



## Acoustic signal sources

What to listen to today?

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# Acoustic signal sources

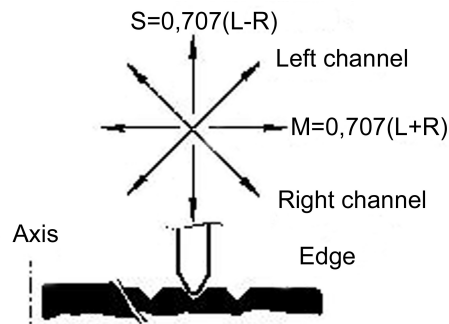
## 1 Introduction

The first link in the acoustic chain for home listening is the source of the acoustic signal. Nowadays, there are many ways to get a signal. The subject of this article is an overview of several available options and a simple evaluation of their basic properties.

## 2 Gramophone



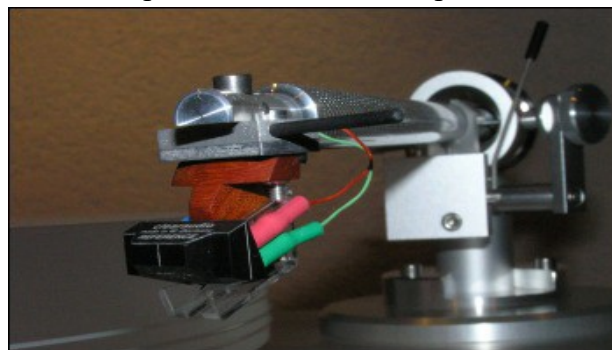
Recording on gramophone records was the first to be commercially accepted. This lasted for many decades until the first tape recorders began to appear. The signal recorded on the gramophone record is an analog signal which is recorded in such a way that the analog signal, after appropriate amplification, is converted via a special electromagnetic circuit to the mechanical movement of a sharp needle which records the analog signal into the gramophone record. In order to avoid as little distortion as possible when dredging the analog recording into the board, the force acting on the needle must be as high as possible and the resistance to dredging the recording as low as possible. Therefore, the recording is made in a soft material (first wax, then a special varnish) and then a metal copy is made from it chemically (by plating). This then serves as a matrix for pressing plastic (vinyl) gramophone records. However, since this would only be a monaural recording, albeit very high quality (only one channel), the listener would hear a quality analog signal, but would not receive any information about the space in which the recording was made and the listener would lose a lot emotions. Because one uses two ears to assess space, it was necessary to record two channels (stereo). In order to be able to record everything on one gramophone record, it was necessary to find a way to record two channels. After many different ways, the method of using two electromagnetic circuits perpendicular to each other, which control one needle, has finally been adopted. One channel is recorded at an angle of  $45^\circ$  to the left of the needle axis and the other channel  $45^\circ$  to the right of the needle axis. Both channels are then recorded and the record then becomes stereo. A gramophone transfer must be performed in a similar way, which evaluates the given recording and becomes a source of an acoustic signal for the listener, which is then further processed in its acoustic chain (see the figure below).



The main physical problem of gramophone recording is its level. Because the recording depth in the turntable is very small, these are very small fractions of a millimeter, the movement of the pickup needle is also small, and the voltages generated by the pickup coils of the turntable as well. Such a small signal is very sensitive to ambient interference and its processing is therefore not easy. To give you an idea, the voltage of one monocell of a standard battery, which perhaps everyone knows, is 1.5 V. However, the highest output level of high-quality turntables is a thousand times lower! But we're talking about recording the signal with the highest level! If you imagine that a symphony orchestra normally manages the dynamics of its performance 90 or 100 dB, and you would like to record such dynamics on a gramophone record, you would have to work with signals even a hundred thousand times lower! But you are already in the levels of tens of nanovolts and this is already the standard noise level of even the best rechargeable batteries!

