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Extraction of Sound using 3D

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Summary

1. Introduction	5
1.1. History.....	5
1.2. 78 rpm discs	5
1.2.1. Dimensions of the groove profile	5
1.3. Optical reading methods.....	6
1.3.1. 2D imaging.....	6
1.3.2. 3D imaging.....	6
1.3.3. Redundancy	7
1.4. Goal of "Extraction of sound using 3D"	7
1.5. Achieved work	8
2. Task list	8
2.1. Getting started	8
2.2. Data acquisition.....	8
2.3. Study of the data acquisition system.....	8
2.4. Algorithms extracting sound from data	9
2.5. Analysis of sound quality.....	9
3. Study of the data acquisition system.....	9
3.1. Overview of the hardware	9
3.1.1. Process to scan an entire disc	10
3.2. Confocal microscope MPLS180	11
3.2.1. The principle of confocal microscopes.....	11
3.2.2. MPLS180 characteristics	12
3.2.3. Performances of MPLS180 on discs	14
3.2.4. Intrinsic error of the probe.....	17
3.2.5. Conclusion of the intrinsic error measurement.....	21
3.2.6. Acquired 3D height map of a disc.....	21
3.3. XPS motion controller	21
3.4. Motion stages	22
3.4.1. Rotation stage	22
3.4.2. Translation stage	22
3.4.3. Focusing stage	23
3.4.4. Error due to rotation.....	23
3.4.5. Error due to translation	25
3.5. Autofocus System	26
3.5.1. Description	26
3.5.2. Error due to autofocus.....	27
3.6. Other imperfection of the system	28
3.6.1. Irregular circle of scan ring	28
3.6.2. Lateral density of points	28
3.7. Conclusion of data acquisition system analysis.....	28
3.7.1. Sampling frequency and angular scanning velocity	28
3.7.2. Decrease number of bad points.....	30
3.7.3. Increase resolution.....	30

3.7.4.	Offset removal.....	31
3.7.5.	Error due to translation and rotation.....	31
3.7.6.	X-position of MPLS180 points	31
3.7.7.	Irregularity of the circle	31
4.	Data acquisition software.....	32
4.1.	Front Panel	32
4.2.	Modes	34
4.3.	Disc scanning mode.....	34
4.3.1.	Sequence for disc data acquisition.....	34
4.3.2.	Order of multiple axial steps and multiple scan passes.....	35
4.4.	Disc test mode.....	35
4.5.	Data file formats.....	35
4.5.1.	The *.pri file	35
4.5.2.	The *.bri File	37
5.	Analysis software	38
5.1.	Choice of programming environment	38
5.2.	Description of "3DSEVEN"	38
5.2.1.	Block diagram	39
5.2.2.	Front panel - GUI	40
5.2.3.	Data visualizing.....	41
5.2.4.	Data statistics.....	41
5.2.5.	Parameter setting	41
5.3.	Data reading.....	42
6.	Data cleaning and filtering algorithms.....	43
6.1.	Offset removal.....	43
6.2.	Averaging of overlapped scans	43
6.2.1.	Point failure percentage.....	44
6.3.	Low pass filter.....	44
6.3.1.	Filter design	45
6.3.2.	Filter implementation.....	46
6.3.3.	Data after filtering	47
6.3.4.	Average in time direction.....	48
7.	Algorithms extracting sound from data.....	50
7.1.	Approximate method to find groove centers.....	50
7.2.	Algorithms to find the exact groove centre	50
7.2.1.	Groove slopes intersection method.....	51
7.2.2.	Constant Line method	51
7.2.3.	Advanced constant line method	52
7.2.4.	Decision	52
7.2.5.	Least Squares Method.....	53
7.3.	Problems occurred during development	54
7.4.	Tracking of the grooves	55
7.5.	Extracting the sound	58
7.5.1.	Derivative Method	59
7.5.2.	High pass filter	60
7.6.	Create the .wav File	60
7.7.	Create a bitmap.....	61
7.7.1.	Error sources	62

7.8.	Histograms (Statistics)	63
7.8.1.	Height.....	63
7.8.2.	Number of points per slope	64
7.8.3.	Sum of Squared Residual	64
7.8.4.	Interpolated Slopes.....	65
7.8.5.	Sound difference	65
7.9.	Cuts	67
7.9.1.	Slope fitting cut.....	67
7.9.2.	Sound cut	67
7.10.	The groove file (*.grv)	68
8.	Analysis of sound	70
8.1.	RIAA equalization.....	70
8.1.1.	Transfer Function	70
8.1.2.	Effect	72
8.2.	SINAD	73
8.3.	THD	73
8.4.	LabVIEW SINAD VI	73
8.5.	Sony Sound Forge 9.0	74
8.6.	Analysis of the constant frequency record transcription	74
8.7.	Analysis of extracted sound of the constant frequency record	75
8.7.1.	Without overlapped scans	76
8.8.	Density of points	77
8.8.1.	Low pass filter.....	79
8.8.2.	Weighted average	80
8.8.3.	Size of virtual stylus	80
8.8.4.	Cuts	82
8.8.5.	Manual sound restoration for improvement.....	83
8.8.6.	Comparison to I.R.E.N.E.....	84
8.8.7.	Conclusion	84
8.9.	Analysis of music extraction	85
8.9.1.	In the evening by the moonlight	86
8.9.2.	Chattanooga blues	87
8.9.3.	I wonder where my easy rider's gone	88
8.10.	Conclusion of the sound analysis.....	88
9.	Scanning and processing time	89
9.1.	Scanning time	89
9.2.	Processing time	89
10.	Further development	89
10.1.	Cuts.....	89
10.2.	Groove detection	90
10.3.	Include varying fiber spacing.....	90
10.4.	45° measurements.....	90
11.	Conclusion.....	91
12.	Acknowledgments.....	92
13.	Figures	93
14.	References	96

1. Introduction

1.1. History

In the digital age, our technology allows us to record and store music and images with devices we always have in our pockets. This has been possible only for a few years, and it took a long time to get to this point.

Leon Scott recorded first sounds with his phonoautograph back in 1857. But the mechanical cylinder phonograph invented in 1877 by Thomas Edison was the first practical sound recording and reproduction device that signified the start of commercial recording, distribution and sale of sound recordings.

Due to its easier manufacture, transport and storage, the phonographic disc introduced in 1889 by Emile Berliner overtook the cylinders by 1910 and became the major recording format in the 20th century before the digitalization of sounds.

In the first half of the 20th century, before the introduction of magnetic tapes, phonographic records were the most common way to store music, and a large part of the world's audio heritage is stored on 78 rpm discs, which was the standard music format in this period.[1]

1.2. 78 rpm discs

First phonographic record discs were mostly made of shellac. The sound recorded on it was stored on a groove spiral from outside to the disc's centre. The principle for playing back the sound was basically that a stylus followed the groove and the sound was produced by the lateral velocity of the needle.

The double-sided 78 rpm disc was first recorded purely acoustically, and after the introduction of sound transducers (microphone and loudspeaker) electrically, which had the advantage that the recorded sound was louder and filtered.[1]

1.2.1. Dimensions of the groove profile

Figure 1 shows the typical dimensions of a groove. The angle of the groove walls is 45 °.

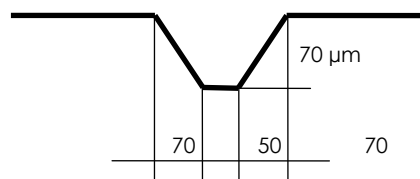


Figure 1: Dimension of a groove

One of the major disadvantages of all the mechanical carriers was that, as the discs were fragile, and as they were played, the grooves deteriorated.

1.3. Optical reading methods

To avoid the abrasion and to extract sound without physical contact, different optical reading methods imaging the carriers have been investigated. By digital processing, it is possible to reconstruct sound from discs that are not playable with a stylus because they are cracked or worn, which sometimes lead to skipping from one groove to the next.

Different optical reading systems were developed at Lawrence Berkeley National Laboratory (LBNL) as well as at the University of Applied Sciences at Fribourg (EIA FR). The main difference between the different projects is the optical scanning method, which retrieves sound data from mechanical carriers. Two- and three-dimensional scanners have been employed.

1.3.1. 2D imaging

The two-dimensional scanning process basically consists of acquiring a high-resolution monochromatic picture of the medium. This picture contains information about the slope at each point, whereas a steep section is represented by dark pixels and a flat section by bright pixels.

The 2D method was employed by Carl Haber and Earl Cornell from LBNL to obtain sound from discs in their so-called project I.R.E.N.E. They were able to extract sound from phonographic records, which in some cases was comparable to the quality of analogue playback systems. [2]

A very similar project called VisualAudio is being investigated at the University of Applied Sciences in Fribourg in Switzerland, to retrieve the audio data from damaged phonographic records that either were cracked or whose lacquer layer has shrunk over the years. By taking high-resolution photographs on a film and then scanning these films, the team at EIA FR was able to save an amount of unplayable records digitally.[3]

1.3.2. 3D imaging

The three-dimensional scanning process consists of acquiring a height map of the sound carrier instead of a picture representing the slopes. The depth of each point is accurately measured by a confocal microscope to represent the surface of the support. By digital processing the stylus movement is simulated virtually to extract the sound.

This method was applied in a former diploma project by two students of University of Applied Sciences Fribourg at LBNL to extract sound from dictation belts with a single-point color-coded confocal microscope.[4] The same probe was used to extract sound from cylinders. Unlike for discs and the dictation belts, where sound is stored as a lateral modulation of the groove, cylinders have vertical groove modulation, which requires the 3D method.

The main problem of the single-point confocal microscope was the long duration of data acquisition. This is the reason why LBNL acquired a multiple-point line sensor (MPLS180), which can measure the depth of 180 points simultaneously, obviously promising much more rapid acquisition.

MPLS180 was employed at LBNL to scan cylinders and extract the sound. Rather than taking 80 hours, this probe is able to scan a whole cylinder in just 20 minutes.

1.3.3. Redundancy

The major difference between mentioned imaging methods is the redundancy. 3D scanning acquires much more information about a groove profile hence redundancy is higher than for a 2D imaging. A 2D scanner basically just says whether a surface is flat or not, whereas a 3D scanner returns the height of a measured point. With 2D scanning a groove is characterized just by 4 edges:

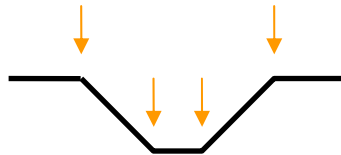


Figure 2: Measurement points of 2D system

With 3D scanning a groove is characterized by the height of lots of points across the groove profile.

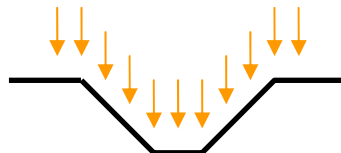


Figure 3: Measurement points of 3D system

When redundancy is higher, more measurements determine a groove center and an error on one point has less weight. But if there are only 4 points determining a groove center the failure of just one point has much more influence. Finally with a 3D scanner the groove center position should be found more accurate and reliable, supposed that the disc images well.

1.4. Goal of “Extraction of sound using 3D”

The goal of the diploma project “Extraction of sound using 3D” was to scan 78’s using MPLS180 and extract sound from them. The data acquisition hardware was similar to that used for I.R.E.N.E., so not a lot of mechanical work had to be done. But the software for acquiring data did not yet exist to use MPLS180 for discs. The digital processing of the acquired data also had to be investigated in order to be able to play sound.

To get satisfying results, the entire hardware had to be studied in detail, by determining the influence of several parameters and measurement errors.

1.5. Achieved work

In the course of "Extraction of sound using 3D", the data acquisition has been studied and evaluated, an MPLS180 data acquisition software has been adapted for disc scanning and a analyzing software called "3DSEVEN" has been created in order to scan discs and retrieve sound.

We got to the point where we could extract sound from discs. This sound was analyzed and compared to I.R.E.N.E.

2. Task list

2.1. Getting started

The first steps of a project are always to acquire information about all the equipment that will be used, as well as becoming acquainted with former similar research that has already been conducted. In the case of "Extraction of sound using 3D," this involves the study of confocal microscopes and the lecture of documentation of former projects such as VisualAudio, I.R.E.N.E, 3D Cylinder sound reconstruction and the diploma project of 2 EIA-FR students at LBNL describing the scanning of dictation belts.

We also have to become accustomed to new software that is suitable for this project. This is **LabVIEW** (for **L**aboratory **V**irtual **I**nstrumentation **E**ngineering **W**orkbench), a platform and development environment for a visual programming language, which will be used for the data acquisition, and **Visual C#** for the data processing algorithms.

To understand how MPLS180 works, we can use the manufacturer's software to perform some initial measurements of the probe on discs, to get an idea of what we can expect the data to look like.

2.2. Data acquisition

Before being able to study the quality and accuracy of the data, we had to scan the data by rotating / translating the disc and saving the readout values of MPLS180 in a file. A LabVIEW program will be designed for this. Such a program to acquire data from cylinders has already been developed. As the task of data acquisition from discs was very similar to the existing software used for cylinders, a few modifications of this program provided satisfactory storage of scanned disc data.

2.3. Study of the data acquisition system

Once data can be acquired and saved in a file, the data has to be analyzed and the influence of parameters upon it has to be studied. With more knowledge about accuracy and errors of the system, we can define measurement criteria, and the quality of acquired data can be improved.

2.4. Algorithms extracting sound from data

The data then has to be treated digitally. Software has to be written in order to clean the scans and filter bad points. This also consists on averaging when multiple measurements of the same point were taken. Further the groove centers have to be found. This can be done with different methods, all coming from different approaches. Different algorithms can be compared. The grooves have to be tracked and their lateral displacement has to be converted into sound, this is basically the concept to extract sound from the acquired data.

2.5. Analysis of sound quality

Being able to extract sound from discs, we can compare different algorithms and analyze their performances. We can calculate signal to noise ratio, harmonic distortion, applied on a test record scan, where we exactly know the content. This analysis can tell us about modification we have to apply to our algorithms.

3. Study of the data acquisition system

It is essential to get the best possible data from the discs, the quality of the extracted sound will depend directly on how accurate the scanned data represents the profile of the grooves. Errors due to bad measurements can never be fixed by the analysis software anymore, they can be treated but will still influence the sound quality negatively.

In order to get the best possible data from discs a good understanding of the system is fundamental. The study of the data acquiring system in this chapter considers the influence of parameters on measurements, accuracy and errors of the system.

3.1. Overview of the hardware

As mentioned earlier in this report, the heart of the data acquiring system is the confocal microscope MPLS180 (MPLS for multiple point line sensor). 2 stages for rotation and translation of the disc platform and a probe focusing stage allow bringing any spot of the disc in the range of the probe. The LAN connected xps motion controller controls the movements of these 3 stages.

Moreover there is a height detector for the auto focusing of the probe to keep measurements in range even if the scanned disc is warped.

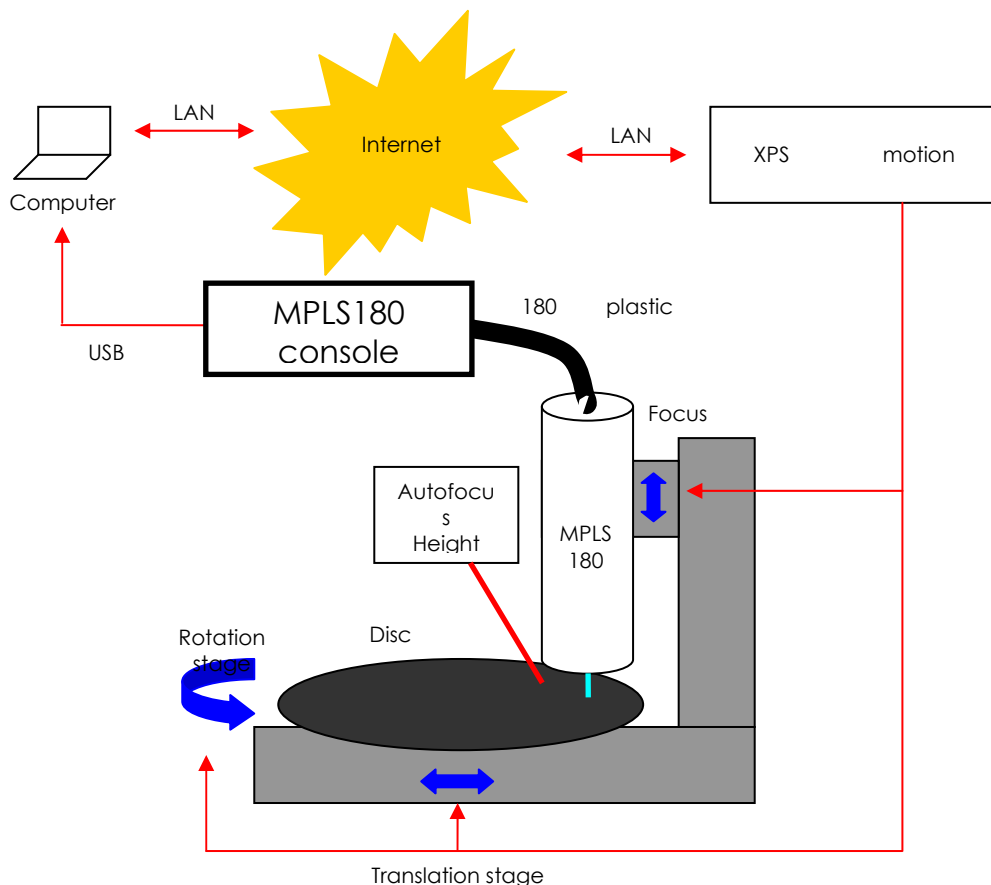


Figure 4: Hardware overview

We are working with the same hardware than I.R.E.N.E. aside from the scanner, thus no big mechanical modification has to be done. The software to control stage movements basically exists, but has to be adapted to scan discs.

3.1.1. Process to scan an entire disc

The hardware above is able to access any spot of the disc and acquire its data. There are a variety of approaches we can take to scanning a disc. A crucial factor is the position of the probe to the groove. We can choose between two positions, the line of the sensor along the grooves or across the grooves.

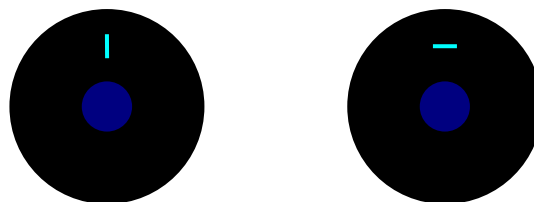


Figure 5: Different scanning directions (across or along the grooves)

And in any position we can tilt MPLS180. Tilting the probe would affect the angle the beams hit the surface what could possibly improve data acquisition

at groove slopes of reflecting discs. A mirror was actually made to hit the disc in an angle of 45° . By turning MPLS180 and arranging the 180 points along the groove, its wall can be detected very well. But issues with auto focusing and rearranging of the data are very complicated to handle.

We rather scan the disc in an intuitional manner. Placing the 180 measure points across the grooves (left drawing) and then rotate the disc, like the historical playback devices. Sequential scanning while rotating the disc by 360° without translation creates data corresponding to a ring of the disc. We then can translate the disc (which comes to the same than moving the probe in the other direction) and spin the disc by another 360° to scan a second ring and so on.

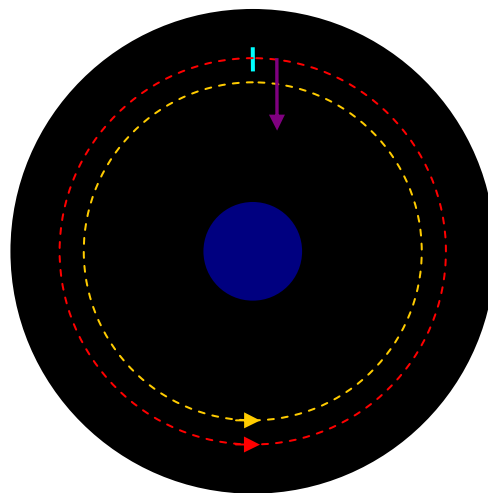


Figure 6: Scanning process with axial steps placing MPLS across the groove

Mentioned hardware is able to rotate/translate the disc and control the stages to execute data acquisition as described.

3.2. *Confocal microscope MPLS180*

The multiple point line sensor is basically a confocal microscope which is able to measure 180 points simultaneously. Confocal microscopes perform 3D measurements, hence they are able to draw a height map of the scanned object. In the case of this project the confocal microscope is used to scan the groove profile of a 78 rpm-disc.

3.2.1. The principle of confocal microscopes

The basic principle is to illuminate only one spot on the sample at a time through a pinhole. The light reflected from the sample is imaged by the objective back to the pinhole. If the sample moves out of focus the reflected light is defocused at the pinhole and will not be detected.

If the lens has a significant chromatic aberration, the wavelength dispersion of light reflected from different depths may be measured and analyzed by a spectrometer to find the wavelength (=color) that is in focus. This

spectrometer works basically as a peak detector, since the intensity of the wavelength that is in focus on the surface will be the highest. [7]

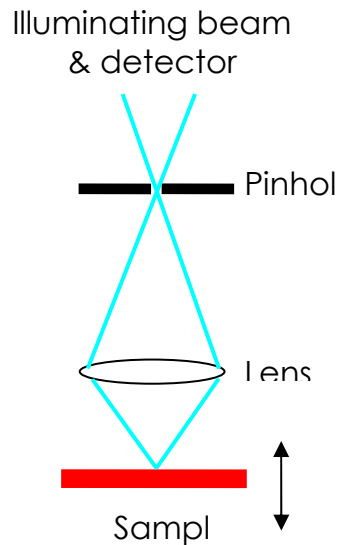


Figure 7: Principle of the confocal microscope

3.2.2. MPLS180 characteristics

A 300 Watt Xenon light source feeds the MPLS180 probe. This lamp is located in the MPLS180 console, which contains also the light detector to measure finally the height of the sample. 180 plastic fibers bound with a black cladding connect the console to the probe and light reaches the sample through a lens with important chromatic aberration. The reflecting light reaches the light detector through 180 fibers after being split from the incoming light by fiber coupling.

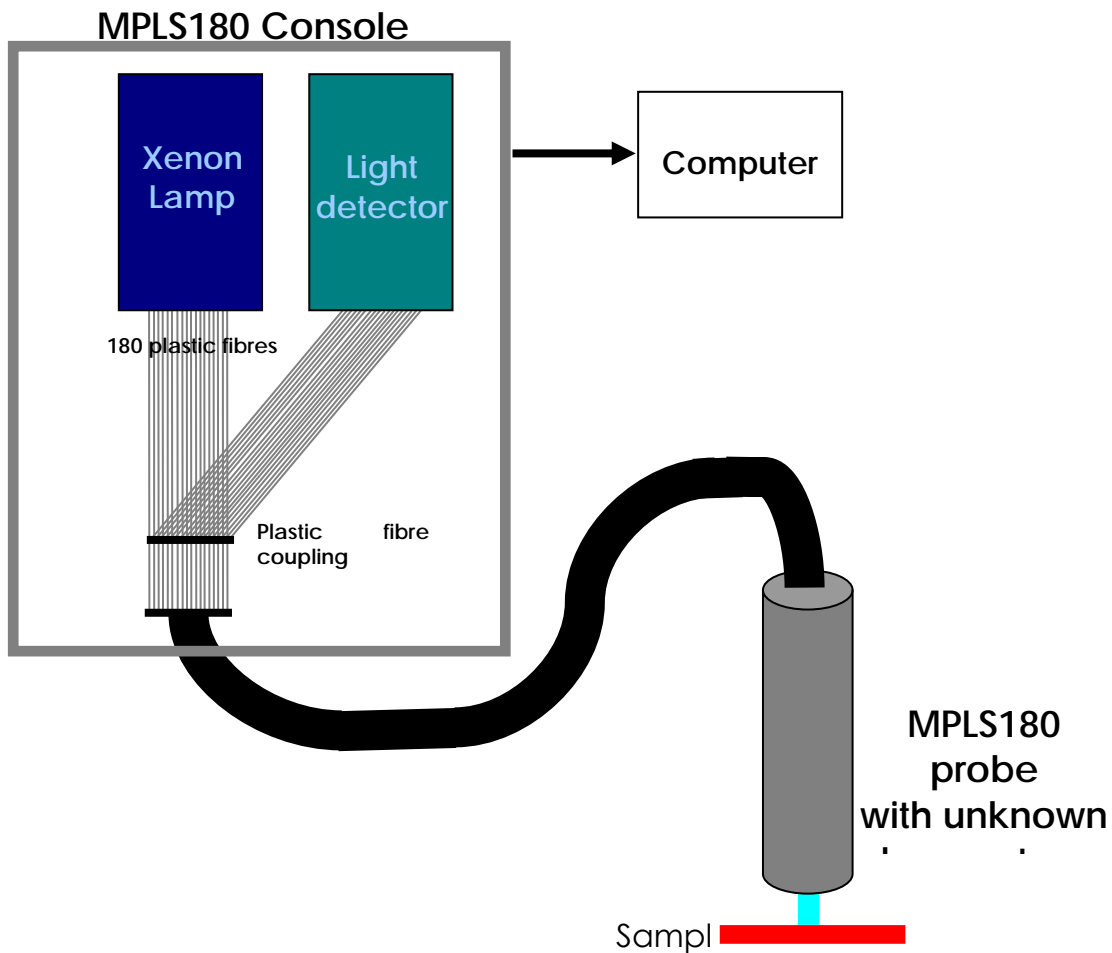


Figure 8: Overview of MPLS180 system

The MPLS180 confocal microscope from STIL images the height of a set of 180 points lying along a line simultaneously. The points are spaced by $10\text{ }\mu\text{m}$, therefore the total length of a scan is 1.8 mm . Each point has a diameter of $3.5\text{ }\mu\text{m}$.

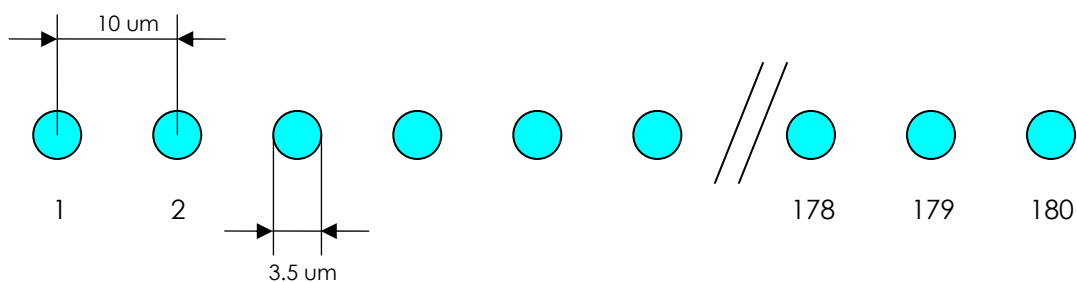


Figure 9: Spacing between points of the MPLS180

The confocal microscope can operate between a frequency of 200 Hz and 1.8 kHz , corresponding to exposure times between 0.55 and 5 ms .

The plastic fibers have a core diameter of $50\text{ }\mu\text{m}$ and a cladding diameter of $125\text{ }\mu\text{m}$. To split the beam, fiber coupling is used. This is a method where light

can be transmitted from one fiber to another by aligning the fibers over a certain distance together.

We do not have a lot of specification about MPLS180. For example we do not know anything about the lens system used. STIL does not want to give any indication about it. [5]

3.2.3. Performances of MPLS180 on discs

The resolution

Resolution in depth:	120 nm
Lateral resolution (spacing of fibers):	about 10 μm [5]

The depth range

MPLS180 is able to keep points in focus in a range of about 400 μm . Although on angled surfaces with marginal light intensity we only get good results towards the end of its depth range (largest readout values = when the probe is closer to the sample). In the lower section of the range, the intensity of the reflected light is often too weak which ends in bad points (0 returned). This range is not sufficient to scan an entire rotation of the disc (the disc is either warped or the platform is not perfectly flat). This is the reason why an auto focusing system is installed. For any height measurement a float value between 0 and 450 is returned.

Exposure time and dark current calibration

MPLS180 operates with an exposure time between 0.55 – 5 ms. To calibrate the confocal microscope for different exposure times the dark current calibration has to be performed.

This calibration measures the offset of each point when out of focus (=dark) and subtracts this offset from the performed measurement. When exposure time is high, lots of fibers are turned off and their value becomes an average of its neighbors. For 2.5 ms exposure time, up to 20 fibers out of 180 fail, for 5 ms up to 50 fibers. Tests have shown that an exposure time of 2.5 ms leads to best results. It is a good compromise between good measurement at groove walls, low noise level and low amount of fiber failures due to the dark current calibration.

Intensity

The light detector in the MPLS180 console measures also the light intensity at each point. The returned value is within the bounds of 0 and 5000. The intensity level hasn't been considered for further digital processing. Bad points correlate directly with the intensity. If the intensity is too weak, a default value (=0) is returned.

Bad fibers

Over the 180 fibers, there are some bad fibers that more often measure bad points than others. It is not random noise, because by repetition of the same measurement, the same fibers will get bad points again. A careful cleaning of the optical system might improve this failure. [6]

X-position

We first assumed that the 180 light beams leave the chromatic aberration lens straight. In fact, this is not the case, they are a little bit spread and the outermost fibers hit the sample with a considerable angle. That results in point spacing depending on the height of the scanned sample. STIL provided us their formula to calculate the x-position in function of the height and the fiber number.

It is especially annoying that the fibers are spread when we want to scan overlapped, as the measurement of the centre fiber comes together with the outside fiber.

