Record Equalization

What it means and how to deal with it

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MOST OF US know that equalization, or Eq, is applied when a record is made and must be counteracted at playback to obtain a flat frequency response .

Very few, however, know much more. What is Eq really? Why is it applied? How do we deal with Eq?

We are sometimes told that without Eq a 12in 78 rpm record would be able to play perhaps only one to 1½ minutes instead of the normal four to 4½ minutes. This is hardly any explanation. It can be compared to the Aristotelian answer to the question why a stone falls to the ground: the ground is the natural place for heavy objects! Not an explanation but a mere observation.

From my experience, many people think the matter too complicated to understand without a solid technical background and consequently they are reluctant to learn and to know. The matter is not complicated and I promise that the reader will understand. Mystical technical mumbo-jumbo will be absent from this paper. So please, do not run away, stay with me and everything will become clear to you; and you will be in a much better position to deal with Eq problems when you are playing your invaluable historic 78 rpm records.

One aspect of the relation between sound and energy is vital to understanding the matter but also this is straightforward, so please follow me on a very short excursion into physics.

Suppose a sound source emanates a constant tone of a frequency, say 500 Hz. It means that 500 complete sound waves, each containing an amount of energy, leave the oscillating object every second. They create a sound pressure, corresponding to a sound level, at the ear or other recipient that they reach. Now imagine that the frequency is lowered to 50 Hz. If the same sound pressure is to be obtained, each wave must now contain 10 times the energy that it contained when the frequency was 500 Hz, i.e. the movements of the oscillating object must be greater.

We can experience that when we put a finger on the cone of a loudspeaker playing at a reasonably loud level. The deep notes are clearly perceived as movements of the cone, the high pitched notes are not. And when our children or grandchildren listen to their loud rock or pop music the movements of the cone reproducing the bass or drum can even be seen.

In the same way, when music is recorded, the movements of the diaphragm in the microphone increase as the frequency decreases. This forces the cutter-head on the recording machine to act likewise, so the excursions in the record groove become proportionally bigger as the frequency to be cut goes lower.

Now we understand why Eq, in this case attenuation of the low frequencies, is used when making records. This forces the user to counteract the attenuation by accentuating the low frequencies by a curve which mirrors the one used at recording.

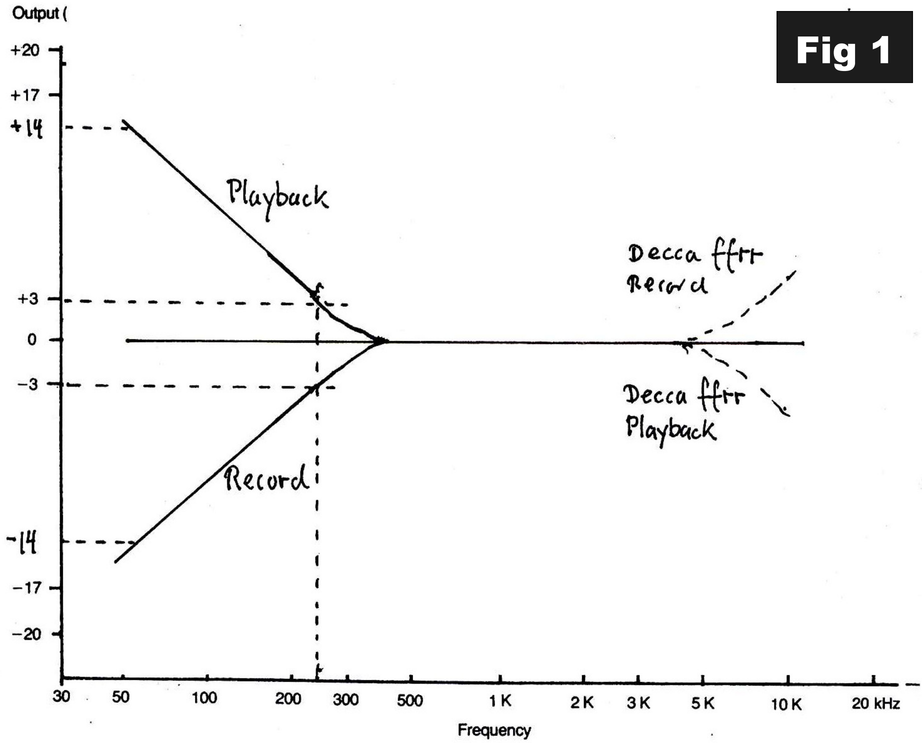
If a curve is to be mirrored it has to be known, and some sort of standardisation in recording is clearly necessary. But when electrical recording was introduced no standards existed, so Eq cannot be expected to be the same for different companies or even for different records in the same company. Thus we might expect total anarchy and chaos. Luckily it never came to that. It quickly became evident that attenuating frequencies above 200–300 Hz did nothing to extend playing time. So Eq was at first not fixed by agreement on a standard, but was determined by the practical experience of the engineers who cut the records. Thus, the turnover frequency in all records before World War II is somewhere between 150 and 300 Hz, usually 200–250 Hz. Figure 1 depicts the recording curve, the playback curve and the resulting flat frequency response of what became the standard 78 rpm record. This curve became standard for all HMV records for more than 25 years.

