Beogram 4000c

Technical Sound Guide

Bang & Olufsen A/S

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Contents

1 History **3**

1.1 The mechanical phonograph . . . 3

1.2 Magnets and Coils . . . 3

2 The physics **4**

2.1 Amplitude vs. Velocity . . . 4

2.2 Surface noise . . . 4

2.3 Mono to Stereo . . . 5

3 The cartridge, stylus, and tonearm **7**

3.1 MMC: Micro Moving Cross . . . 7

3.2 Signal Levels . . . 8

3.3 Tip shape . . . 10

3.4 Bonded vs. Nude . . . 12

3.5 Tracking force . . . 12

3.6 Effective Tip Mass . . . 12

3.7 Compliance . . . 13

3.8 Soundsmith SMMC20CL . . . 13

4 Audio Specifications **14**

4.0.1 Magnitude Response . . . 14

4.0.2 Rumble . . . 14

4.0.3 Rotational speed . . . 14

4.0.3.1 Accuracy . . . 14

4.0.3.2 Stability . . . 14

4.0.3.3 Measurement and Weighting . . . 15

4.0.3.4 Expressing the result . . . 16

5 Beogram 4000c Specifications **17**

6 Further Reading **18**

History

1.1 The mechanical phonograph

In 1856, Édouard-Léon Scott de Martinville invented a device based on the basic anatomy of the human ear. It consisted of a wooden funnel ending at a flexible membrane to emulate the ear canal and eardrum. Connected to the membrane was a pig bristle that moved with it, scratching a thin line into soot on a piece of paper wrapped around a rotating cylinder. He called this new invention a “phonautograph” or “self-writer of sound”.

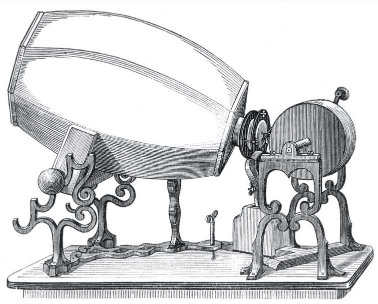

Figure 1.1: The phonautograph.

This device was conceived to record sounds in the air without any intention of playing them back, so it can be considered to be the precursor to the modern oscilloscope.[1] However, in the late 1870’s, Charles Cros realised that if the lines drawn by the phonoautograph were photo-engraved onto the surface of a metal cylinder, then it could be used to vibrate a needle placed in the resulting groove. Unfortunately, rather than actually build such a device, he only wrote about the idea in a document that was filed at the Académie des Sciences and sealed. Within 6 months of this, in 1877, Thomas Edison asked his assistant, John Kruesi, to build a device that could not only record sound (as an indentation in tin foil on a cylinder) but reproduce it, if only a few times before the groove became smoothed.[2]

It was ten years later, in 1887, that the German-American inventor Emil Berliner was awarded a patent for a sound recording and reproducing system that was based on a groove in a rotating disc (rather than Edison’s cylinder); the original version of the system that we know of today as the “Long Playing” or “LP” Record.

Figure 1.2: An Edison “Blue Amberol” record with a Danish 78 RPM “His Master’s Voice” disc recording X8071 of Den Blaa Anemone.

Early phonographs or “gramophones” were purely mechanical devices. The disc (or cylinder) was rotated by a spring-driven clockwork mechanism and the needle or stylus rested in the passing groove. The vibrations of the needle were transmitted to a flexible membrane that was situated at the narrow end of a horn that amplified the resulting sound to audible levels.

1.2 Magnets and Coils

In 1820, more than 30 years before de Martinville’s invention, the Danish physicist and chemist, Hans Christian Ørsted announced the first link made between electricity and magnetism: he had discovered that a compass needle would change direction when placed near a wire that was carrying an electrical current. Nowadays, it is well-known that this link is bi-directional. When current is sent through a wire, a magnetic field is generated around it. However, it is also true that moving a wire through a magnetic field will generate current that is proportional to its velocity.

[1] It should be said that some “recordings” made on a phonoautograph were finally played in 2008. See www.firstsounds.org for more information.
[2] see “Reproduction of Sound in High-fidelity and Stereo Phonographs” (1962) by Edgar Villchur

The physics

2.1 Amplitude vs. Velocity

It is this second interaction that is at the heart of almost every modern turntable. As the stylus (or "needle"[1]) is pulled through the grove in the vinyl surface, it moves from side-to-side at a varying speed called the *modulation velocity* or just the *velocity*. An example of this wavy groove can be seen in the photo in Figure 2.1. Inside the housing of most cartridges are small magnets and coils of wire, either of which is being moved by the stylus as it vibrates. That movement generates an electrical current that is analogous to the shape of the groove: the higher the velocity of the stylus, the higher the electrical signal from the cartridge.

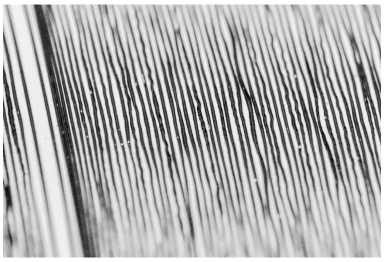

Figure 2.1: The groove in a late-1980's pop tune on a 33 1/3 RPM stereo LP. The white dots in the groove are dirt that should be removed before playing the disc.

However, this introduces a problem because if the amplitude remains the same at all frequencies the modulation velocity of the stylus decreases with frequency; in other words, the lower the note, the lower the output level, and therefore the less bass. This is illustrated in the graph in Figure 2.2 in which three sine waves are shown with different frequencies. The blue line shows the lowest frequency and the orange line is the highest. Notice that all three have the same amplitude (the same maximum "height"). However, if you look at the slopes of the three curves when they pass Time = 0 ms, you'll see that the higher the frequency, the higher the slope of the line, and therefore the higher the velocity of the stylus.

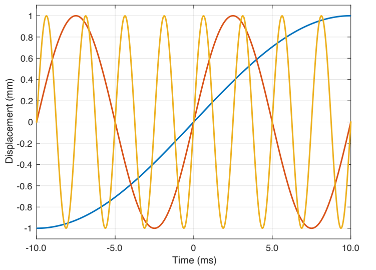

Figure 2.2: Three sine waves of different frequencies (from low to high: blue, red and orange curves), but with the same amplitude.

In order to achieve a naturally flat frequency response from the cartridge, where all frequencies have the same electrical output level, it is necessary to ensure that they have the same modulation velocity, as shown in Figure 2.3. In that plot, it can be seen that the slopes of the three waves are the same at Time = 0 ms. However, it is also evident that, when this is true, they have very different amplitudes: in fact, the amplitude would have to double for every halving of frequency (a drop of 1 octave). This is not feasible, since it would mean that the stylus would have to move left and right by (relatively) huge distances in order to deliver the desired output. For example, if the stylus were moving sideways by $\pm$ 0.1 mm at 1,000 Hz to deliver a signal, then it would have to move $\pm$ 1 mm at 100 Hz, and $\pm$ 10 mm at 10 Hz to deliver the same output level. This is not possible (or at least it's very impractical).

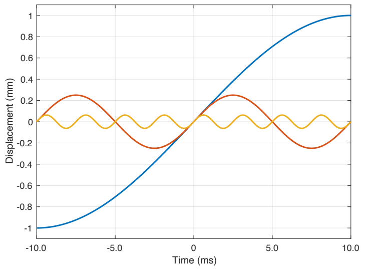

Figure 2.3: Three sine waves of different frequencies (from low to high: blue, red and orange curves), but with the same modulation velocity.