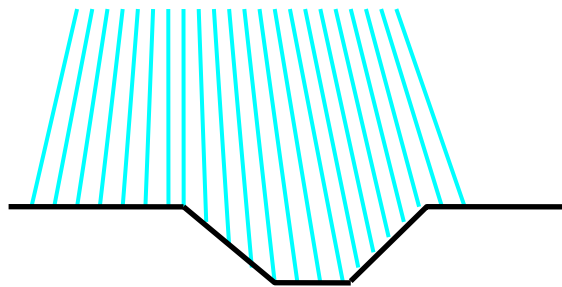


Figure 10: Spreading of fibers

Problem at the groove walls

As described, a confocal microscope measures the height by detecting the wavelength that is in focus. The intensity of the returned light depends a lot on the surface integrity of the sample. As long as the sample is flat, a reflecting surface returns the light best, because it behaves basically like a mirror. Matt surfaces reflect less light, although measurements have proven that MPLS180 is capable to detect the height of both matt and brilliant flat samples accurately.

But the problem is at the groove walls of a disc. The geometrical optics say that light leaves a reflecting surface with the same angle than it was beamed on.



Figure 11: Angle of incidence equals angle of reflection

When the beam of the confocal microscope now hits on the 45° slope of a brilliant disc's groove, exactly the same effect will happen.

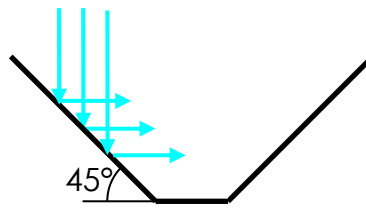


Figure 12: Reflection in a slope section

Hence no light is returned to the detector and we get a bad measure point. Obviously a reflecting disc is not a perfect mirror and therefore sometimes the intensity of the returned light is still high enough to be detected, this permits a height measurement at that point.

Old black Joe

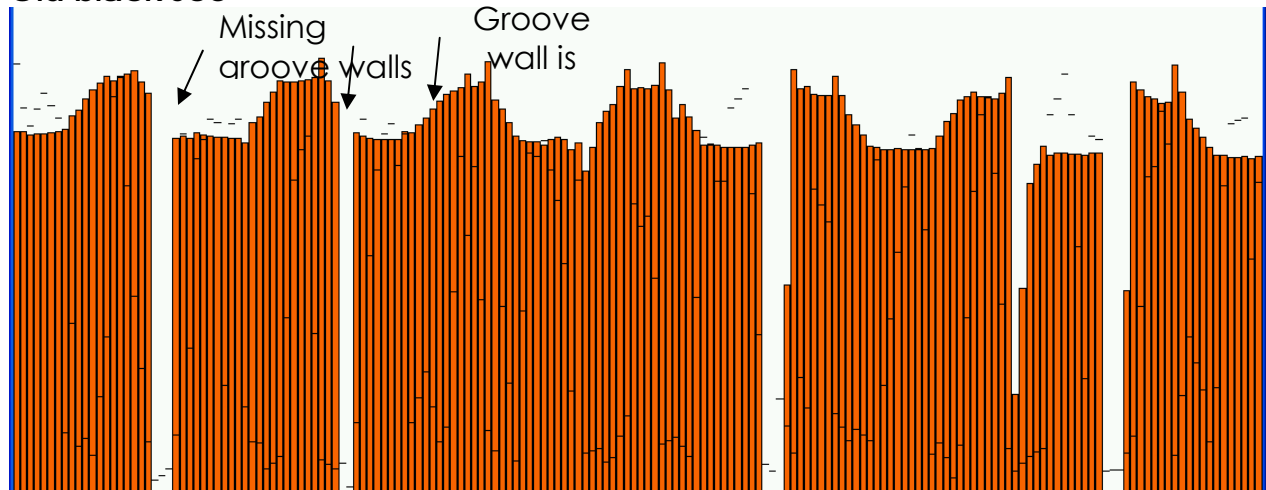


Figure 13: Acquired height profile with missing groove walls

Note: This plot is from the software that comes with MPLS180, it shows the distance of the sample to the probe. The groove bottom corresponds to the higher lobes, the groove top to the lower samples. The groove profile is upside down!

For matt discs the beam will be diffused in any direction and light intensity can be detected and finally the height can be determined. With both high exposure time (2.5 – 5 ms) and matt discs like “In the evening by the moonlight” over the 180 points just a few or none of the fibers fail.

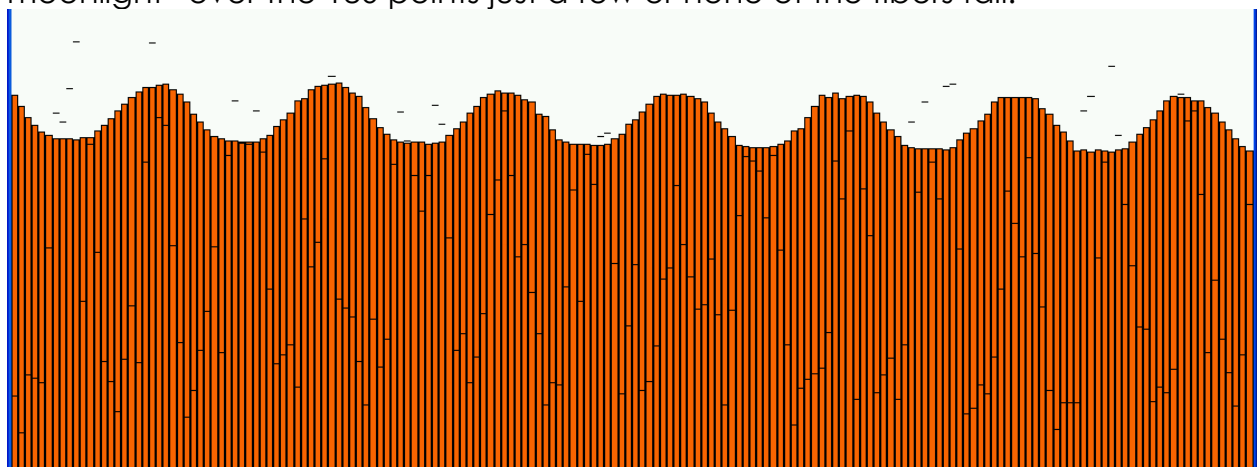


Figure 14: Acquired height profile in good quality

3.2.4. Intrinsic error of the probe

To know more about the performance of MPLS180 on discs, we evaluate its intrinsic error. We scan discs with different surfaces at different location such as groove top, groove bottom and on both groove walls by varying the exposure time.

At these different spots of the disc, the mean value, the root mean square value and the standard deviation of the height measurement were calculated using the LabVIEW data acquisition program in disc test mode (see chapter 4.2).

Different disc surfaces:

Pictures below show the discs we have most scanned on. The tone test record with constant frequency will be used for signal to noise ratio and total harmonic distortion determination.



In the evening by the moonlight
(Matt surface)



Test sound disc
(Mostly matt surface)



Old man blues (Reflecting surface)



Chattanooga Blues (Red matt surface)

Figure 15: Comparison of the phonographic records

Example of a measurement:

Disc	Moonlight
Label	Columbia
Number	A2683
Radial position	0
exposure time	2.5 ms

filenames	Disc-moon-intrinsic1-4			
Number of samples	18000			
fibers disabled by dark calibration	16			
	bottom of groove	top of groove	left slope of groove	right slope of groove
bin #	17	57	62	98
Mean [μm]	84.2455	137.444	106.378	110.483
Std [μm]	0.125532	0.110641	9.08402	0.523289
RMS [μm]	84.2455	137.444	106.765	101.484

Table 1: Measurement table for the „In the evening by the moonlight“ disc

Where: Mean : $\bar{x} = \frac{1}{n} \sum_{i=1}^n x_i$ EQ (1)

Std: $\sigma = \sqrt{\frac{1}{n} \sum_{i=1}^n (x_i - \bar{x})^2}$ EQ (2)

RMS: $x_{RMS} = \sqrt{\frac{1}{n} \sum_{i=1}^n x_i^2}$ EQ (3) [1]

Measurements of the mean value (EQ 1), the standard deviation (STD, EQ 2) and the root mean square (EQ 3) were repeated with exposure times of 1 ms, 1,25 ms, 1,66 ms, 2,5 ms and 5 ms. Shorter exposure times like 0,5 ms were not even worth to try because preliminary test gave bad results.

These measurements were performed on following discs:

Title	Label	Number	condition
In the evening by the moonlight	Columbia	A2683	Very old and worn, matt surface Images very good
Constant tone frequency	Columbia	10003-M	Matt, good condition images well
Chattanooga blues	Paramount	12063-B	Red (matt) surface, images well, very worn and cracked disc
Wabash Cannonball	Columbia	37008	Reflecting surface, images bad at slopes
Les bourgeois blues – Lead Belly	Musicraft	227-B	Reflecting surface, images bad at slopes
I'm looking for a girl	Edison	58912-R	Very reflecting

named Mary	Record		surface, images bad, grooves almost not recognizable
Easy Rider	Bluebird	B-10029- A	Matt surface, images well

Table 2: Characteristics of investigated discs

3.2.5. Conclusion of the intrinsic error measurement

Influence of exposure time

The shorter the exposure time is, the less light is reflected to the detector. For flat surfaces (groove top and groove bottom) this is no problem and the standard deviation is small. But at groove walls the STD increases, there is much more noise even if there are no bad points. This effect is especially important when the exposure time is short. Data acquisitions with an exposure time below 2.5 ms are useless.

Influence of surface

As mentioned in prior chapters, reflecting discs behave badly at the groove walls. We obtain a much higher number of bad points, therefore the standard deviation is big. But at the groove top and bottom, height measurements are accurate (STD low), even better than for matt surfaces. For the very clean and reflecting surface of "I'm looking for a girl named Mary" the grooves could not be recognized.

3.2.6. Acquired 3D height map of a disc

By moving the confocal microscope over the disc, a 3D height map can be acquired.

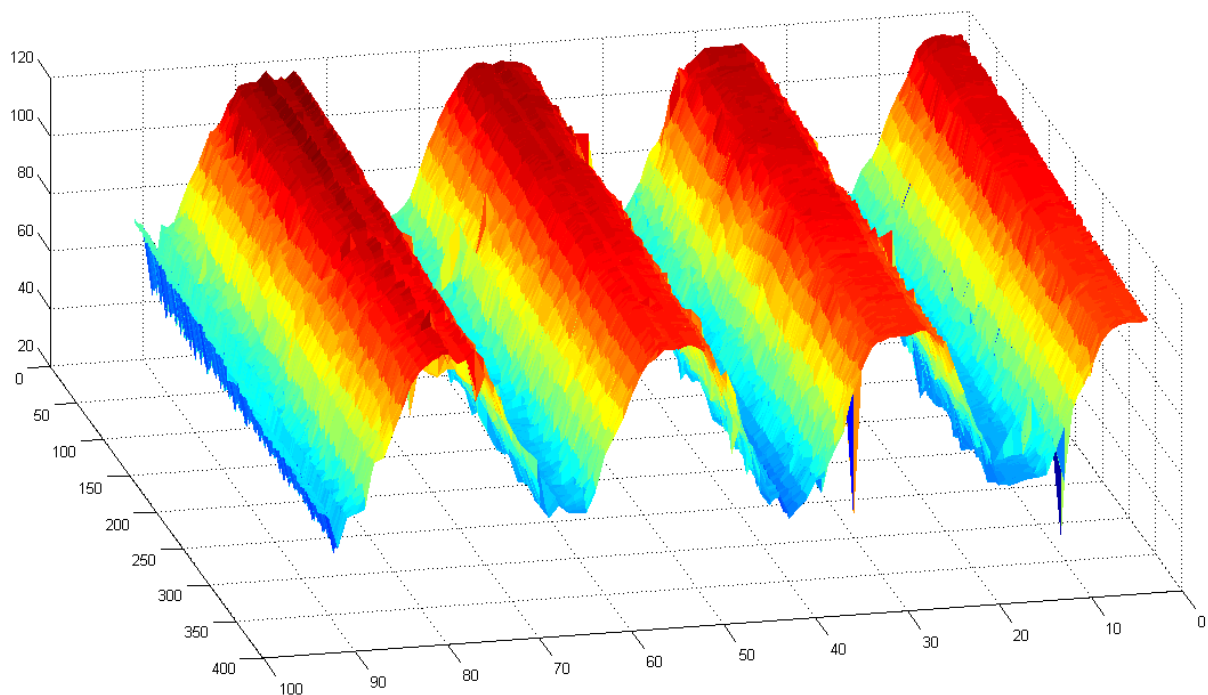


Figure 87: 3D height map of a phonographic disc with grooves

3.3. XPS motion controller

The XPS is a high-performance, easy to use, integrated motion controller offering high-speed communication through Ethernet, good accuracy and programming functionality. It is capable of driving up to 8 different stages powered by a 500 W supply. [8]

6 stages are connected on our XPS motion controller, whereof 3 concerning our project for scanning discs. The other 3 stages are used to scan cylinders.



Figure 16: XPS Motion Controller

3.4. Motion stages

To scan any spot of the disc and bring it into focus, we need 3 stages: a disc rotation, a disc translation and a probe focusing stage.

3.4.1. Rotation stage

For the disc rotation we use Newport's RGV 100 that they describe as "an ultra-low profile direct-drive rotation stage that provides ultra-fast rotations with very high resolution and outstanding positioning performance."

Its specification:

Travel Range	360° continuous
Resolution	0.0001°
Uni-directional repeatability	0.0002° typical, 0.0003° guaranteed
Reversal value (Hysteresis)	0.0002° typical, 0.0004° guaranteed
Accuracy	0.004° typical, 0.007° guaranteed
Maximum speed	720°/s

Table 3 : Specifications of the rotation stage [9]

Formulas describing the calculation of the values above can be found in the appendix.

For our application a constant velocity is important not to produce distortion of the sampling frequency as well as the uni-directional repeatability (when we want to scan 2 passes, we have to step at exactly the same initial point).

3.4.2. Translation stage

For the disc translation we use Newport's linear stage ILS250 CC that "offers 250 mm travel range and combines fast, sub-micron resolution motion in a

highly stiff and robust package", which is especially optimized to avoid bending effects due by thermal expansion.

Its specification:

Travel Range	250mm
Resolution	0.5 μm
Uni-directional repeatability	0.7 μm typical, 1.5 μm guaranteed
Reversal value	0.4 μm typical, 1 μm guaranteed
Bi-directional repeatability	0.6 μm typical, 2 μm guaranteed
On-axis accuracy	2.5 μm typical, 5 μm guaranteed
Maximum speed	100 mm/s

Table 4: Specifications of the translation stage [10]

The uni-directional-repeatability is especially important when we want to overlap to scans. For a 10 μm beam spacing of MPLS180 5 μm is a lot.

3.4.3. Focusing stage

To get the image into focus we have to move MPLS180 up and down. Newport's LTA-HS allows very accurate position.

Its specification:

Travel	50 mm
Resolution	0.035 μm
Minimum incremental Motion	0.1 μm
Uni-directional repeatability	0.15 μm typical, 0.5 μm guaranteed
Bi-directional repeatability	0.6 μm typical, 2 μm guaranteed
On-axis accuracy	5 μm typical, 15 μm guaranteed
Maximum speed	5 mm/s

Table 5: Specifications of the focusing stage [11]

This is probably the stage where accuracy is the least important of the 3 stages. The image just has to become into focus. As the focus range of MPL180 is about 400 μm and the groove depth about 70 μm , there is some tolerance.

With these 3 stages we are able to scan any position of the disc.

3.4.4. Error due to rotation

The data acquisition of an entire disc requires multiple axial steps. Any time one ring was scanned it is indispensable to return to the initial angular position, not to lose any data of the scan and that multiple ring scans fit perfectly together.

The rotation error was measured by affixing a sticker on the disc and acquiring the same data ring several times. MPLS180 detects the sticker in height. For every scan pass this variation in height should occur at the same position.

The profile of the entire sticker, approximately 120 μm high, is shown below. Whereas height measurements of 3 scan passes (green, blue and red) are represented.

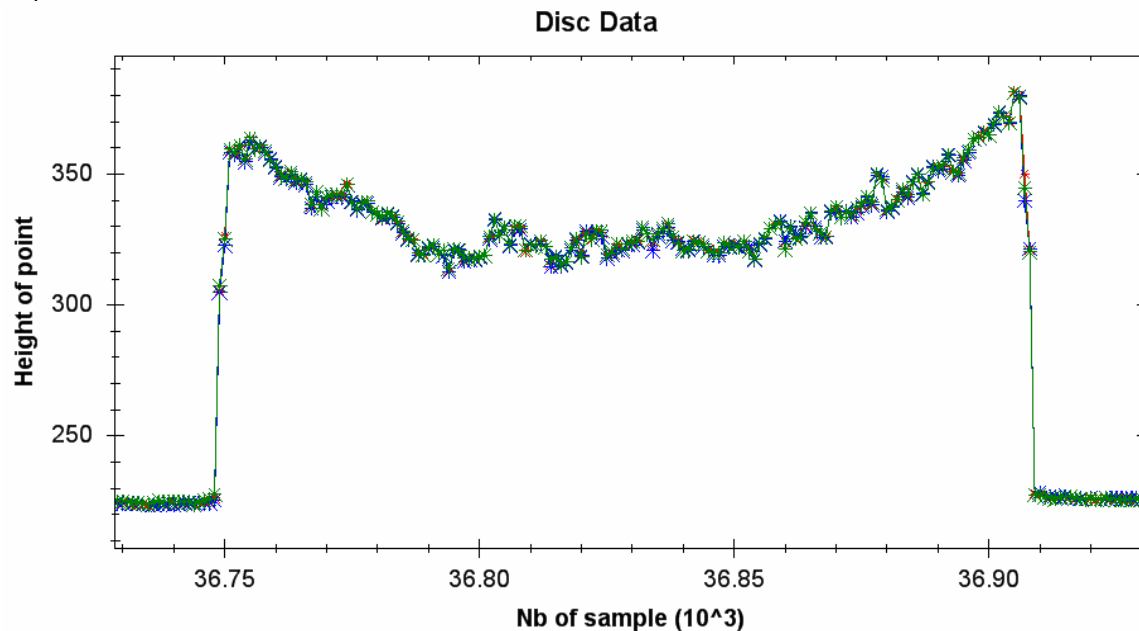


Figure 17: Measured height profile of the sticker in all three repetitions

A more detailed graphic shows the begin of the sticker:

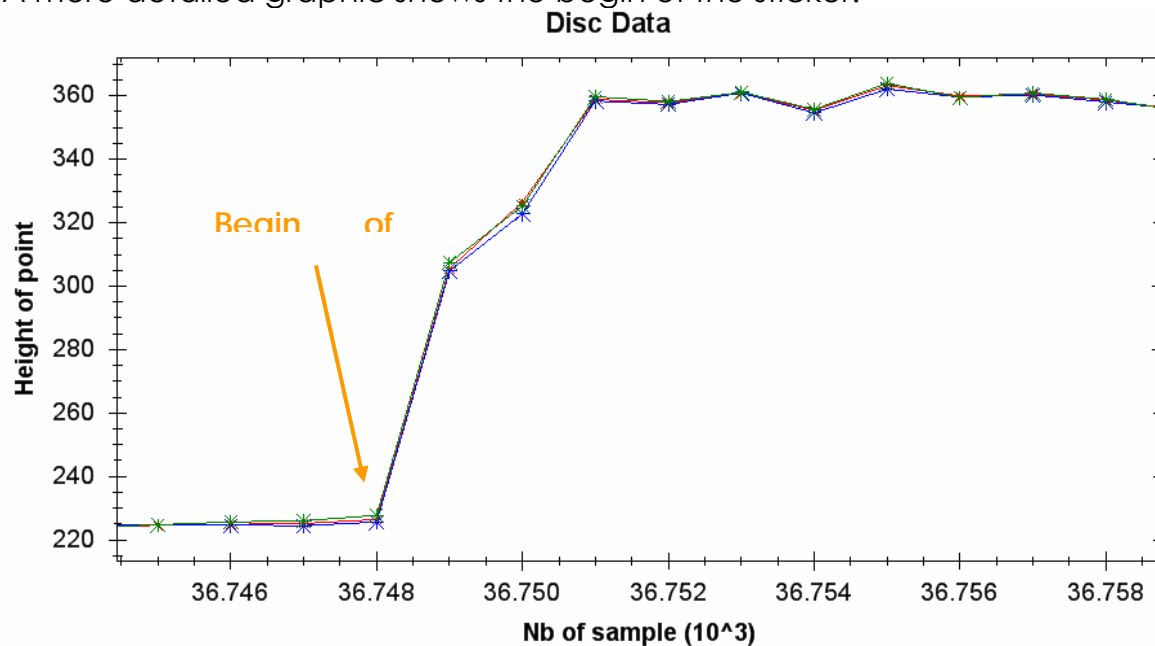


Figure 18: Zoomed height profile from the beginning of the sticker

This graphics show that the sticker was detected in each scan at exactly the same place.

Slight height differences occur due to measurement noise and the auto focus system, which will be considered later. The distance between 2 samples is 0.009° . At a radius of 12 cm, where the measurement was taken, it comes to about 20 μm .

In summary, rotation accuracy is very good and shouldn't affect sound quality negatively.

3.4.5. Error due to translation

Data acquisition with multiple axial steps also requires a very accurate translation of the disc. There shouldn't be any gap nor overlapping between different scan rings not to lose sound.

The error due to translation was measured by scanning a profile with just one groove. By averaging over a certain number of scan repetitions we obtained a mean value (like for the intrinsic error)

Then the disc platform was moved to the end of the translation range and put back at the initial position (about 250 mm of travel). And the measurement of the mean value was repeated. By determination of the groove centre of each repetition the approximate translation accuracy can be calculated.

For the determination of the groove centre, the same algorithm was used, as for the sound discs. It will be described later in this paper.

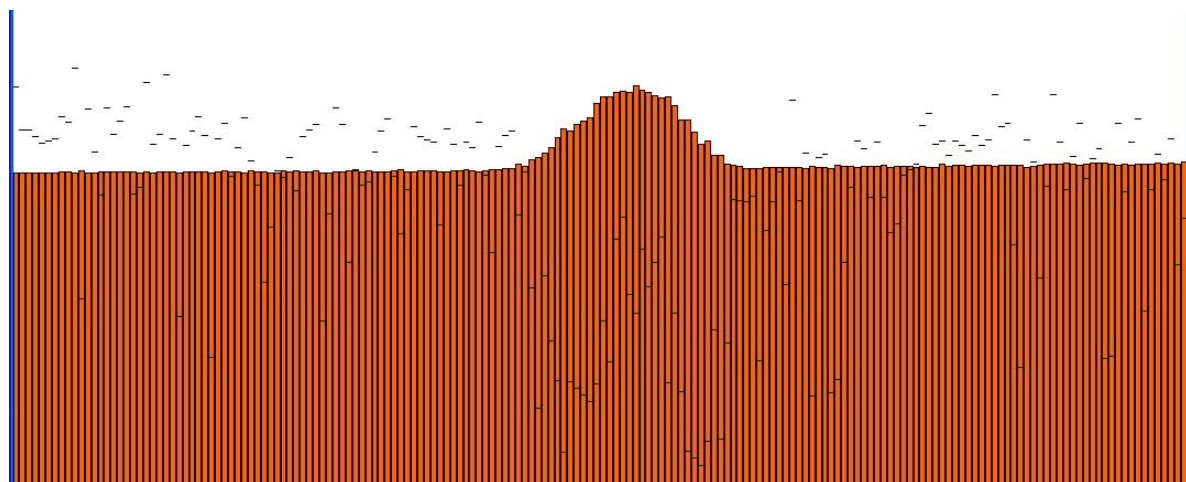


Figure 19: Height profile showing the acquired groove

Again, the groove is upside down!

Results:

The groove centre position was calculated on 10000 180-point-acquisitions; by taking the average of all the 10000 centers and by comparing them with the same measure repeated after shifting and coming back to the same place, we obtained these results:

	1st	2nd	3rd	4th
--	-----	-----	-----	-----

	measurement	measurement	measurement	measurement
Groove centre position [um]	87.15	87.15	87.35	87.45

Table 6: Results of the repetitive measurements

The difference between these 4 measurements is 0.4 μm . The typical uni-directional repeatability of the stage is 0.7 μm . Thus we are within the tolerance.

Compared to MPLS180's spot diameter of 3.5 μm the translation error can be neglected. It would be very complicated to remove it actually. The effect on the sound, when the translation is not accurate is a simple click when the groove passes from one scan ring to the next.

3.5. Autofocus System

As the scanned discs are over an entire rotation always a little warped, even though they are clipped on the platform, their height cannot always be kept in the 400 μm range of MPLS180. This is the reason why an autofocus system has to be lined up. I.R.E.N.E. already used an auto focus system. A few mechanical modifications fulfill making work it with MPLS180.

3.5.1. Description

A height detector measures the height of the disc just a couple of degrees in front of MPLS180's line. The knowledge about the discs height that will be scanned is then fed to the xps motion controller. A feedback control system controls the focusing stage and keep measurements always in the range of MPLS180, taking into account the angular advance of the height detector.

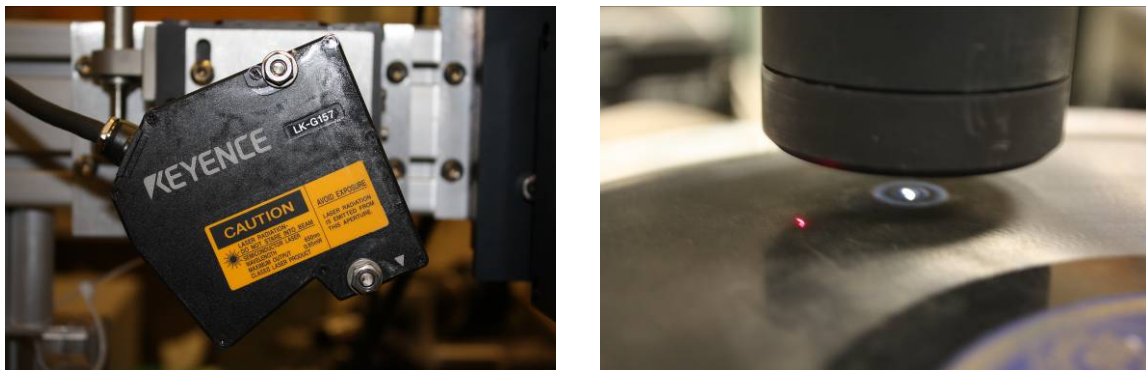


Figure 20: Keyence Height Detector (Left) and its measuring spot (Right)

3.5.2. Error due to autofocus

When we want to measure overlapped or/and interleaved and average data of different scan passes, imprecision of autofocus influences data quality. The autofocus control system will never control the focusing stage exactly the same for overlapped scans as the height detector's measurement spot will not be at the same axial position of the disc.

The autofocus error was measured by scanning a flat surface of a disc and overlapping two rings (because this will be the concept to scan discs).

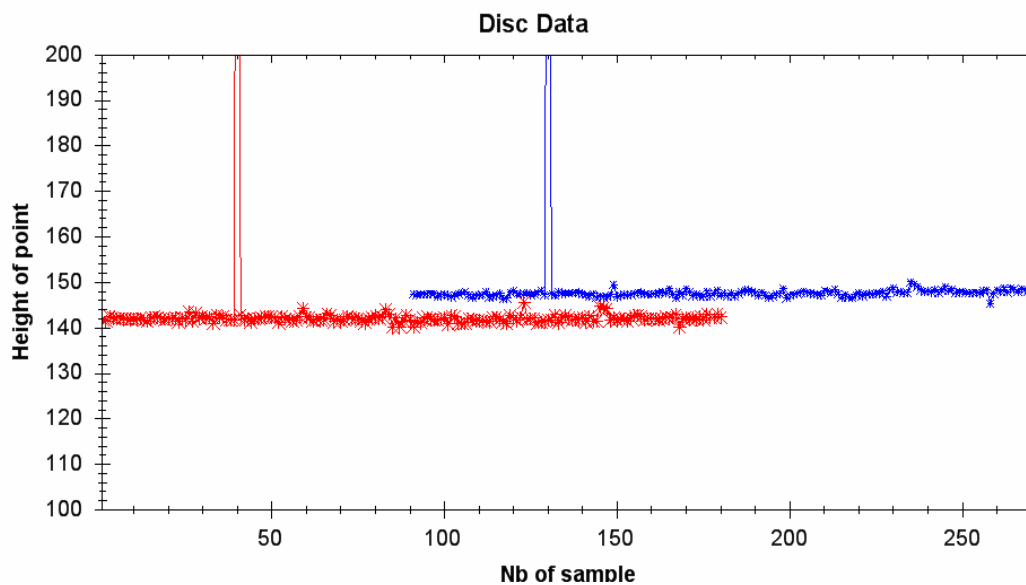


Figure 21: Error due to auto focus system of 2 overlapped passes

The red points represent the first pass, while the blue the second pass. As seen, error due to autofocus is considerable especially for overlapped scans and has to be treated.

3.6. Other imperfection of the system

3.6.1. Irregular circle of scan ring

Sometimes the grooves on the disc have a slightly elliptical form instead of being round. The scan ring becomes therefore irregular. Grooves pullout the rings and come in later. Grooves move on the scan ring basically like a sine wave with a frequency of 1.3 Hz (=78 rpm / 60 s) due to the ellipse. This frequency does not come into account in sound domain, as audible frequency is between 20 Hz and 20 kHz.

