Bello> robot, ready to be fully suspended for a performance of Antheil's Ballet Mecanique in the village of Mechelen, march 2022:


YAMAHA

<u>Midi implementation:</u>



- Channel 13
- Note-on commands: the velocity byte steers the force wherewith the bells are struck:
- Note-Off or Note-on with velo = 0 turns the bell off
- Key-pressure commands are used to set the repetition speed of the bell strokes. These repetition speeds (from 1Hz up to 60Hz) are individually programmable for each bell. When set to zero, the bells will not repeat but produce a single stroke. Note that the key-pressure commands are sticky and are memorised for each note.
- The two red lights in front are mapped on notes 126 and 127, the velocity byte steers the flashing speed.
- The red light underneath the front is mapped on note 119, the velocity byte steers the brightness.
- Controller 30: This controller can be used to set all the repeat frequencies to one and the same value for all bells. It is a quick way to set the repetitions rates and an alternative for the key-pressure commands. However, the time between sending this controller many times should be kept reasonably long, as it requires reprogramming of all lookup tables in the firmware. It is advised to send this controller at a time when no bells are sounding. If this is not done, glitches and irregular performance may become audible. Note repetition rate is controlled by the parameter value. To switch repetition off, a zero value should be send.
- Controller 66: enables (>0) or disables (=0) bell operations. CC66 = 0 resets all key-pressure values to zero.
- Controller 123: all notes off, stops all bells, preserving the key pressure values and thus the repetition rates..

Balmec project: <Propellers>

George Antheil (08.07.1900 - 12.02.1959) was a great admiror of the futurists ideas. Hence he introduced quite a variety of industrial noises in his orchestral music. The ballet mecanique, a ballet for machines, prescribes no less than three airplane propellers on stage together with a battery of seven industrial electric bells, some 16 player pianos, a siren and percussion. It was written between 1923 and 1924. All too often, orchestras performing this music fake these essential components either by subsituting them with percussion instruments, or even worse, by sampled sounds reproduced on loudspeaker channels. Needless to say that this goes against the composers original intentions, although during his entire lifetime he never got these components working as conceived. Just like Igor Strawinsky in his original version of Les Noces, he even never got the player piano's to play in sync... Nowadays it ought to become possible to realize all those ancient dreams, even though the composer during his lifetime has compromised his own ideas on many occasions. In some performances of the Ballet, large fans have been used and -as these devices are pretty noiseless- the composer had the players hold sticks against the blades... This clearly doesn't lead to anything like an airplane sound However, having working real airplane propellers on stage was and still is, not a trivial undertaking. The construction of these elements, such that they can safely be used in performances, was confined to us as a collaborative project with the local Ictus ensemble.