The solution for this limitation was to use low-frequency audio compensation filters, both at the recording and the playback stages. When a recording is mastered to be cut on a disc, the low frequency level is decreased; the lower the frequency, the lower the level. This results in a signal recorded on disc with a constant *amplitude* for signals below approximately 1 kHz.

Of course, if this signal were played back directly, there would be an increasing loss of level at lower and lower frequencies. So, to counteract this, a filter is applied to the output signal of the turntable that boosts the low frequencies signals to their original levels.

2.2 Surface noise

A second problem that exists with vinyl records is that of dust and dirt. If you look again at the photo in Figure 2.1, you can see white specks lodged in the groove. These look very small to us, however, to the stylus, they are very large bumps that cause the tip to move abruptly, and therefore quickly. Since the output signal is still proportional to the modulation velocity, then this makes the resulting cracks and pops quite loud in relation to the audio signal.

In order to overcome this problem, a second filter is used, this time for

[1] Some authors reserve the term "stylus" for the device that is used to cut the groove during mastering, and the term "needle" for the device used to play a phonographic record. However, the two terms are used interchangeably in this document.

higher frequencies. Upon playback, the level of the treble is reduced; the higher the frequency the lower the output. This reduces the problem of noise caused by surface dirt on the disc, however it would also reduce the high frequency content of the audio signal itself. This is counteracted by increasing the level of the high-frequency portion of the audio signal when it is mastered for the disc.

This general idea of lowering the level of low-frequencies and/or boosting highs when recording and doing the opposite upon playback is a very old idea in the audio industry and has been used on many formats ranging from film "talkies" to early compact discs. Unfortunately, however, different recording companies and studios used different filters on phonographs for many years.[2] Finally, in the mid-1950s, the Recording Industry Association of America (the RIAA) suggested a standard filter description with the intention that it would be used world-wide for all PVC "vinyl" records.

Figures 2.4 and 2.5 show the responses of the RIAA filters used in both the mastering and the playback of long playing vinyl records. Although there are other standards with slightly different responses, the RIAA filter is by far the most commonly-used.

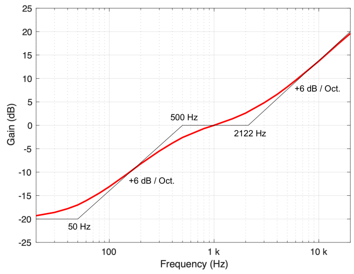

Figure 2.4: The "pre-emphasis" filter to be used in the mastering to disc, as described by the RIAA standard. The black line shows the simplified description, and the red curve shows the real-world implementation.

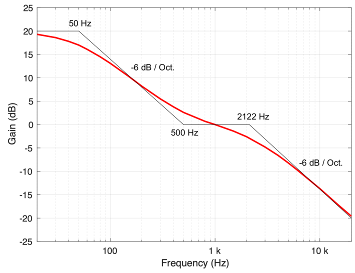

Figure 2.5: The "de-emphasis" filter to be used for playback as described by the RIAA standard. This standard filter response is integral in what is now commonly called a "RIAA preamp".

It may be of interest to note that typical descriptions of the RIAA equalisation filter define the transition points as time constants instead of frequencies. So, instead of 50 Hz, 500 Hz, and 2122 Hz (as shown in the response plots), the points are listed as 3180 μs, 318 μs, and 75 μs instead. If you wish to convert a time constant (Tc) to the equivalent frequency (F), you can use the equation below.

$$F = 1 / (2 \pi Tc)$$

2.3 Mono to Stereo

In Edison's first cylinder recordings, the needle vibrated up and down instead of left and right to record the audio signal. This meant that the groove cut into the surface of the tin foil was varying in depth, and therefore in width, as shown in Figure 2.6.

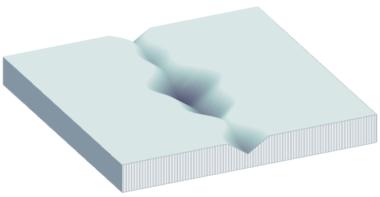

Figure 2.6: Example of an audio signal encoded using a vertical cutting system.

There are some disadvantages to this system, such as the risk of the needle slipping out of the groove when it is too shallow, or suffering from excessive wear if the groove is too deep. In addition, any vertical variation in the recording surface (such as a cylinder that is not quite round, or mechanical vibrations in the player caused by footsteps in the room) becomes translated into unwanted noises upon playback.[3]

Figure 2.7: An Edison cylinder player, on display in the Struer Museum.

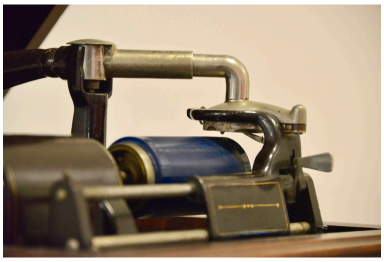

Figure 2.8: A closeup of the Edison player. Notice that the needle is mounted to move vertically, modulating a membrane located at the end of the tonearm (the bent pipe).

Berliner's Gramophone used a different system, where the needle vibrated sideways instead. This lateral cut system produced a groove on the disc with a constant depth, thus avoiding some of the problems incurred by the vertical cut recording system.

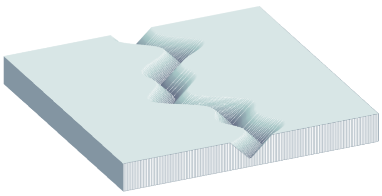

Figure 2.9: Example of an audio signal encoded using a lateral cutting system.

[2] see the Manual of Analogue Sound Restoration Techniques (2008), by Peter Copeland

[3] Some 78 RPM discs use a vertical cutting system as well, including those made by Edison Disc Records and Pathé.

However, both of these systems were only capable of recording a single channel of audio information. In order to capture 2-channel stereo audio (invented by Alan Blumlein in 1931) the system had to be adapted somehow. The initial challenge was to find a way of making a disc player that could reproduce two channels of stereo audio, while still maintaining compatibility with lateral-cut discs.

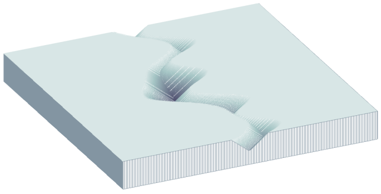

Figure 2.11: Example of two different signals encoded on the two channels of a stereo groove.

The solution was to rotate the modulation direction by 45°, so the two walls of the groove are used to record the two separate audio channels. This means that the stylus moves in two (theoretically independent) axes as shown in Figure 2.10. When the same signal is applied to both channels (better known as a "dual-mono" or "in-phase" signal), then the stylus moves upwards for the left while moving downwards for the right channel (or down-left & up-right), for example. This means that signals that are identical in both channels move the stylus laterally, exactly as in earlier monophonic discs.[4]

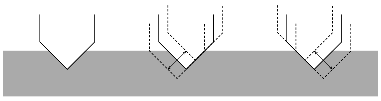

Figure 2.10: An over-simplified depiction of how the two audio channels are encoded in the groove. From left to right: No modulation (silence); Left channel signal modulates the groove's left wall; Right channel signal modulates the right wall.

As a result, if you look at the groove in a modern two-channel stereo LP, it appears that it simply wiggles left-to-right (as can be seen in Figure 2.1). However, if you inspect the same groove with extreme magnification, you can see that the modulations in the two sidewalls of the groove are slightly different, since the audio signals on the left and right channels are not identical.