3.6.2. Lateral density of points

The lateral density of MPLS180 is about 10 μm . About 7 points hit the groove wall. As seen, sometimes lots of these points fail. If only 2 points fail, we already lose a lot of information about the groove walls.

To get the groove profile in a higher resolution and make bad points less important, we want to scan the same ring with several passes, spaced by either 2.5 μm or 5 μm . So we get much more points all over the groove profiles and better resolution. With more points any fitting will be more accurate.

3.7. Conclusion of data acquisition system analysis

All the imperfections of the data acquisition system force us to take several measures. Although we have to classify them in order to get rid of the ones with the biggest importance.

3.7.1. Sampling frequency and angular scanning velocity

The sampling frequency depends on the MPLS180's acquisition frequency and the rotation speed of the disc.

The disc spins at 78 revolutions per minute, which comes to a frequency of:

$$f = \frac{60s}{78rpm} = \underline{\underline{0.769Hz}} \Rightarrow T = \frac{78rpm}{60s} = \underline{\underline{1.3s}} \quad \text{EQ (4)}$$

Human hear frequencies up to 20 kHz. To respect the sampling theorem we need at least a sampling frequency of 40000 kHz. Because filtering of spectral repetitions due to sampling is never perfect we want to sample at a higher frequency. To keep some tolerance we sample at 52 kHz.

This frequency corresponds to:

$$\text{Number of samples on one disc revolution: } \# = \frac{f}{T} = \frac{52kHz}{1.3s} = \underline{\underline{40000}} \quad \text{EQ (5)}$$

$$\text{Sample spacing: } dphi = \frac{360^\circ}{40000} = \underline{\underline{0.009s}} \quad \text{EQ (6)}$$

With an exposure time of 2.5 ms, which gave best data acquisition, angular scanning velocity comes to:

$$\omega = dphi * \frac{1}{\text{exposure time}} = 0.009^{\circ} * \frac{1}{2.5 \text{ ms}} = \underline{\underline{3.6^{\circ}/s}} \quad \text{EQ (7)}$$

3.7.2. Decrease number of bad points

Bad points occur at angled surfaces. But the probability for a point failure is not the same for every fiber. It does not depend on the scanned object, even by 2 consecutive equal scans the probability to obtain bad points at the same place is big. Because of this reason we want to measure the same point on the sample with a different fiber.

We scan overlapped in order to be able to average between 2 reasonable values. Throw one away, when it is bad and keep one good measurement. When both points are bad, we cannot determine the accurate height. Hence a bad point persists.

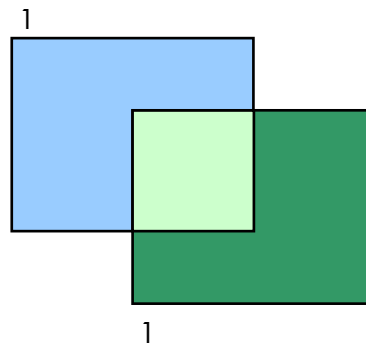


Figure 22: Principle of an overlapped scans

3.7.3. Increase resolution

If we want an accurate high-resolution groove profile, we need a big amount of points. MPLS180's point spacing is $10\text{ }\mu\text{m}$, so we just get about 7 points on the groove slopes and about 5 on the bottom. By spacing different scan passes by 2.5 or $5\text{ }\mu\text{m}$ (which comes to 4 respectively 2 passes instead of one) we multiply the number of points by 4, respectively 2. Interleaving of points allows a higher resolution of groove profiles and determination of the groove centre becomes more precise.

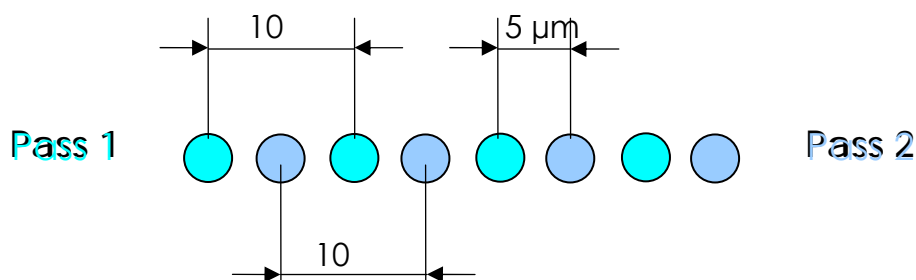


Figure 23: Principle of increasing resolution by point interleaving

As we also overlap scans, the density comes to 2 points spaced by 2.5 respectively $5\text{ }\mu\text{m}$.

3.7.4. Offset removal

Height differences due to autofocus are important when averaging points from overlapped scans. The strategy to get rid of absolute height differences of points is to calculate the offset of the groove profile. By subtracting the offset, we rearrange the groove profile around 0. A groove top will be above 0, a groove bottom below 0.

Of course bad points shouldn't affect the calculation of the mean value (= offset), they are just skipped.

After removing the offset, average of overlapped scans can be taken. The absolute height of the points that depends on the autofocus position does not come into account any longer.

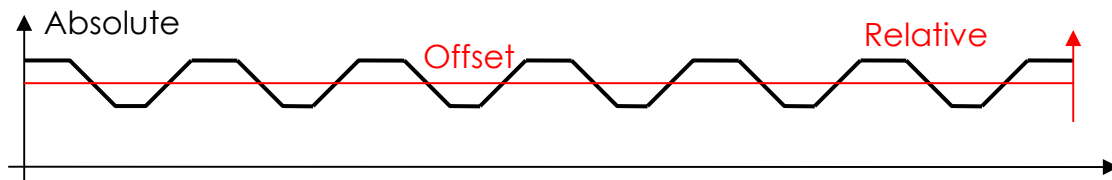


Figure 24: Explication of offset removal with absolute and relative height

3.7.5. Error due to translation and rotation

These errors can be considered as negligible. They shouldn't affect the data too badly, so none measures have to be taken.

3.7.6. X-position of MPLS180 points

The x-position of the beams of MPLS180 depends on the fiber number and the height because of the fact that the beams are spread. We are going to disregard the irregular spacing of the fibers. Digital treatment of data would be complicated, as every point also would additionally have a position and not only a height. This effect was also discovered at an advanced progress of the project and would have changed the whole concept.

3.7.7. Irregularity of the circle

To get rid of the irregularity of the circle, we can apply a high pass filter at 20 Hz in time direction. As already mentioned, music corresponds to the stylus velocity. Velocity is the derivative of the position. By taking the derivative we also get rid of the circle irregularity of the disc.

This defect is not annoying because it is easy to remove. Only that we need probably 1 more axial scan pass all over the disc, because the groove position from the centre of the disc varies elliptically.

4. Data acquisition software

We need powerful data acquisition software that controls and synchronizes all the equipment. Input parameters must describe exposure time, scanning speed, initial and end position of the scan, overlapping of scans, interleaving of points, autofocus settings, etc. It has to take all parameters into account that are relevant for the data acquisition. LabVIEW offers a good environment for visual programming of such software.

For a prior project using MPLS180 to scan phonographic cylinders a LabVIEW data acquisition software has been developed at LBNL. As scanning discs is very similar – there is also a rotation, a translation and a focusing stage – with a few modifications, the LabVIEW program can be used to scan discs as well.

A LabVIEW program (usually called VI for virtual instrument) is made in 2 windows. The Front Panel window is the GUI (graphical user interface) where the user can enter and set parameters as well as view solutions in indicators and plots. The other window is the block diagram that contains program code in a graphical form (such as icons, wires, etc.)

4.1. Front Panel

The front panel of “3D-control” lets the user control the data acquisition with MPLS for cylinders and discs.

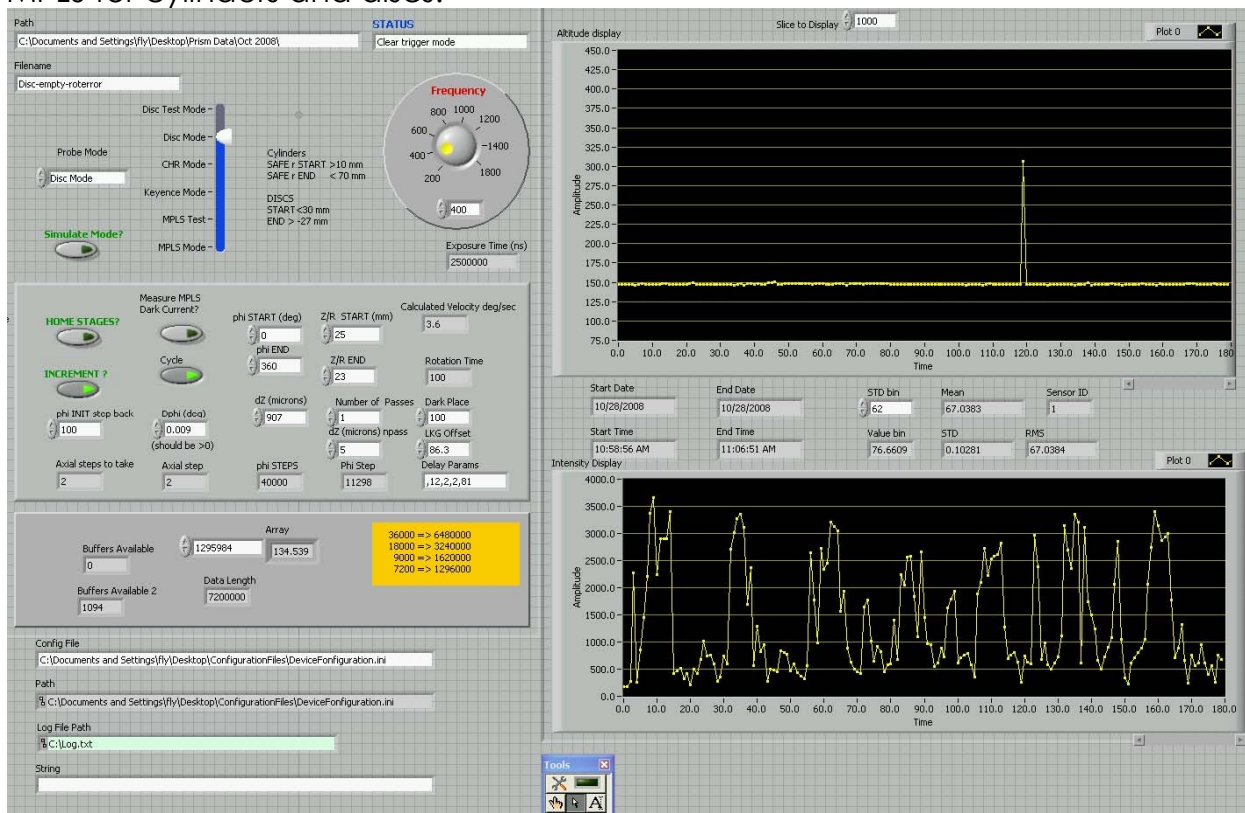


Figure 25: The LabVIEW front panel used for data acquisition

The most important input parameters are on the left side of the front panel.

Path:	The folder that the file will be written in
Filename:	Name of the written file
Mode:	Choice of the mode, see next chapter
Frequency:	Choose frequency i.e. exposure time of MPLS180
Dphi:	the angular difference between two 180-point-scans, defines the rotation speed
Phi START:	angular start position of the scan (in degrees)
Phi END:	angular end position of the scan (in degrees)
R START:	radial start position
R END:	radial end position
dZ / dR (microns):	difference between 2 radial steps (for overlapping scans)
Number of passes:	of the same data ring
dZ / dR (microns) npass:	radial step between multiple passes
LKG Offset and Delay Params :	Autofocus parameters
Config File:	Location of the configuration file for MPLS180

STD bin (on the right side): fiber choice, indicates on what fiber mean, RMS and STD values should be calculated on

Output plots:

On the right side there are first of all 2 plots:

Altitude display:	shows the altitude measured by MPLS180 of the 180 points
Intensity display:	shows the intensity level measured by MPLS180 of the 180 points

Fields on the left:

Axial steps to take:	calculated number of axial steps to take to scan from R START to R END
Axial step:	indicates actual axial step during execution
phiSTEPS:	number of 180-point-scans of a scan ring
Phi step:	number of 180-point-scan that is scanned right now
Start and End date/time:	indicates exact start and end time/date
Value bin:	shows actual value of chosen fiber
Mean:	indicates calculated mean value for chosen fiber over the number of scans indicated by "phi steps"
RMS:	indicates the calculated root mean square value for the chosen fiber over same number of scans than mean
STD:	corresponding standard deviation

4.2. Modes

This LabVIEW program can run in 6 different modes:

Modes	
0	MPLS mode for scanning cylinders
1	MPLS test mode -> stationary data acquisition for tests on cylinders
2	Keyence mode - NA
3	CHR mode - NA
4	Disc mode for scanning discs
5	Disc test mode -> stationary data acquisition for tests on discs

Table 7: Listing of the acquisition modes

Important for sound extraction from discs are modes 4 and 5, the ones that were created during this project. The other modes have already existed.

4.3. Disc scanning mode

This is the mode that we actually use to acquire data from discs. It executes the scanning process like described in the prior chapter taking into account all the required parameters and plots the results of height and intensity for MPLS's 180 points in the displays. It then writes the data containing the height in a *.pri file (format description later in this paper and the intensity in a different file of type *.bri.

4.3.1. Sequence for disc data acquisition

Once the VI is running in disc scanning mode (mode 4) it executes following sequence

- 1) initialize MPLS180
 - set exposure time
 - set into trigger mode
 - start acquisition
- 2) Dark current calibration if desired
- 3) Move to starting position
- 4) Set pulse mode for RGV (rotation stage)
- 5) Loop for data acquisition in jog mode
 - RGV to start position
 - set into trigger mode
 - start RGV JOG
 - read data from MPLS180
 - end RGV jog
 - shift translation stage XM (for multiple axial steps and multiple scan passes)
- 6) Disable trigger mode & stop pulse mode for RGV

4.3.2. Order of multiple axial steps and multiple scan passes

Above mentioned, overlapping and interleaving of points drives to multiple axial steps and multiple scan passes.

Data acquisition starts with the first axial step. For the first axial step, multiple scan passes for point interleaving are then executed before moving to the next axial step. And then all the scan passes for the second axial step are executed. And so forth.

4.4. *Disc test mode*

This mode is only for stationary test measurements. It measures for one of the 180 points the mean value, the root mean square and the standard deviation of a height measurement over a chosen number of points and indicates the results in the corresponding fields. It also writes acquired height data in a *.pri-file and intensity data in a *.bri file.

4.5. *Data file formats*

For further treatments of the data the file format has to be known. The LabVIEW program writes 2 files containing height and intensity data.

4.5.1. The *.pri file

This file contains the height data of the scanned sample. The structure of this file is very simple with a header at the beginning of the file followed by the whole data.

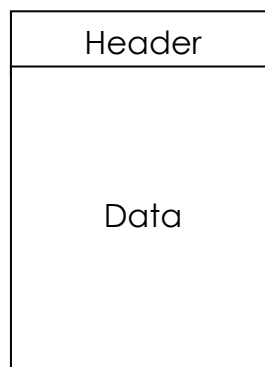


Figure 26: General structure of a *.pri file

Header

All information concerning the scanning method is written in the header. Its size is fixed to 10 floating point values containing the input parameters before scanning and information about obtained data size. The values indicate exactly the same than the LabVIEW input parameters and output indicators.

ID is the identifier of the file. Its default value is 99999.

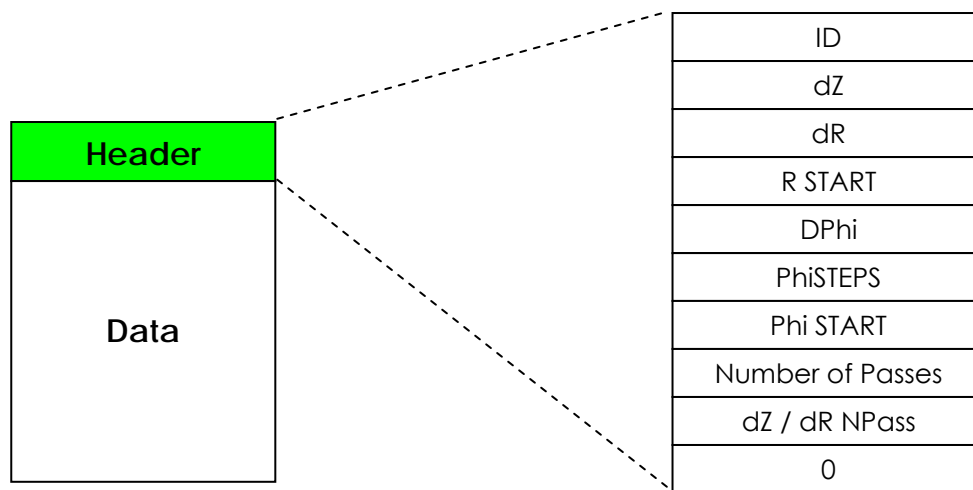


Figure 27: Detailed view of the *.pri file header

Data

Each scan returns values for 180 points. These values will be written consecutively into the file. Finally there is a data containing the height of all the points for all axial steps and scan passes.

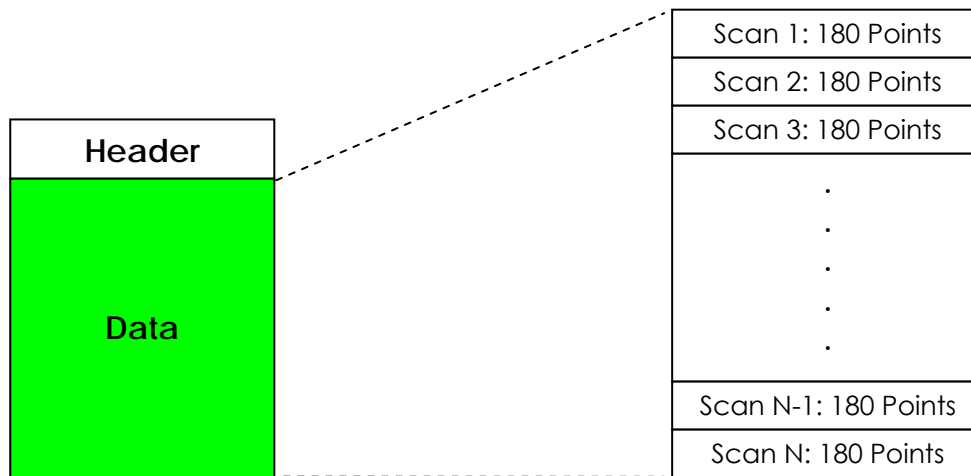


Figure 28: Detailed view of the data part in a *.pri file

4.5.2. The *.bri File

MPLS180 not only measures heights, it also indicates the intensity of each point. The intensity is saved in the *.bri file. The structure and the header are exactly the same as for the *.pri file. Instead of the height value, the intensity value is written into the file.

5. Analysis software

One of the main goals of the project is to write software that reads the data files, analyzes them and finally extracts the sound. This software is written in C#.

5.1. *Choice of programming environment*

At the beginning we had different possibilities how we can proceed to write our analysis software. There are two existing programs. One is a program in C# called PRISM written by Earl Cornell, a collaborator at LBLN. This program was designed to analyze cylinders. Another one was written some years ago by Noé Lutz and Michel Yerly, both former students from the EIA-FR, in their diploma project. The purpose of their program was to analyze data captured from dictation belts. We decided to write our own program in C# instead of modifying one of the mentioned programs and used them as an inspiration. This decision was taken after looking on both programs. It is very difficult to understand someone else's algorithms, even more, when the code is uncommented and variable names do not say anything about what they contain.

5.2. *Description of "3DSEVEN"*

"3DSEVEN" is powerful software to analyze data and extract sound from 78 rpm discs scanned with MPLS180. It contains several algorithms for cleaning, filtering and characterizing the data, finding groove centers and finally extracting sound. Descriptions of these algorithms and the concept of extracting sound are described in the next chapters.

"3DSEVEN" means **3** dimensional sound extraction, **very** effective noise reduction and is beside that a combination of the names of its authors.

5.2.1. Block diagram

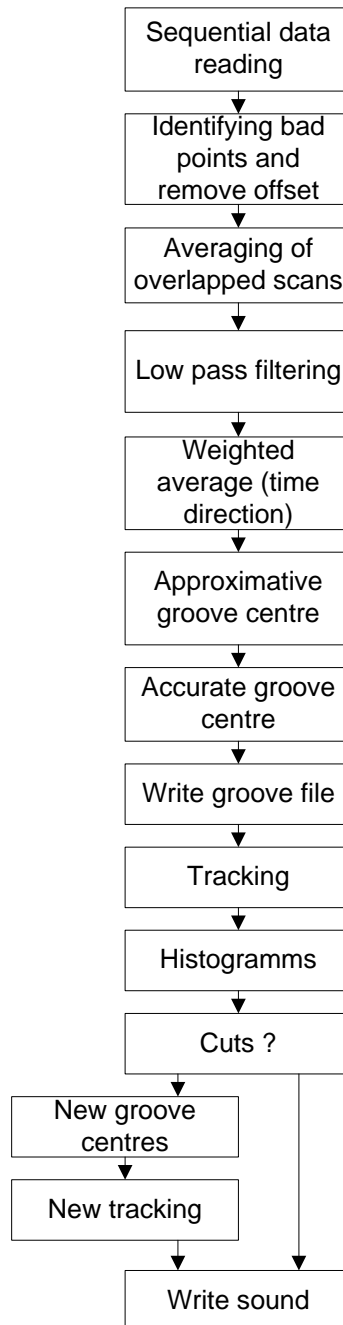


Figure 29: Bloc diagram of the program 3DSEVEN

5.2.2. Front panel - GUI

The program uses a GUI (Graphical User Interface) in order to facilitate the communication between the user and the program.

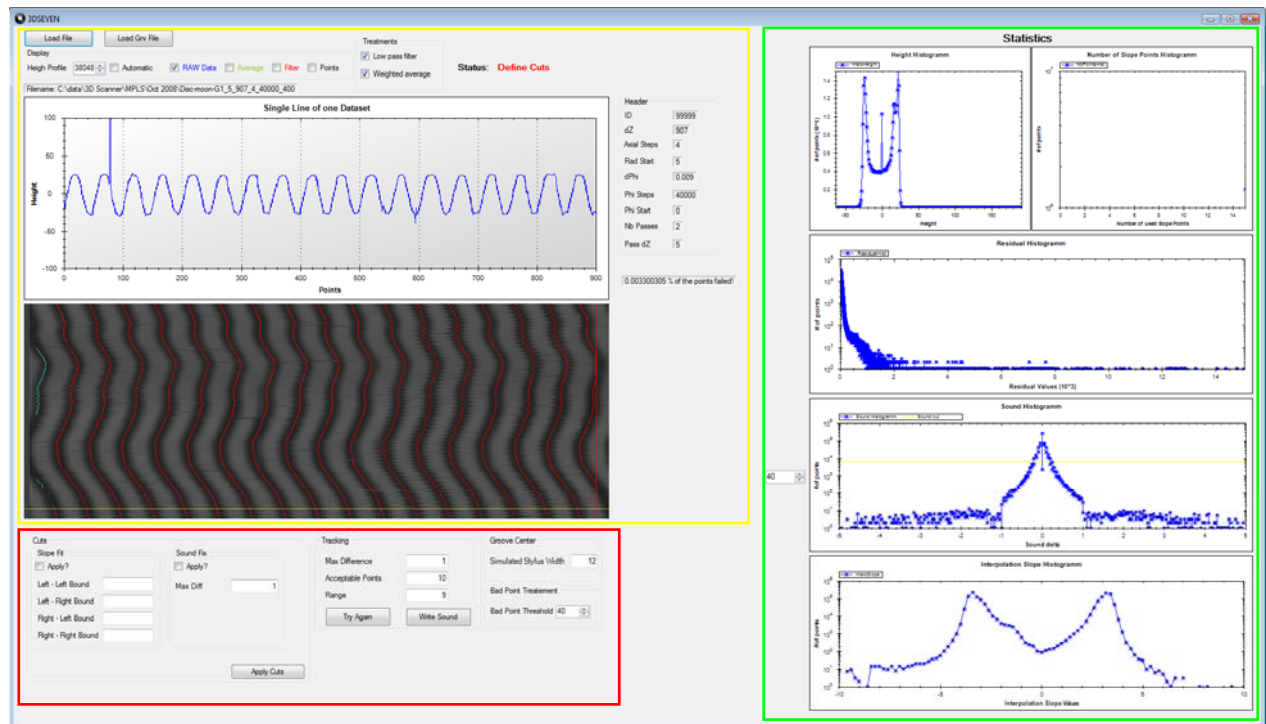


Figure 30: GUI of the program 3DSEVEN and division in three sections

This GUI is divided into different sections, the **data visualizing** section, the **data statistics** section and a **parameter setting** section. The next chapters introduce this GUI in a brief way. For more detailed information please refer to the user manual in the appendix.

5.2.3. Data visualizing

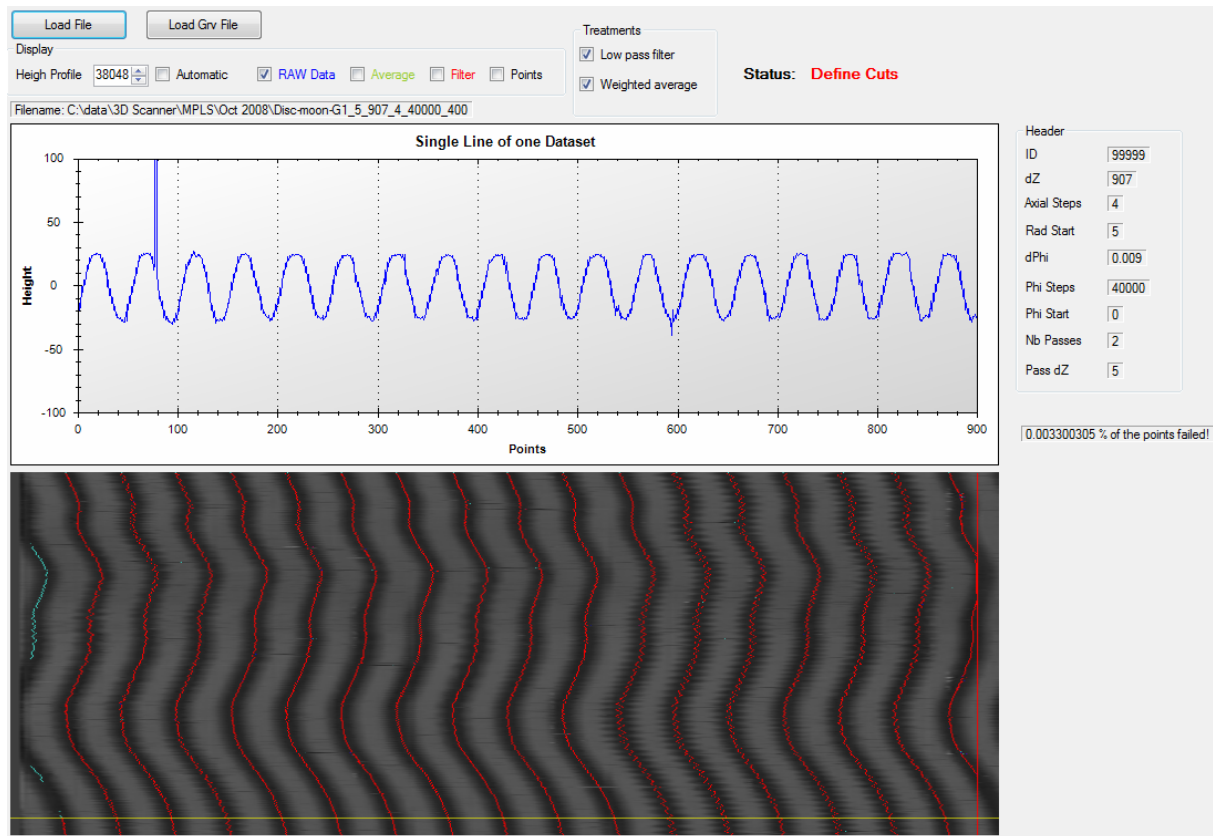


Figure 31: Data visualization section of the GUI

The data visualizing section makes scanned data visible for the user. On the right side the header information is shown. The 2D plot is used to display an entire height profile. Above the plot some checkboxes allow to change the displayed data. Below, the height image with the grooves and additional tracking information is shown. A status indicates the current operation of the program.

5.2.4. Data statistics

All the histograms characterizing raw data, groove center search and extracted sound are displayed in this part. What the histograms correspond to will be shown in chapter 7.7.

5.2.5. Parameter setting

The settings in this section of the GUI are very important for a good data analysis. The parameters affect the treatment of the data and therefore the quality of the reconstructed sound. Thresholds for cuts can be set, what they refer to will be mentioned later on.

For the tracking some parameters can be set. When the tracking failed (visible on the height map), a "Try Again"-button takes different input values into account for a new tracking.

Once tracking with the input parameters succeeded sound can be written by pressing the "Write Sound" button.

An input box allows to set the width of the simulated virtual stylus. The principle of this is mentioned in the chapter "Find Groove Centers". Finally a bad point threshold lets the user specify when a point is treated as a bad point.

Figure 32: Parameter setting part of the GUI

5.3. Data reading

To apply algorithms on the data stored in files, it has to be copied into the workspace and rearranged to fit together. By reading the file headers we obtain knowledge about the scanning method and the size of the acquired data which is fundamental, because the data has to be treated dependent on how it was acquired.

After reading the header and saving the parameters, the data appropriate to the discs height profile has to be read out. Here we have to differ mainly:

- For overlapped scans, the offset has to be removed and the average of overlapped points has to be taken.
- For interleaved data points, the data has to be aligned correctly.