Another big problem is the transmission characteristic of turntables, because the low frequency level is about ten times lower than the medium and high frequency level, so the received signal must always be first processed by a correction preamplifier to balance the frequency response and then substantially amplified to the level processable by the power amplifiers. The pickups are either crystalline (piezoelectric principle, they require higher forces per tip - they wear gramophone records more) or with moving magnets not moving coils (these have up to ten times lower pressing forces and have established themselves for their quality). However, each amplifier stage always distorts the signal a bit and adds a certain level of noise and hum to it. No solution can avoid it. Gramophone signal processing is therefore a very complex process with very high demands on quality. However, it is always a 100% analog signal (although due to the laws of physics a little corrected for dynamics) so the resulting sound is very high quality. The cost of acquiring a high quality analog signal source such as a turntable is very high. They always go into the tens of thousands of crowns and it is not a problem to find systems for many hundreds of thousands of crowns. Although the world went on to use tapes, compact discs, then DVDs, SACDs, and blue-ray discs, records were still alive and still being produced.

However, people with technocratic basics have arguments, because the signal-to-noise ratio rarely gets over 70 dB and gramophone records do not have an infinite lifespan and wear out even when used. Channel separation is even up to a hundred times worse than the above signal-to-noise ratio. But it plays and very well. Gramophone records are still produced and sold.



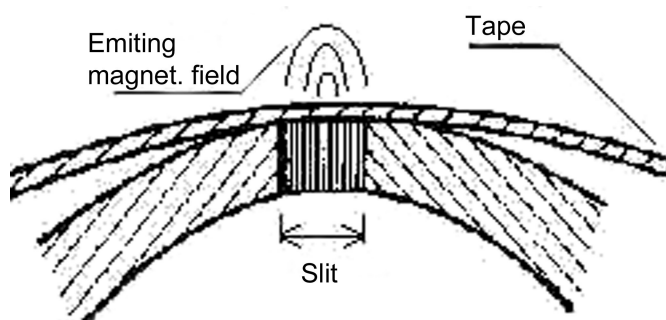
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### 3 Tape recorder



In the next step, but mainly with the advent of semiconductors, the commercial market began to turn more to tape recorders. The tape recorder had one basic advantage over turntables, and that was that the user could record whatever he wanted. A tape is a plastic tape on which a material is applied that can be magnetized as needed. By dragging it over the scanning head, the acoustic signal is recorded, which was converted into electrical and recorded on the tape (the tape was magnetized according to the signal - see the picture below).

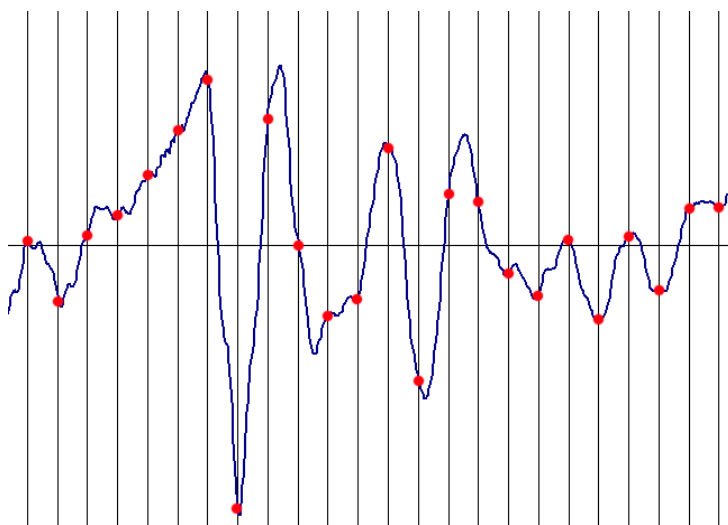


Tape recording also requires correction of the output signal being read, because the upper band transmitted frequencies is limited mainly speed tape. So the demands on internal electronics are also high. Thus, the so-called disc tape recorders appeared on the market, and the higher-quality ones enabled the use of speeds of around 19 m / min, and thus their quality was very high for its time. For commercial purposes, however, began to produce tape recorders with half speed and thus with worse parameters, and eventually came on the market so-called cassette tape recorders, which not only reduced the speed to 4.5 m / min and narrowed the magnetic tape to about half and their quality began to clash greatly with the physical capabilities of the materials used. Various support circuits such as Dolby-B, C, etc. began to appear, but all the improvements were always at the expense of something. Their only advantage was size, so they were very successful for young people. The audiophiles either got studio tape recorders with higher speeds or stayed with the turntables. Although this acoustic source was of good quality in the case of tape recorders, it is

gradually dying. The biggest problem is that the tapes have completely disappeared from the market, so if you do not have enough supplies, this acoustic source for home listening is practically dead.