After the war the picture becomes more blurred because Eq was now attempted also at high frequencies, in the hope that, by accentuating treble levels in recording and attenuating them during playback, record noise could be reduced.

But in the treble region there is no physical factor determining the choice of turnover frequency and amount of Eq. In the USA much higher treble boosts were commonly applied than in Europe, making it apparently more complicated to compensate for different treble Eq. Luckily the problem is simpler than it seems.

One must understand that even though it seems logical to try to reduce noise by boosting treble during recording and attenuate during playback, it is not without costs. In audio engineering free meals are very rare.

Here is also a price to pay: higher demands are put to the cutter-head where the stylus must now move considerably faster at high frequencies and so must the pickup stylus during playback. Tracking the signal becomes more complicated and the inevitable result is higher distortion. It is argued that harmonic distortion of high frequencies is not that important since the distortion products will be outside our hearing range. The third harmonic of 5kHz is 15kKz, which only



young persons hear, and loudspeakers of the period were not able to reproduce frequencies above some 8–10 kHz. But the coin has another side. A record very rarely contains only one frequency, and high harmonic distortion at high frequencies produces very nasty interferences between different frequencies, and difference tones, not harmonically related to anything in the music, are the result. The situation is aggravated as the program reaches the inner grooves close to the label. The information stored in the groove of one revolution here often has less than one third of the length that was available at the beginning of the record. Large amplitudes of high frequencies tend to be cramped together and the obvious result is a harsh and distorted sound.

We have not yet defined what exactly we understand by the turnover point. If we look at Fig 1 we see the curve in the low frequencies beginning to turn from the straight line somewhere around 500 Hz, but the change is at first so gradual as to be inaudible. Not until this curve reaches 250 Hz can the ear detect a difference, and at that point the difference is 3 dB. Thus, as the smallest change to be clearly audible, 3 dB was chosen as the universal standard defining the point of turnover. It is as simple as that. In the present case, the difference between

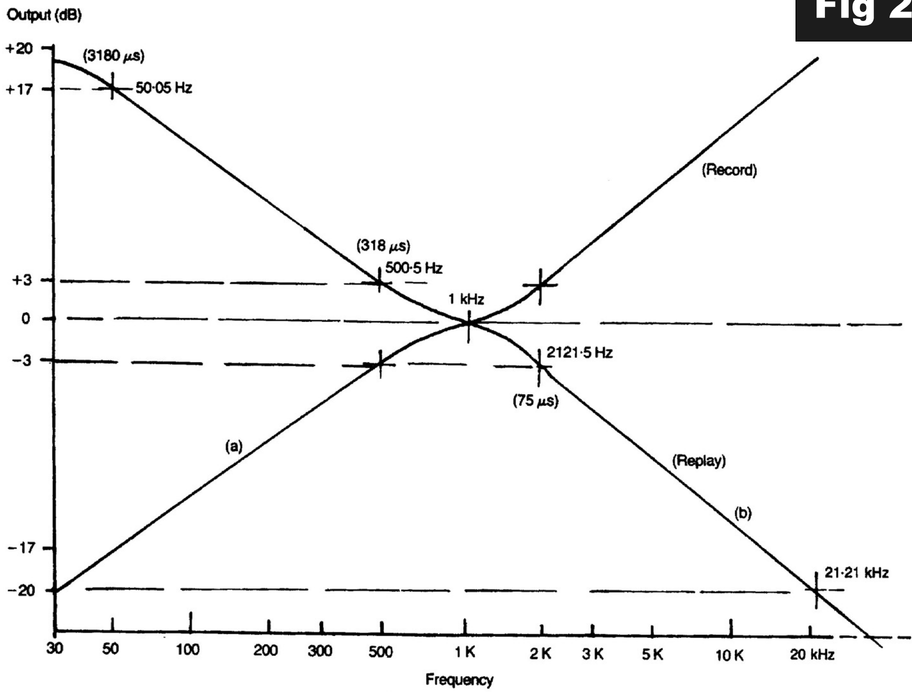


Fig 2

the turnover point, 250 Hz, and the point where the slope begins changing, 500 Hz, equates to about one octave.