As we aim to do both overlapped and interleaved scans, offset removal, averaging and point spacing has to be considered in any case.

The idea is to write the data in an array that represents the height profile of an entire data acquisition through all axial steps like a picture. As the order of the written data in the file does not correspond to the image some treatment has to be done:

- offset removal
- averaging of overlapped scans
- realigning the data points that they match together and represent data like an image

These algorithms clean and filter bad data points and make advantage of overlapping and interleaving of scans. Figure 34 below shows the data after joining axial steps and multiple passes together and executing cleaning and filtering algorithms (Figure 40).

6. Data cleaning and filtering algorithms

6.1. Offset removal

Offset is removed to get rid of the error due to the auto focus system to rearrange data around 0. After offset subtraction the groove profile is described by the relative height to the mean value (= offset) of a 180 point scan.

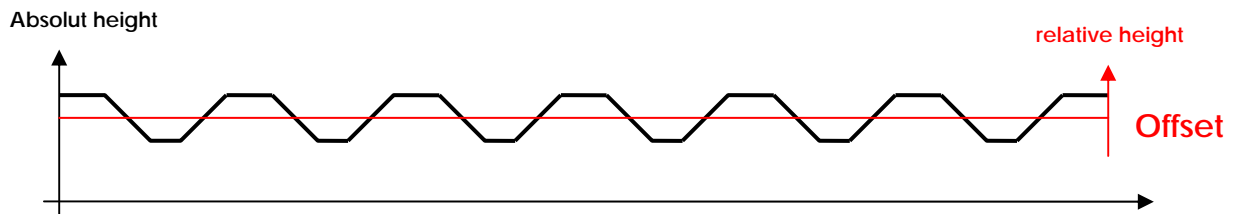


Figure 33: Principle of the offset removal process

For the calculation of the offset of a 180 point scan, bad points should not come into account, because they would adulterate the result strongly. They are identified and for the calculation of the offset value simply skipped. The average is then taken on the number of reasonable points, the result represents the mean value (=offset).

6.2. Averaging of overlapped scans

As we measure each point twice by overlapping scans, we can get rid of a lot of bad points. If one of the 2 points is bad, we can take the value of the other one, if both points have a reasonable value, we can take the average of the two. Only in the case of 2 bad points there is no choice to average, a bad point persists and we don't know the accurate height of the point.

This method of averaging used together with the offset removal is a good way to make profit of double point measurement and to clean the data up.

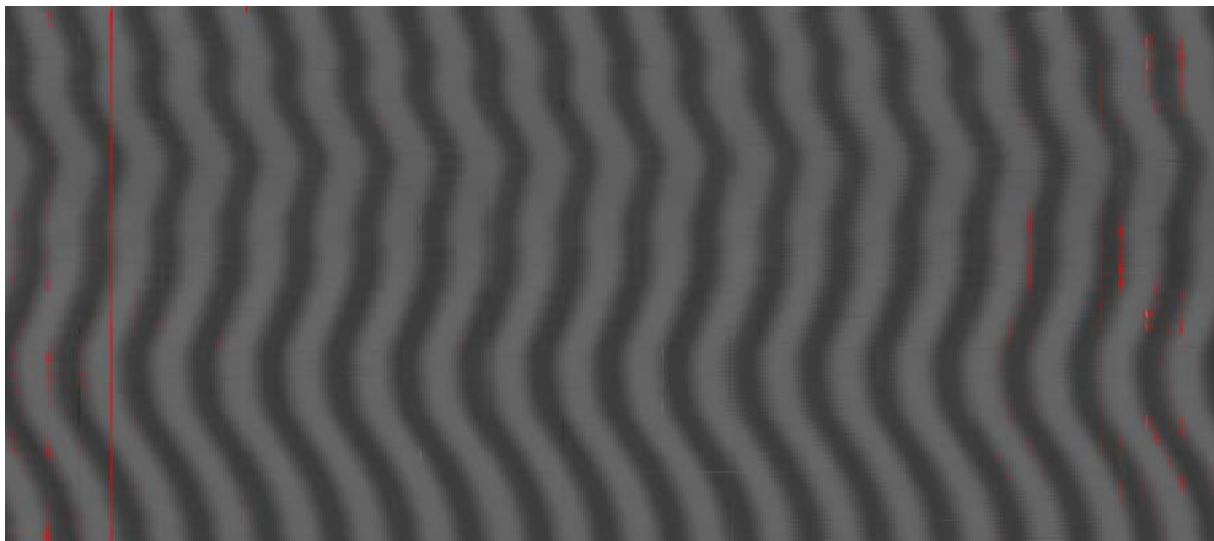


Figure 34: Disc height image with bad points (red dots) in the non-overlapped part

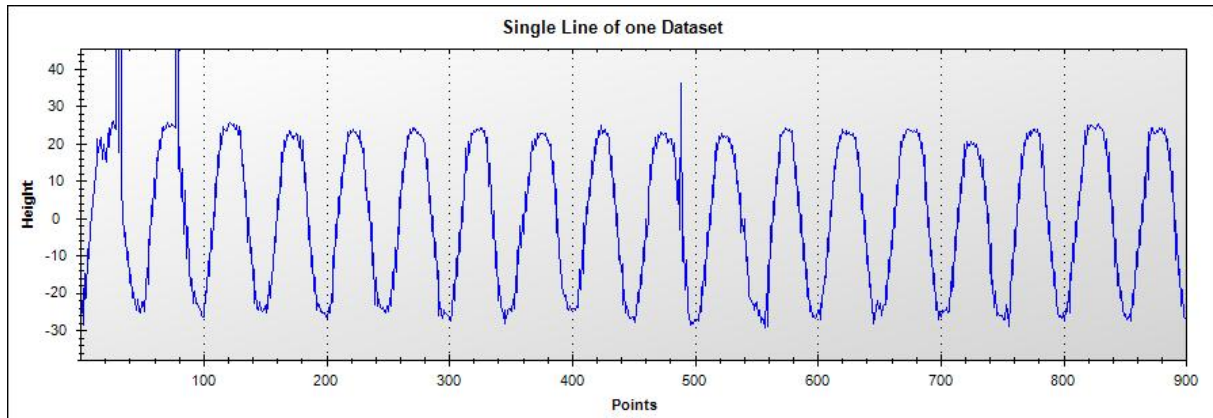


Figure 35: A single height profile with bad points on the left side (Area without overlapping)

On the left and the right border region, there is no overlapping of scans. Hence just one single point indicates the height of the disc. The occurrence of red points (bad points) at the outer regions is clearly denser which proves that double point measurement improves the height profile.

6.2.1. Point failure percentage

Scan overlapping decreases the number of bad points. For the record that images best ("In the evening by the moonlight") with overlapping, only **0.33 %** of the points are bad, without overlapping about 10 times more points fail (**3.59 %**).

6.3. Low pass filter

By zooming in on the height profile we recognize, that even with the offset removal the height adjustment between points can't be made very accurate and a lot of undulation across the points persist. This undulation has in comparison to the groove profile a high frequency (in space direction across the grooves). A low pass filter lends itself to filter these high frequency undulations (Figure 36). With a good choice of filter type and cut off frequency, we can get rid of most undulations.

Note: This is not a filter in time space, but in distance space. Units are [m]!

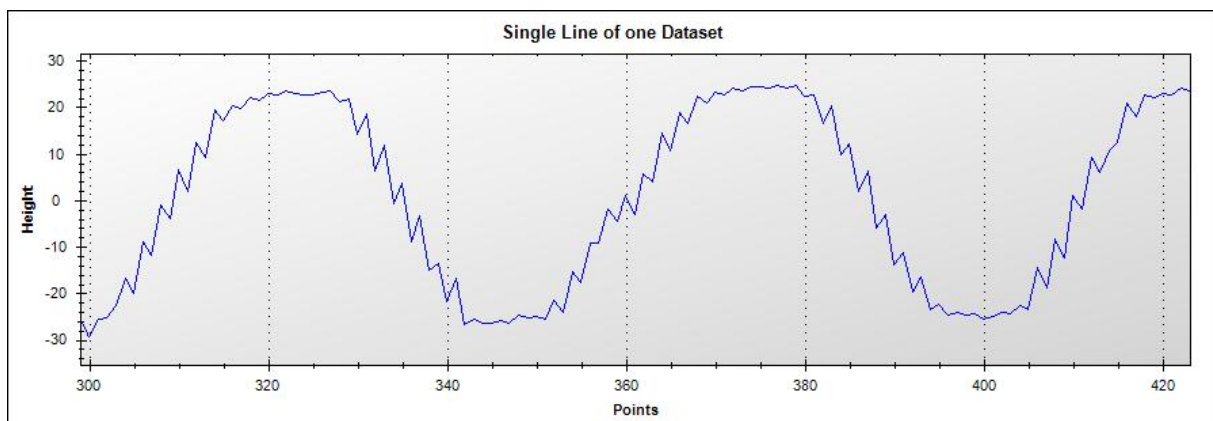


Figure 36: High frequency undulation due to the auto focus system

6.3.1. Filter design

A good filter choice is fundamental. First we have to choose between **IIR** (infinite impulse response, recursive filter) and **FIR** (finite impulse response, non recursive filter). The IIR filters offer a much better attenuation than the FIR for the same order. This is basically the reason why the IIR was chosen. As we filter data with bad points, output values are affected by less bad points when the order is lower.

We need a filter without ripples in the pass band. The most common IIR filter types accomplishing these requirements are Butterworth or Tchebychev 2. Second one has a steep transition from the pass band to the blocking frequency band. But it has ripples in the blocking band and attenuates very high frequencies less than the Butterworth. That is why a Butterworth filter has been chosen. [12]

This is not a filter in time direction, but in space direction! The usual unit time [s] comes to a distance [m] respectively [Hz] to [1/m]. For a point spacing of 5 μm , the sampling "distance frequency" is $1 / (5 \mu\text{m}) = 200'000 [1/\text{m}]$.

Specification of IIR filter type Butterworth:

Order:

6

Normalized Cut-off frequency: 0.3 (equivalent to 60'000 [1/m])

These values were found by approach and analyzing how treated data looked like.

Filter frequency response:

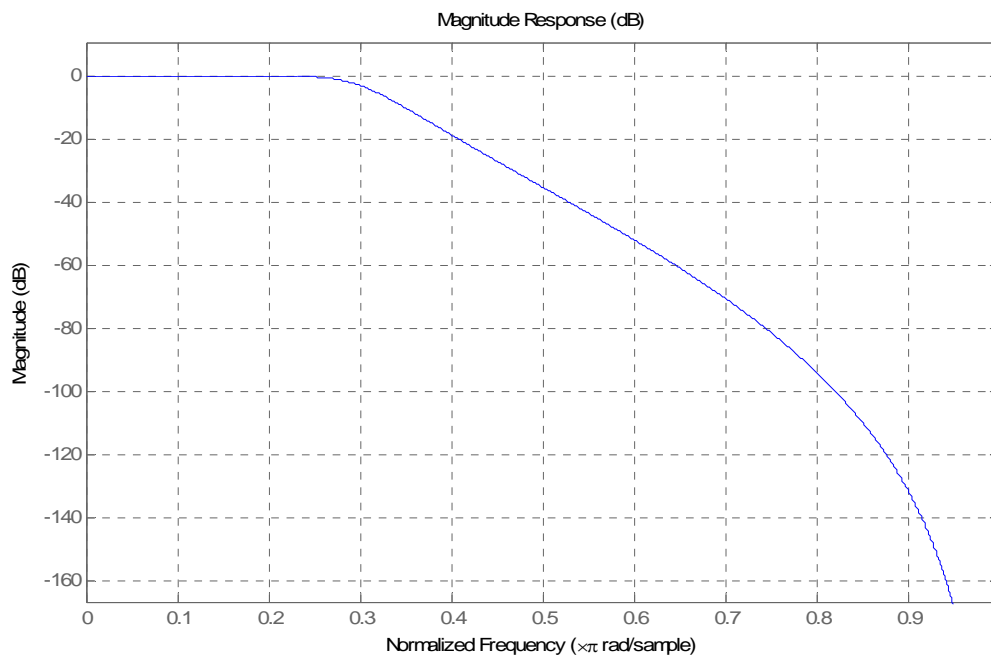


Figure 37: Frequency response of the implemented IIR low pass filter

MATLAB includes a filter tool which is a good environment to design digital filters. It shows magnitude response, phase delay, etc and finally generates the coefficients.

6.3.2. Filter implementation

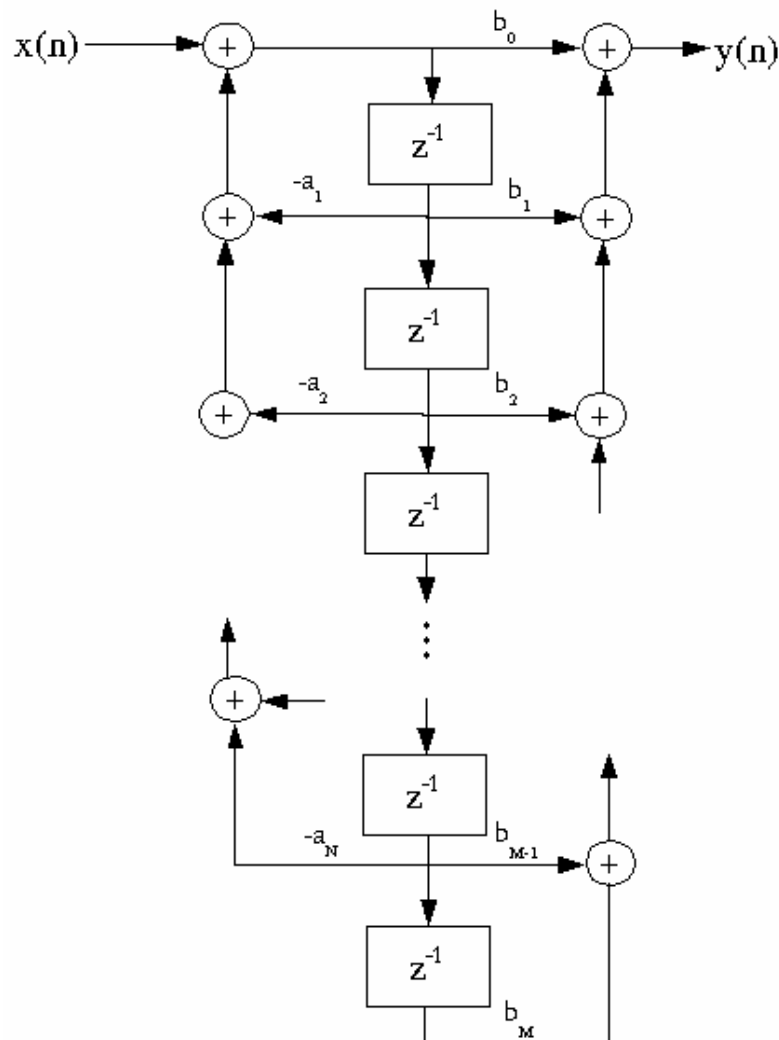


Figure 38: Structure of the IIR filter

The schematic above shows an IIR filter structure. An output value always depends on the actual input value, the prior input values and also, as it is recursive, on the prior output values. This structure has been coded in C#. We didn't convert it into second order cells, because the execution would be slower as every value would have to be treated 3 times, and as we are working in 64-bit double precision, rounding and chopping of numbers should not affect values for a 6th order filter implementation.

6.3.3. Data after filtering

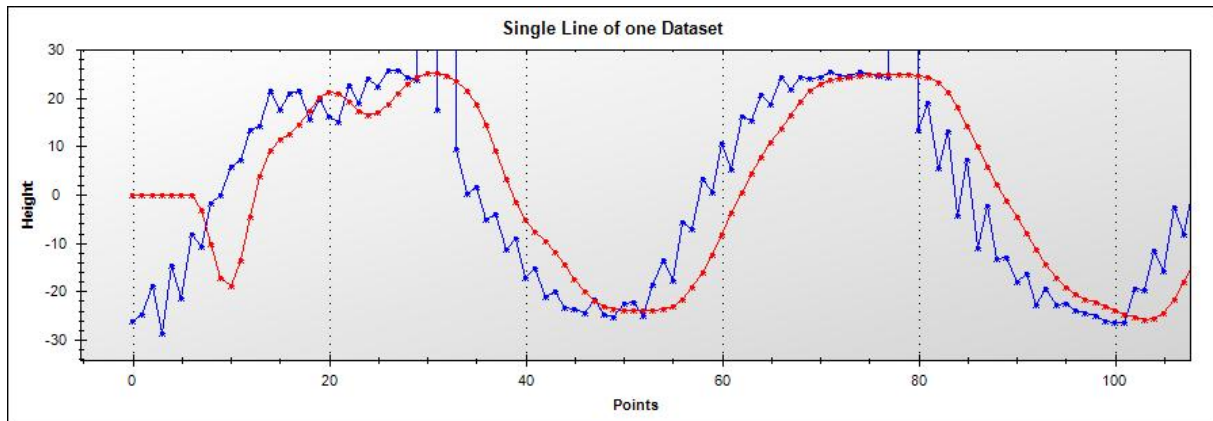


Figure 39: Comparison of height profile before (blue) and after (red) filtering



Figure 40: Height image after filtering

Figures above show the data after filtering. The red curve shows filtered data, while the blue curve represents raw data. The low pass filter removes high frequency undulation introduced by overlapping and averaging. Of course bigger irregularities still remain.

The low pass filter also introduces a phase delay. The groove centers will be shifted. As all the centers are displaced by the same amount, this should not affect music badly.

6.3.4. Average in time direction

Another approach to clean the data is to compare samples in time direction. To improve the quality of the acquired data, an average is implemented. For each point in a current height profile, its neighbors in the time direction will be taken into account to calculate the average. The average works like a low pass filter. Increasing the average length would decrease the cutoff frequency. We investigated two different kinds of averages: A standard average and a weighted average, where the neighbors affect the result less.

The following calculation shows the DFT of the weighted average. Each neighbor is taken into account by a factor of $\frac{1}{2}$.

$$y[k] = 0.25x[k-1] + 0.5x[k] + 0.25x[k+1] \quad \text{EQ (8)}$$

$$X(f) = \sum_{k=-1}^1 x(k) \cdot e^{-j2\pi f k} \quad \text{EQ (9)}$$

$$X(f) = 0.25 \cdot e^{j2\pi f} + 0.5 + 0.25 \cdot e^{-j2\pi f} \quad \text{EQ (10)}$$

$$X(f) = 0.5 + 0.5 \cos(2\pi f) \quad \text{EQ (11)}$$

This DFT calculation concerns an average where every sample affects identically.

$$y[k] = \frac{1}{3}x[k-1] + \frac{1}{3}x[k] + \frac{1}{3}x[k+1] \quad \text{EQ (12)}$$

$$X(f) = \sum_{k=-1}^1 x(k) \cdot e^{-j2\pi f k} \quad \text{EQ (13)}$$

$$X(f) = \frac{1}{3} \cdot e^{j2\pi f} + \frac{1}{3} + \frac{1}{3} \cdot e^{-j2\pi f} \quad \text{EQ (14)}$$

$$X(f) = \frac{1}{3} + \frac{2}{3} \cos(2\pi f) \quad \text{EQ (15) [13]}$$

Finally, figure 41 illustrates the differences:

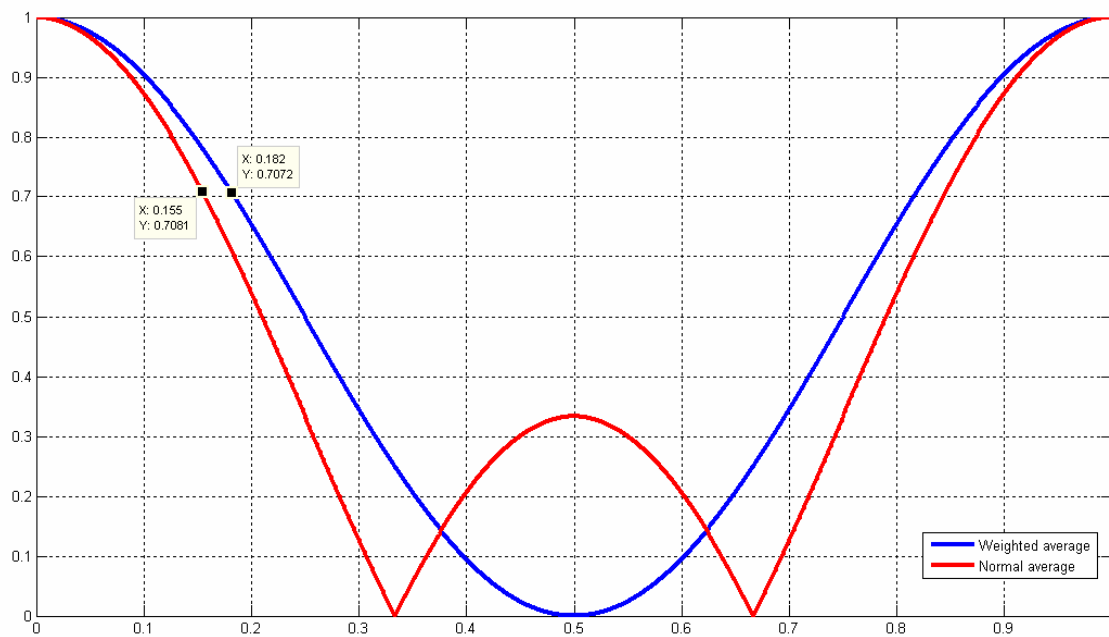


Figure 41: Comparison of frequency response between weighted and standard average

Every DFT has a periodicity of 1. For normalized frequencies, 1 corresponds to the sampling frequency. The advantage of the weighted average is evident. It has a higher cutoff frequency and a better HF reject as well.

Average is taken on 3 samples in time direction. The sampling frequency of the system is 52 kHz. Therefore the cutoff frequency is 9.5 kHz for the weighted average and 8 kHz for the standard average. We are aware that the audio band is up to 20 kHz and therefore a quality loss is inevitable. But there is a possibility offered by the measurement system to increase the sampling frequency up to 104 kHz.

The program allows the user to switch off the average. Comparison using the average or not is treated in a further chapter.

7. Algorithms extracting sound from data

7.1. *Approximate method to find groove centers*

For further algorithms it is necessary to have an approximate position of the groove center. Due to the offset removal at the beginning of processing, the height profile is located around zero. An easy way to find the center of the groove is to count all the consecutive values below zero. First we use a zero crossing detection to find the point where the falling slope becomes negative. This is the start point. The end point is located where the rising edge of the same groove gets positive. The number of points between the start point and end point is then counted. By taking the center between start and end point the approximate groove position is determined.

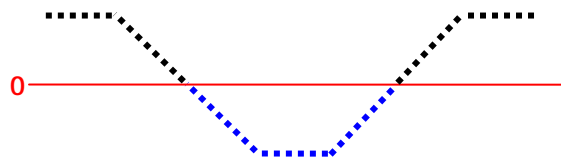


Figure 42: Principle of the approximate method for groove center detection

Due to bad data, multiple zero crossing in the same slope can appear and must be taken into account.

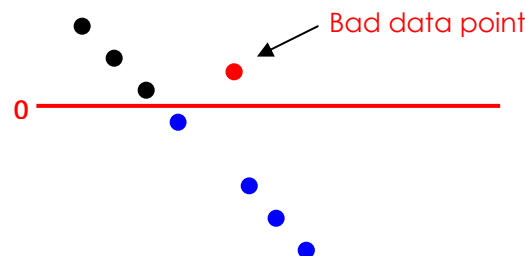


Figure 43: Problem of multiple zero crossing

This kind of issue can be fixed using a constant minimal distance between the start and the end point. It is important to choose an adapted value for this constant, using the physical characteristics of a phonographic record.

7.2. *Algorithms to find the exact groove centre*

Finding the exact position of each groove is the most important issue in the whole program. The quality of the reconstructed sound depends mostly on the accuracy of the groove center position. Bad data points can introduce errors, so that not all groove center positions are detected. Therefore a reliable bad-data treatment is essential because every groove center that is not detected reduces the quality of the sound.

There are a lot of different approaches to determine the groove center positions. Here are some methods we've investigated:

7.2.1. Groove slopes intersection method

The points measured in the groove walls become interpolated to a linear function of 1st order, one for each groove wall. Afterwards the intersection of these two functions returns the groove center position. In a program, such calculations are often much more difficult to implement than it might first appear.

More concretely by using the approximate groove center position, we have to decide which points belong to the slopes (groove walls), to the groove bottom and to the groove top. The points belonging to the slopes become interpolated using the Least Squares Algorithm. The mathematical background of this algorithm is explained later on in chapter 7.2.5. This algorithm provides the coefficients of the linear 1st order functions on both sides of the groove. Finally, we can calculate the intersection between those functions to get the position of the groove center.

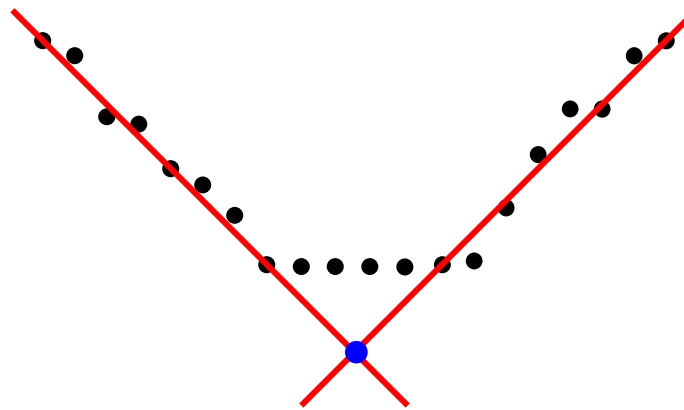


Figure 44: Principle of the groove slope intersection method

Advantages

The algorithm is very easy and consequently needs a short calculation time.

Disadvantages

A single bad point can significantly influence the center position, because it deflects the slope of the interpolated line and the intersection point changes. Even a small change of one slope can introduce a significant error for the groove center position.

7.2.2. Constant Line method

The principle of this algorithm is very similar to the groove slopes intersection algorithm. Instead of taking the intersection of the two linear functions, a horizontal line with a constant length is dropped into the groove. Once both ends of this line intersect with the linear functions of the slopes, the left and the right intersection position are registered. By averaging these two points, the groove center position is calculated.

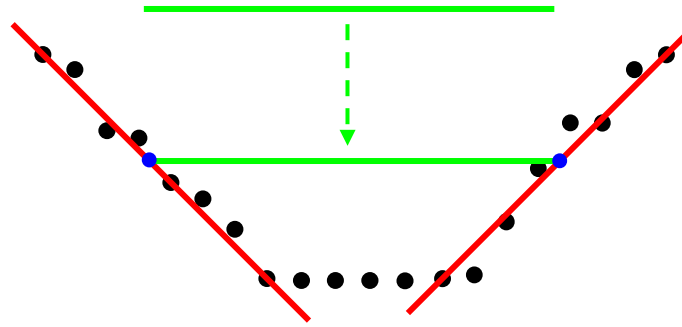


Figure 45: Principle of the constant line method

Advantages

A bad point in a slope has a less important effect on the center position.

Disadvantage

This algorithm needs much more time to calculate. For each position on the slope, the point on the other slope with the same height has to be found. This requires much more calculation.

7.2.3. Advanced constant line method

The principle is similar to the constant line method. Instead of dropping only one horizontal line with a constant length, multiple lines, each with a different length are dropped into the groove. The lines will intersect with the slopes at different heights and therefore multiple center positions are detected. The final center position is the average of all center positions.

Advantage

If one slope is deflected due to a bad point, it affects the groove center position less, because of the average.

Disadvantage

An average function is tricky, because a completely wrong value affects the final result.

This algorithm takes even longer to calculate.

7.2.4. Decision

Finally, we decided to use the constant line method. This provides the best compromise between calculation time and expected accuracy. Additionally, the line width is a parameter that introduces another possibility of improving the quality of the reconstructed sound.

7.2.5. Least Squares Method

The least squares method is an algorithm for fitting a polynomial through data points. When the sum of squared residuals between the polynomial and the data points is minimal, the best fit is achieved. A residual is the difference between the original value and the fitted line. [1]

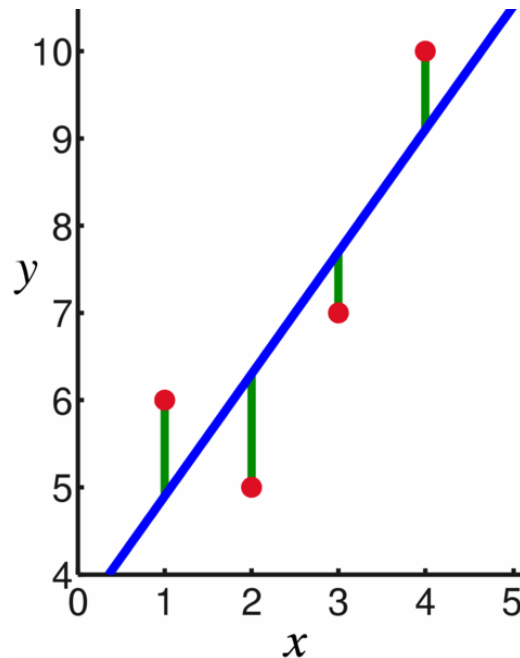


Figure 46: Basics of the Least Squares Method

The red dots are the original data values. The blue line is the best-fit line and the green bars show the residuals.