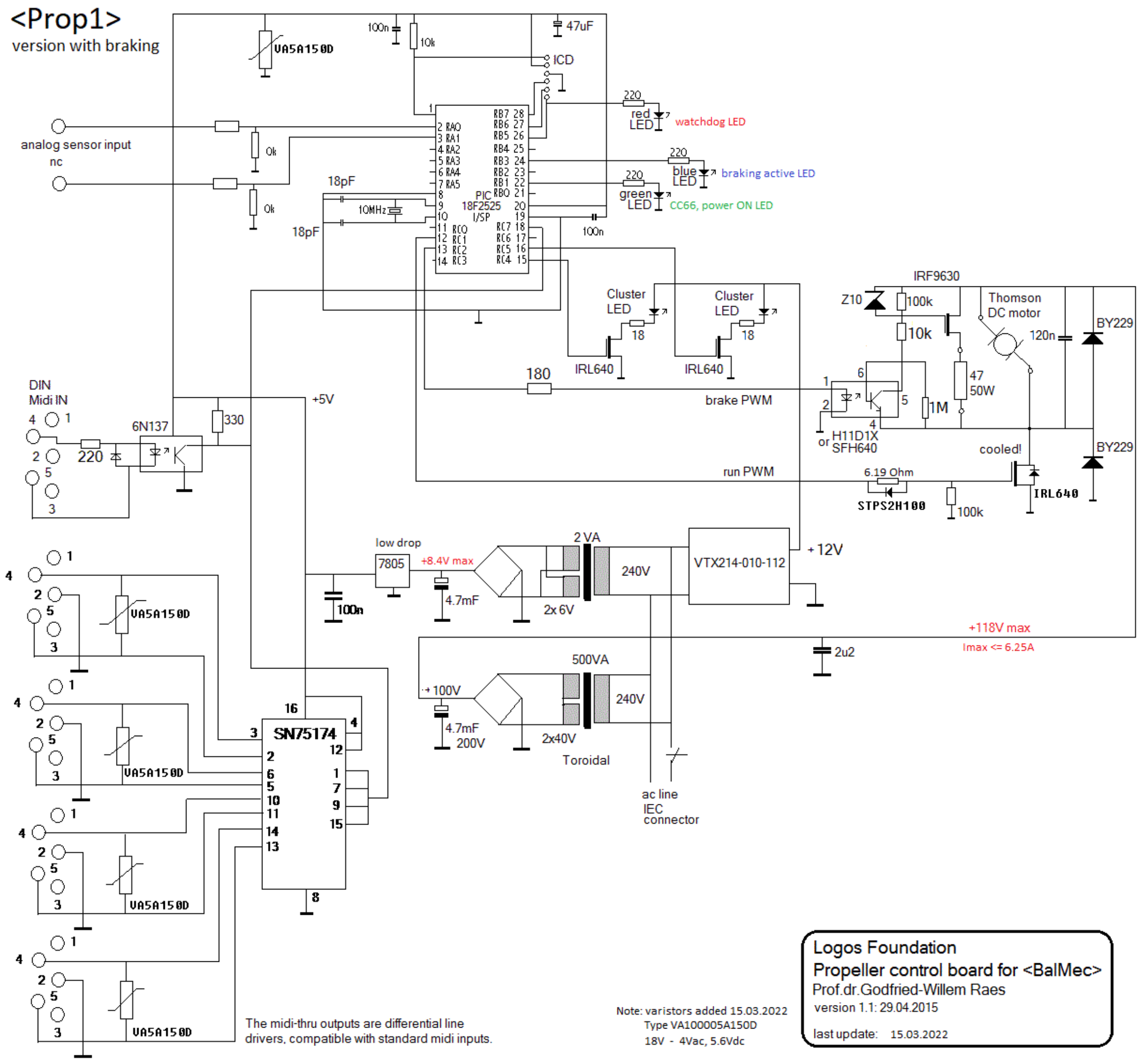
Propellers:

We started off by tracing suitable real air plane propellers -not fan blades- and studying the mathematics and physics of their behaviour, as obviously having them rotate at the normal speed as on an air plane would entail very high thrust forces to be developed. Prohibitively dangerous. Hence we designed the motors such that all forces developed are in a safe range. Also we designed the structures such that they produced blowing wind, instead of sucking wind as in aircraft. Doing so, the forces are always developed backwards. The artistic problem is that at too low speeds, the propellers do not make the air plane noise requested and wanted by the composer. This lead us into researching the shaping of the blades such as to make them produce more noise. Another research topic was to consider the possibility to let the fan blades closely cross the edge of low pitched resonator tubes, thus provoking the typical low frequency noise of an aircraft propeller. This also makes possible the rhythmic notation used in the score. On propeller 1 such a resonator was build. The result is very convincing. A similar design on the large propellers was impractical as the resonators would become physically to large. Maybe Helmholtz resonators could be applied here.

Propeller 1:

This is a rather small propeller, span 660 mm (26"), carved from wood. The motor used here is a DC motor, making precise control relatively easy. The motor rotation is transmitted to the propeller axle with a V-belt. The gear ratio can be changed by mounting different pulleys. An adaptor piece was turned on the lathe to make the propellers central hole fit the axle. This was fabricated from a piece of nylon, outer diameter 30.0 mm, inner diameter 14.5 mm, length 45 mm. The propeller axle is mounted on a steel holder made from a piece of HEA 100 x 100 profile. The axle is mounted on this part with four M10 x 35 bolts. The motor base -in construction steel as well- is cut out from a 450 mm long piece of 100 x 50 x 4 rectangular profile, also serving as a resonator. At the back end of this tubing, we constructed an acoustic horn to amplify the sound in the 80Hz to 130Hz range. The horn is folded, with the opening pointing to the audience. This yields a quite convincing airplane sound, although below the sound pressure level of a real prop-engine.The motor power was calculated to stay below 10% of the nominal power required for use on an airplane (estimated at some 5 to 8 kW). The whole structure was firmly welded together.

