******* 5-star review –** *A vinyl enthusiast's reference book with many historical anecdotes. Visually beautiful photographs and charts. The author is a recognized professional audio engineer and musician. The hobby is full of misinformation which cause people to waste barrels of money…Get this book. It's anti-BS…*

******* 5-star review –** *Information from the author's professional experience as well as research and study. I learned more about the stylus shape in getting the most out of the record grooves without distortion. The author's no-nonsense approach is refreshing in this age of fantasy thinking by hi-fi hucksters selling snake oil to innocent audiophiles. I'm glad he shared his insights, wisdom and information in this nice book…*

******* 5-star review –** *A great gift for the enthusiast and those new to vinyl. Highly recommend!*

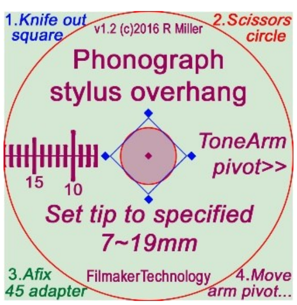
******* 5-star review –** *Essential reading for anyone who wants to get the best from their vinyl – and older shellac – records…by carefully setting up your system. Construction chapters show you how to make your own equipment. I like this book because it is well written, easy to understand and has useful illustrations. There's also online updates. What makes this book so good is that it shows how great vinyl playback does not need thousands of dollars…*

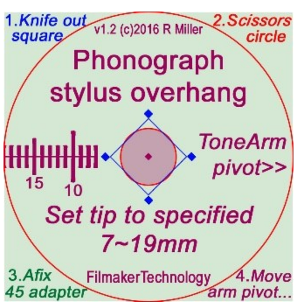
******* 5-star review –** *I intend to build both the book's preamp and tonearm. A very thoughtful book on the turntable and its science; I would recommend…*

******* 5-star review –** *Perfect for a techno-music-lover, it's a profusely illustrated guide, with knowledge that's harder to find in print than you'd think…*

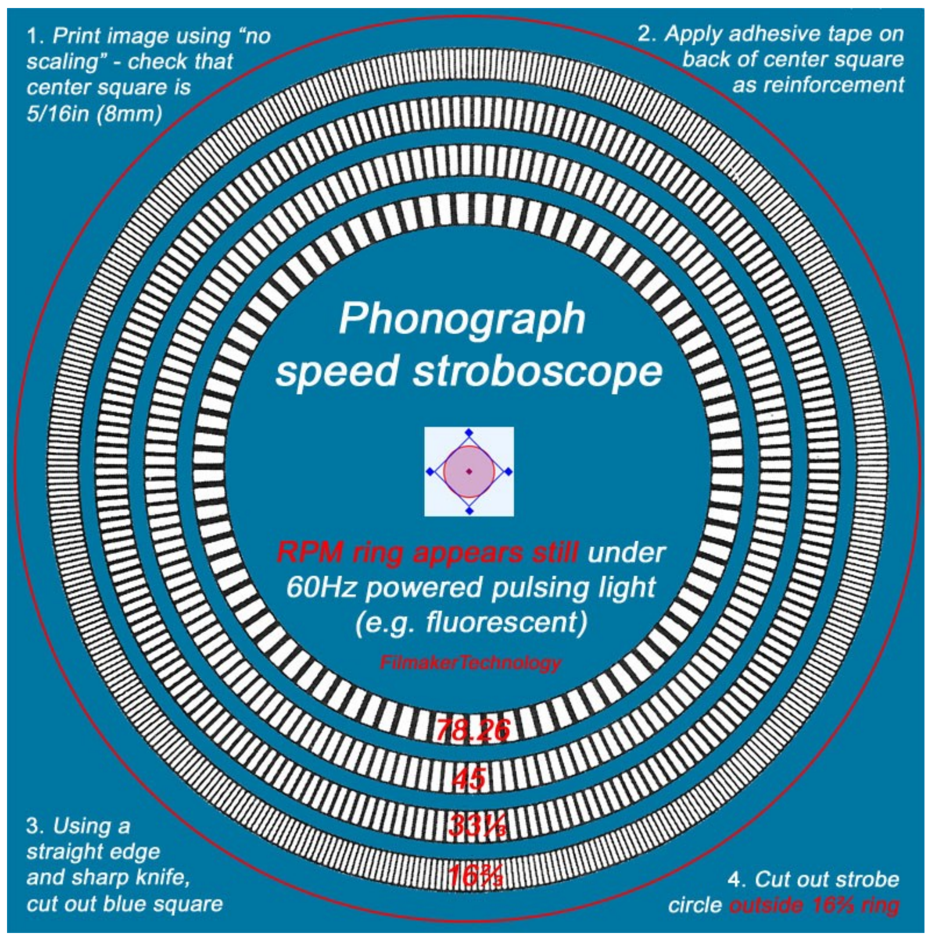
[Comments edited for space.]