7.3. Problems occurred during development

During the sound analysis we recognized a bug which prohibits calculating the groove centers correctly. The bug occurred only during processing big data sets. After a certain scan width, the groove centers were not detected anymore. The analysis showed that the quality of data was the same for this part and that there was no reason that the groove detection did not work correctly.

Finally, we located the problem in the calculation of the interpolated line. After the critical scan width, the offset value for the line becomes too big for the type of number used. This overflow leads to completely wrong values for the interpolated line and consequently to a non-detected groove center.

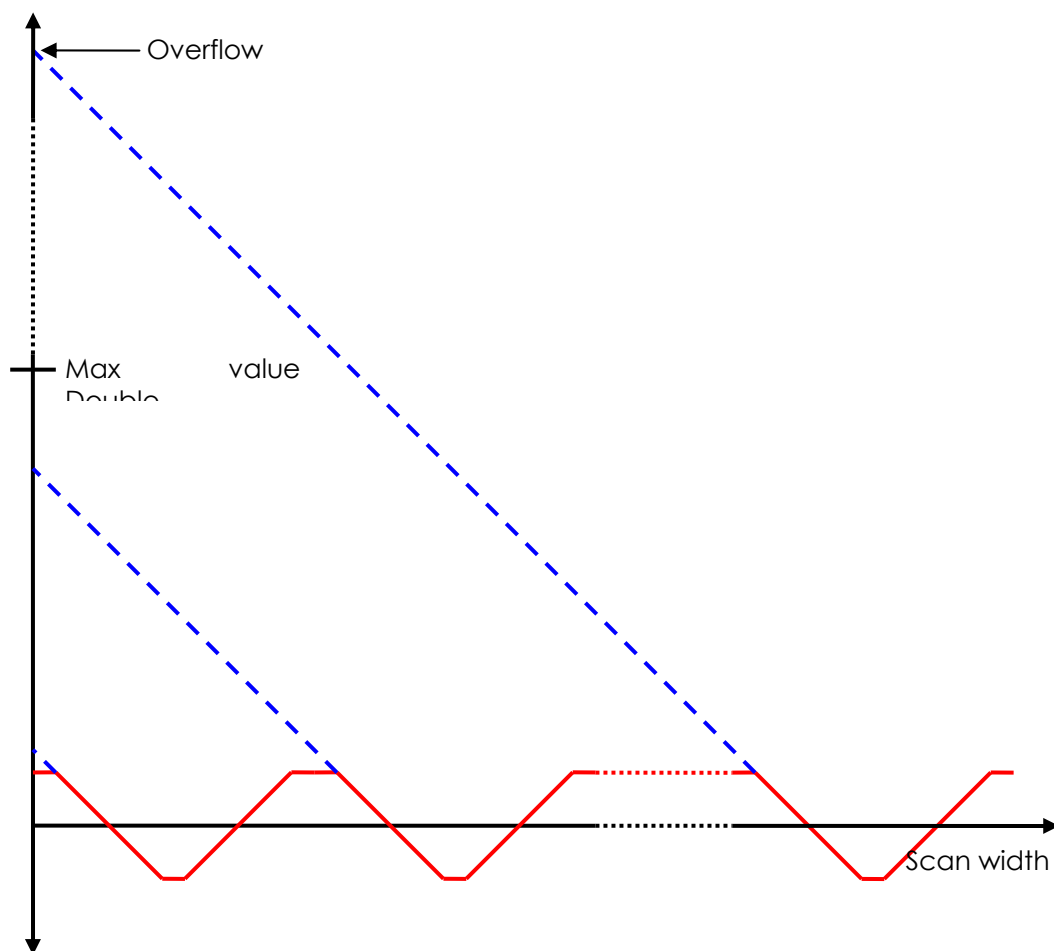


Figure 47: Problem of the data type overflow

7.4. Tracking of the grooves

The tracking process is equivalent to the stylus following the grooves. During this process, we are trying to match all the groove center positions together to determine the lateral displacement of the grooves. The result of this operation is a stream of all groove center positions aligned in the right way. This stream finally allows us to get the sound from the data. An additional difficulty is matching up the groove segments in the right sequence in order to assure the continuity of the sound.

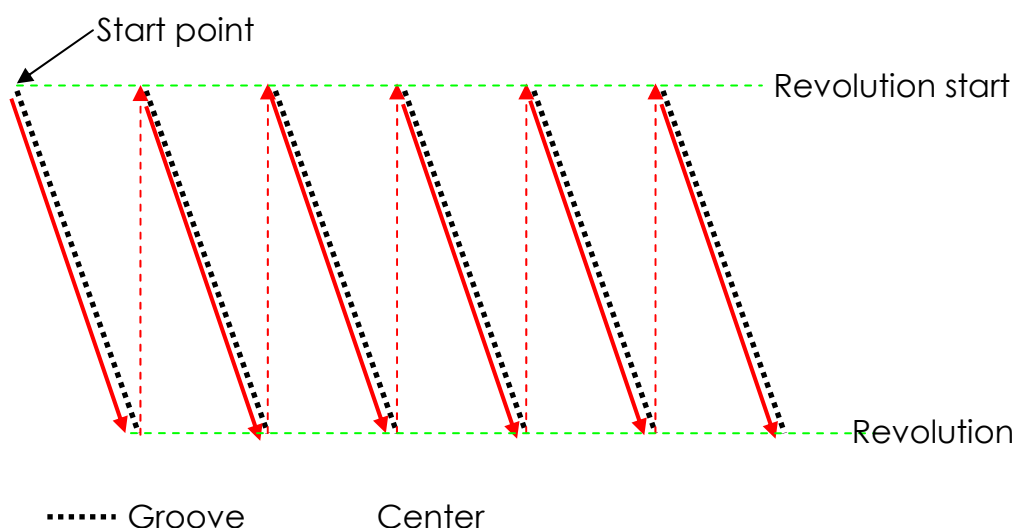


Figure 48: Principle of the tracking process

This figure shows the principle for one revolution. With our probe we are able to measure multiple grooves during one revolution. Because the grooves on a record are aligned in a spiral, the end of the groove at the bottom of the image is linked to the start of the next groove at the top of the image.

For comparison the following image shows the scanned height profile of a disc. The darker the image is, the deeper the point. Therefore the dark parts of the images represent the grooves bottom.

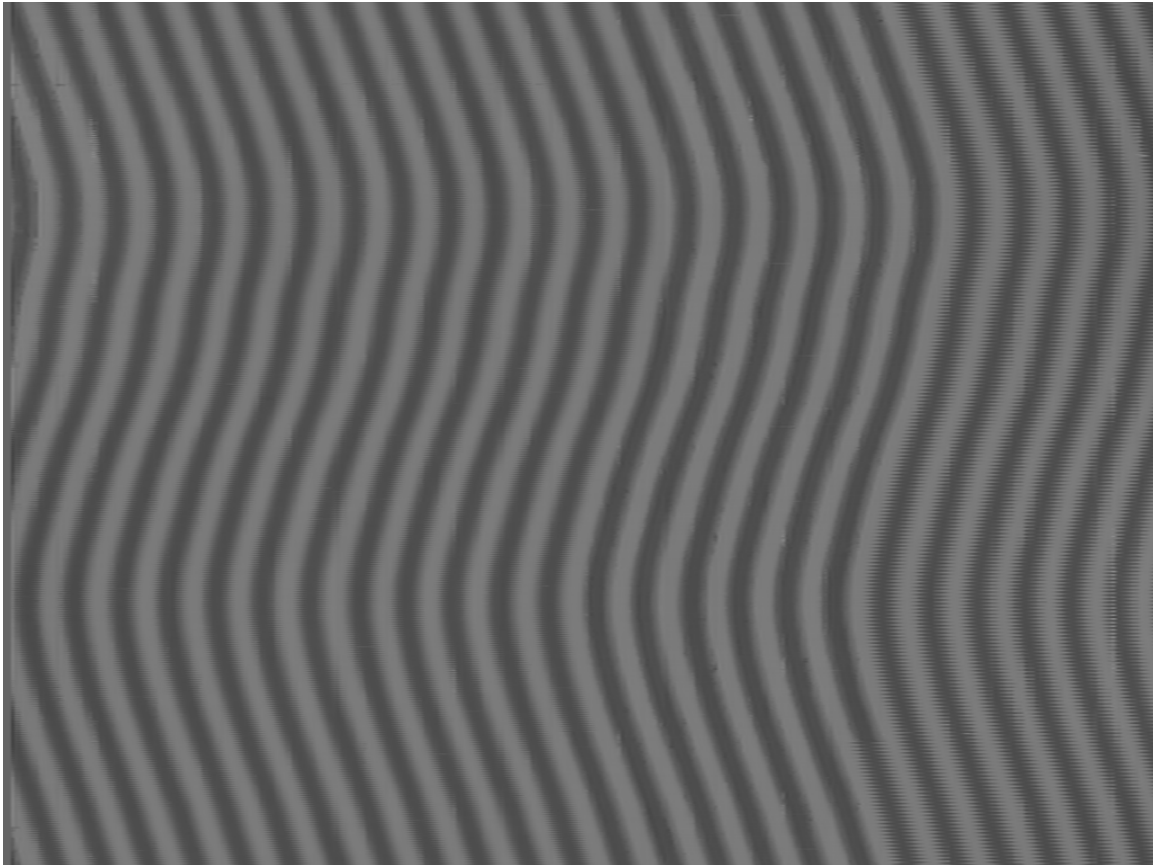


Figure 49: Image of the discs height profile

How it works

The first issue is to select the start point for the tracking. How this works is explained later on. Once the start point is determined, the program jumps to the next line and looks if there is a groove center position that matches.

The condition for a match is that the difference between two consecutive groove centers has to be smaller than a compare value. This value is based on the physical limitations. A groove can change only within a certain lateral distance. Additionally the groove direction changes slowly.

There is a bound effect to consider. Once the last center position of a revolution is determined, the tracking has to jump to the beginning of the next revolution.

In case the program can't find any groove center in the next line, additional features are included to search the next possible groove center. This case is covered later.

The start point

Due to the effect that the reference hole of the phonogram record is not perfectly in the rotation center, the way of the groove is curvy between the beginning and the end of a revolution instead of being straight. This causes that the groove can get lost on the left bound of the image even when the starting point of this groove is inside the image. The following image shows this case:

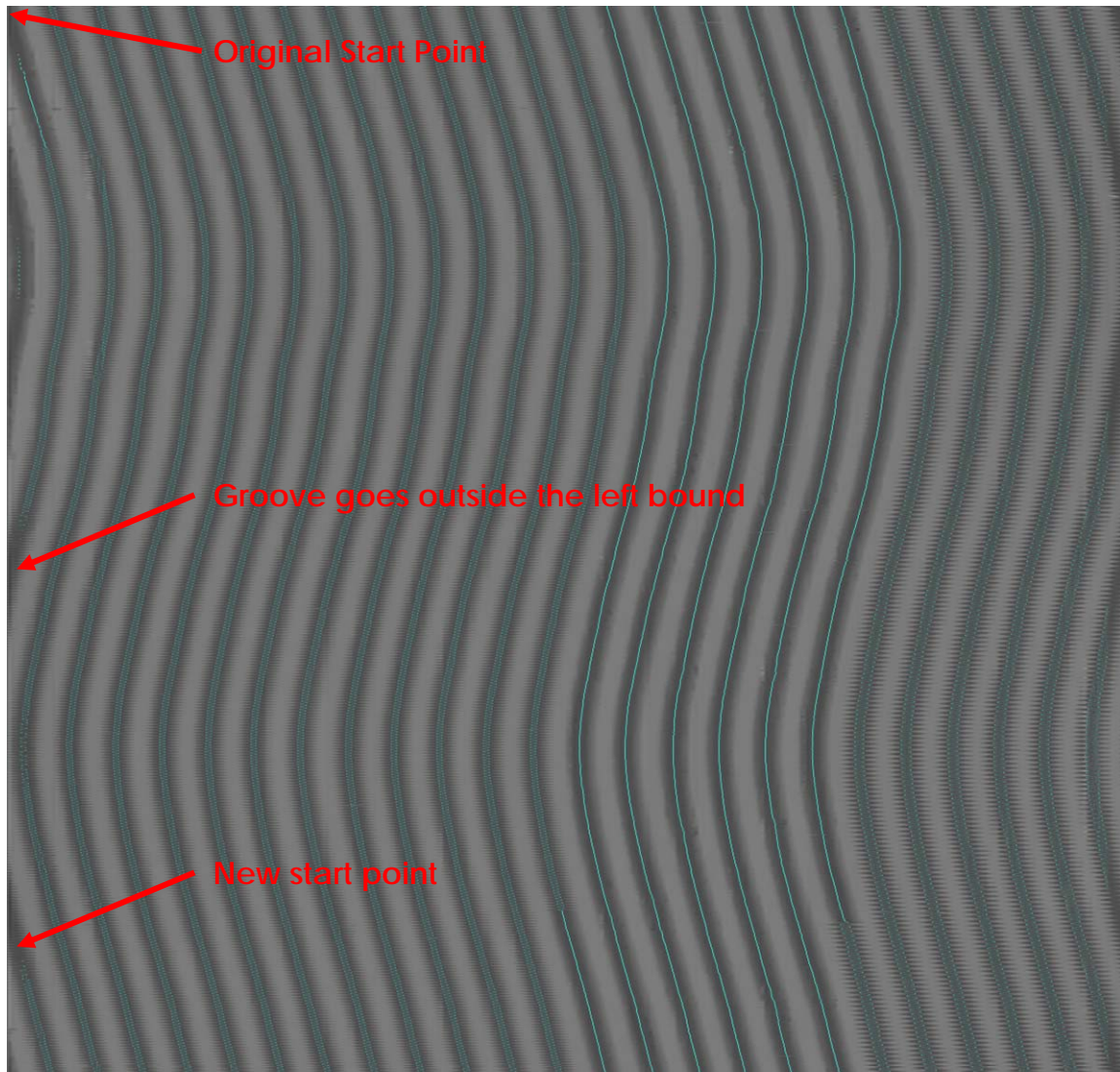


Figure 50: Start position problem

Therefore the start position has to be modified to another point where the future way of the groove never gets outside of the left bound. Our program uses a very easy solution. When the program has found all the groove centers and is ready for the tracking, the height image of the scan appears including all center positions. The user itself can decide where the start position should be by clicking with the mouse to the chosen position. Of course this is not the perfect solution, but the user can see the quality of the data. If the quality is bad, a tracking does not make sense. Instead of spending some more calculation time on tracking, the user can directly abort the program and try another file.

Bad or missing groove centers

Whenever the program cannot match a groove center in the following line of the image, there is a missing or a bad groove center. This case needs a particular treatment that is very reliable. Once a bad groove center is matched with a correct, the way of the groove is lost and tracking failed. To avoid this, all consecutive lines were checked for a groove center that

matches. Since the number of forward looking lines can achieve a considerable value, the condition for a match needs to be adapted.

The program creates a zone depending on the last found groove center. The next groove center and a defined amount of consecutive groove centers have to be located in this zone.

This avoids using bad groove centers in the zone. The following examples show this process.

If this problem occurs in a silent part the zone does not need to be wide. The additional condition avoids that the few bad groove centers in the zone are considered.

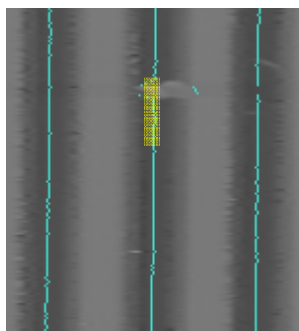


Figure 51: Search for next groove center in a silent part

In a wavy part we can really see why a zone is necessary. The track is recovered after a few bad groove centers. If the zone is too small it takes a lot more lines to find a new groove center.

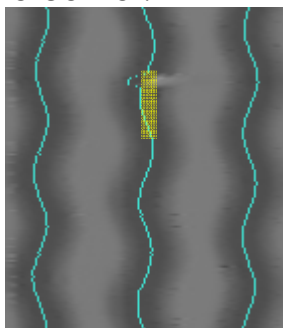


Figure 52: Search for the next groove center in a wavy part

Once the next groove center has been found, we take into account that the position where the accurate groove center is missing also needs a value. Otherwise the sampling frequency of the sound changes for this section. Therefore the missing center positions take the value of the last detected groove center position.

7.5. Extracting the sound

The tracking provides a stream with all the groove centers aligned. There are different methods to extract the sound from this stream. The stream itself indicates each center position of the groove. The following figure shows the stream in the case when the center hole is perfectly centered.

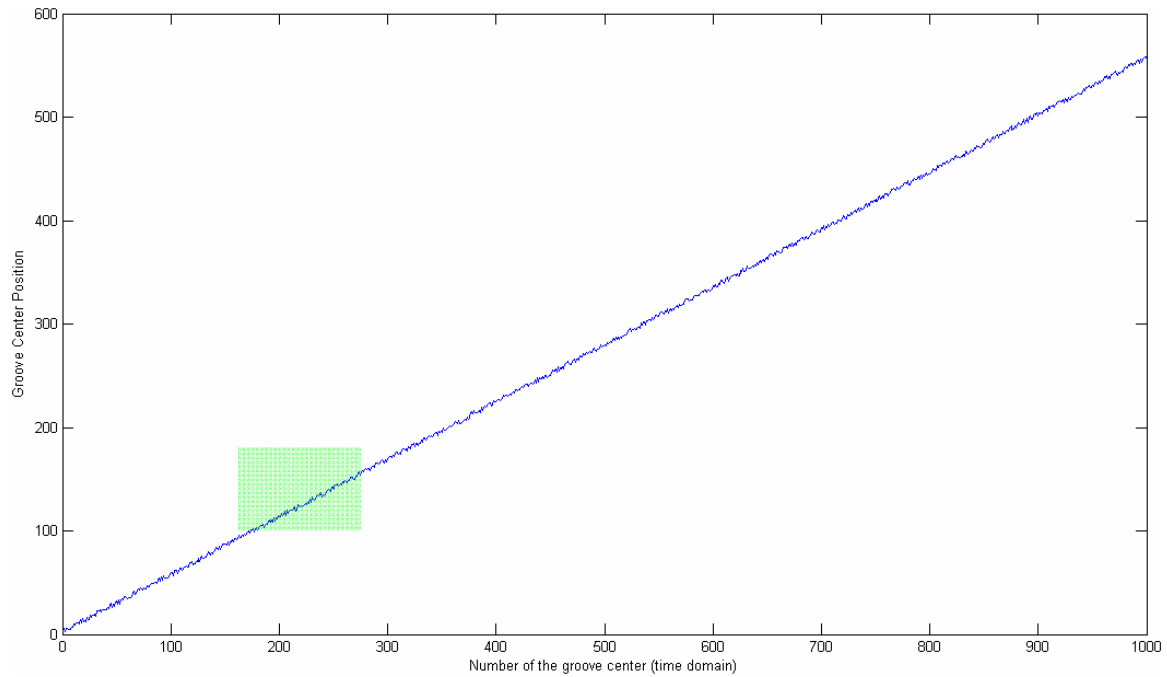


Figure 53: Stream of all groove centers

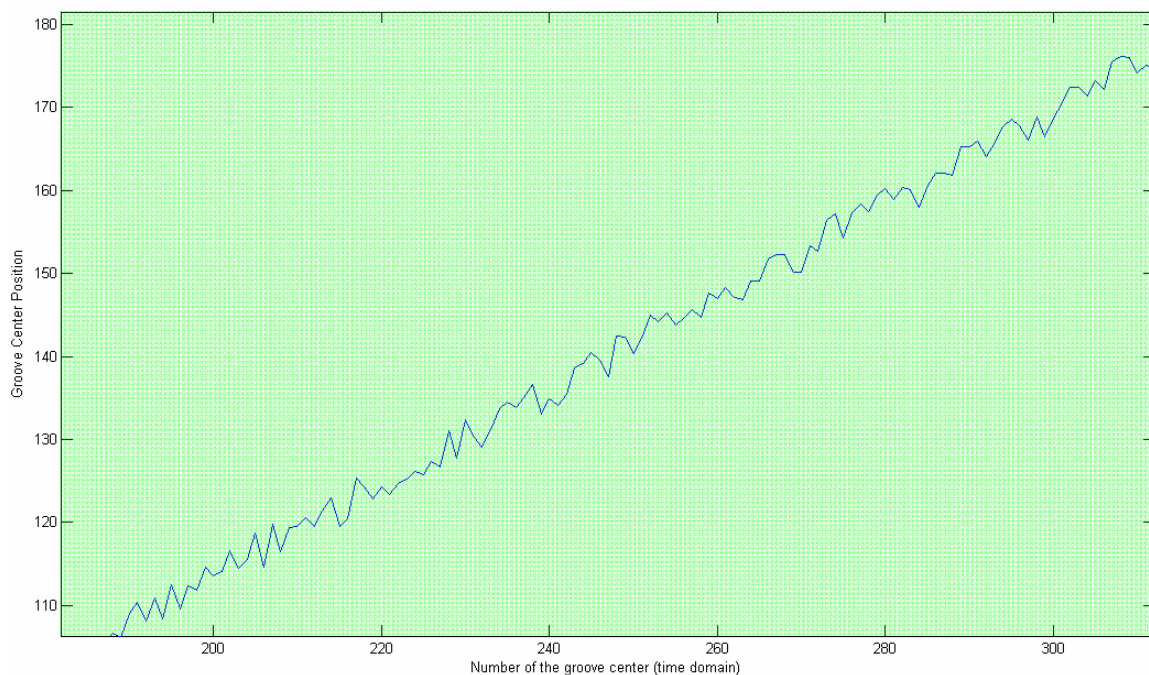


Figure 54: Detailed view of this stream showing the overlaid signal that represents the sound
The sound is superposed on the stream. There are different possibilities to extract the sound from this stream.

7.5.1. Derivative Method

The simplest way to get out the sound is to take the difference between two consecutive samples (i.e. groove center positions). This corresponds to take the derivate in the continuous world.

The algorithm corresponds to the physical effect when reading the disc. Sound is produced by the lateral velocity of the stylus, which is the derivative

of the position. If we take the difference between 2 groove center positions we simulate the stylus velocity.

Supplementary we do not have to take care of the ramp or its slightly different shape because the derivative eliminates the offset from the ramp. The following signal shows the result after applying the derivative method to the signal mentioned above.

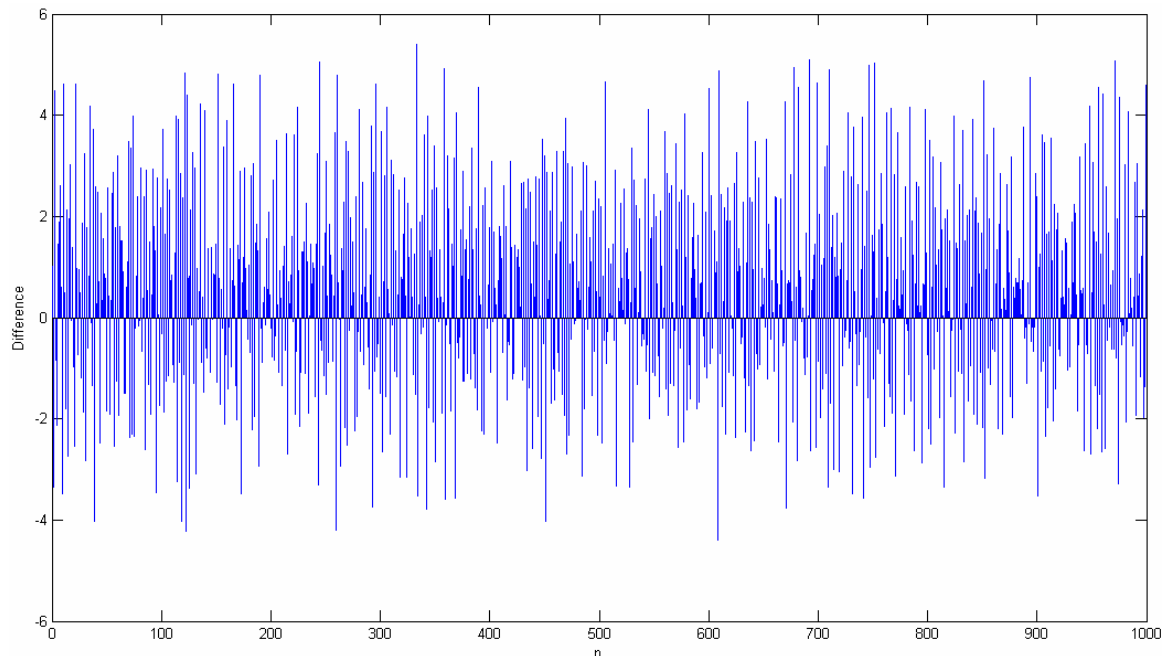


Figure 55: Stream after applying derivative method is equal to the sound

After the derivative method of the sound stream, sound has been extracted only has to be written in a audio file. (*.wav)

7.5.2. High pass filter

Another method to extract sound from the data stream is to apply a high pass filter with a cutoff frequency at 20 Hz. By applying this filter the ramp due to the spiral and eventual undulation due to the irregularity of the disc are suppressed. But the derivative still has to be taken into account in order to simulate the stylus velocity and not the stylus position. An 8th order Butterworth high pass filter split into biquadratic sections was implemented, but was never really used to extract sound because the derivative method seems to be much easier.

7.6. *Create the .wav File*

Once the sound is extracted, it needs to become playable on a computer. Therefore a wave file is created. The advantage of a wave file is that the sound is not compressed at all. In contrast the file size is much bigger than with a compressed music files like mp3.

Before the wave file is written, equalization of the volume has to be done. This prevents to have same music files with different amplitudes. The difficulty is to find the maximum value of the sound amplitude in order to adjust the sound

level to this reference. A big noise peak causes lower amplitude of the sound signal and should be treated before writing the file.

Each wave file needs a header. This header is in a standard format and contains information about the wave format, sampling frequency, channels (mono, stereo), length, resolution (Bits per sample) and so on. Then all the values of the sound stream are added to the file. The stream values are all integer values on 16 bits. So the amplitude range of the samples is from -32768 to 32767. [1]

7.7. Create a bitmap

The analysis of the disc is based on a height profile of a phonogram record. This height profile can be visualized in an image showing the records surface in height. The user can use this image to analyze the condition of a record. Beside the grooves, all scratches, dust particles and other detriments are visible on this image.

The program itself creates two different images. One shows the proper surface of the scanned phonogram record and the other one shows the same image with the tracking information overlapped. This allows a visual check if the tracking was well done, preventing a loss of information during the tracking process.

All created images are saved in bitmap format. The images are non-compressed and consequently of big size. Due to this the images are scaled in the height. A standard acquisition provides an image height of 40000 pixels and a variable image width depending on the number of axial steps scanned. To reduce the size, just every second line is included in the bitmap.

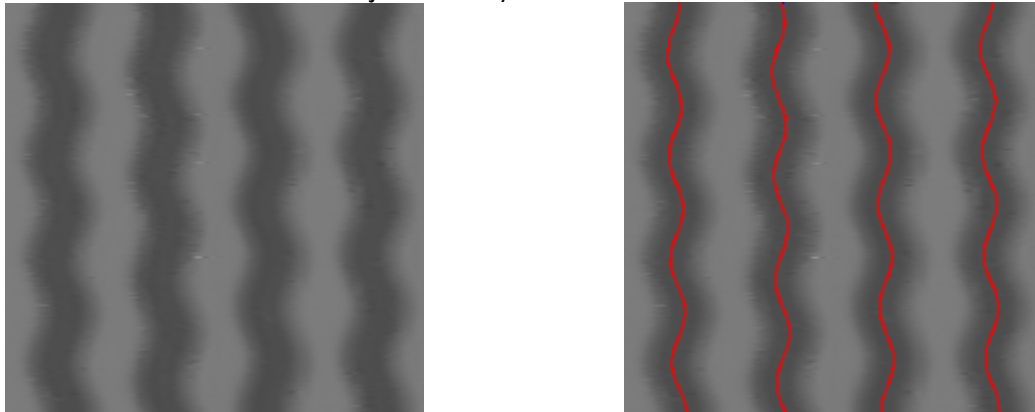


Figure 56: Detail of the first (left) and second (right) image created by the program

7.7.1. Error sources

Dust particle

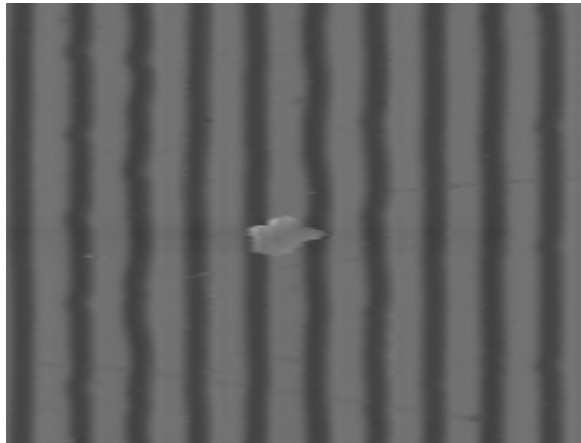


Figure 57: Image of a discs height profile including a dust particle on the discs surface

Dust particles are always on the top of the surface. Due to the grayscale height profile, dust particles are always bright on an image. It is advisable to clean the phonogram record before scanning with compressed air to get rid of dust particles.