[4] For a more correct explanation of this movement, see http://rfcafe.com/references/radio-electronics/stereo-disc-july-1958-radio-electronics.htm

The cartridge, stylus, and tonearm

3.1 MMC: Micro Moving Cross

As mentioned above, when a wire is moved through a magnetic field, a current is generated in a wire that is proportional to the velocity of the movement. In order to increase the output, the wire can be wrapped into a coil, effectively lengthening the piece of wire moving through the field. Most phono cartridges make use of this behaviour by using the movement of the stylus to either:

- move tiny magnets that are placed near coils of wire (a *Moving Magnet* or *MM* design) or
- move tiny coils of wire that are placed near very strong magnets (a *Moving Coil* or *MC* design)

In either system, there is a *relative* physical movement that is used to generate the electrical signal from the cartridge. There are advantages and disadvantages associated with both of these systems, however, they will not be discussed here.

There is a third, less common design called a *Moving Iron* (or *variable-reluctance*[1]) system, which can be thought of as a variant of the Moving Magnet principle. In this design, the magnet and the coils remain stationary, and the stylus moves a small piece of iron instead. That iron is placed between the north and south poles of the magnet so that, when it moves, it modulates (or varies) the magnetic field. As the magnetic field modulates, it moves relative to the coils, and an electrical signal is generated. One of the first examples of this kind of pickup was the Western Electric 4A reproducer made in 1925.

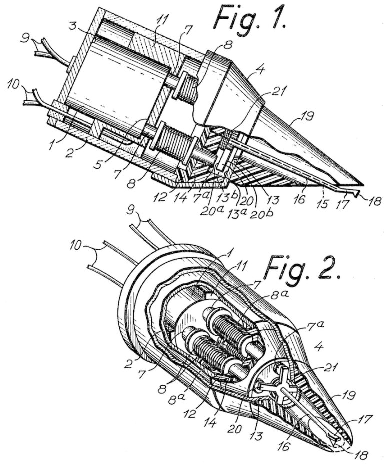

Figure 3.1: Figures from Rørbaek Madsen's 1963 patent for a Stereophonic Transducer Cartridge.

In 1963, Erik Rørbaek Madsen of Bang & Olufsen filed a patent for a cartridge based on the Moving Iron principle. In it, a cross made of Mu-metal is mounted on the stylus. Each arm of the cross is aligned with the end of a small rod called a "pole piece" (because it was attached to the pole of a magnet on the opposite end). The cross is mounted diagonally, so the individual movements of the left and right channels on the groove cause the arms of the cross to move accordingly. For a left-channel signal, the bottom left and top right cross arms move in opposite directions - one forwards and one backwards. For a right-channel signal, the bottom right and top left arms move instead. The two coils that generate the current for each audio channel are wired in a push-pull relationship.

Figure 3.2: Erik Rørbaek Madsen explaining the MMC concept.

There are a number of advantages to this system over the MM and MC designs. Many of these are described in the original 1963 patent, as follows:

- "The channel separation is very good and induction of cross talk from one channel to the other is minimized because cross talk components are in phase in opposing coils."
- "The moving mass which only comprises the armature and the stylus arm can be made very low which results in good frequency response."
- "Hum pick-up is very low due to the balanced coil construction"
- "... the shielding effect of the magnetic housing ... provides a completely closed magnetic circuit which in addition to shielding the coil from external fields prevents attraction to steel turntables."
- Finally, (although this is not mentioned in the patent) the

[1] reluctance is the magnetic equivalent of electrical resistance
[2] Sound Recording Handbook", ed. Glen Ballou

push-pull wiring of the coils "reduces harmonic distortion induced by the non-linearity of the magnetic field."[2]

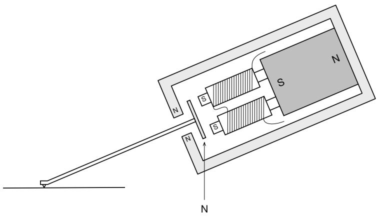

Figure 3.3: The magnetic circuit representation of the MMC cartridge, showing the diagonal pair of pole pieces for one of the two audio channels.

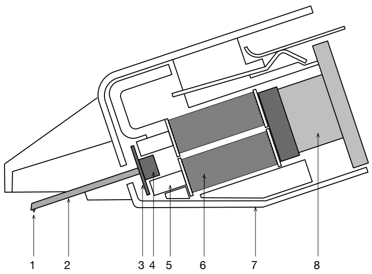

Figure 3.4: The Micro Moving Cross MMC 4000 cartridge design. 1. Nude Pramanik diamond, 2. Low mass beryllium cantilever, 3. Moving micro cross, 4. Block suspension, 5. Pole pieces (4), 6. Induction coils, 7. Mu-metal screen, 8. Hycomax magnet

Figure 3.5: Large-scale models of the MMC cartridges used for past demonstrations.

3.2 Signal Levels

Every audio device relies on a rather simple balancing act. The "signal", whether it's speech, music, or sound effects, should be loud enough to mask the noise that is inherent in the recording or transmission itself. The measurement of this "distance" in level is known as the *Signal-to-Noise Ratio* or *SNR*. However, the signal should not be so loud as to overload the system and cause distortion effects such as "clipping", which results in what is commonly called *Total Harmonic Distortion* or *THD*.[3] One basic method to evaluate the quality of an audio signal or device is to group these two measurements into one value: the *Total Harmonic Distortion plus Noise* or *THD+N* value. The somewhat challenging issue with this value is that a portion of it (the noise floor) is typically independent of the signal level, since a device or signal will have some noise regardless of whether a signal is present or not. However, the distortion is typically directly related to the level of the signal.

In modern digital PCM audio signal (assuming that they are correctly-implemented and ignoring any additional signal processing), the noise floor is the result of the dither that is used to randomise the inherent quantisation error in the encoding system. This noise is independent of the signal level, and entirely dependent on the resolution of the system (measured in the number of bits used to encode each sample). The maximum possible level that can be encoded without incurring additional distortion that is inherent in the encoding system itself is when the maximum (or minimum) value in the audio signal reaches the highest possible signal value of the system. Any increase in the signal's level beyond this will be clipped, and harmonic distortion artefacts will result.

Figure 3.6 shows two examples of the relationship between the levels of the signal and the THD+N in a digital audio system. The red line shows a 24-bit encoding, the blue line is for 16-bit.

The "flat line" on the left of the plot is the result of the noise floor of the system. In this region, the signal level is so low, it's below the noise floor of the system itself, so the only measurable output is the noise, and not the signal. As we move towards the right, the input signal gets louder and raises above the noise floor, so the output level naturally increases as well. However, in a digital audio system, we reach a maximum possible input level of 0 dB FS. If we try to increase the signal's level above this, the signal itself will not get louder, however, it will become more and more distorted. As a result, the distortion artefacts quickly become almost as loud as the signal itself, and so the plots drop dramatically.

This is why good recording engineers typically attempt to align the levels of the microphones to ensure that the maximum peak of the entire recording will just barely reach the maximum possible level of the digital recording system. This ensures that they are keeping above the noise floor as much as possible without distorting the signals.