Alignment tools - print a copy of this page exact size





Use alignment table in 2nd edition p60 (or p68 in 1st edition).



Left: ***Stylus Overhang Checker*** for pivot-to-tip alignment for lowest tracking distortion using tonearms 8½~12in (218~305mm) cf. p68 – attach to a flat 45rpm adapter. View stylus' cantilever tangent to groove to angle the cartridge body. Reducing other playback distortions are throughout the book. *Right:* ***Phonograph Speed Stroboscope*** to check turntable pitch 60Hz (50Hz versions online). 78.26rpm resolved the historic range of 65~100+*!*

*This book's **2nd edition** is now available under the title

Better Sound from your Phonograph: *How come? How-to!*

Completely rewritten, with new micro-photos, and an Index.

UPDATES covers 1st & 2nd editions, so 2nd ed. owners please skip p4~9 and redundancies after.

Contents – UPDATES & extras (hyperlinked to content) – 1st & 2nd book ed's

As in the Afterword of Better Sound from your Phonograph

The author might come off in both the book and following pages as passionate about his long-time and in-depth study and publications on audio reproduction, from the benefits of using stereo subwoofers to full sphere 3D recording & reproduction (minimum of 10 speakers), especially for music, movies, & gaming. However, it does not mean that tomorrow he might not discover an error (see *Errata*), or new knowledge, and change his tune. This describes the scientific method of a curious engineer by training & experience, both begun in the hi-fi era of the 1950s and ongoing, toward what might be next. It's why this *UPDATES & extras* exists as a companion to the book, to be revised as needs arise [version date is in each footer].

If you too are more curious about cause & effect than shopping & swapping, I invite you to peruse either of the book's editions, in any order relevant to you, to implement its knowledge, and to reference in future. [These UPDATES are for both 1st & 2nd editions, so 2nd edition owners please skip any redundancies.] When making an equipment purchase, the book can prepare you, and save you *buyer's remorse*. Pull the trigger *after* doing homework rather than impulsively, or based on uninformed gossip. The internet can be a great resource; also a quagmire of poppycock. I pull no punches busting myths, so as to be responsible to both inquiring audiophiles and those new to the phonograph about how to get truly higher quality sound.

Contrary to what some believe, audio technology, electronics particularly, have improved in reproducing sound to within picking a nit. This is particularly true of digital audio, which technically is far superior to prior methods. Most changed over audio's 140+ years, audio electronics can be characterized completely by modern scientific measurements (frequency flatness, non-linearity distortions that add tones not in the original recording, noise, timing anomalies such as flutter and jitter [Winer 2012]). Of course individual subjective hearing perceptions cannot be measured directly, and are influenced by disparate preferences for timbral colorations ("euphonious distortion"), and biases reinforced by cool looks and high prices.

This leaves electro-mechanical transducers (phono pickups, loudspeakers) and listening room acoustics as the perennial frontiers. Acoustic treatment, while difficult, is doable by hobbyists; pushing limits making transducers is challenging for manufacturers. Explored in depth both in the book and here, a phonograph pickup is limited in reproduction accuracy largely by the shape of its tip tracing the modulated groove – it is fair to say *the sound of vinyl is the sound of its stylus.* Unless it mimics the profile of the chisel that cut the record, it will be pinched at the groove's midpoint, and trace its curves eccentrically, generating distortion.

Fixed in the groove is program content – the outcome of musicianship and the recording engineer, who in vinyl's Golden Age developed skills to work in spite of primitive (now "vintage") tools. In their attempts in vain to "*re*-produce" the record in pursuit of a sound they ephemerally prefer, audiophiles fuss about what fanzines call "euphonious distortions." A close second is an unearned discovery when the bad sounding frequency response of one component accidentally compensates the opposite error in another [Pickering]. There's greater return on your investment of effort & money by reading before spending either.

Whether the hobby is music or audio – or both – there is nothing that can be done by enthusiasts to "improve" the recordings as produced and distributed: The song, arrangement, performance, capture, mix, and mastering, often embedding the influences of lousy control room speakers & acoustics. But for its replay, we have plenty to do. The *raison d'être* for collecting (and cleaning old) analog LPs – and buying, maintaining, or restoring the equipment to play them – is to enjoy the music recorded on them, which hold up well compared to successor tape and streaming distribution media. Compared to any other, properly played, phonograph records are a high quality archive in sound of our culture over audio's 140+ years!