In the mid-1950s the Long Playing Record had become the normal way to distribute music and an international agreement for standardisation of the recording curve and therefore the playback curve, which is the mirror, was reached. This is what is today known as the RIAA curve (RIAA is short for Record Industry Association of America). When this practice quickly established itself and pickups had become much better, having a close to linear frequency response, the manufacturers of playback equipment had something real to deal with. Amplifiers with correction input stages for pickups became standard so the record-playing public could forget about the problem (if it was ever aware of it!). Things just worked.

The RIAA curve prescribes three turnover frequencies, 2122Hz, 500Hz and 50 Hz. This more commonly expressed as 75, 318 and 3180 microseconds. You can turn these microseconds into the corresponding frequencies by dividing them into 159000 but for our purpose there is no need to stray into this. Suffice to say that this way of expressing the frequencies makes calculation of corrective networks easier.

The result is that during playback treble attenuation has its turnover point at 2122 Hz, the bass accentuation has 500 Hz as turnover point, and at 50 Hz the bass lift flattens out. But as the reader will remember the turnover frequency is defined as the point where the level has already changed by 3 dB. So the final curve ends up without any frequency band where the level is constant (see Figure 2)

HMV worked officially from the start of the electrical era with a turnover point of 250 Hz, meaning that at this frequency the playback curve must be 3dB up. Above 500 hz the level was constant.

When a 78 rpm record is played back through an amplifier that corrects in accordance with the RIAA curve which has no part with constant level, major damage to the sound is done. It seems to be common belief that these severe faults can be corrected by means of an ordinary tone control. They can not. The onset of the bass lift is about an octave too high and if a standard bass control is adjusted to give correct level at very low frequencies the mid bass will wrong and if the control is set to give correct mid bass the bottom end will be wrongly treated. The result is either a muddy or a thin low end and a far too much attenuated high end. As mentioned earlier the high end was not changed during recording at all. But opposed to the bass control, the treble control can do a fairly good job in transferring the RIAA curve into the standard 78rpm curve, not accurate, but acceptable and it is absolute useful when it comes to playback of the records with different amounts of treble lift during recording after WW II. Of these records the best known in our part of the world is the Decca ffrr (full frequency range recording). The Decca treble boost was moderate. The curve is among the only curves available to the public and it remained unchanged for 10 years. It can also be seen in Figure 1. But there were many others and in the USA the treble lift was often taken much further than we did in Europe. It must however be said that attenuating the treble by some 12dB at 10 kHz in the RIAA input stage and then bring the level back by accentuating treble by a corresponding amount with a tone control is, from a technical point of view, a very bad idea. Not only is it debatable how accurate this way is, but if it is not completely correct (and it most certainly is not) transient response will deteriorate and clarity is impaired.

The low end presents the biggest problem. If one reads articles dealing with record playback problems in T*he Wireless World* (a highly respected British magazine) from the 1950s, it is clearly seen that the problems were severe and of much interest. It has to be realized that many pickups of that period had far from linear frequency response so one had to deal with two Eq problems at the same time: the frequency response of the pickup and the Eq curve of the record to be played. In the time before the RIAA curve was adopted, one finds frustration in many of the articles about this. One has the title "Stylus in Wonderland" and the author bemoans the state of things, telling us that even though many amplifiers now had selectors for both bass and treble compensation, adjusting them often took more time than the record played. He also complains that even major record companies kept their recording curves a secret and they were also guilty of changing them without notice. But it must be admitted that things were not easy for the companies either. Before the time came when recordings were made on tape, prior to transferring them to matrices, metal matrices were the only way programs could be exchanged between countries and affiliated companies. When, for example, HMV issued a recording made in America it was by no means certain that it was recorded to British standards. Later. tapes could be transferred to matrices differently in different countries, so it is not uncommon for pressings from different places to sound different in the early 1950s.

The RIAA standard set things straight. A curve was also proposed and adopted for 78rpm records but this never became significant since the shellac record was to become obsolete within a few years.