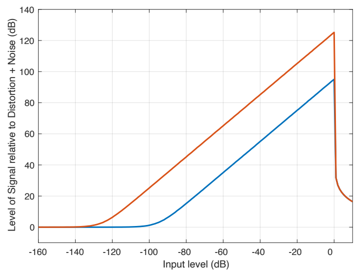

Figure 3.6: Two examples of the relationship between the levels of the signal and the THD+N in a digital audio system. These are idealised calculations, assuming TPDF dither in a "perfect" LPCM system. The red line shows a 24-bit encoding, the blue line is for 16-bit.

Audio signals recorded on analogue-only devices generally have the same behaviour; there is a noise floor that should be avoided and a

[3]The assumption here is that the distortion produces harmonics of the signal, which is a simplified view of the truth, but one that is easy to measure.

maximum level above which distortion will start to increase. However, many analogue systems have a slightly different characteristic, as can be seen in the idealised model shown in Figure 3.7. Notice that, just like in the digital audio system, the noise floor is constant, and as the level of the input signal is increased, it rises above this. However, in an analogue system, the transition to a distorted signal is more gradual, seen as the more gentle slopes of the curves on the right side of the graph.

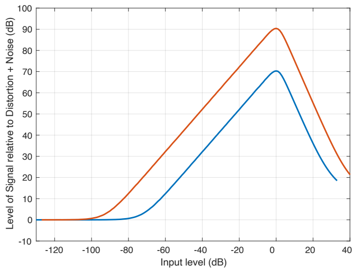

Figure 3.7: Two examples of the relationship between the levels of the signal and the THD+N in a simplified analogue audio system, showing two different maximum SNRs.

As a result, in a typical analogue audio system, there is an “optimal” level that is seen to be the best compromise between the signal being loud enough above the noise floor, but not distorting too much. The question of how much distortion is “too much” can then be debated – or even used as an artistic effect (as in the case of so-called “tape compression”).

If we limit our discussion to the stylus tracking a groove on a vinyl disc, converting that movement to an electrical signal that is amplified and filtered in a RIAA-spec preamplifier, then a phonograph recording is an analogue format. This means, generally speaking, that there is an optimal level for the audio signal, which, in the case of vinyl, means a modulation velocity of the stylus, converted to an electrical voltage.

Although there are some minor differences of opinion, a commonly-accepted optimum level for the groove on a stereo recording is 35.4 mm/sec for a single audio channel at 1,000 Hz. In a case where both audio channels have the same 1 kHz signal recorded in phase (as a dual-monophonic signal), then this means that the lateral velocity of the stylus will be 50 mm/sec.[4]

Of course, the higher the modulation velocity of the stylus, the higher the output of the turntable. However, this would also mean that the groove on the vinyl disc would require more space, since it is being modulated more. This means that there is a relationship between the total playing time of a vinyl disc and the modulation velocity. In order to have 20 minutes of music on a 12” LP spinning at 33 1/3 RPM, then it the standard method was to cut 225 “lines per inch” or “LPI” (about 89 lines per centimetre) on the disc. If a mastering engineer wishes to have a signal with a higher output, then the price is a lower playing time (because the grooves much be spaced further apart to accommodate the higher modulation velocity) however, in well-mastered recordings, this spacing is varied according to the dynamic range of the audio signal. In fact, in some classical recordings, it is easy to see the louder passages in the music because the grooves are intentionally spaced further apart, as is illustrated in Figure 3.8.

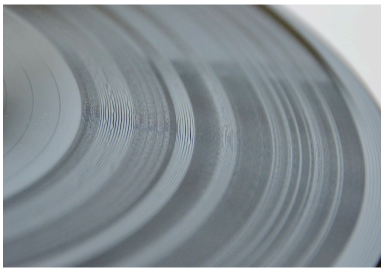

Figure 3.8: An extreme example of a disc in which the groove spacing has been varied to accommodate louder passages in the music. One consequence of this is that this side of the disc contains a single piece of music only 15 and a half minutes long.

As can be seen in the Technical Specifications at the end of this document, the Beogram 4000c is factory-calibrated so that a standard reference modulation velocity of 35.4 mm/sec on one channel at 1 kHz will produce an output of 354 mV RMS at the output. For a sine wave, this corresponds to a peak level of 500 mV. It can safely be connected to a Line input of any audio device.

However, it should be noted that the maximum possible output level of the turntable is 8.0 V peak, which may, in fact, be reached with some discs. Consequently, it should be noted that playing these recordings with higher modulation velocities (and therefore higher output levels) may result in the Beogram 4000c clipping the Line input stage of a device connected “downstream” (depending on its maximum allowed input level).

If you are connecting the Beogram 4000c to the RCA Line input of a Beolab 90 or Beolab 50, it is recommended that you set the Maximum Input Level of that input on the loudspeaker to 4.0 V RMS (which corresponds to 5.7 V peak) or 6.5 V RMS (9.2 V peak) using its Input Setup menu. This will ensure that you maintain adequate headroom for playback.

A large part of the performance of a turntable is dependent on the physical contact between the surface of the

[4] (35.4*2) / $\sqrt{2}$ because the two channels are modulated at an angle of 45° to the surface of the disc.

vinyl and the tip of the stylus. In general terms, as we're already seen, there is a groove with two walls that vary in height, almost independently and the tip of the stylus traces that movement accordingly. However, it is necessary to get down to the microscopic level to consider this behaviour in more detail.

When a record is mastered (meaning, when the master disc is created on a lathe) the groove is cut by a heated stylus that has a specific shape, shown in Figure 3.9. The depth of the groove can range from a minimum of 25 μm to a maximum of 127 μm, which, in turn varies the width of the groove. [5]

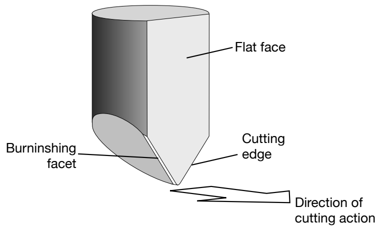

Figure 3.9: The cutting stylus used to create the groove in the master disc.

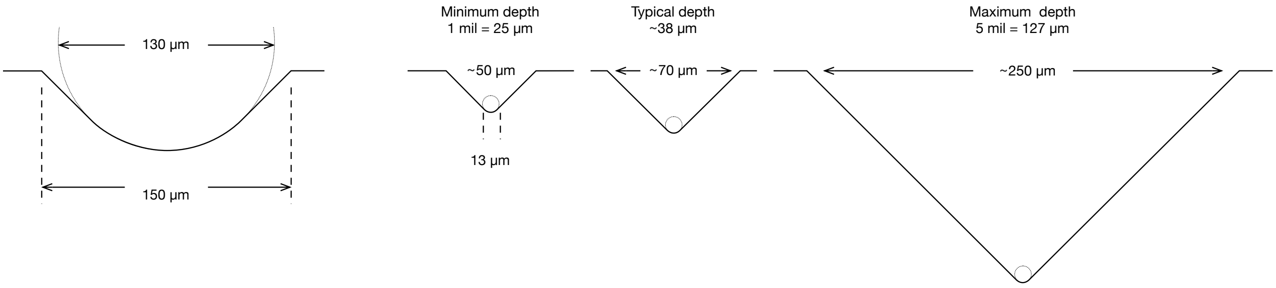

Figure 3.10: Dimensions of record grooves, drawn to scale. The figure on the left is typical for a 78 RPM shellac disc. The three grooves on the right show the possible variation in a 33 1/3 "microgroove" LP.

The result is a groove with a varying width and depth that are dependent on the decisions made by the mastering engineer, and a modulation displacement (the left/right size of the "wiggle") that is dependent on the level of the audio signal that is being reproduced.