The electric control of the propeller is not as easy as it might seem. As long as we only have to cope with very slow changes of rotation speed, we can live with just variable voltage control on the motor. However, if we want relatively fast braking, we have to deal with the problem that the motor -due to the inertia of the propeller- will become generative. This imposes the use of braking resistors and precise electronic control. Hence the PIC microprocessor (an 18F2525) needs two PWM controlled output channels: one for speed control, one for braking. Also the analog input channels can be used to monitor motor -and thus propeller- behaviour at all times. This is the circuit we designed for the MIDI control of the propeller:

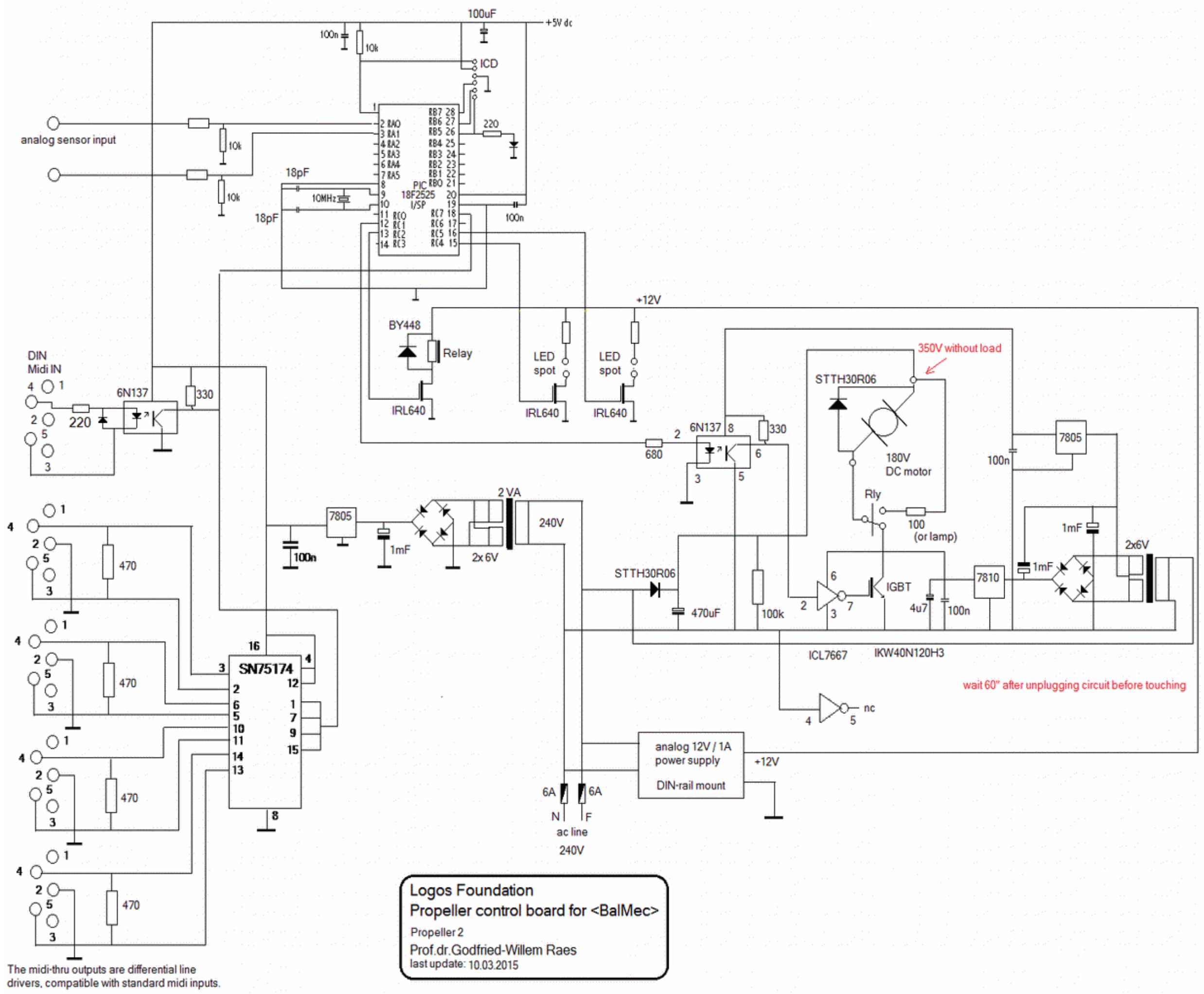


As we anticipate that in practical use, the distance between the different components of the setup might become relatively large, possibly exceeding the 5 to 10 meter limit for MIDI cabling, we provided in differential line drivers on all MIDI boards. With these, cables up to 100 meter in length can be used. For reliable performance, screened twisted pair cable should be used. DMX cable, properly terminated works very well.