Returning to *Wireless World*, one finds a wide range of articles dealing with how to compensate for the characteristics of many different pickups, correct compensation for a bewildering number of recording Eq standards, and down-to -earth solutions, even though not perfect, nevertheless able to improve playback results immensely.

One of the pioneers in the hi-fi field was P J Baxandall. His famous tone control described in 1951 amounted to a revolution. It is still used all over the world for many applications. This control immortalized his name and established his authority in the field of, especially, pre- and correction amplifiers.

In a famous article from January 1955, just before the RIAA curve became standard for LPs, Baxandall takes a very pragmatic attitude to the record Eq problem, describing an amplifier which, if used in conjunction with a pickup of linear frequency response and a standard tone control such as his own, will do the job well for virtually all records.

Before looking into his solution time has come to make a few observations about playback of pre-electrical records. This means all records made before 1925 or ’26 and some, not many, made in the following few years.

It is often maintained that Eq was not applied during acoustic recording and consequently the records should also be played back without any compensation at all. It is of course true that electrical Eq was not applied, but Eq can be applied mechanically as well. And mechanical Eq definitely was used: without it, playing times of four minutes for a 12in record could not have been achieved.

Eq was applied intentionally but also unintentionally, simply by the laws of physics. The size of the recording horn limits the lowest frequency which can be processed and increasing the size of the horn tends to weaken the treble. Then the tension of the diaphragm and the springs between the levers linking the diaphragm to the cutting stylus all tend to minimise the larger movements—which, as we have seen, is what we can expect in the bass region. Also the quality of the wax used for cutting and the temperature played a major role. So some playback Eq can absolutely be healthy for even pre-electric records, where much more spread in frequency response is to be expected. This also applies to vertical-cut records, e. g. Pathé.

Now that we know what EQ is and understand why it is applied, we can now be stringent when we state the necessary demands to the pick-up playback correction amplifier.

1. The amplifier must be able to correct for the bass attenuation by boosting low frequencies. The turnover point need not be variable. A 3dB point a close to 250 Hz works fine even when the actual turnover point differs from this point. In practice it never differs by more than half an octave and experience shows that deviations of this magnitude are insignificant. But the boost should not continue down to subsonics. There is little of interest below 50 Hz, and boosting frequencies below this not only accentuates rumble from the gramophone but also rumble recorded from the cutting machine—which can be rather annoying. Ideally the bass lift should go only down to 50 Hz and then flatten out, or, better still, decrease. The latter is however not a simple task in designing an amplifier and it is not necessary, even for the more serious amateur. Flattening is fine.
2. The amplifier must have a control that can apply a variable treble attenuation to the signal. This is adjusted to give a pleasant tonal balance of the program being reproduced. It is impossible to give information about the curves used by the companies after WW II but a variable control, even if it cannot give correction with 100pc accuracy, is able to produce very good results.
3. The amplifier must have the overall gain to produce a signal of a magnitude which will comply with the sensitivity of a standard auxiliary or radio input of a normal amplifier. Furthermore it must have an input resistance of 47 kOhms, which is the standardised load for all modern pickups.

4, The amplifier must be able to deal with signals substantially larger than the signal produced by the music on the record. Scratches and specs of dust can produce signal peaks many times higher than the modulation in the grooves. If the input stage is overloaded by such peaks, it can be paralysed for a short time during recovery, i.e. when an overloading peak upsets the working conditions of the amplifier, subsequent signals could be blocked by temporary amplifier paralysis for, perhaps, a few milliseconds. The resulting so-called transient intermodulation, as this kind of distortion is commonly called, is a major contributor to listening fatigue. The input stage must be able to cope with signals at least 10 times, or still better 20 times, the musical peak signals. "*for* a short time"; i.

5. The amplifier should have a reasonably low inherent noise and—sadly, it needs to be said—must be completely free from mains interference or, put plainly, hum.