In a perfect situation, the stylus that is used to play that signal back on a turntable would have exactly the same shape as the cutting stylus, since this would mean that the groove is traced in exactly the same way that it was cut. This, however, is not practical for a number of reasons. As a result, there are a number of options when choosing the shape of the playback stylus.

3.3 Tip shape

The earliest styli were the needles that were used on 78 RPM gramophone players. These were typically made from steel wire that was tapered to a conical shape, and then the tip was rounded to a radius of about 150 μm, by tumbling them in an abrasive powder.[6]. This rounded curve at the tip of the needle had a hemispherical form, and so styli with this shape are known as either *conical* or *spherical*.

The first styli made for "microgroove" LP's had the same basic shape as the steel predecessor, but were tipped with sapphire or diamond. The conical/spherical shape was a good choice due to the relative ease of manufacture, and a typical size of that spherical tip was about 36 μm in diameter. However, as recording techniques and equipment improved, it was realised that there are possible disadvantages to this design.

Remember that the side-to-side shape of the groove is a physical representation of the audio signal: the higher the frequency, the smaller the wave on the disc. However, since the disc has a constant speed of rotation, the speed of the stylus relative to the groove is dependent on how far away it is from the centre of the disc. The closer the stylus gets to the centre, the smaller the circumference, so the slower the groove speed.

If we look at a 12" LP, the smallest allowable diameter for the modulated groove is about 120 mm, which gives us a circumference of about 377 mm (or 120 * π). The disc is rotating 33 1/3 times every minute which means that it is making 0.56 of a rotation per second. This, in turn[7], means that the stylus has a groove speed of 209 mm per second. If the audio signal is a 20,000 Hz tone at the end of the recording, then there must be 20,000 waves carved into every 209 mm on the disc, which means that each wave in the groove is about 0.011 mm or 11 μm long.

[5] "The High-fidelity Phonograph Transducer" B.B. Bauer, JAES 1977 Vol 25, Number 10/11, Oct/Nov 1977
[6] ibid.
[7] pun intended

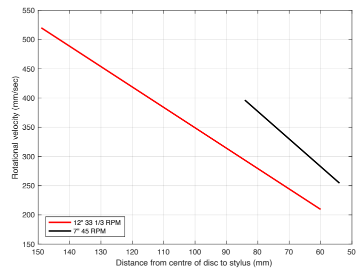

Figure 3.11: The relative speed of the stylus to the surface of the vinyl as it tracks from the outside to the inside radius of the record.

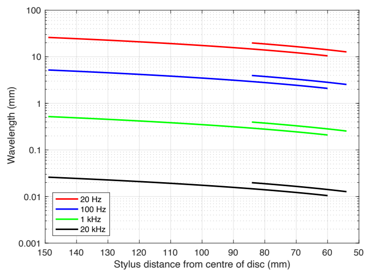

Figure 3.12: The wavelengths measured in the groove, as a function of the stylus's distance to the centre of a disc. The shorter lines are for 45 RPM 7"discs, the longer lines are for 33 1/3 RPM 12" LPs.

However, now we have a problem. If the "wiggles" in the groove have a total wavelength of 11 μm, but the tip of the stylus has a diameter of about 36 μm, then the stylus will not be able to track the groove because it's simply too big (just like the tires of your car do not sink into every small crack in the road). Figure 3.13 shows to-scale representations of a conical stylus with a diameter of 36 μm in a 70 μm-wide groove on the inside radius of a 33 1/3 RPM LP (60 mm from the centre of the disc), viewed from above. The red lines show the bottom of the groove and the black lines show the edge where the groove meets the surface of the disc. The blue lines show the point where the stylus meets the groove walls. The top plot is a 1 kHz sine wave and the bottom plot is a 20 kHz sine wave, both with a lateral modulation velocity of 70 mm/sec. Notice that the stylus is simply too big to accurately track the 20 kHz tone.

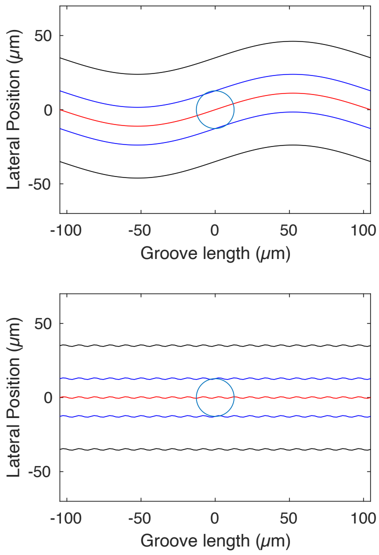

Figure 3.13: Scale representations of a conical stylus with a diameter of 36 μm in a 70 μm-wide groove on the inside radius of a 33 1/3 RPM LP, looking directly downwards into the groove. See the text for more information.

One simple solution was to "sharpen" the stylus; to make the diameter of the spherical tip smaller. However, this can cause two possible side effects. The first is that the tip will sink deeper into the groove, making it more difficult for it to move independently on the two audio channels. The second is that the point of contact between the stylus and the vinyl becomes smaller, which can result in more wear on the groove itself because the "footprint" of the tip is smaller. However, since the problem is in tracking the small wavelength of high-frequency signals, it is only necessary to reduce the diameter of the stylus in one dimension, thus making the stylus tip *elliptical* instead of conical. In this design, the tip of the stylus is wide, to sit across the groove, but narrow along the groove's length, making it small enough to accurately track high frequencies. An example showing a 0.2 mil x 0.7 mil (10 x 36 μm) stylus is shown in Figure 3.14. Notice that this shape can track the 20 kHz tone more easily, while sitting at the same height in the groove as the conical stylus in Figure 3.13.

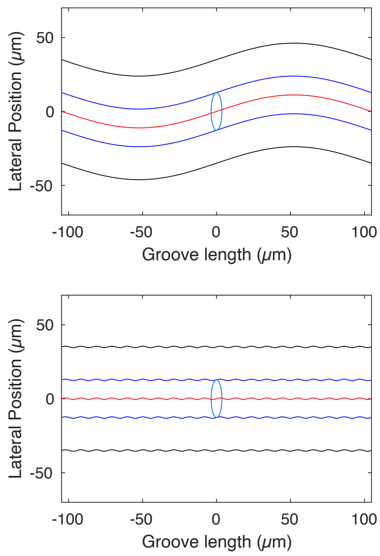

Figure 3.14: Scale representations of an elliptical stylus with diameters of 10 x 36 μm in a 70 μm-wide groove on the inside radius of a 33 1/3 RPM LP, looking directly downwards into the groove. See the text for more information.

Both the conical and the elliptical stylus designs have a common drawback in that the point of contact between the tip and the groove wall is extremely small. This can be seen in Figure 3.15, which shows various stylus shapes from the front. Notice the length of the contact between the red and black lines (the stylus and the groove wall). As a result, both the groove of the record and the stylus tip will wear over time, generally resulting in an increasing loss of high frequency output. This was particularly a problem when the CD-4 Quadradisc format was introduced, since it relies on signals as high as 45 kHz being played from the disc. In order to solve this problem, a new stylus shape was invented by Norio Shibata at JVC in 1973. The idea behind this new design is that the sides of the stylus are shaped to follow a much larger-radius circle than is possible to fit into the groove, however, the tip has a small radius like a conical stylus. An example showing this general concept can be seen on the right side of Figure 3.15.