Propeller 2:

This propeller is also made of wood, but coated with polyester and carefully balanced in the Hofman propeller factory. The wind span is 1890 mm. Size of the axle mounting hole: 58 mm, provisions made for flange mounting with six M12 bolts. The motor used to drive it is a GPM90, 0.75kW DC motor designed for 180 V DC operation at 1500 rpm. This motor is powered from the mains single side rectified voltage directly. This is the circuit for the control:

< Prop 2>



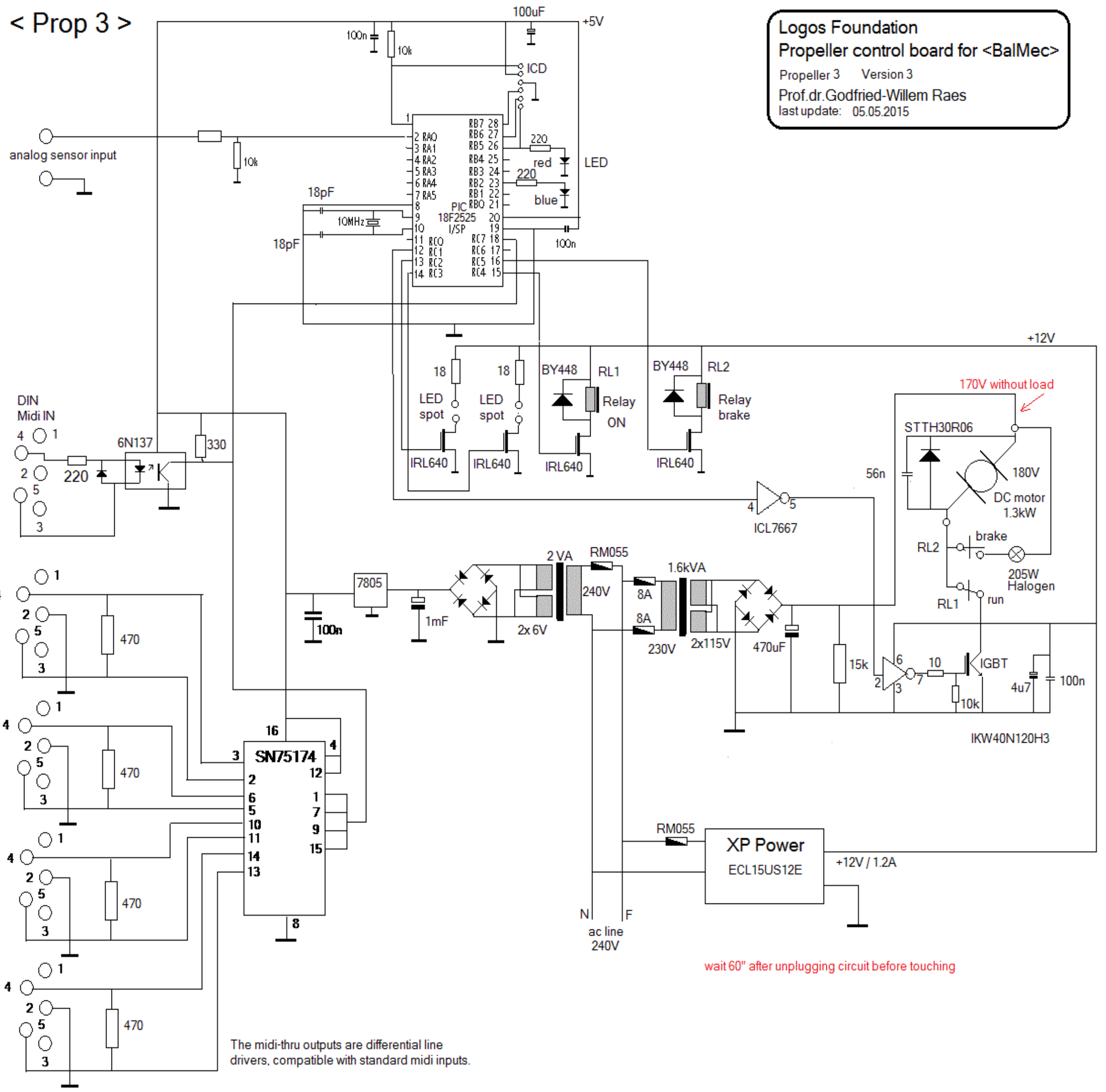
As parts of the PCB are directly coupled to the mains voltage a word of warning may not be misplaced here: this board does carry high voltages! Do not touch. There is galvanic isolation between input and output, so using the circuit involves no danger. The steel structure itself is properly grounded. There are two automatic fuses on the machine making it possible to cut all power, however these fuses should not be used as a switch taking into account that being in such close proximity to the propeller entails a danger in its own, For safety reasons we advise users to use a switch in the power wire at least 5 meters away from the engine.