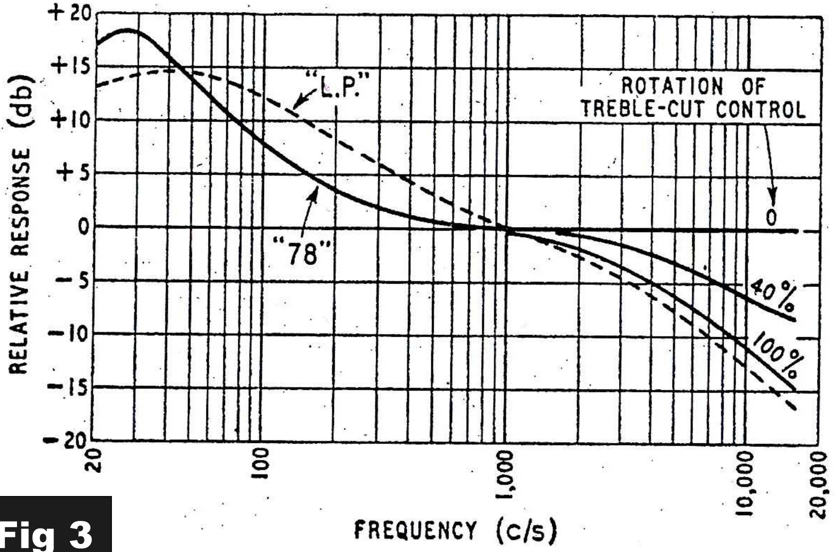


Fig 3

The Baxandall correction amplifier complied with most of these demands. Pickups of the period had a much higher output than pickups today so a smaller gain was required and the load to the pickup was not then standardised. If however the signal from a modern pickup is passed through an amplifier with a gain of around 20 or 26dB which also presents a 47 kOhm load, it works extremely well—even today, 54 years later.

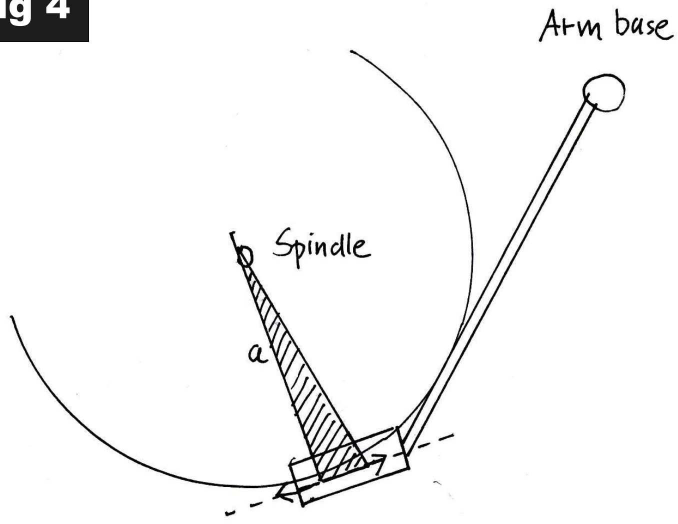
The basic principle was a corrective network around a single valve (semiconductors were just born and not even baptised in 1955). This network could be changed from LP correction to 78 correction with a single switch (the LP correction was not yet the RIAA version, but fairly close). In the 78 position a fixed bass lift with a 250Hz turnover was applied together with a potentiometer providing continuous control of treble attenuation. This must even today be considered a brilliant approach and Figure 3 shows the possible Baxandall curves. It can be seen that he even managed to reverse the bass lift for very low frequencies and thus reduced rumble.

The next part of this paper will describe a simple amplifier which almost anyone can make . It uses less than a handful of cheap standard components available everywhere and it complies with all the demands listed above.

Even out of the scope of the topic I shall conclude this part taking a quick look at the gramophone to be used for playing 78 rpm records.

The turntable must of course be able to run stable at 78 revolutions per minute, but not all records run at exactly this speed. In the early years the spread could

Fig 4



be fairly big, from about 70 to 85 rpm, so a turntable with variable speed is to be preferred. The actual correct speed for a particular record can kindle heated debates and I try to keep clear of these mined waters whenever possible. I can only say that the Lenco models 75 and 78, which are continuously variable from 15 to almost 90 revolutions per minute, cover all needs and they are very good machines too.

Paradoxically the demands to the pickup arm from the old 78 rpm records are more serious than they are from the more refined LP records. Larger movements of the stylus tend to provoke arm resonances and torsions. Loose bearings will rattle, so a good, stiff arm is a must. Unfortunately the Lenco arms are not ideal . . . .

The pickup element must be mounted firmly in the headshell and the stylus must be in the correct position to track optimally. It should not only look correct it should *be* absolutely correct, and tested with a so-called protractor. Such a device may not be easy to come by, but an absolutely precise method is to use a right-angled triangle made of light cardboard like a postcard (see Figure 4).