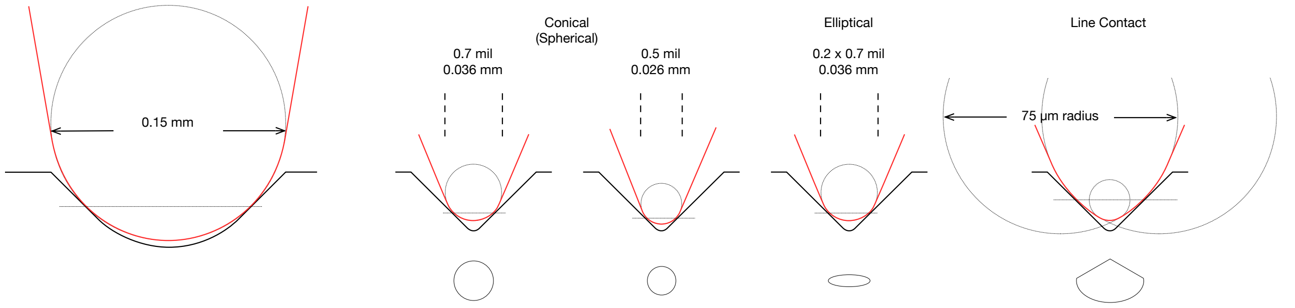

Figure 3.15: Dimensions of example styli, drawn to scale. The figure on the left is typical for a 78 RPM steel needle. The four examples on the right show different examples of tip shapes. These are explained in more details in the text. (For comparison, a typical diameter of a human hair is about 0.06 mm.)

There have been a number of different designs following Shibata's general concept, with names such as MicroRidge (which has an interesting, almost blade-like shape "across" the groove), Fritz-Geiger, Van-den-Hul, and Optimized Contour Contact Line. Generally, these designs have come to be known as *line contact* (or *contact line*) styli, because the area of contact between the stylus and the groove wall is a vertical line rather than a single point.

Originally, the Beogram 4002 was supplied with an MMC 6000 cartridge, which featured a stylus tip designed by Subir K. Pramanik, an engineer at Bang & Olufsen. This became known as the Pramanik diamond, and was designed to ensure maximum surface area with the groove wall on its vertical axis while maintaining a minimum contact along the horizontal axis.

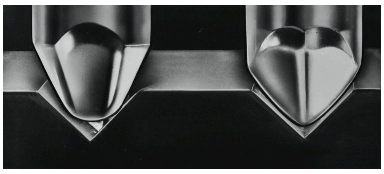

Figure 3.16: An example of an elliptical stylus on the left vs. a line contact Pramanik grind on the right. Notice the difference in the area of contact between the styli and the groove walls.

3.4 Bonded vs. Nude

There is one small, but important point regarding a stylus's construction. Although the tip of the stylus is almost always made of diamond today, in lower-cost units, that diamond tip is mounted or *bonded* to a steel pin which is, in turn, connected to the cantilever (the long "arm" that connects back to the cartridge housing). This bonded design is cheaper to manufacture, but it results in a high mass at the stylus tip, which means that it will not move easily at high frequencies.

Figure 3.17: Scale models (on two different scales) of different styli. The example on the left is bonded, the other four are nude.

In order to reduce mass, the steel pin is eliminated, and the entire stylus is made of diamond instead. This makes things more costly, but reduces the mass dramatically, so it is preferred if the goal is higher sound performance. This design is known as a *nude* stylus.

3.5 Tracking force

In order to keep the stylus tip in the groove of the record, it must have some force pushing down on it. This force must be enough to keep the stylus in the groove. However, if it is too large, then both the vinyl and the stylus will wear more quickly. Thus a balance must be found between "too much" and "not enough".

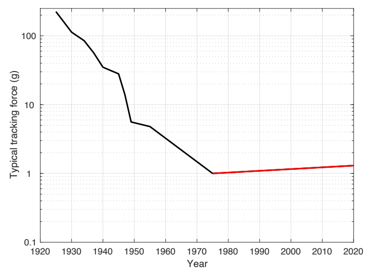

Figure 3.18: Typical tracking force over time. The red portion of the curve shows the recommendation for Beogram 4002 and Beogram 4000c.

As can be seen in Figure 3.18, the typical tracking force of phonograph players has changed considerably since the days of gramophones playing shellac discs, with values under 10 g being standard since the introduction of vinyl microgroove records in 1948. The original recommended tracking force of the Beogram 4002 was 1 g, however, this has been increased to 1.3 g for the Beogram 4000c in order to help track more recent recordings with higher modulation velocities and displacements.

3.6 Effective Tip Mass

The stylus's job is to track all of the vibrations encoded in the groove. It stays in that groove as a result of the adjustable tracking force holding it down, so the moving parts should be as light as possible in order to ensure that they can move quickly. The total *apparent* mass of the parts that are being moved as a result of the groove modulation is called the *effective tip mass*. Intuitively, this can be thought of as giving an impression of the amount of inertia in the stylus.

It is important to not confuse the tracking force and the effective tip mass, since these are very different things. Imagine a heavy object like a 1500 kg car, for example, lifted off the ground using a crane, and then slowly lowered onto a scale until it reads 1 kg. The “weight” of the car resting on the scale is equivalent to 1 kg. However, if you try to push the car sideways, you will obviously find that it is more difficult to move than a 1 kg mass, since you are trying to overcome the inertia of all 1500 kg, not the 1 kg that the scale “sees”. In this analogy, the reading on the scale is equivalent to the Tracking Force, and the mass that you’re trying to move is the Effective Tip Mass. Of course, in the case of a phonograph stylus, the opposite relationship is desirable; you want a tracking force high enough to keep the stylus in the groove, and an effective tip mass as close to 0 as possible, so that it is easy for the groove to move it.

3.7 Compliance

Imagine an audio signal that is on the left channel only. In this case, the variation is only on one of the two groove walls, causing the stylus tip to ride up and down on those bumps. If the modulation velocity is high, and the effective tip mass is too large, then the stylus can lift off the wall of the groove just like a car leaving the surface of a road on the trailing side of a bump. In order to keep the car’s wheels on the road, springs are used to push them back down before the rest of the car starts to fall. The same is true for the stylus tip. It’s being pushed back down into the groove by the cantilever that provides the spring. The amount of “springiness” is called the *compliance* of the stylus suspension. (Compliance is the opposite of spring stiffness: the more compliant a spring is, the easier it is to compress, and the less it pushes back.)

Like many other stylus parameters, the compliance is balanced with other aspects of the system. In this case it is balanced with the effective mass of the tonearm (which includes the tracking force[8]), resulting in a resonant frequency. If that frequency is too high, then it can be audible as a tone that is “singing along” with the music. If it’s too low, then in a worst-case situation, the stylus can jump out of the record groove.

If a turntable is very poorly adjusted, then a high tracking force and a high stylus compliance (therefore, a “soft” spring) results in the entire assembly sinking down onto the record surface. However, a high compliance is necessary for low-frequency reproduction, therefore the maximum tracking force is, in part, set by the compliance of the stylus.