The motor for this propeller is a flanged type, so it was a lot of work to construct a well fitting flange to fit the motor on the HEA220 profile base. The centre hole has to be 130 mm diameter and the M12 mounting bolts have to be countersunk types.

When braking, the motor becomes a generator. To make reasonably fast braking possible we provided a braking resistor switched over the motor windings on a stop command. In fact this resistor is a 205 W halogen bulb (Osram). The normal resistance would be 258 Ohms, but when cold this value is down to 25 Ohms. It is absolutely normal that this lamp will never glow in this application.

Propeller 3:

This is a heavy duty propeller made in metal, presumably a magnesium-aluminium-titanium alloy. The wing span is 1740 mm and the axle hole is 58 mm. As on propeller 2, it is also designed to be mounted on a six hole flange.
The motor used to drive it is a GPM90, 1.3kW DC motor designed for 180 V DC operation at 1500 rpm. The circuit for the control is almost identical to the circuit used for propeller 2, but here we used a separation transformer avoiding the trouble we had with a first version using single side rectified mains voltage directly. Thus many improvements were added to the PCB design as well.. This is the circuit drawing:



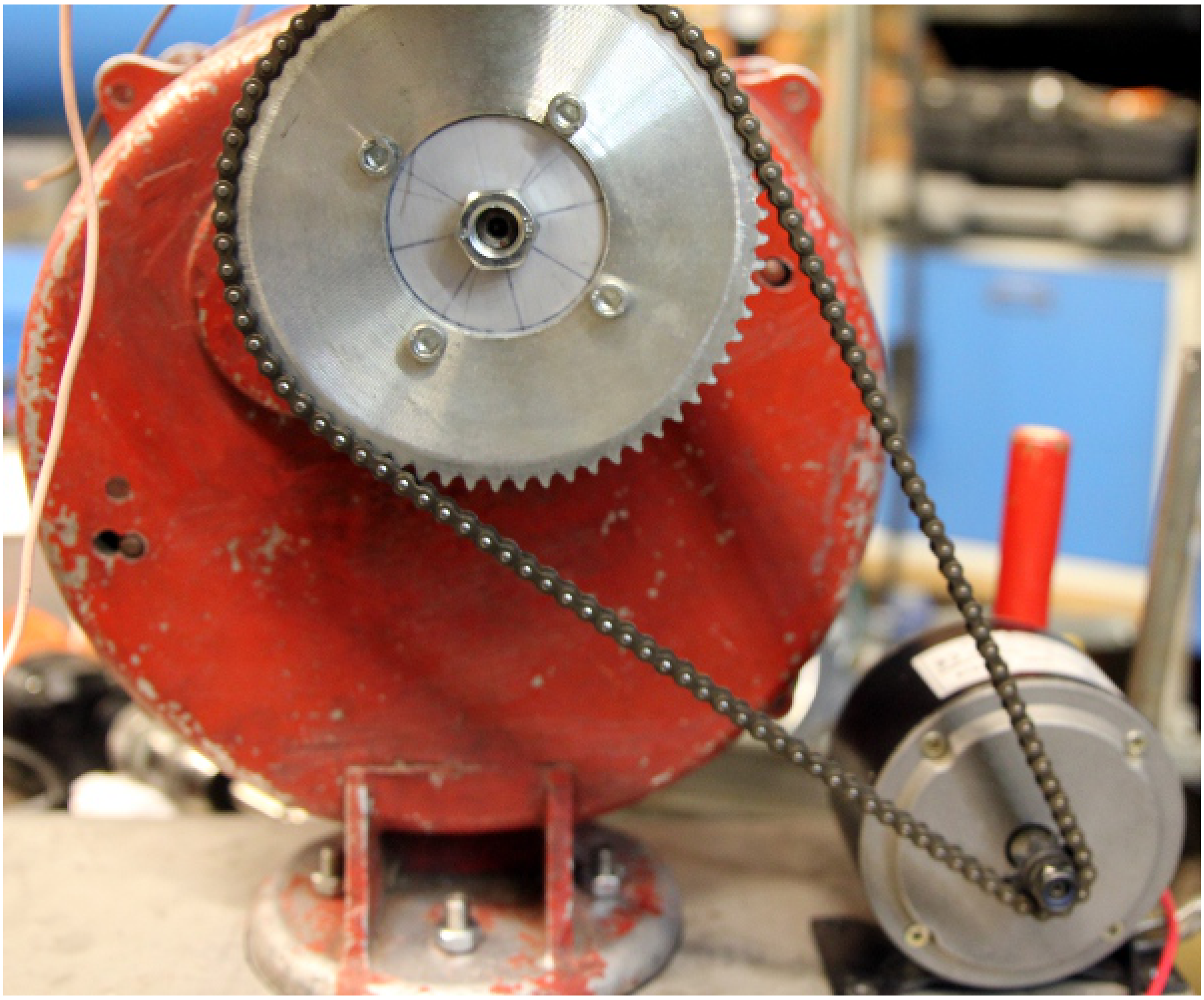
As it is the case with the notation in the score for the siren, it is unclear at what speed the propellers are supposed to sound. It is technically impossible to start/stop propellers fast. We found a solution by providing a switchable resonator for the propellers that can switched on very fast. Thus it would no longer be required to have the propellers themselves to change speed rapidly. As yet, this feature is under study.