The triangle should be placed over the headshell as shown, with the short side following the centre-line of the shell, i.e. parallel with the cartridge element. When the pick-up is taken across the grooved area of the disc, the triangle is moved back and forth and the position found where the line “a” deviates least from the centre of the spindle during travel. Directly under line “a” is the correct position of the stylus.. For a well-designed arm, mounted correctly, it should be possible to achieve, that line “a” never leaves the spindle, or only just. If that is not possible, the arm is either of inferior design or the distance between the arm-base and the spindle is incorrect. That could easily be the case if the arm on a gramophone is replaced with a new by someone not aware that the mounting distance is of paramount importance, and just a few millimetres of misplacement makes correct tracking impossible. In such a case, even the best arm could prove worse than the original.

The pickup cable should be of good quality and not longer than necessary. A separate ground lead is advantageous. When such a lead is present, none of the four leads from the element should have any contact to the machine´s ground. Again the Lenco needs a better cable and a separate ground lead.

The pickup should be wired for mono i.e. the L and the R leads should be connected together as should the LG and the RG leads. By mono wiring, signals caused by vertical movements cancel, thereby reducing rumble.

At last the stylus. At least two styli are required by the serious amateur, one 65 micron and one 90 micron.*

Nearly all pre-WWI records play better with a broader stylus than later records. The styli should preferably have truncated tips so that they will normally not reach the bottom of the groove. The modulation is in the walls. The bottom only contains noise. Elliptical styli are also available. They are in some cases better at the inner groves but in general they are more noisy, probably because they can-better follow the grains of the shellac owing to their smaller contact area. Alas, again no free meals!

After these brief considerations concerning playback equipment let me conclude this part by a few comments on the loudspeakers to be used. Throughout the 78rpm era, small extensions of frequency response gradually became possible. The first big step came with the introduction of the electrical recording. Before that a range from 100 Hz to 3 or 3.5 kHz,, more or less, was possible. But from 1926 the bass response improved about an octave and the treble region expanded to around 4 kHz. Improvements in the years leading to World War II were small, with 5 kHz the *de facto* maximum. But some records were marginally better, especially experimental ones . Pickups of the period showed no improvement, so before the introduction of the lightweight pickup after the war, there seemed little use for a broader frequency response; thus the priority was on reduction of distortion.

After the war the top frequency moved towards 10 kHz, but the signal to noise ratio at these high frequencies was very poor because of the grainy material used for the records. In this department improvements was also taking place, but one cannot talk about a breakthrough.

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* Styli of these dimensions must, regrettably, be obtained overseas.. Cartridges or stylus assemblies need to be sent for retipping to the well known Expert Stylus Company in England: PO Box 3, Ashtead, Surrey, UK, KT21 2QD (telephone 0372276604, fax 0372276147).

In the design of loudspeakers today, everything is done to extend the frequency response to above 18 kHz. When we use a modern loudspeaker to reproduce old records, the top octave of the loudspeaker will reproduce only noise because there is no signal is present.

I remember a former colleague from the Royal Academy in Aarhus who said that he found his old records sounded more natural when he used the loudspeaker in what he described as “an old wooden radio”

He was in my opinion right. The records were made to be heard from exactly that, and it seems worthwhile to contemplate a special speaker for this purpose alone, Not a two- or three-way speaker but a single-speaker system with a unit having a response to around 8 or maximum 10 kHz, and without any crossover filter. Smaller bass units intended for two-way systems are obvious candidates. Of course a suitable cabinet is necessary, but in recycling stores loudspeakers can often be had for almost nothing. They can be cannibalized for the cabinets or, if one is lucky, the bass speaker unit could be just the one required. It can be argued that the same could be achieved using a low-pas filter somewhere in the playback chain. But a filter that has a steep roll-off, has always severe phase shifts around the turnover point, causing coloration of the sound. In my opinion a roll-off steeper than 12dB per octave can easily do more harm than good. A perfect design is impossible and a good design is complicated, so with a dedicated speaker we get not only a natural roll-of but we make the signal path simpler and shorter compared to the filter approach.

Today we almost always use two speakers (stereo) and this is also a very good idea when playing mono: the sound gets more transparent and less aggressive. So *some* improvements can be gained from modern approach to sound reproduction.

Relevant articles from *The Wireless World*

O. J. Russell: Stylus in Wonderland, October 1954

R. L. West & S. Kelly: Pick-up Input Circuits, November 1950

“Cathode Ray”: Equalization, February 1952

P. J. Baxandall: Negative-Feedback Tone Control, October 1952

P. J. Baxandall: Gramophone and Microphone Preamplifier, January & February 1955

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