If you are comparing the specifications of different cartridges, it may be of interest to note that compliance is often expressed in one of five different units, depending on the source of the information:

- “Compliance Unit” or “cu”
- mm/N
  millimetres of deflection per Newton of force
- μm/mN
  micrometres of deflection per thousandth of a Newton of force
- x 10^{-6} cm/dyn
  hundredths of a micrometre of deflection per dyne of force
- x 10^{-6} cm/10^{-5} N
  hundredths of a micrometre of deflection per hundred-thousandth of a Newton of force

Since

- mm/N = 1000 μm / 1000 mN
- 1 dyne = 0.00001 Newton

Then this means that all five of these expressions are identical, so, they can be interchanged freely. In other words:

20 CU
= 20 mm / N
= 20 μm / mN
= 20 x 10^{-6} cm / dyn
= 20 x 10^{-6} cm / 10^{-5} N

3.8 Soundsmith SMMC20CL

Today, the Beogram 4000c is supplied with an SMMC20CL cartridge made by Soundsmith. This is a nude contact line diamond stylus with a solid sapphire cantilever. It is a variable reluctance design, based on the original Bang & Olufsen Micro Moving Cross construction.

The SMMC20CL has an effective tip mass of 0.32 mg, lower than the 0.5 mg of the original MMC 4000 cartridge, but slightly higher than the 0.22 mg in the MMC 6000 cartridge.

[8] On the Mechanics of Tonearms, Dick Pierce

Audio Specifications

4.0.1 Magnitude Response

The magnitude response[1] of any audio device is a measure of how much its output level deviates from the expected level at different frequencies. In a turntable, this can be measured in different ways.

In the case of the Beogram 4000c, the frequency response is measured from a standard test disc with a sine wave sweep ranging from at least 20 Hz to at least 20 kHz. The output level of this signal is recorded at the output of the device, and the level is analysed to determine how much it differs from the expected output. Consequently, the measurement includes all components in the audio path from the stylus tip, through the RIAA preamplifier, to the line-level outputs.

4.0.2 Rumble

In theory, an audio playback device only outputs the audio signal that is on the recording without any extra contributions. In practice, however, every audio device adds signals to the output for various reasons. As was discussed above, in the specific case of a turntable, the audio signal is initially generated by very small movements of the stylus in the record groove. Therefore, in order for it to work at all, the system must be sensitive to very small movements in general. This means that any additional movement can (and probably will) be converted to an audio signal that is added to the recording.

This unwanted extraneous movement, and therefore signal, is usually the result of very low-frequency vibrations that come from various sources. These can include things like mechanical vibrations of the entire turntable transmitted through the table from the floor, vibrations in the system caused by the motor or imbalances in the moving parts, warped discs which cause a vertical movement of the stylus, and so on. These low-frequency signals are grouped together under the heading of *rumble*.

A rumble measurement is performed by playing a disc that has no signal on it, and measuring the output signal's level. However, that output signal is first filtered[2] to ensure that the level detection is not influenced by higher-frequency problems that may exist.

If the standard being used for the rumble measurement is the DIN 45 539 specification, then the resulting value is stated as the level difference between the measured filtered noise and a the standard output level, equivalent to the output when playing a 1 kHz tone with a lateral modulation velocity of 70.7 mm/sec.

4.0.3 Rotational speed

Every recording / playback system, whether for audio or for video signals, is based on the fundamental principle that the recording and the playback happen at the same rate. For example, a film that was recorded at 24 frames (or photos) per second (FPS) must also be played at 24 FPS to avoid objects and persons moving too slowly or too quickly. It's also necessary that neither the recording nor the playback speed changes over time.

A phonographic LP is mastered with the intention that it will be played back at a rotational speed of 33 1/3 RPM (Revolutions Per Minute) or 45 RPM, depending on the disc. (These correspond to 1 revolution either every 1.8 seconds or every 1 1/3 seconds respectively.) We assume that the rotational speed of the lathe that was used to cut the master was both very accurate and very stable. Although it is the job of the turntable to duplicate this accuracy and stability as closely as possible, measurable errors occur for a number of reasons, both mechanical and electrical. When these errors are measured using especially-created audio signals like pure sine tones, the results are filtered and analyzed to give an impression of how audible they are when listening to music. However, a problem arises in that a simple specification (such as a single number for "Wow and Flutter", for example) can only be correctly interpreted with the knowledge of how the value is produced.

Accuracy

The first issue is the simple one of accuracy: is the turntable rotating the disc at the correct *average* speed? In the Beogram 4000c, this speed is governed by a tachometer built into the drive motor, that (like on almost all turntables) can be adjusted by the user using the controls shown in Figure 4.1. These adjustments allow for a usable range of $\pm$ 3%, which in music terms is equivalent to $\pm$ half of a semitone.

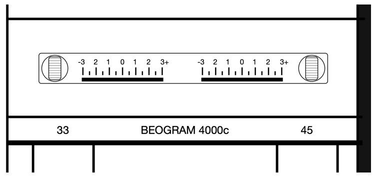

Figure 4.1: The fine adjustment controls for the 33 1/3 and 45 RPM settings.

Stability

Like any audio system, regardless of whether it's analogue or digital, the playback speed of the turntable will vary over time. As it increases and decreases, the pitch of the music at the output will increase and decrease proportionally. This is unavoidable. Therefore, there are two questions that result:

[1] This is the correct term for what is typically called the "frequency response". The difference is that a magnitude response only shows output level vs. frequency, whereas the frequency response would include both level and phase information.

[2] The characteristics of the filters are defined in internal standards such as DIN 45 539 and IEC98-1964.

- How much does the speed change?
- What is the rate and pattern of the change?

In a turntable, the amount of the change in the rotational speed is directly proportional to the frequency shift in the audio output. Therefore for example, if the rotational speed decreases by 1% (for example, from 33 1/3 RPM to exactly 33 RPM), the audio output will drop in frequency by 1% (so a 440 Hz tone will be played as a 440*0.99 = 435.6 Hz tone). Whether this is audible is dependent on different factors including

- the rate of change to the new speed
  (a 1% change 4 times a second is much easier to hear than a 1% change lasting 1 hour)
- the listener's abilities
  (for example, a person with "absolute pitch" may be able to recognise the change)
- the audio signal
  (It is easier to detect a frequency shift of a single, long tone such as a note on a piano or pipe organ than it is of a short sound like a strike of claves or a sound with many enharmonic frequencies such as a snare drum.)

In an effort to simplify the specification of stability in analogue playback equipment such as turntables, four different classifications are used, each corresponding to different rates of change. These are drift, wow, flutter, and scrape, the two most popular of which are wow and flutter, and are typically grouped into one value to represent them.

Drift

Frequency *drift* is the tendency of a playback device's speed to change over time very slowly. Any variation that happens slower than once every 2 seconds (in other words, with a *modulation frequency* of less than 0.5 Hz) is considered to be drift. This is typically caused by changes such as temperature (as the playback device heats up) or variations in the power supply (due to changes in the mains supply, which can vary with changing loads throughout the day).

Wow

Wow is a modulation in the speed ranging from once every 2 seconds to 6 times a second (0.5 Hz to 6 Hz). Note that, for a turntable, the rotational speed of the disc is within this range. (At 33 1/3 RPM: 1 revolution every 1.8 seconds is equal to approximately 0.556 Hz.)

Flutter

Flutter describes a modulation in the speed ranging from 6 to 100 times a second (6 Hz to 100 Hz).

Scrape

Scrape or *scrape flutter* describes changes in the speed that are higher than 100 Hz. This is typically only a problem with analogue tape decks (caused by the magnetic tape sticking and slipping on components in its path) and is not often used when classifying turntable performance.