## 4 CD players and other audio disc players

However, developments continued and this area of audio was also affected by digitization. CD players were the first to appear and the audio signal began to be sampled according to the agreed standards in the so-called Red Book. It was agreed that the sampling frequency would be 44.1 kHz and each sample would have 16 bits. The figure below shows signal sampling.



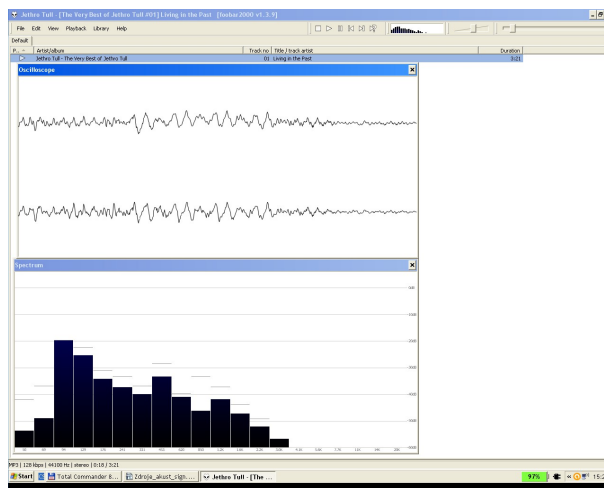
The blue curve shows the acoustic signal and the red dots from where the signal sample is taken. From this figure, the difference between the analog signal, which is a blue curve, and the digital signal, which is the numerical information about the position of the red dot, both vertically (by a certain number of bits) and horizontally (by the selected sampling frequency), is clear. For example, the turntable described above records exactly the entire shape of a blue curve as it extrudes it into a soft matrix. The CD player only receives information about the positions of the individual red dots and must connect them together to recreate the recorded analog signal. So it is clear that the larger the number of samples, the better the result.

Because at a sampling frequency of 44.1 kHz used for CD players, the maximum possible transmitted frequency is 22.05 kHz and the transmitted audio signal should be up to 20 kHz, leaving only 2 KHz left to filter out the upper part of the frequency spectrum. The upper part of the spectrum, ie 22.6 to 44.1 kHz, is actually a mirror-image of a true copy of the required spectrum from 0 to 22.05 kHz, so it must be filtered out as perfectly as possible. However, this is an impossible task for a classic analog filter. So digital filters began to be used. According to the available theories, they should meet the expected parameters. But the audiophiles didn't like the sound of the CD players. Development went on to produce higher resolution discs than CDs, and Audio DVDs, HDCDs, and SACDs appeared. 96 khz sampling began to be used, and the individual samples had a larger number of bits in this case 24. These acoustic sources achieved much better parameters than a standard CD, but the audiophiles still heard that was not it. As development progressed, on the one hand, Blue Ray discs began to appear with even finer sampling, but a lot of energy was also devoted to exploring the reasons why this is still not the case. Because the

capabilities of today's electronics are amazing compared to those a few years ago, and today's measurement technology can be unseen, the reason why digital players still had problems. In the case of a CD player, it has been found that the 16-bit vertical resolution is not as restrictive as the selected sampling frequency of 44.1 kHz.