Scratch

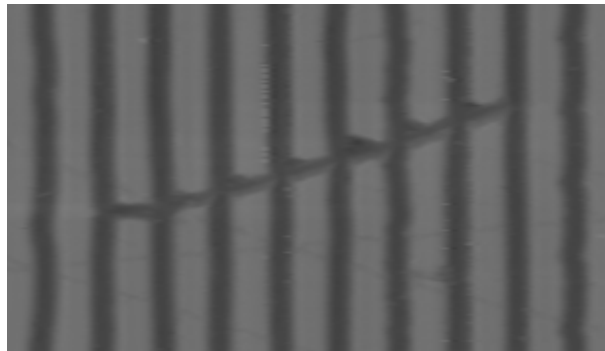


Figure 58: The appearance of a scratch on a disc surface

Old phonogram records mostly have a lot of scratches because they were used a lot over the years. Such scratches can cause a swap of the groove during the tracking process, because any groove center can be found if the groove wall is lost.

Crack

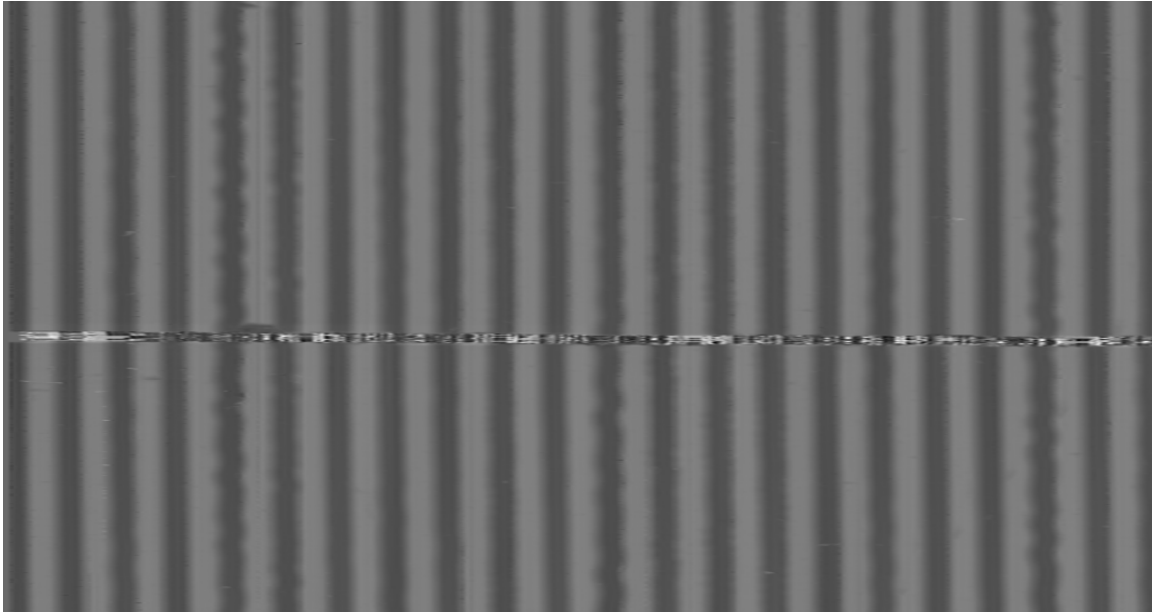


Figure 59: A disc surface of a cracked disc

If a phonographic record is cracked, it can't be played with a conventional player. Therefore an optical reading method is necessary to reconstruct the sound of such a record. A crack affects the tracking like a scratch. It's important to consider this during the tracking process, otherwise sound gets lost due to a groove swap.

7.8. Histograms (Statistics)

One of the most powerful tools to improve the quality of sound is to use statistics about the scanned disc. Histograms illustrate the statistical distribution of the measured data. Therefore we implemented some histograms to characterize data so that we finally can treat the data based on this statistics.

7.8.1. Height

The height histogram illustrates the different heights that appear in the whole height image. Normally there should be two peaks (groove bottom and groove top) and a continuous curve in between the peaks. If there are more than two peaks, the data is not good.

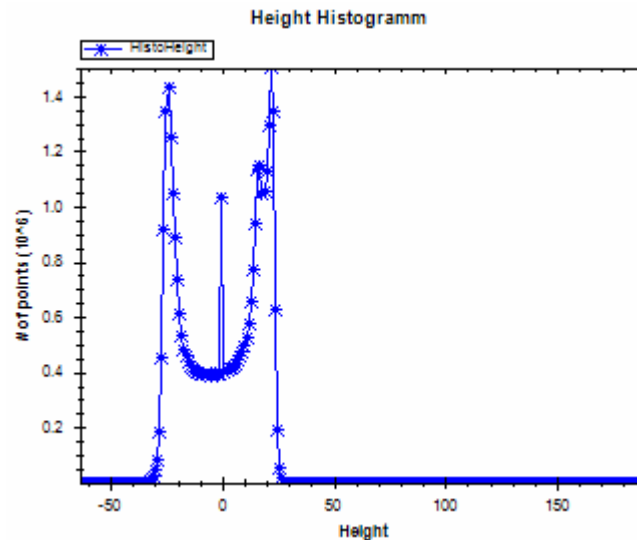


Figure 60: The histogram of the all the height over the whole surface

The example above shows a dataset with 3 peaks. The highest peak is usually the groove top, because physically the groove top has a larger surface than the bottom.

Here a third peak appears at zero. In this case the dataset has a part where all values are zero due to a malfunction of the acquisition system.

7.8.2. Number of points per slope

Each groove wall becomes interpolated during the analysis. This histogram shows how many points were taken into account to calculate the interpolated function. Based on this histogram, slopes with insufficient number of points may be rejected.

7.8.3. Sum of Squared Residual

A residual is a statistical value when interpolation is used. The difference between the original value of the point and its value after interpolation is called residual.

For each slope the sum of squared residuals is calculated.

$$e_i = y_i - \hat{y}_i \quad \text{EQ (16)}$$

$$SR = \sum_{i=0}^N e_i^2 \quad \text{--}$$

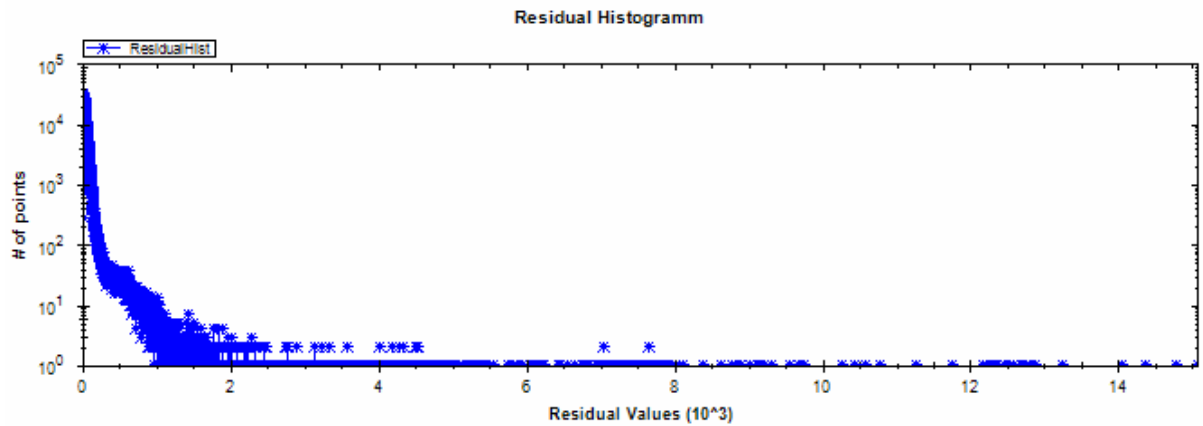


Figure 61: The histogram showing the distribution of the sums of squared residuals
 This histogram allows rejecting bad slopes in case the sum of squared residuals of a slope is over the limit.

7.8.4. Interpolated Slopes

Each slope becomes interpolated to a linear function of the form:

$$y = \alpha \cdot x + \beta \quad \text{EQ}$$

The slope α is used in this histogram to determine the statistical distribution of all slopes calculated. Due to the fact that each groove has two different slopes, two peaks are expected.

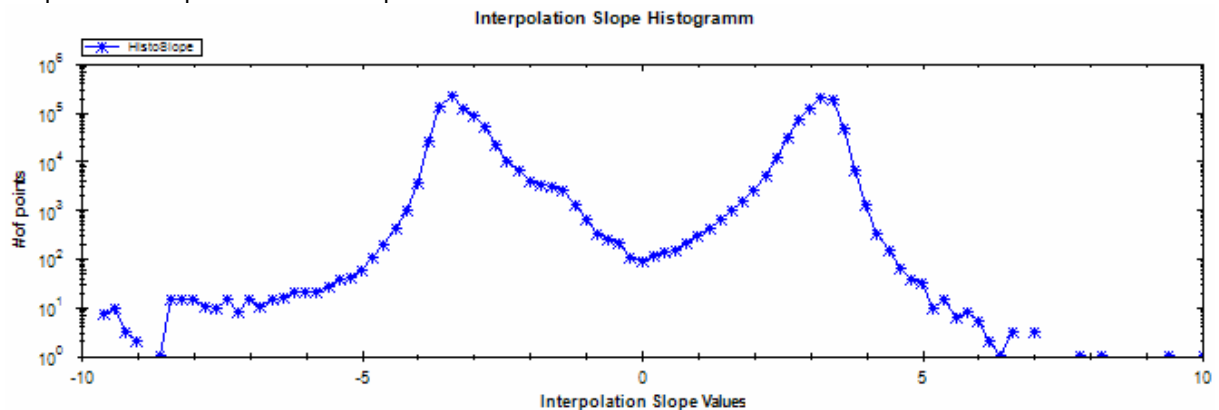


Figure 62: Histogram of the calculated slopes of interpolated functions

Once the histogram is created, slopes with values outside certain limits can be rejected.

7.8.5. Sound difference

Once the tracking is done, a stream with all the groove centers appears. The sound is the difference between two consecutive groove center position. This difference is used for this statistical distribution.

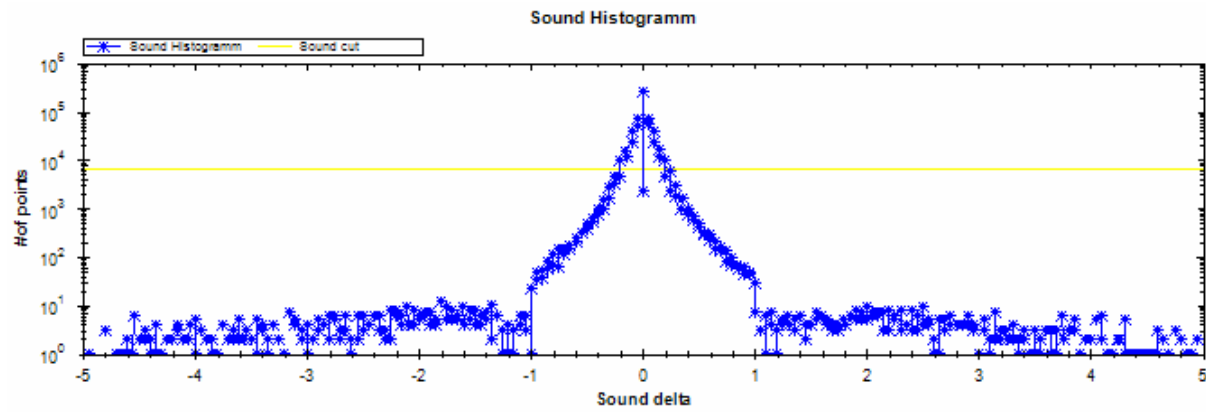


Figure 63: Histogram of the calculated differences between two groove centers

Big differences cause clicks and increase the noise level.

7.9. Cuts

Based on the histograms, a user can determine the limits for rejecting groove center positions and differences between 2 consecutive sound samples. The operation of removing this bad data is called cutting. In our case, the user can enter the cutting values and do these cuts.

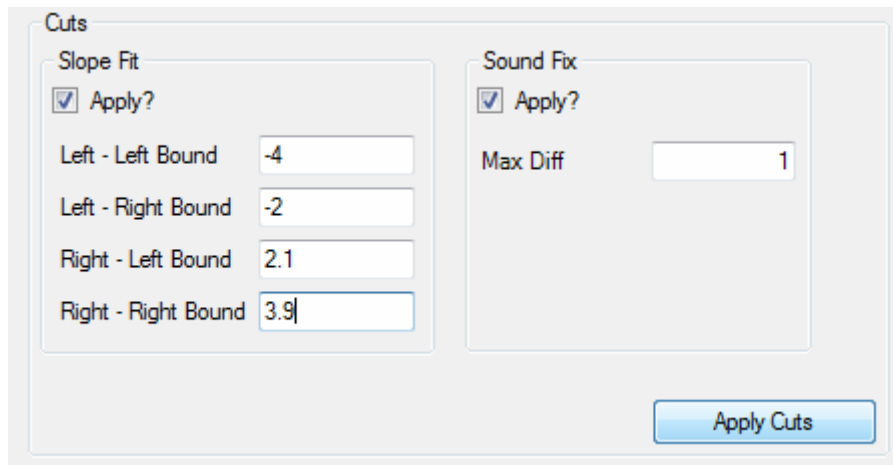


Figure 64: Applying cuts with the program

7.9.1. Slope fitting cut

The slope fitting cut is based on the interpolated slope histogram. For each peak, a left and a right bound determine the accepted range of interpolation slopes. All the other slopes are treated again. Before a slope is rejected, another algorithm is trying to find a slope in the specified range. The principle is very simple. The point of the slope with the biggest residual is rejected. Then the new fit for the slope is calculated. When this slope comes into the range specified by the cuts, the slope is does not have to be rejected any more. The algorithm does this procedure maximally 3 times. If the slope is still beyond the cut limits, it will be rejected definitively.

7.9.2. Sound cut

This cut is based on the sound difference histogram. The user specifies the maximal accepted difference between two consecutive groove centers. Differences that are bigger than this threshold are rejected. The rejection means that the amplitude of the corresponding sound sample is set to half range.

Instead of setting these points to half range, an interpolation would be much better. Unfortunately there is often a series of rejected points, so the interpolation becomes very difficult to realize. It can also be interpolated later with audio software (for example Sony Sound Forge 9.0).

7.10. The groove file (*.grv)

When scanned data was processed and groove center positions were found a *.grv file is written. This file contains information about the groove center positions and the groove walls.

When the same data set is analyzed a second time, for example when different cuts want to be applied, the *.grv file can be loaded and the entire data does not have to be processed a second time.

The major advantage using a *.grv file is that the calculation time for all the groove center calculation drops out and therefore the new results are present quicker.

The general structure of the *.grv file is almost like for the other files mentioned before: a simple header followed by the data. One difference is that a file terminating string indicates the end of the file. The data is written in string format. One line of the file corresponds to one variable.

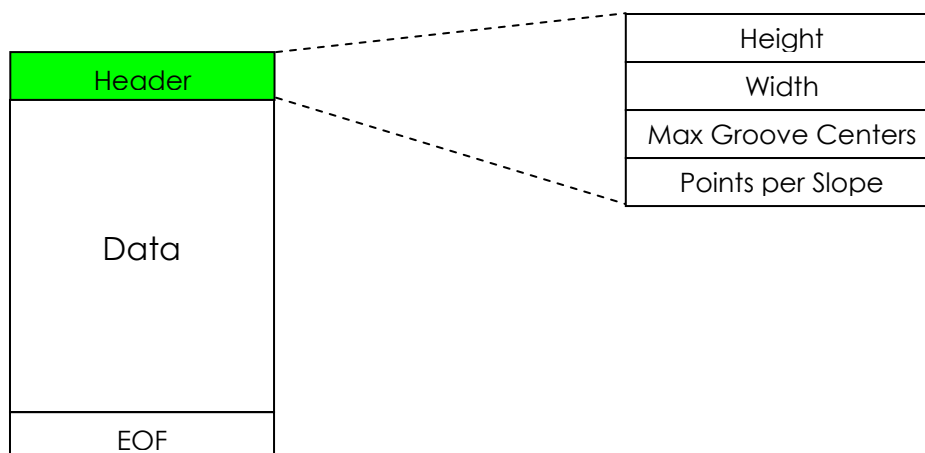


Figure 65: Detail view of the *.grv file header

All the header variables are useful for the file reader. Height and width indicates the dimension of the whole acquired data. Max Groove Centers provides the maximal number of groove centers found in one line of the dimension width (horizontal direction). Points per slope indicate the maximal value of points in a groove slope used to fit in a linear function.

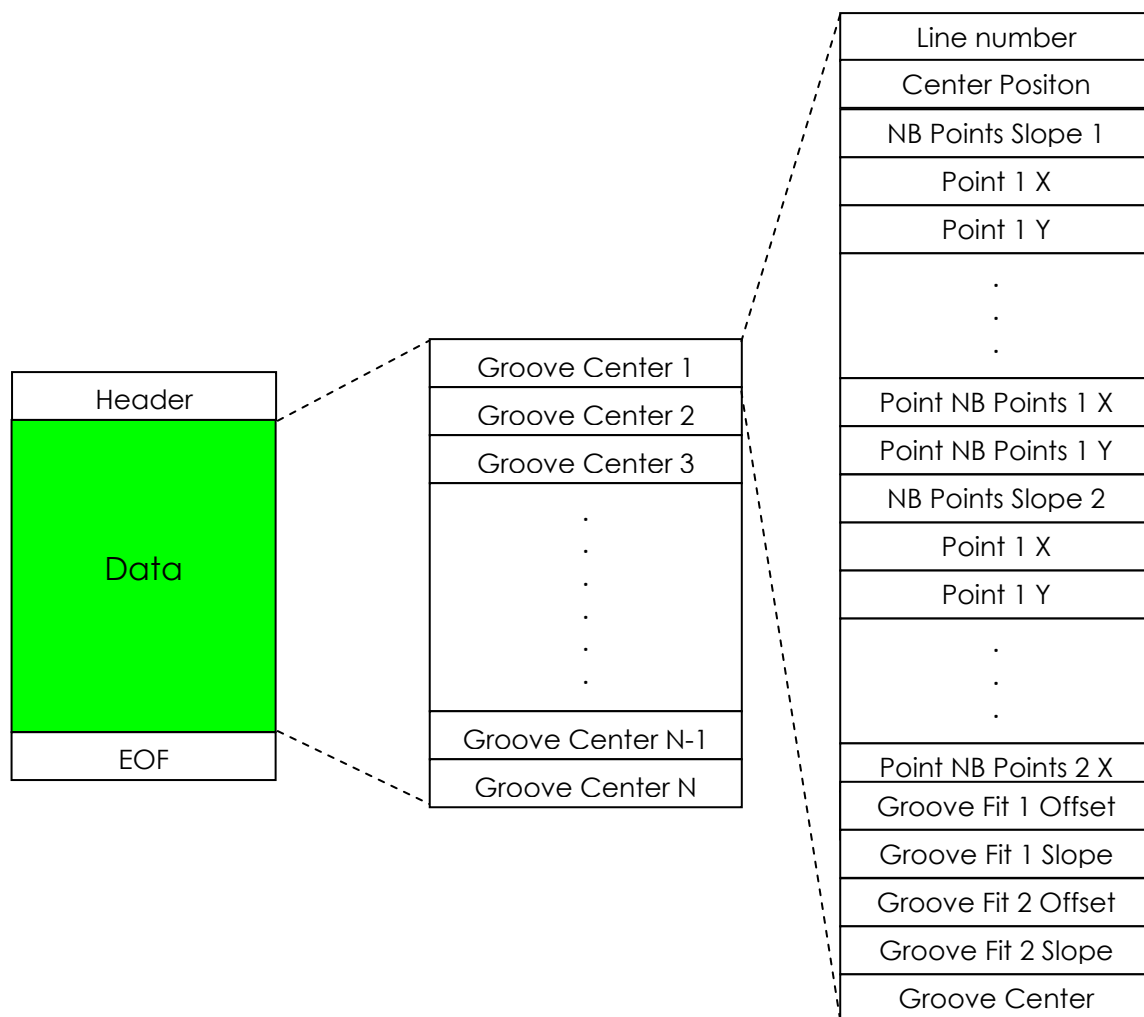


Figure 66: Detail view of the data part in a .*grv file

Every parameters of a groove is saved in this file. **Line number** determines the vertical position of the groove center in the image whereas the **Center position** indicates the number of the groove center for the vertical position. Further all the slope points from both groove walls are written into this file. The variable **NB Points Slope** contains for each groove side the total number of points used. Additionally, both coefficients from the linear fitted line ($c_0 + c_1 * x$) of each slope were written to the file. Finally the calculated **groove center** position complements the groove parameters.

In order to determine the end of the file, a predefined string "EOF" at the end of a file is implemented. It's an easy, effective and common method.

8. Analysis of sound

To evaluate the performance of our algorithms, and to understand which parameter settings lead to best sound extraction, we have to analyze the sound. As it is difficult to perform any measurements on music, we scanned a constant frequency test tone record. On this record we know exactly what frequencies it contains and we even have an analogue transcription of it for comparisons. The test tone record unfortunately did not image very good, although with overlapping and interleaving measurement it was possible to acquire good data.

The first section of the sound analysis is mostly on the sound test disc. On the extracted sound of this record, we calculate the signal to noise and distortion ratio (SINAD) as well as the total harmonic distortion (THD) with a LabVIEW program.

Before any sound is analyzed it was equalized with a RIAA filter, to take record equalization into account.

8.1. RIAA equalization

The Recording Industry Association of America (RIAA) established a specification for a correct playback of phonographic records. The purpose of this specification is to permit greater playback times, to improve sound quality, and to limit the disc damages. This is done during recording by attenuating low frequencies and boost high frequencies. It is called RIAA equalization.[1]

Not to distort the sound, this equalization of frequencies has to be included after extracting the sound. Conventional players also contain an RIAA equalizer.

8.1.1. Transfer Function

The RIAA equalization is based on the following transfer function:

$$H = \left| \frac{(1 + P \cdot t_3)}{(1 + P \cdot t_1) \cdot (1 + P \cdot t_2)} \right| \quad \text{EQ (19) [14]}$$

With $P = 2\pi f$ and some basic math functions we finally obtain to the frequency response:

$$H_{dB}(f) = 10 \cdot \log(1 + 4 \cdot \pi^2 \cdot t_3^2 \cdot f^2) - 10 \cdot \log(1 + 4 \cdot \pi^2 \cdot t_1^2 \cdot f^2) - 10 \cdot \log(1 + 4 \cdot \pi^2 \cdot t_2^2 \cdot f^2) \quad \text{EQ (20)}$$

Where t_1, t_2, t_3 are time constants with the following values:

$$t_1 = 3,18 \text{ ms}$$

$$t_2 = 75 \text{ } \mu\text{s}$$

$$t_3 = 318 \text{ } \mu\text{s}$$

The transfer function is illustrated by the graphic below:

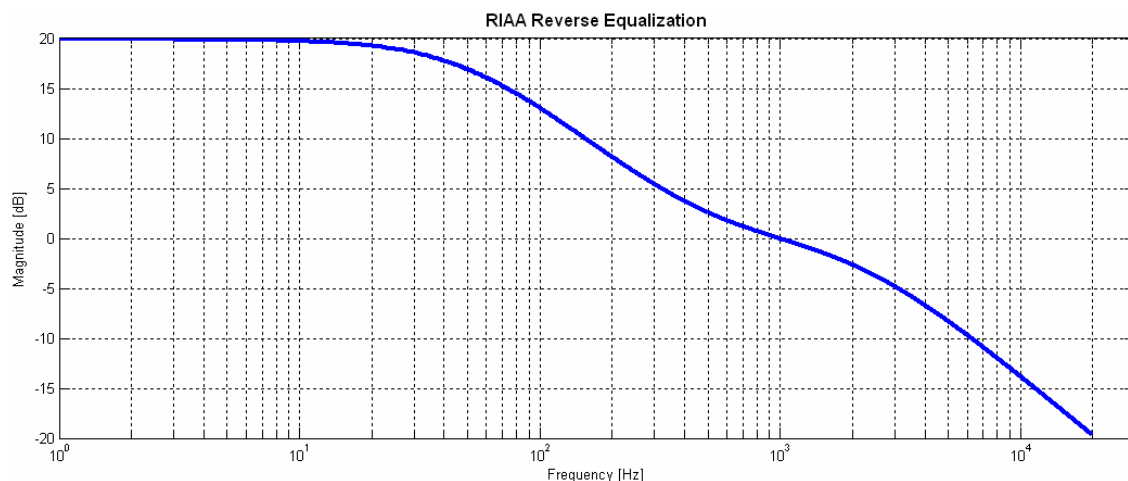


Figure 67: Frequency response of the RIAA equalization

In our sound analysis tool, Sony Sound Forge 9.0, we created an equalization filter using the calculated values of this transfer function.

8.1.2. Effect

The spectrum below illustrates the effect of the RIAA equalization:

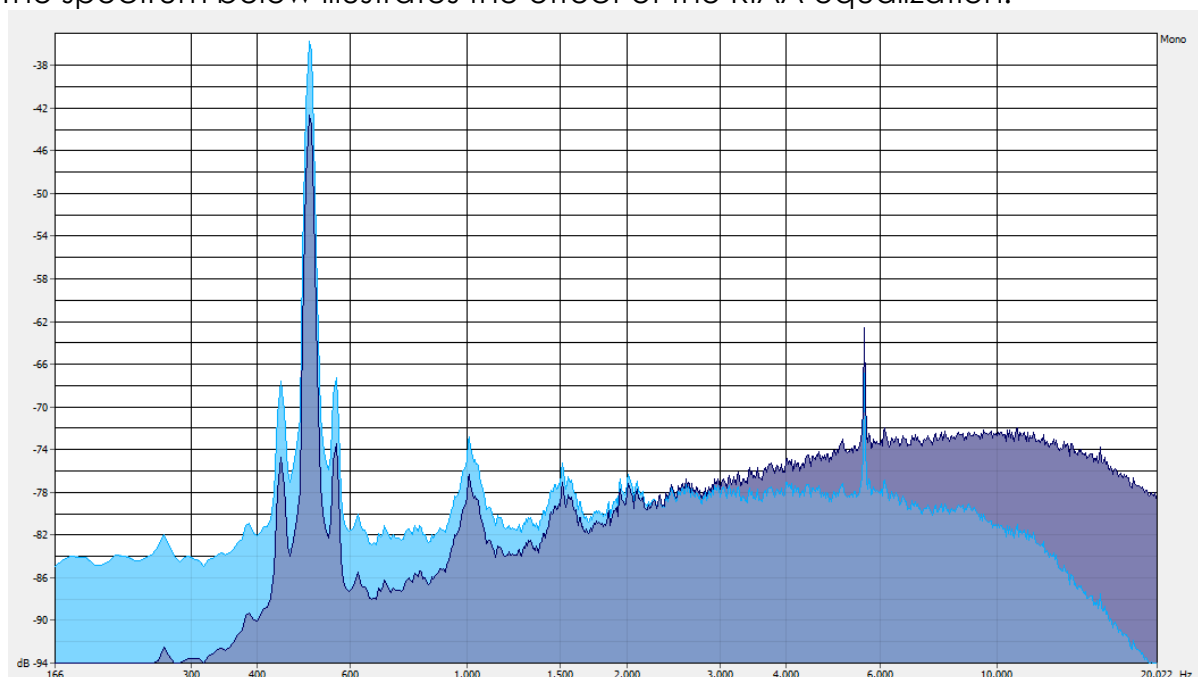


Figure 68: Comparison of the spectrums before (violet) and after (blue) RIAA equalization

The spectrum before equalization (violet) has a much bigger attenuation in low frequencies and the high frequencies are boosted. After the RIAA equalization (blue) the spectrum is almost flat. Due to the equalization, the high frequency noise is attenuated too, what provides an improvement on the sound quality.

8.2. SINAD

SINAD (**S**ignal to **n**oise and **d**istortion ratio) represents the ratio between the power of the signal and the power of the noise and distortion. To compare our output signal, we suppose the recorded sine wave as perfect, which is in fact not the case, but we need a reference to be able to compare to something.

The constant tone frequency disc is in a very good condition. Therefore the transcription of the analogue record is in a good quality, it gives certain indication, what performance we can await. [1]

8.3. THD

The **t**otal **h**armonic **d**istortion (THD) of a signal is a measurement of the harmonic distortion present and is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency.

THD is only representative when we measure on a tone test record with constant frequency. This record is supposed to contain only one frequency without harmonics. A record with sound always contains harmonics. Because harmonics make the difference between tone colors, they let the human ear differ between instruments and make music so multifarious.

For the calculation of the THD we are neglecting that the peaks have a certain width. The voltage ratio in percent is calculated with the formula below:

$$THD = 100 * \frac{\sqrt{\Sigma \text{harmonic powers}}}{\sqrt{\text{fundamental power}}} = 100 * \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + V_5^2 + \dots + V_n^2}}{V_1} \quad \text{EQ (21)}$$

The MatLab code for the THD calculation is in the appendix. [1],[15]

8.4. LabVIEW SINAD VI

To determine the SINAD we use a LabVIEW virtual instrument. This program reads a wave file and performs a Signal to Noise and Distortion analysis. It finds the fundamental frequency and simulates a perfect sine wave at this frequency.