Measurement and Weighting

The easiest accurate method to measure the stability of the turntable's speed within the range of Wow and Flutter is to follow one of the standard methods (of which there are many, but they are all similar[3]). A special measurement disc containing a sine tone, usually with a frequency of 3150 Hz is played to a measurement device which then does a frequency analysis of the signal. In a perfect system, the result would be a 3150 Hz sine tone. In practice, however, the frequency of the tone varies over time, and it is this variation that is measured and analysed.

There is general agreement that we are particularly sensitive to a modulation in frequency of about 4 Hz (4 cycles per second) applied to many audio signals. As the modulation gets slower or faster, we are less sensitive to it, as was illustrated in the example above: (a 1% change 4 times a second is much easier to hear than a 1% change lasting 1 hour).

So, for example, if the analysis of the 3150 Hz tone shows that it varies by $\pm$1% at a frequency of 4 Hz, then this will have a bigger impact on the result than if it varies by $\pm$1% at a frequency of 0.1 Hz or 40 Hz. The amount of impact the measurement at any given modulation frequency has on the total result is shown as a "weighting curve" in Figure 4.2.

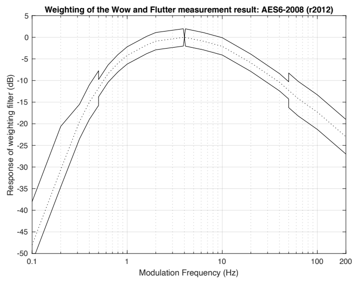

Figure 4.2: Weighting applied to the Wow and Flutter measurement in most standard methods. See the text for an explanation.

As can be seen in this curve, a modulation at 4 Hz has a much bigger weight (or impact) on the final result than a modulation at 0.315 Hz or at 140 Hz, where a 20 dB attenuation is applied to their contribution to the total result. Since attenuating a value by 20 dB is the same as dividing it by 10; a $\pm$ 1% modulation of the 3150Hz tone at 4 Hz will produce the same result as a $\pm$10% modulation of the 3150 Hz tone at 140 Hz, for example.

This shows just one example of why

[3]Examples of these standards are AES6-2008, CCIR 409-3, DIN 45507, and IEC-386

comparing one Wow and Flutter measurement value should be interpreted very cautiously...

Expressing the result

When looking at a Wow and Flutter specification, one will see something like <0.1%, <0.05% (DIN), or ±0.1% (AES6). Like any audio specification, if the details of the measurement type are not included, then the value is useless. For example, "W&F: <0.1%" means nothing, since there is no way to know which method was used to arrive at this value.[4]

If the standard is included in the specification (DIN or AES6, for example), then it is still difficult to compare wow and flutter values. This is because, even when performing identical measurements and applying the same weighting curve shown in Figure 4.2, there are different methods for arriving at the final value. The value that you see may be a peak value (the maximum deviation from the average speed), the peak-to-peak value (the difference between the minimum and the maximum speeds), the RMS (a version of the average deviation from the average speed), or something else.

The AES6-2008 standard, which is the currently accepted method of measuring and expressing the wow and flutter specification, uses a "2σ" or "2-Sigma" method, which is a way of looking at the peak deviation to give a kind of "worst-case" scenario. In this method, the 3150 Hz tone is played from a disc and captured for as long a time as is possible or feasible. Firstly, the average value of the actual frequency of the output is found (in theory, it's fixed at 3150 Hz, but this is never true). Next, the short-term variation of the actual frequency over time is compared to the average, and weighted using the filter shown in Figure 4.2. The result shows the instantaneous frequency variations over the length of the captured signal, relative to the average frequency (however, the effect of very slow and very fast changes have been reduced by the filter). Finally, the standard deviation of the variation from the average is calculated, and multiplied by 2 (hence "2-Sigma", or "two times the standard deviation"), resulting in the value that is shown as the specification. The reason two standard deviations is chosen is that (in the typical case where the deviation has a Gaussian distribution) the actual Wow & Flutter value should exceed this value no more than 5% of the time.

The reason this method is preferred today is that it uses a single number to express not only the wow and flutter, but the probability of the device reaching that value. For example, if a device is stated to have a Wow and Flutter of "1% (AES6)", then the actual deviation from the average speed will be less than 1% for 95% of the time you are listening to music. The principal reason this method was not used in the 1970s when the Beogram 4002 turntable was released is that it requires statistical calculations applied to a signal that was captured from the output of the turntable, an option that was not available 45 years ago. The older DIN method that was used showed a long-term average level that was being measured in real-time using analogue equipment such as the device shown in Figure 4.3.

Figure 4.3: Bang & Olufsen WM1, analogue wow and flutter meter.

Unfortunately, however, it is still impossible to know whether a specification that reads "Wow and Flutter: 1% (AES6)" means 1% deviation with a modulation frequency of 4 Hz or 10% deviation with a modulation frequency of 140 Hz – or something else. It is also impossible to compare this value to a measurement done with one of the older standards such as the DIN method, for example.

[4] Similarly, a specification like "Frequency Range: 20 Hz to 20 kHz" means nothing, since there is no information about the levels used to define the range.

Beogram 4000c Specifications

Turntable	Maximum	Typical
Output Level, 35.4 mm/sec, 1 channel, 1 kHz*		354 mV RMS
Maximum Output	$>$ 8.0 V Peak	
Deviation from RIAA standard, 20 - 20,000 Hz	$< \pm$ 1.0 dB	$\pm$ 0.6 dB
Channel difference, 20 - 20,000 Hz	$< \pm$ 0.5 dB	$\pm$ 0.2 dB
Speed drift (IEC-386)	$<$ 0.1 %	0.04 %
Wow and flutter (AES6-2008)	$<$ 0.15%	0.09%
Rumble (DIN 45 539)	$>$ 50 dB	$>$ 59 dB
Interchannel phase deviation at 1 kHz, 12” LP	$<$ 0.1 $^\circ$	
Pickup Cartridge		Soundsmith SMMC20CL
Stylus		Nude, 0.12 mm square
Radius of curvature		Contact Line
Recommended tracking force		1.3 g
Compliance		28 μm/mN
Effective tip mass		0.32 mg
General information		
Automatic speed selection		Yes
Speeds		33 1/3 and 45 RPM
Speed control range		$\pm$ 3%
Dial for speed		2 pointers
Tonearm		Tangential
Lift system		Electronic
Automatic pickup movement		Yes
Automatic record size		Yes
Motor		Controlled synchronous
Drive system		Belt
Dustcover		Hinged and detachable
Power supply		110, 130, 220, 240 volts
Frequency		50, 60 Hz
Dimensions W x H x D		49 x 10 x 38 cm
Weight		12 kg

* Measurement includes Soundsmith SMMC20CL

Further Reading

Audible Effects of Mechanical Resonances in Turntables; Brüel and Kjær Application Note (1977)

Audio Measurement Handbook; Bob Metzler, Audio Precision Inc. (1993)

Basic Disc Mastering; Larry Boden (1981)

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How the Stereo Disc Works; Radio-Electronics, (July 1958)
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Manual of Analogue Sound Restoration Techniques; Peter Copeland (2008)

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- 45 541: Frequency Test Record St 33 and M 33 (33 1/3 rev/min; Stereo and Mono)
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- 45 543: Frequency Response and Crosstalk Test Record
- 45 544: Rumble Measurement Test Record St 33 and M 33 (33 1/3 rev/min; Stereo and Mono)
- 45 545: Wow and Flutter Test Records, 33 1/3 and 45 rev/min
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