As mentioned above, this format requires very demanding filtering to pass a frequency of 20 kHz and suppress frequencies above 22 KHz at most. Careful measurements revealed that the necessary digital filter would theoretically handle this, but would have to have a source of sampling frequency with physically unattainable jitter. Jitter is the value of the pulse width change of the sampling frequency. Theoretically, it should be zero, but no frequency source is so perfectly stable. It should be noted that if the crystal has a slightly different frequency but is stable, the result is perfect. But if its frequency fluctuates (has high jitter) then the result is bad. You can find a lot of information about jitter on the Internet today. I first encountered this in an article by Ryohei Kusunoki. It simply explains what the maximum jitter can be acceptable for the CD format to avoid errors when converting from digital to analog signal. For a standard CD player with a digital filter with eight times re-sampling, the minimum jitter is 21.6 ps. But standard sampling signal sources with a crystal oscillator have jitter at the level of several hundred ps !! This is the biggest problem of commercially produced players. Since it is not a problem to build or buy a source of sampling signal at the level of ps units, it is quite easy to improve the quality of your CD or DVD player by adjusting its frequency base. Tested on many standard CD and DVD players and the benefit is very good for a few crowns. A big advantage over a turntable is the basic output voltage behind the converter itself. The standard output voltage in the CD player is 2V. That's 1000 times more than a turntable. This means that the effect of basic physical noise on the acoustic signal is 1000 times smaller, so it means 60 dB smaller than with a turntable !!

## 5 Music files on PC



With the advent of computers, it became possible to use their computing power, and music recordings began to appear in the form of files that could be processed by personal computers. Subsequently, the great expansion of MP3 players began, because they are portable, they can hold a large number of MP3 files and, especially in terms of their price, they are available to a very wide group of consumers. From the point of view of a lover of quality music, the worst thing is that due to their unbeatable availability they are very popular from childhood and cause children to get used to using them and if they do not play a musical instrument or are not from a musical family to experience sound Acoustic musical instruments see no reason to listen to anything else. With the

advent of mobile phones, which today all MP3 players have, their position has been significantly strengthened. Today, everyone has them since childhood.

However, MP3 coding is the so-called lossy coding. The basic premise for its use is the idea that one concentrates on the incoming sound at the appropriate maximum sound pressure level (volume) and does not observe what happens at levels several tens of decibels lower (much quieter sounds). It is therefore possible to delete this information and reduce the size of the acoustic files. The higher the compression (suppressing signals at a lower volume), the smaller the files themselves and the more they can fit on a given MP3 player. They are sent faster from mobile phones or other portable devices to friends, and if it is their other use, such as their use as a ring-tone or notification tone, they are unbeatable there.

Over time, many other ways of encoding audio recordings have emerged, but since virtually everyone is on the Internet today, they usually have the ability to download decoding applications for virtually anything from the Internet.

From the point of view of a lover of quality listening, however, it is an odd path. With the loss of information, the spatial information at the place of recording is practically completely lost and, of course, the color of the tones changes, the musical instruments sound different, etc. Fortunately, even for these listeners, the possibilities are great today. There are already many lossless compressions to reduce acoustic recording files, and their quality has mostly been perfect lately. In addition, professional DA converters now commonly work with an output voltage of up to 9V, so the problems with the natural level of physical noise are even 3.5 less than were even for DA converters for CD players. When it comes to signal sampling itself, the level of recordings 192kHz / 24 bits (studio resolution) in lossless files is commonly available on the Internet today, such as:

Audio Interchange File Format (AIFF)

Apple Lossless Audio Codec (ALAC)

Free Lossless Audio Codec) etc.

However, files with a resolution of 385kHz / 24 and 32 bits are already available somewhere. So today's audiophile has amazing possibilities. All he needs is a computer with internet access, a high-quality DA converter (not a standard internal or external sound card), a high-quality power amplifier, high-quality speakers, high-quality cabling and a suitably arranged listening room, and the result is incredibly great. It is common today to make the same recording in several resolutions and listen to the differences. I have tried from your children and acquaintances that you don't even need to be a listening expert and you will definitely know the differences. The human brain still knows what sounds are natural and natural and what is artificial.

I highly recommend listening to it. Words can never put it right. Stop by and you can hear it.