The SINAD is then defined as the ratio of the RMS energy of **the input sound** to the RMS energy of noise and distortion. The RMS energy of noise and distortion is the RMS energy of **the input sound** less the energy in the fundamental. As the ratio usually is expressed in [dB], we take $10 \cdot \log$ of it.

$$SINAD = 10 \cdot \log \left(\frac{W_{RMS\ IN}}{W_{RMS\ IN} - W_{fundamental}} \right) \quad \text{EQ (22)}$$

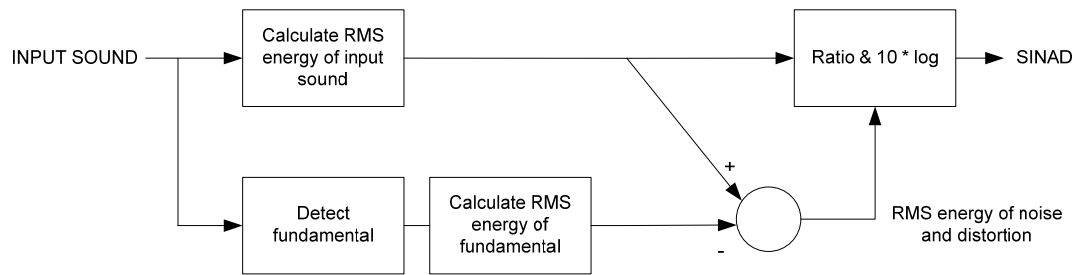


Figure 69: Bloc diagram of the LabView VI calculating SINAD

8.5. Sony Sound Forge 9.0

For the sound analysis we used Sony's audio editing software Sound Forge 9.0. It has an amount of possibilities to process sound. We basically used the Equalizer to do the RIAA equalization, the function of drawing the sound power spectrum, the waveform window, where the sound curve can be analyzed and the volume adjustment. It offers also possibilities for sound restoration, noise reduction and click removal for manual treating of the sound. We are using these tools at the end, when we try to improve our sound the best possible.

For the spectrum analysis it is important **always** to use the same window, in order to be able to compare between different sound versions. We used a Blackman-Harris window with a length of 32'768.

8.6. Analysis of the constant frequency record transcription

The quality of extracted sound with "3DSEVEN" is not only dependent on different algorithms used but also on the quality of the record and the condition of the disc.

This record contains sine waves at different frequencies: 10, 9, 8, 7, 6, 5, 4, 3, 2, 1.5, 1 kHz and 800, 500, 300, 200, 150, 100, 70, 50 Hz.

We first want to characterize the constant tone frequency record. We define SINAD and THD of this record and analyze its spectrum.

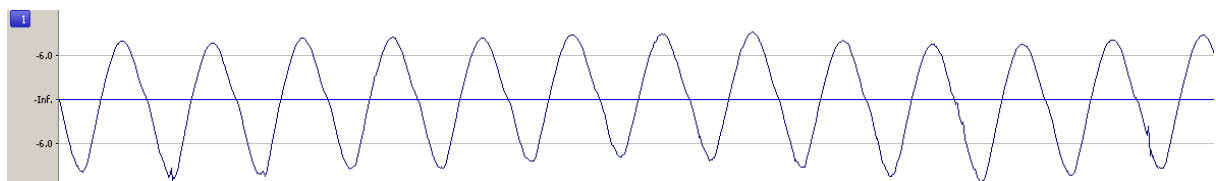


Figure 70: 500 Hz sine wave from the transcription

This is the spectrum of the 500 Hz sine wave with a peak at -25 dB.

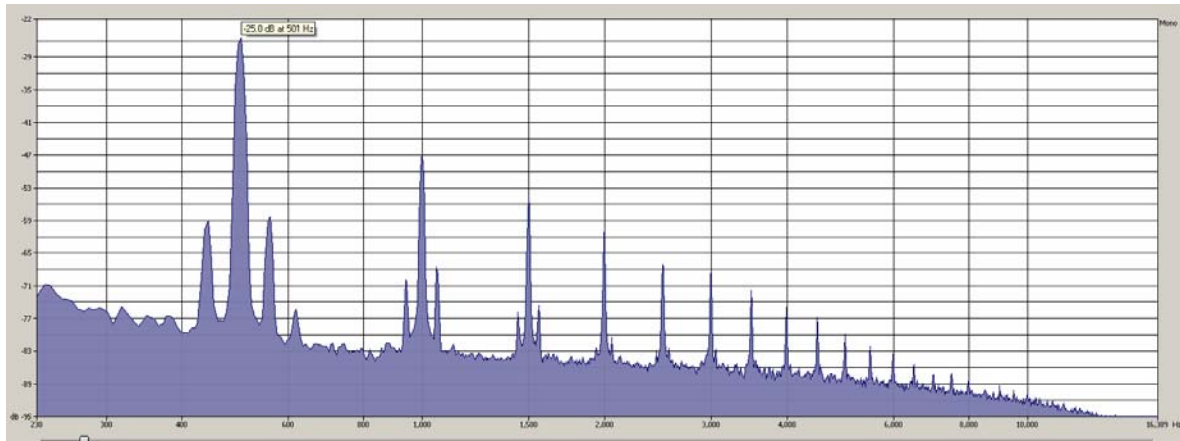


Figure 71: Spectrum of the 500 Hz transcription signal

On the spectrum above we detect an intermodulation of the sine wave with 60 Hz signal. There are 2 peaks on either side of the fundamental in a distance of 60 Hz 33 dB lower than the fundamental. This is the frequency of the electricity network in the United States, where this record was recorded. During recording there must have been some vibrations done to the power supply.

The harmonics are quite important, the first has a peak 22 dB lower than the fundamental, the second 30 dB lower. Power of fundamental and harmonics in dB are in table 9.

Fundamental	1 st	2 nd	3 rd	4 th	5 th	6 th	7 th	8 th	9 th	10 th
-25 dB	-47.1	-54.7	-62	-67.8	-70	-73.8	-76.1	-78.1	-81.8	-83.6

Table 8: Power of fundamental and harmonics

We obtain by measurement of the 500 Hz frequency:

SINAD [dB]	18.66 dB
THD [%]	8.69 %

Table 9 : Results of SINAD and THD

The quality of the constant frequency tone record is good, but the intermodulation with the supply power is important. The recording could have been done better by getting rid of this modulation.

8.7. Analysis of extracted sound of the constant frequency record

In this chapter we analyze how different algorithms and parameters have influence on the sound quality. In the "3DSEVEN" software different algorithms can be turned on and off (low pass filter, weighted average and cuts). In this chapter we analyze whether these algorithms improve sound quality or not. We also analyze how point spacing and overlapped scans affect the sound.

8.7.1. Without overlapped scans

Without overlapped scans and interleaving of points there is only one single height measurement every 10 μm for the groove profile. Bad points will directly affect the data and can only be cleaned by the low pass filter.

This is the fastest method to scan the disc, but as the performance of MPLS180 leaves a lot to be desired and did not image the constant frequency record very well, the sound quality is low.

SINAD [dB]	2.10 dB
THD [%]	81.33 %

Table 10: Results of SINAD and THD

The image after tracking also shows that groove centers could not be found everywhere very accurately and tracking did not work very well.

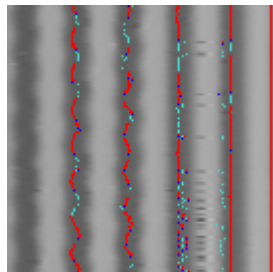


Figure 72: Detail of the image after tracking

These results prove that it was essential to overlap scans, where we obtain following values:

SINAD [dB]	3.2 dB
THD [%]	28.4 %

Table 11: Results of SINAD and THD

On the spectrum in figure 73 we see that the noise level for the version with overlapped scans in blue is lower than for the spectrum in violet without overlapping.

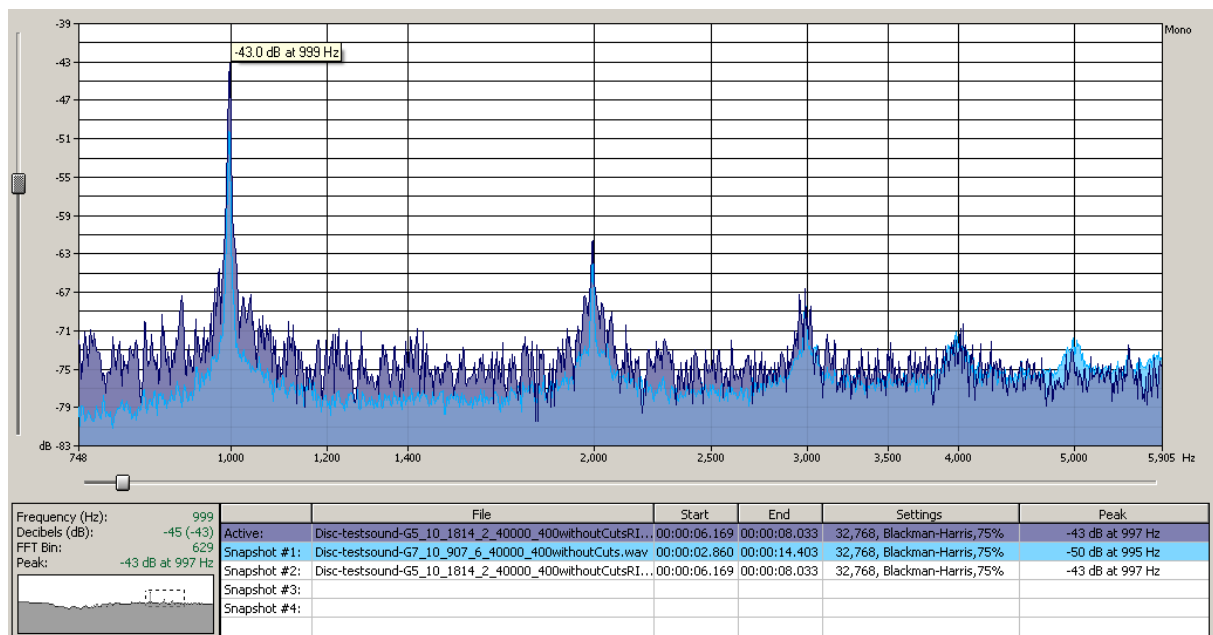


Figure 73: Spectrums showing the effect of the overlapping

8.8. Density of points

By interleaving points the resolution of the groove profile increases. We have a height measurement every 5 μm . As there is no overlapping yet, a lot of bad points remain in the data.

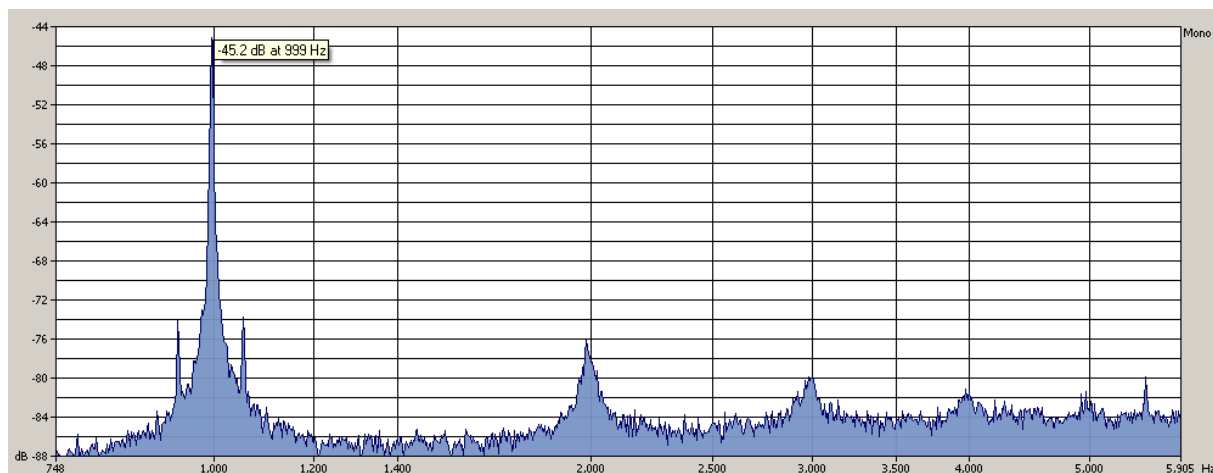


Figure 74: Spectrum showing the effect of interleaving points

SINAD [dB]	6.64 dB
THD [%]	3.92 %

Table 12: Results of SINAD and THD

When we now interleave even more points, spaced by 2.5 μm , moreover we overlap the scans, we are approaching to the quality of transcription.

SINAD [dB]	12.52 dB
THD [%]	2.12 %

Table 13: Results of SINAD and THD

Figure 75 shows the spectral comparison between transcription and interleaved (2.5 μm spacing) and overlapped version.

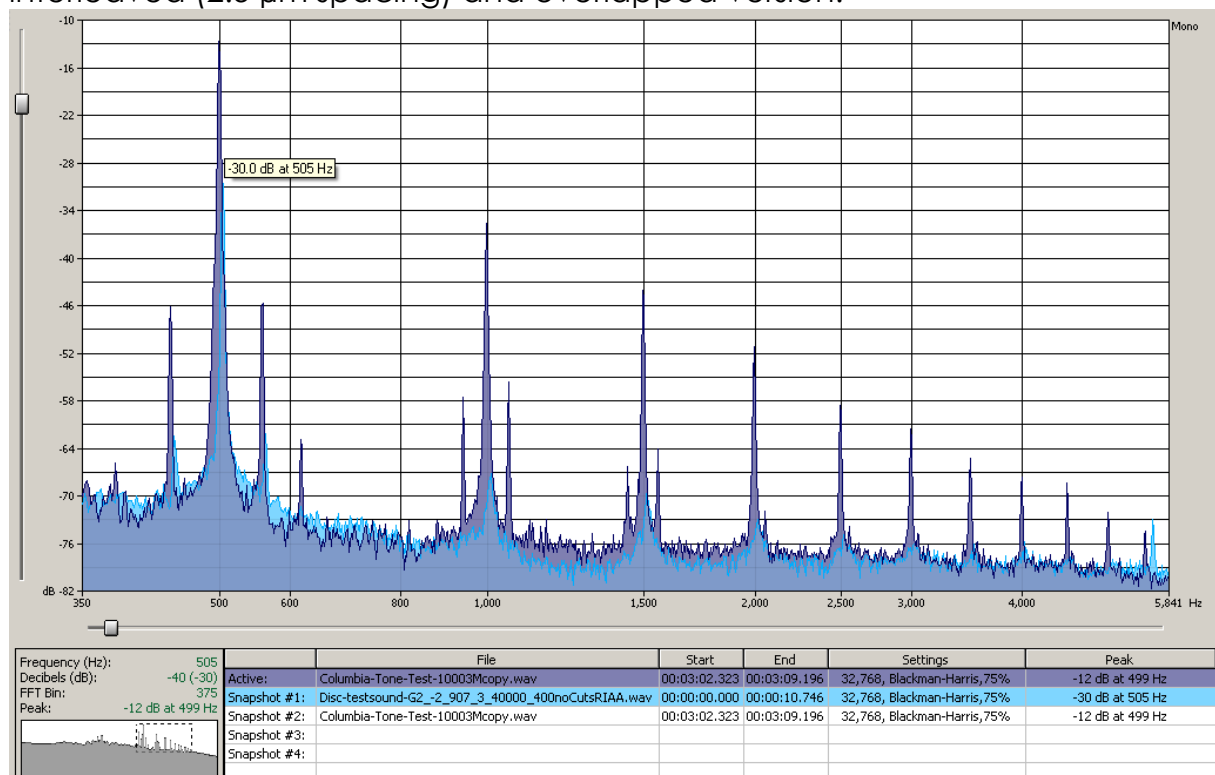


Figure 75: Comparison of spectrums between 2.5 μm (blue) and 5 μm (violet) spacing

Although there is a certain improvement with point spacing of 2.5 μm to 5 μm , most of our datasets were taken with a spacing of 5 μm , because a 2.5 μm point spacing produces huge data sets and scanning and digital treatment takes a long time. 5 μm spacing gives also quite satisfying results. Further measurements were all performed with a spacing of 5 μm and overlapped scan.

8.8.1. Low pass filter

Without low pass filtering, the groove profile has a lot of undulation, and bad points are not filtered. It is still possible to extract sound, but the performance is not very good.

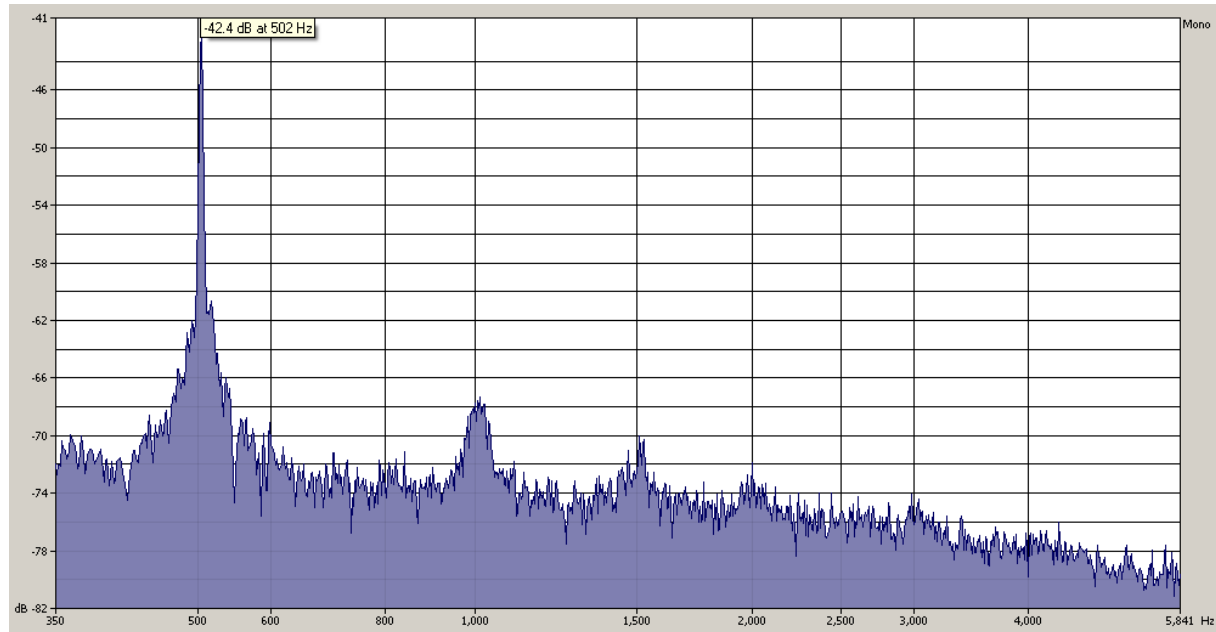


Figure 76: Spectrum of extracted sound without any low pass filtering

SINAD [dB]	6.59 dB
THD [%]	8.13 %

Table 14: Results of SINAD and THD

The values of SINAD and THD as well as the spectrum probably do not seem to be bad, but the subjective quality for the human ear is not good. A lot of groove centers could not be found due to undulation or bad points. This ends in silent parts and therefore the sound is not continuous.

8.8.2. Weighted average

The weighted average affects mostly the high frequencies. Therefore it will not affect the THD. But the SINAD will be affected because averaging attenuates high frequencies.

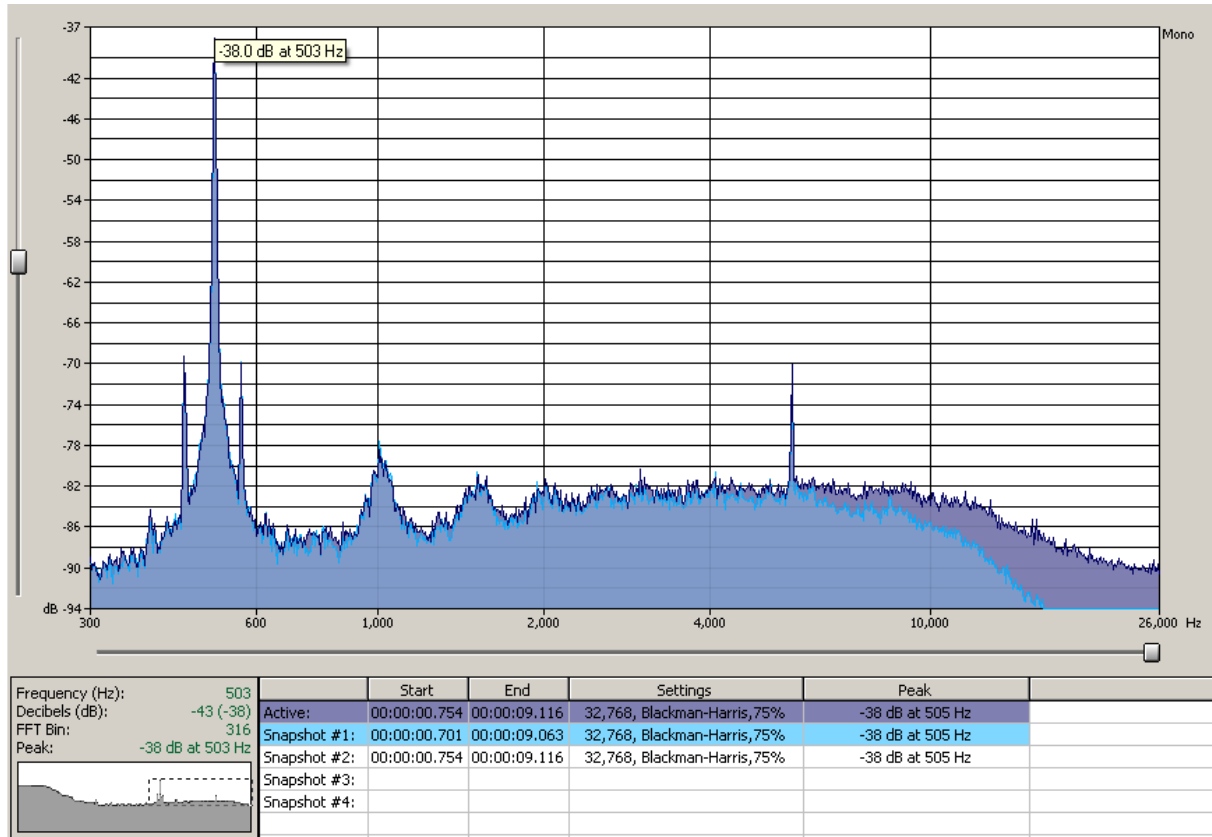


Figure 77: Effect of applying the weighted average (blue) and without average (violet)

With average	weighted	SINAD [dB]	14.63 dB
Without average		SINAD [dB]	12.43 dB

Table 15: SINAD values depending on average

8.8.3. Size of virtual stylus

With "3DSEVEN" it is possible to vary the size of the virtual stylus. This parameter sets the depth at what the groove center will be found and it basically defines the length of the line that is dropped into the groove. If the groove walls have different slopes, this parameter has influence on sound.

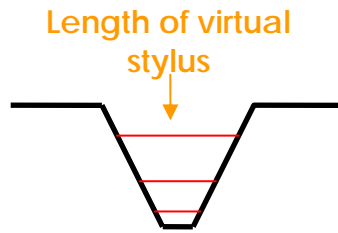


Figure 78: Principle of changing the virtual stylus length

Sound extraction of 3 different stylus sizes was compared. This parameter can be set in a range of 5 to 20, whereas one unit corresponds to 10 μm . The parameter was set to 8, 12, 16, hence a virtual stylus with a width of 80 μm , 120 μm and 160 μm was simulated. The difference for several stylus sizes is very small. By listening to the sound, no difference can be made. Also the spectrum is very similar.

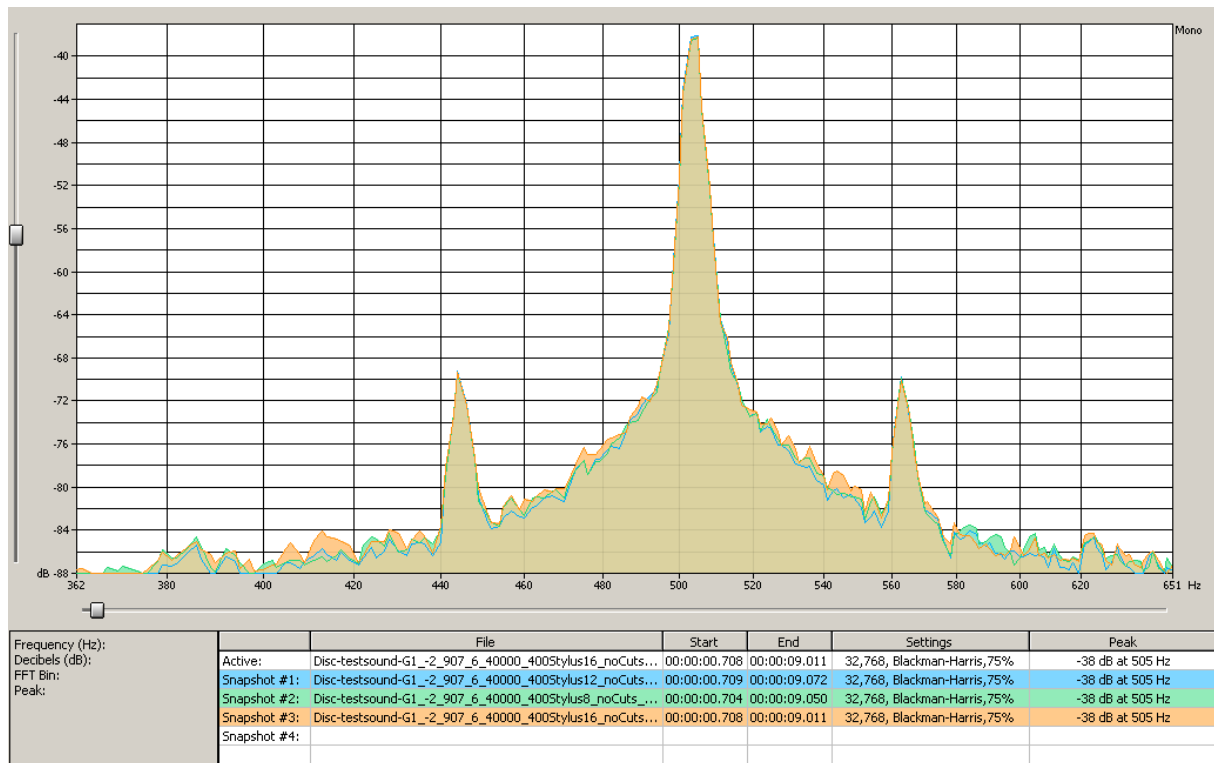


Figure 79: Comparison of the extracted sound spectrums using different lengths for the virtual stylus

The 2 peaks on the left and the right side of the fundamental correspond to intermodulation of the power supply during recording.

8.8.4. Cuts

Applying cuts permits to improve bad slope fitting. When the fitted line after cuts did not become better, the groove center is rejected. This should lead to more accurate groove center positions.

For the constant frequency tone record applying of cuts actually did not improve the sound.

In the spectrum in figure 80 the blue curve after applying cuts has much more noise in low frequencies in comparison with the green curve without cuts.

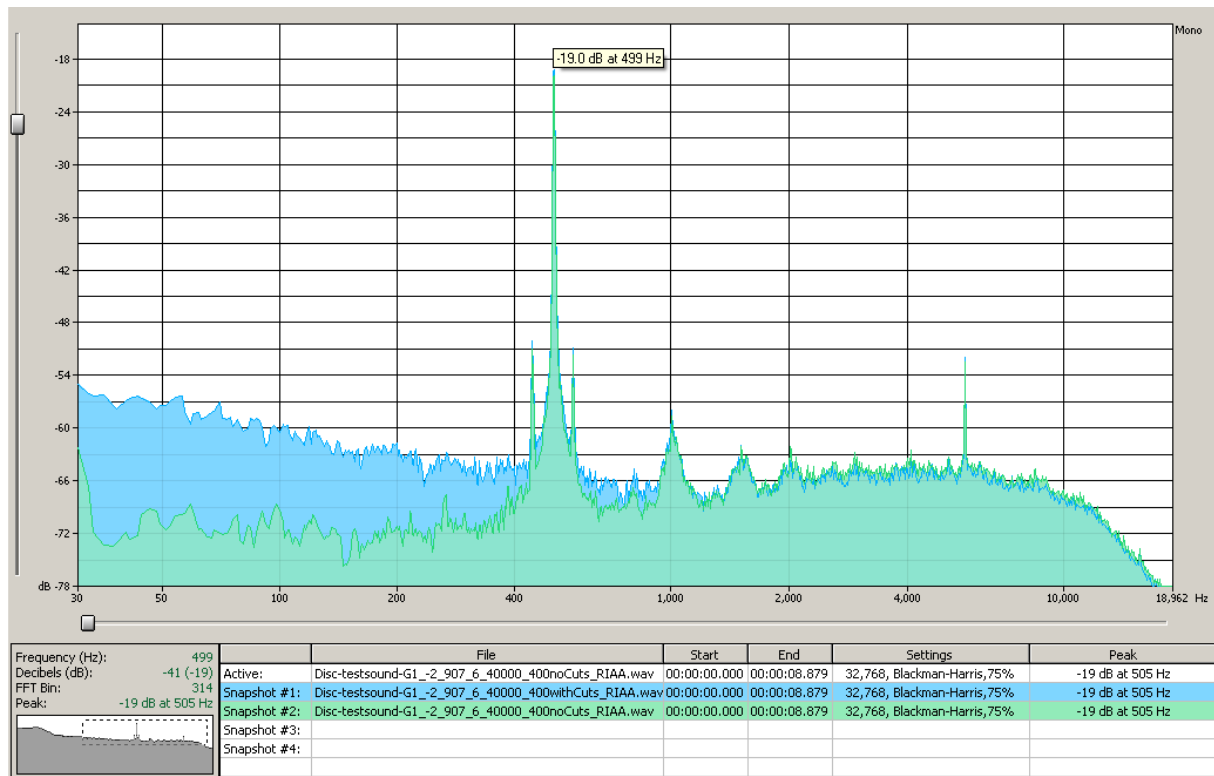


Figure 80: Comparison of sound spectrums before (green) and after (blue) cuts

	Without cuts	With cuts
SINAD [dB]	14.63 dB	12.84 dB
THD [%]	1.45 %	1.42 %

Table 16: Comparison of SINAD and THD with and without Cuts

8.8.5. Manual sound restoration for improvement

Several tools in Sound Forge allow improving sound quality manually. One tool acquires a noise print of a silent section of the record. To remove the noise, Sound Forge subtracts the noise print from the entire sound signal which contains also the noise. After this noise removal in the frequency domain, only the sound remains in the spectrum.