Balmec project: <Balsi>

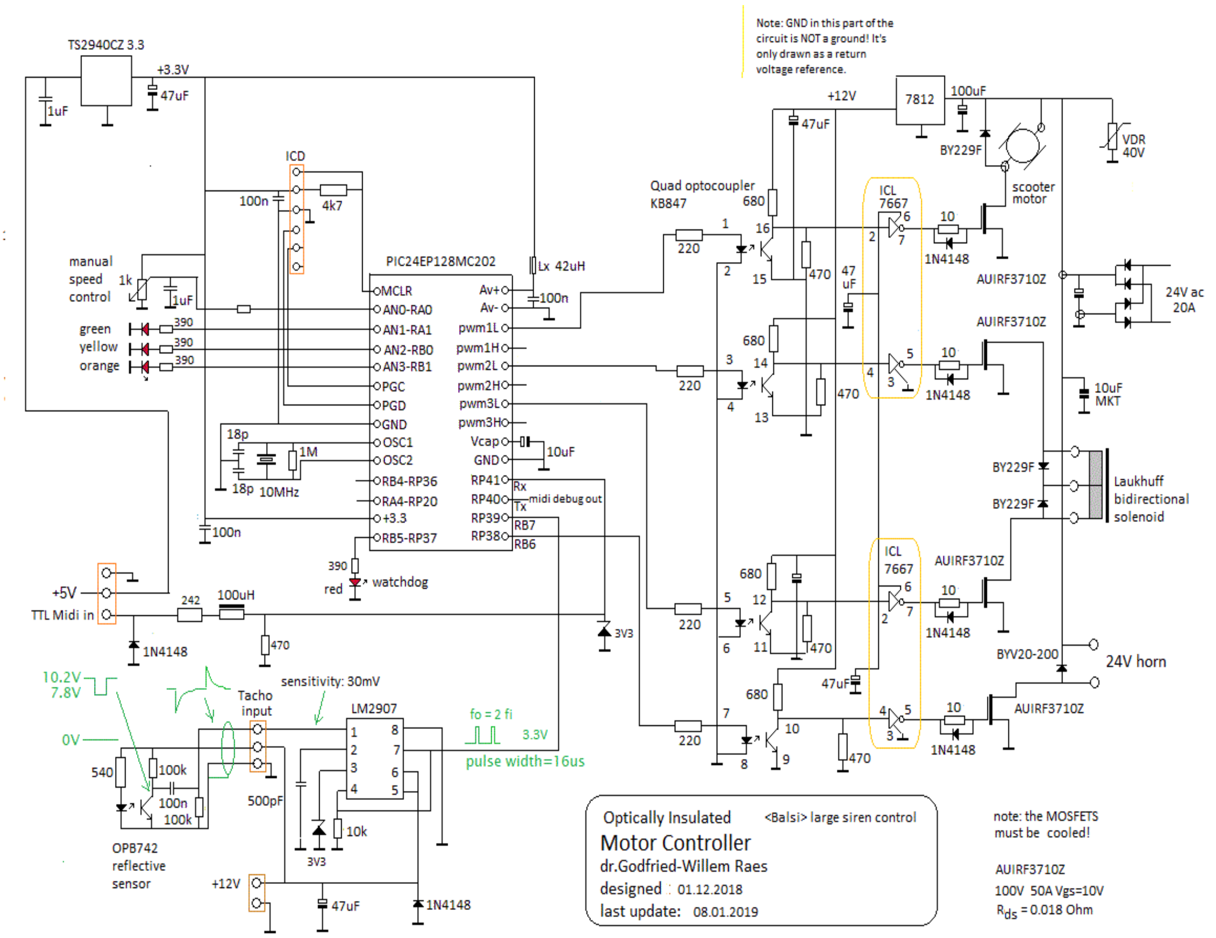
The instructions in the score render it impossible to use a standard crank driven siren, as it is detrimental to the gears in these devices to be started and stopped fast. So an electrically driven mechanical siren with safe braking possibilities or fast sound control has to be designed. The score is very unclear as to the pitches the sirens are supposed to sound. In the score they appear notated as percussion instruments. The siren we used as a starting point for this automation project, before we changed its mechanical construction, looked like this:

It is a heavy Polish made military siren we acquired on the local flea market in Ghent. The handgrip and the crank were removed first. The mount for the handgrip was modified to accommodate a bidirectional solenoid to drive a damper mechanism.

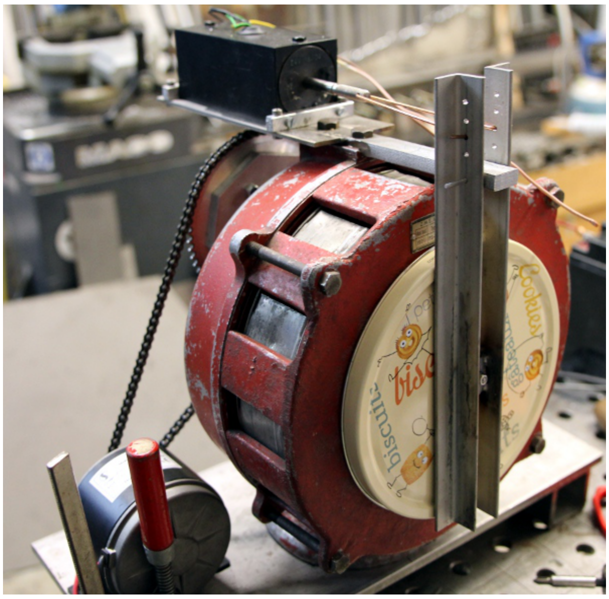
Of course, from a mechanical point of view, driving the sound producing rotator of the siren directly with a motor would seem the easiest solution. After all, this is how electrically driven sirens generally work. However, starting from an existing and historical crank driven siren, this would require an almost complete redesign and balancing of the instrument as we would have to remove the system of dented wheels inside. If we estimate the maximum speed of rotation on the crank as 3 rotations per second, and if we choose a standard motor with 2750 RPM - that is ca. 46 rotations per second, we need belts or gears with a speed down proportion of ca. 1:15. So, if we take a small V-belt wheel on the motor, diameter 40 mm, the driven wheel needs to have 600 mm in diameter. That's way larger than what's readily available on the market... Moreover, frictional losses would become quite large. So, a two step gear, two times 1:4, looked like a better design at first...
Before we tackled this project, we made already a few siren driven robots: <Sire> , a robot using 24 small sirens as well as the large siren integrated in <Springers>. In these earlier projects, we used DC motors and PWM control to drive the sirens. There was no reliable way to control the produced pitch precisely though. After many unsuccessful experiments with gears and AC motors to drive this new siren, we came across a motor from an electric scooter. This motor had a dented wheel and drove the back-wheel of the scooter with a chain. It looked like a perfect solution to the problem at hand here. Here is a detail of the chain solution as set up for the experiment:



Here is the circuit to control this motor:



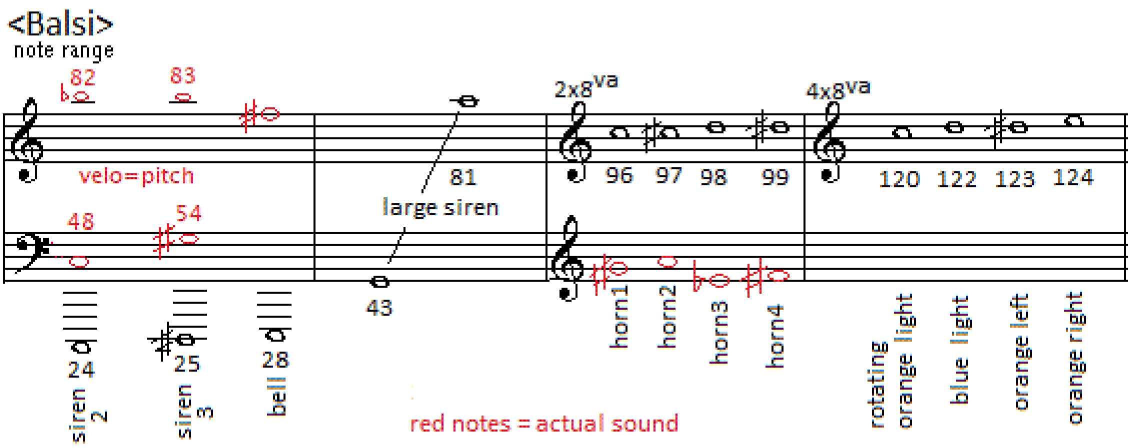
A novel component in this design is the addition of a damper mechanism. An often inconvenient property of sirens in music, is that the sound volume is always proportional to the pitch produced. To overcome this inconvenience to a great extend, we made a damper consisting of a circular plate that can cover the suction side of the siren. The plate is driven by a bidirectional solenoid, mounted in top of the siren. The construction is shown in the picture:



Experiments with the siren running and the damper quickly made us encounter some problems: as the speed of the siren goes up, the suction force exserted on the damper plate rises considerably. To such an extend even, that the solenoid is not strong enough the open the damper any more. Thus it became mandatory to use the solenoid on an over-driven voltage and to provide the firmware with some intelligence to make the solenoid force a function of the siren speed. Another effect we noticed, is that the pitch of the siren becomes a function of the damper position. With the damper closed, and the siren driven with a same voltage, the pitch can be up the a fourth higher. To make precise control of the produced pitches possible, we added a tacho circuit, using the classic LM2907 chip. Here the chip is used in a non-standard frequency doubling configuration. The output pulses are fed to an external interrupt input on the microprocessor for period calculation. A PID regulator was to be implemented in the firmware.

During the design and construction process we decided to add a few more automated components in this <Balsi> robot, to make a more universally useable machine. Thus we added two smaller motor driven sirens. One of them is a universal motor driven siren, still quite loud but way softer than the large siren. The range for this siren is a lot higher than what can be reached with the large siren. The third siren is a 24V DC motor driven siren, capable of reaching 1600Hz. Also we found place to add four car horns and a motor driven electric fire alarm bell. For the large siren, we implemented very precise pitch control using PID regulation and a sensor. Some visual components were added as well: two rotating flash-lights, one orange, one blue and two orange pinker lights recycled from a motor bike.

The complete midi-mapping of all <Balsi> components is here:



So far, all performances of 'Ballet Mecanique' wherefor we collaborated, were done using a MIDI-sequencer program with the tracks for the player pianos as well as tracks for the <Bello> robot and de <Balsi> robot. It would not require a lot of extra efforts, to fully automate the entire performance: an automated and suitable xylophone <Xy> is already available in our robot orchestra, as well as automated drums. Impossible to prove this,. But nevertheless we are convinced Antheil himself would have loved such a fully automated and truly futuristic performance.