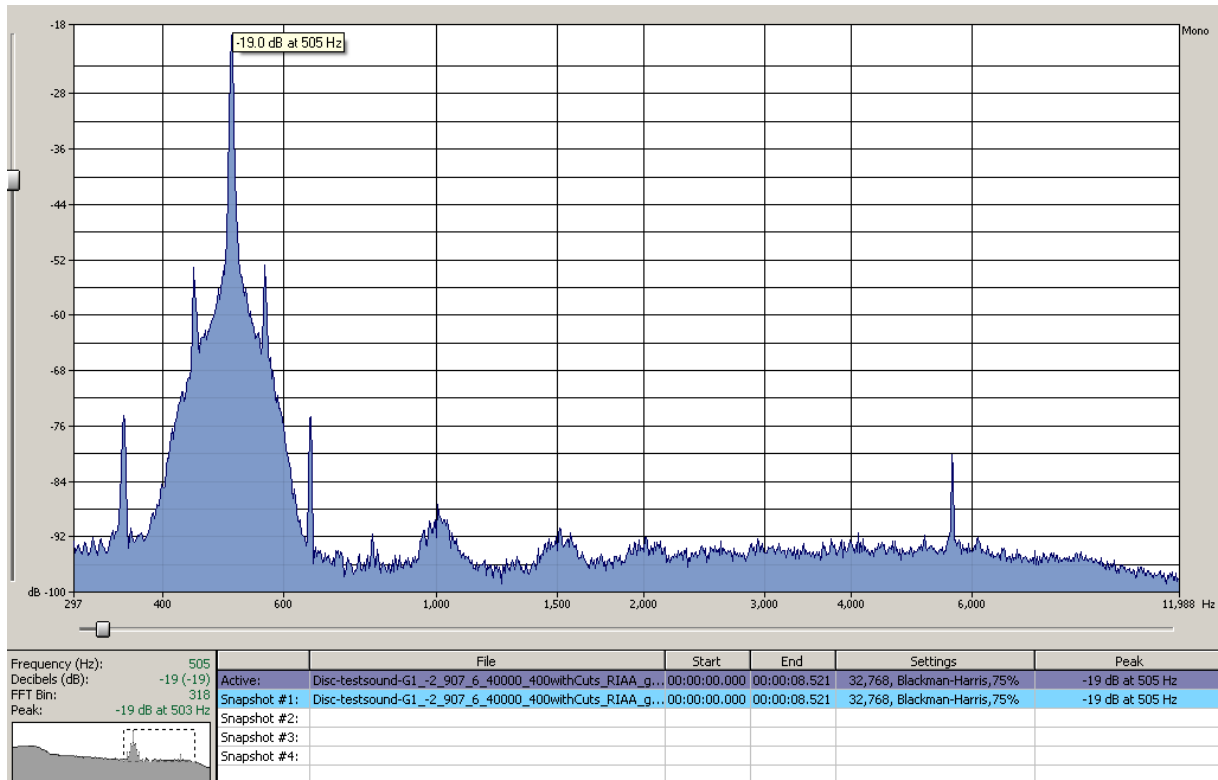


Figure 81: Sound spectrum after the noise removal

SINAD [dB]	27.9 dB
THD [%]	0.05 %

Table 17: Results of SINAD and THD

Although the noise print tool performs a very good effect on the tone test record, it is useless to apply on music records. It adds to much harmonic distortion and the tone color will not be the same anymore.

8.8.6. Comparison to I.R.E.N.E

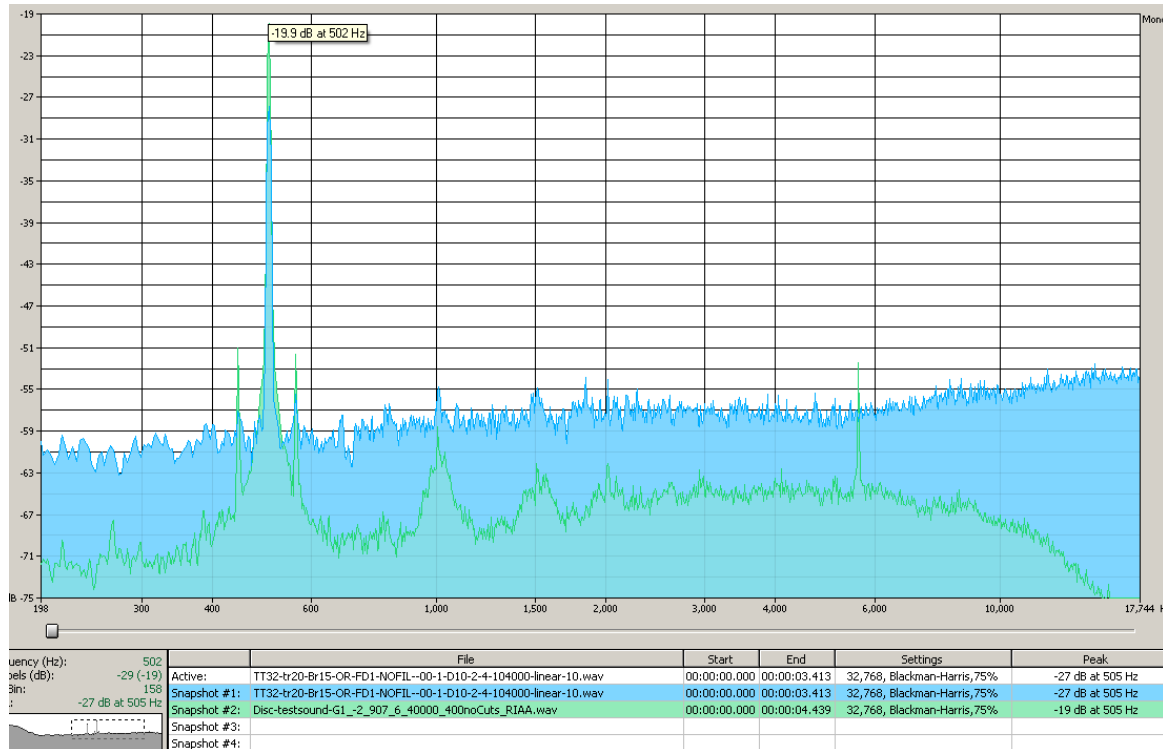


Figure 82: Comparison between I.R.E.N.E (blue) and MPLS180 (green)

The spectrum above shows a comparison of the sound extraction of MPLS180 (in green) and IRENE (in blue). The MPLS180 3D scanner imaged much better on the test tone record than the 2D camera of I.R.E.N.E. Therefore the extracted sound is of much better quality. In the I.R.E.N.E. version the high noise level hides the 60 Hz intermodulation and the harmonics. **Note:** This is not very authentic reference, because the test tone record imaged unusual bad with the 2D camera.

8.8.7. Conclusion

	SINAD [dB]	THD [%]
Transcription	18.66	8.69
No overlapping, no interleaving	2.10	81.33
Overlapping, no interleaving	3.20	28.4
Overlapping and interleaving (5um)	6.64	3.92
Overlapping and interleaving (2.5um)	12.52	2.12
Without low pass filter ¹	6.59	8.13
Low pass filter without weighted average ¹	12.43	1.45
Low pass filter with weighted average ¹	14.63	1.45
Without cuts ¹	14.63	1.45
With cuts ¹	12.84	1.42
Manual sound restoration ¹	27.9	0.05

I.R.E.N.E. ²	0.50 ²	NA
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Table 18: Final comparison of SINAD and THD

1: With overlapping and interleaving (5 μ m point spacing)

2: Not relevant, because the 2D scanner images the tone test record unusual bad

To extract sound from 78 rpm records it is definitely necessary to scan overlapped and interleaved. To really make profit of this multiple point measurement the low pass filter has to be applied. The effect of the weighted average is not very important and it could also be applied later on in the sound domain.

The cuts did not improve the sound quality for the test record. But as we will see later, applying them on real music discs has positive effects.

Extracted sound with MPLS180 has less harmonic distortion than the analog transcription.

8.9. Analysis of music extraction

Among the disc stock at LBNL we found some music records that MPLS180 imaged well. We scanned 3 records and extracted sound from them. Data of them was acquired by overlapping scans with a point spacing of 5 μ m. We applied all of the data cleaning algorithms (low pass filter and weighted average) and will now compare the influence of cuts.

For these records it is not possible to determine neither the SINAD nor the THD. To determine the SINAD we would have to possess a transcription. The measurement of the THD is not relevant for music since there is always harmonic distortion to produce different tone colors.

All the *.wav sound files are on the disc in attachment to this paper. A playback of the different sound versions gives probably a better impression.

8.9.1. In the evening by the moonlight

This is a very old record dating from 1908. It was recorded acoustically, therefore the sound level is very weak and the noise level we suppose to be quite considerable already on the disc.

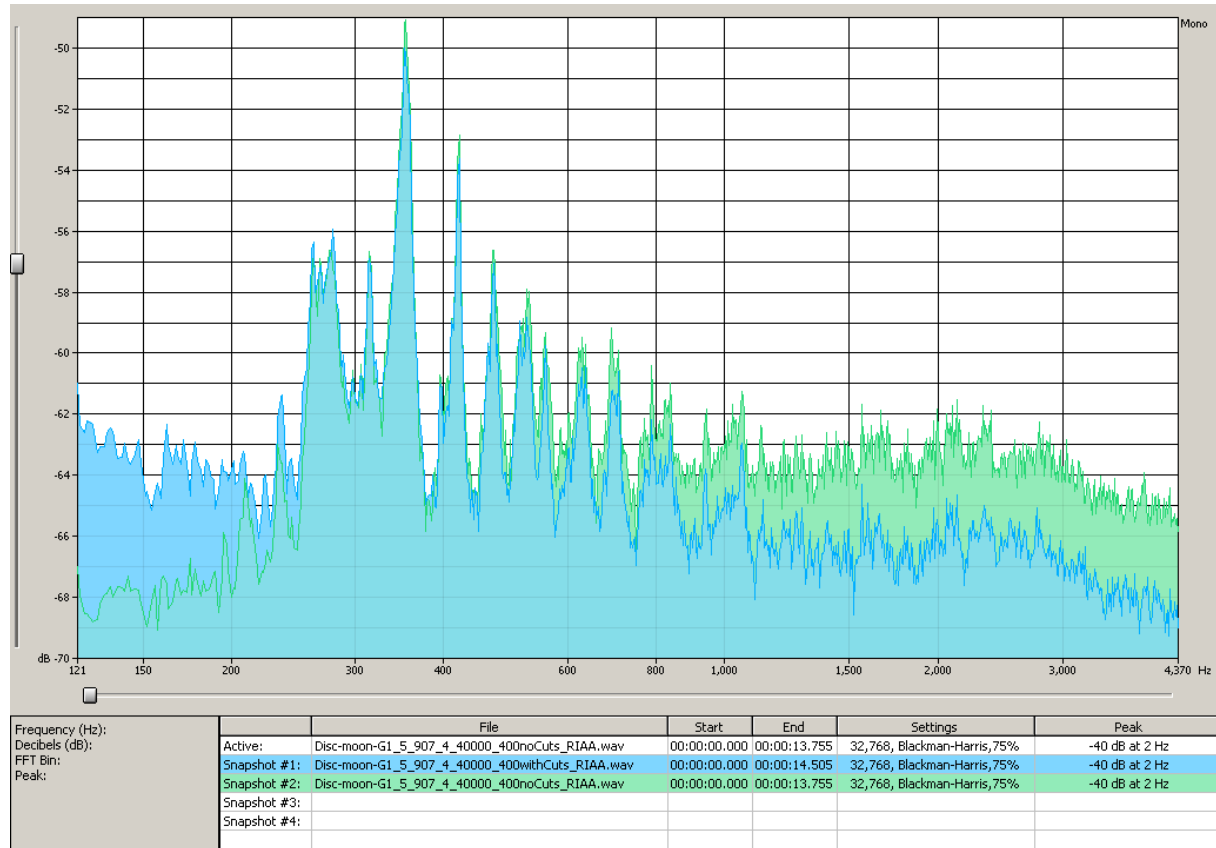


Figure 83: Effect of the cuts

For this disc, applying cuts leads to better signal to noise ratio. The blue spectrum after cuts is merely worse for low frequencies than the green before any cuts.

There is neither an I.R.E.N.E. version nor a transcription existing, why we cannot compare the extracted sound to any other reference.

8.9.2. Chattanooga blues

The Chattanooga Blues record is a cracked disc and very worn. This disc was also scanned with I.R.E.N.E. and we have a transcription to compare on.

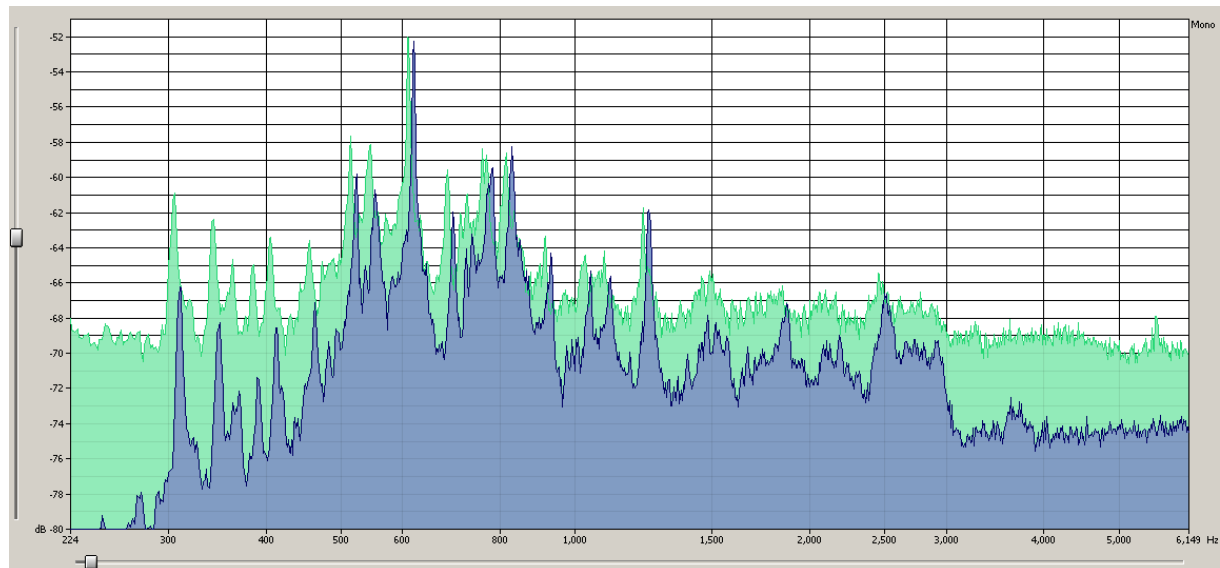


Figure 84: Comparison between MPLS180 (green) and transcription version (violet)

The transcription in blue has a much lower noise level all over the frequency band than the MPLS180 version in bright green. The transcription quality could not be reached, although our version does not have drawn sections due to worn grooves. There is a slight frequency shift visible. The rotation speed during the transcription must have been marginally too fast, we suppose our rotation stage to be more accurate than an old player.

Compared to I.R.E.N.E. without applying cuts for both versions, extracted sound is very similar. The noise level might be a little higher for our MPLS180 version, but peaks in the spectrum of the 3D version are sometimes narrower.

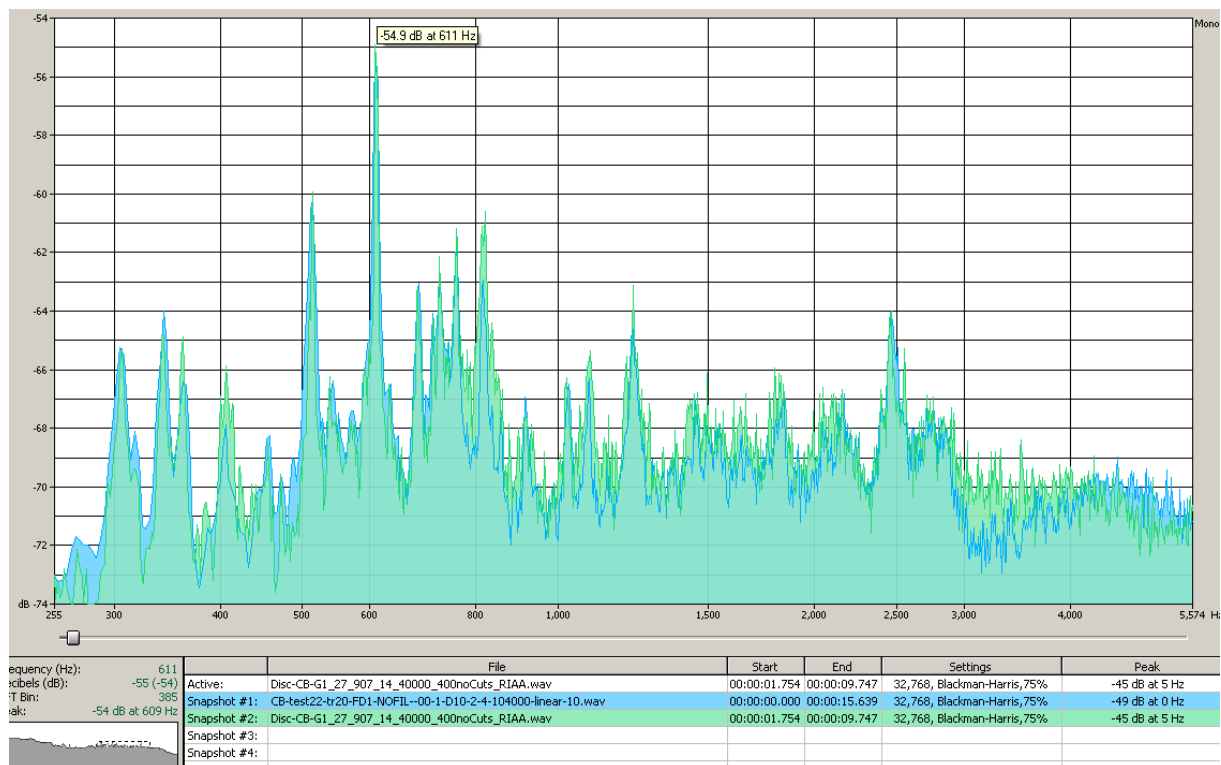


Figure 85: Comparison between IRENE (blue) and MPLS180 (green)

8.9.3. I wonder where my easy rider's gone

We have reconstructed sound in a good quality from this record. There is no possibility to compare because neither a transcription nor an I.R.E.N.E version exists. The sound file is on the CD.

8.10. Conclusion of the sound analysis

The quality of the analogue transcription is often difficult to reach with an optical system but it can be approached very close. For worn or cracked discs where a conventional playback might fail (groove skipping) sound can still be played with an optical reading system such as I.R.E.N.E. or MPLS180. Hence lost music can be played back again without any deterioration due to stylus contact.

I.R.E.N.E. and MPLS180 perform measurement of almost opposite quality depending on the scanned surface. The two dimensional camera images better on reflecting discs whereas MPLS180 images only well on matt and worn discs.

The software designed for the 2D camera definitely is more sophisticated to apply cuts, which affects sound positively. "3DSEVEN" is able to apply cuts, but this part of the program is not developed enough yet. As there is much more data in a three dimensional groove profile, there are more possibilities to improve data. Nevertheless it is more difficult to make profit of it, algorithms are more complicated and data treatment takes more time.

MPLS180 performed only good measurements of matt and worn discs. Sound quality on these records is already quite low (acoustic record, deterioration by stylus). The quality of extracted sound cannot be better than on the record.

9. Scanning and processing time

The sound analysis has shown that satisfying sound extraction needs scan overlapping, point interleaving and high exposure times. Obviously this arises the scanning the processing time.

9.1. Scanning time

A disc contains sound in a band of a width of 7 cm. With an exposure time of 2.5 ms scanning and buffering of the data of a single ring takes about 2.5 minutes. Without overlapping and interleaving the scanning of an entire disc (approximately 40 scan rings) takes about 100 minutes. When data is overlapped and point spacing is 5 μm we need 4 times more scan rings and scanning time comes up to 400 minutes (= 6h 40 min).

With I.R.E.N.E. scanning of an entire disc took only about 20 minutes, MPLS takes much more time.

9.2. Processing time

“3DSEVEN” needs quite a lot of resources, with a very powerful machine which arrived at LBNL during the project digital processing an entire disc takes about 1.5 hours. (estimated, we’ve never processed an entire disc, only half of a disc).

It might be possible to process data faster by adapting the algorithms, but this requires more knowledge about program coding. But as long as the processing time is inferior to the scanning time, our concentration was not on accelerating the processing.

10. Further development

10.1. Cuts

The experience of the IRENE 2D project has shown that various statistical distributions, and cuts on these quantities, can have a significant effect on impulse and other transient noise features.

In the case of 3D scanning some of the basic histograms have been collected but at present the cut levels have not been studied in detail to determine what amount of noise reduction can be achieved. This is an aspect for further research and development in the 3D scanning and data analysis effort.

10.2. Groove detection

Different algorithms for the groove detection might be implemented to obtain the position of the groove with different approaches.

10.3. Include varying fiber spacing

As mentioned before MPLS180's fiber spacing depends on the fiber number and the height of the sample, due to the spread of the 180 fibers. Points do not have a regular spacing across the groove profile any longer. Digital data processing becomes much more complicated, because every point does not only have a height but also a relative lateral position. This changes the whole concept for matching data together and averaging between overlapped scans. Including this aspect should lead to more accurate groove profiles.

10.4. 45° measurements

As MPLS180 does not image well on groove walls of records with reflecting surface, another approach is to scan the grooves in an angle of 45°.

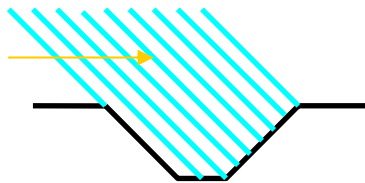


Figure 86: Principle of 45° measurement

This concept has big issues with data realigning and the autofocus system, but preliminary measurements have shown that better sound quality might be achieved and reflecting discs can be scanned with the multiple line sensor.

11. Conclusion

“Extraction of sound using 3D” showed that it is possible to scan discs with a multiple line sensor and extract sound from old records without contact. Although a lot of measures have to be taken in order to get satisfying results because the MPLS180 has difficulty to image discs.

Extracted sound can be compared with sound retrieved with other optical reading systems, but a further development including sophisticated bad data rejection needs to be implemented to really make profit of the 3D acquisition system. Varying fiber spacing has to be taken into account, and a 45° measurement has to be evaluated. The time was too short to include these developments in this diploma project.

It is difficult to compare the sound scanned with MPLS180 with other reading methods. It is more complementary to I.R.E.N.E than comparable, because MPLS180 imaged well on matt surfaces, opposite to the 2D scanner that imaged well the reflecting discs. In comparison to an analogue playback the sound quality of a optical reading system is probably less good, but discs that were not playable with the conventional method can be played again.

This project definitely gave us a larger horizon about how digital signal processing can be used. We acquired a lot of knowledge about new software such as LabVIEW and Visual C# and improved our program coding skills. “Extraction of sound using 3D” was based on a lot of algorithm development, which is not the main task of an electrical engineer. But we probably have more ease for the physical comprehension of the system, the signal processing, the issues about sampling frequency, digital filtering and sound analysis (SINAD & THD). This project could have been proposed for a computer scientist together with an electrical engineer, but we still were able to do it.

A project like this could be continued forever, there are always new ideas and approaches which might yield to improvement. But the time for this project was limited and the presentation of results (writing documentation and analyzing sound) was more important than developing until the end, this is the reason why we could not advance any further.

It was undoubtedly a unique chance to do this project abroad in Berkeley that we are happy not to have missed.

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13. Figures

Figure 1: Dimension of a groove	5
Figure 2: Measurement points of 2D system.....	7
Figure 3: Measurement points of 3D system.....	7
Figure 4: Hardware overview.....	10
Figure 5: Different scanning directions (across or along the grooves)	10
Figure 6: Scanning process with axial steps placing MPLS across the groove .	11
Figure 7: Principle of the confocal microscope	12
Figure 8: Overview of MPLS180 system.....	13
Figure 9: Spacing between points of the MPLS180	13
Figure 10: Spreading of fibers	15
Figure 11: Angle of incidence equals angle of reflection	15
Figure 12: Reflection in a slope section.....	16
Figure 13: Acquired height profile with missing groove walls	17
Figure 14: Acquired height profile in good quality	17
Figure 15: Comparison of the phonographic records.....	18
Figure 16: XPS Motion Controller	22
Figure 17: Measured height profile of the sticker in all three repetitions	24
Figure 18: Zoomed height profile from the beginning of the sticker.....	24
Figure 19: Height profile showing the acquired groove.....	25
Figure 20: Keyence Height Detector (Left) and its measuring spot (Right).....	26
Figure 21: Error due to auto focus system of 2 overlapped passes	27
Figure 22: Principle of an overlapped scans.....	30
Figure 23: Principle of increasing resolution by point interleaving.....	30
Figure 24: Explication of offset removal with absolute and relative height	31
Figure 25: The LabView front panel used for data acquisition.....	32
Figure 26: General structure of a *.pri file	35
Figure 27: Detailed view of the *.pri file header	36
Figure 28: Detailed view of the data part in a *.pri file.....	37
Figure 29: Bloc diagram of the program 3DSEVEN	39
Figure 30: GUI of the program 3DSEVEN and division in three sections.....	40
Figure 31: Data visualization section of the GUI	41
Figure 32: Parameter setting part of the GUI	42
Figure 33: Principle of the offset removal process.....	43
Figure 34: Disc height image with bad points (red dots) in the non-overlapped part	43
Figure 35: A single height profile with bad points on the left side (Area without overlapping).....	44
Figure 36: High frequency undulation due to the auto focus system.....	44
Figure 37: Frequency response of the implemented IIR low pass filter	45
Figure 38: Structure of the IIR filter	46
Figure 39: Comparison of height profile before (blue) and after (red) filtering	47
Figure 40: Height image after filtering	47
Figure 41: Comparison of frequency response between weighted and standard average	49
Figure 42: Principle of the approximate method for groove center detection	50

Figure 43: Problem of multiple zero crossing	50
Figure 44: Principle of the groove slope intersection method	51
Figure 45: Principle of the constant line method.....	52
Figure 46: Basics of the Least Squares Method.....	53
Figure 47: Problem of the data type overflow.....	54
Figure 48: Principle of the tracking process.....	55
Figure 49: Image of the discs height profile	56
Figure 50: Start position problem.....	57
Figure 51: Search for next groove center in a silent part	58
Figure 52: Search for the next groove center in a wavy part	58
Figure 53: Stream of all groove centers	59
Figure 54: Detailed view of this stream showing the overlaid signal that represents the sound	59
Figure 55: Stream after applying derivative method is equal to the sound	60
Figure 56: Detail of the first (left) and second (right) image created by the program.....	61
Figure 57: Image of a discs height profile including a dust particle on the discs surface	62
Figure 58: The appearance of a scratch on a disc surface	62
Figure 59: A disc surface of a cracked disc	63
Figure 60: The histogram of the all the height over the whole surface.....	64
Figure 61: The histogram showing the distribution of the sums of squared residuals.....	65
Figure 62: Histogram of the calculated slopes of interpolated functions	65
Figure 63: Histogram of the calculated differences between two groove centers	66
Figure 64: Applying cuts with the program	67
Figure 65: Detail view of the *.grv file header.....	68
Figure 66: Detail view of the data part in a *.grv file	69
Figure 67: Frequency response of the RIAA equalization	72
Figure 68: Comparison of the spectrums before (violet) and after (blue) RIAA equalization.....	72
Figure 69: Bloc diagram of the LabView VI calculating SINAD.....	74
Figure 70: 500 Hz sine wave from the transcription	74
Figure 71: Spectrum of the 500 Hz transcription signal	75
Figure 72: Detail of the image after tracking	76
Figure 73: Spectrums showing the effect of the overlapping	77
Figure 74: Spectrum showing the effect of interleaving points.....	77
Figure 75: Comparison of spectrums between 2.5 um (blue) and 5 um (violet) spacing	78
Figure 76: Spectrum of extracted sound without any low pass filtering	79
Figure 77: Effect of applying the weighted average (blue) and without average (violet)	80
Figure 78: Principle of changing the virtual stylus length	81
Figure 79: Comparison of the extracted sound spectrums using different lengths for the virtual stylus	81
Figure 80: Comparison of sound spectrums before (green) and after (blue) cuts	82

Figure 81: Sound spectrum after the noise removal	83
Figure 82: Comparison between I.R.E.N.E (blue) and MPLS180 (green)	84
Figure 83: Effect of the cuts	86
Figure 84: Comparison between MPLS180 (green) and transcription version (violet)	87
Figure 85: Comparison between IRENE (blue) and MPLS180 (green)	88
Figure 86: Principle of 45° measurement	90
Figure 87: 3D height map of a phonographic disc with grooves.....	18

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