Both editons of this book are for the less nonsense phonograph hobbyist, whether on a budget or not. A return, if you will, to Golden Age of the hi-fi hobby in the 1950s, when the author was bit by the bug of focused listening to recorded music. No high-end solution can magically make up for poor recordings. So with the possible exception of speakers and listening room acoustic treatment, wasted are efforts and gear expenditures in excess of extracting what any recording medium has to offer. As I said early in both 1st and 2nd editions, much audiophilia has devolved into "reverse alchemy," but why "turn your hard-earned gold into no better sounding lead?" When the "secrets" are in books like the one in your hands.

We begin with updating the 1st edition's Table of Contents, followed by a table of 15 stylus profiles to help you understand and choose your next (2nd edition owners, skip to page 8)…

New Table of Contents (for 1st edition) – **Not applicable to 2nd ed.** – *skip p4~9 and redundancies after*

As needed, print the next two pages back-to-back, and fit them in your 1st edition….

Table of Contents

Improves the Wear v. Cutoff chart, some of the most researched and useful information in this book…

Groove & stylus wear v. high freq. "f_3 cutoff"						Data	Calc'd		©2017 Robin Miller 8/24/17d	
Stylus mil	conWµm	conHµm	conAµm²	VTFg	press.lb/in²	speed	rel. wear **		f_3 kHz ***[Miller cf.*]	
Sph 3.0x3.0	19.8	19.8	308	5	16,338	78	0.7	↑ if shellac	1.5	SP
2.0x2.0	13.2	13.2	137	4	29,408	33.3	0.5		1.0	radio ET 16in
1.0x1.0	6.6	6.6	34	3	88,224	33.3	1.6	HF erasure>	1.97	[*Goldmark]
.7x.7	3.8	3.8	11	2	177,425	33.3	3.1	HF erasure>	3.4	
.6x.6	3.5	3.5	10	1	104,572	33.3	1.9	HF erasure>	3.7	stereo range
.5x.5	3.3	3.3	9	1	117,631	33.3	2.1	HF erasure>	3.9	
Ellip .5x3.0	3.8	18.0	65	4	61,603	78	2.6	↑ if shellac	8.0	SP
.4x.7	3.0	4.5	12	3	260,854	33.3	4.6		4.3	
.3x.7	2.25	4.5	9	2	222,560	33.3	3.9		5.8	stereo range
.2x.7	1.5	4.5	6	1	160,477	33.3	2.8		8.7	
Line StHd	2.25	8.4	18	1	56,457	33.3	1.0	≡ wear ref	5.8	D81 .3x.7x2.8
DJ StHd	2.25	8.4	18	3	169,372	33.3	3.0	<2.8 at 2¾g	5.8	D6800SL "
.2 StHd ii	1.5	9.0	13	1	77,257	33.3	1.4	<1.8 at 1¼g	8.7	D81Sii .2x.7x3.0
.13 SAS	1.0	9.0	9	1	114,467	33.3	2.0		13	JICO .13x.7x3.0
.10 Quad	0.8	8.4	6	1	162,742	33.3	2.9		17	Pickg D4500Q

**varies w/VTF, speed, friction. Size data courtesy JICO, or interpolat'd/Miller. ***Outer groove f_3 ~double.

Tabulated above are 15 stylus profiles revealing performance at high frequency, implied distortions, and relative wear of both the stylus and the records it plays. Choose a replacement tip informed by this table. LP or SP side scanning radii range 0.1~3mil (2.5~76µm). Spherical or fat elliptical tips might do for some content, or dirt-encrusted scratchy record. But for clean, wide range content, sharper tips sound better. Their stories begin with column headings, L to R:

Stylus mil categorizes spherical, elliptical, and "line-contact" cross-sections by radial dimensions: "tracing"\"scanning" along each wall x "bearing" wall-to-wall, in mils (1/1,000in, =25.4µm);

conWµm is contact width W *along* each groove wall, the groove wall indentation in microns (µm);

conHµm is contact height H (inclined 45º), the indentation *up-down* each wall, in microns (µm);

conAµm² calculates each side's contact area A, indented in the groove wall, in square microns (µm²);

VTFg is the tonearm's set vertical tracking force pressing the stylus into the groove, in grams (g);

press.lb/in² is the share at 45º on each wall of instant pressure from the VTF, in pounds per sq in;

speed is the record speed in rpm, either 33⅓ or 78 [Groove speed about halves across the disk.];

rel.wear estimates a factor for wearing of both stylus and record grooves (due to tip shape, VTF, friction, & speed) relative to the low-wearing D81 line-contact stylus, assigned the value 1.0;

f_3 kHz is the so-called "cutoff" frequency above which a given shaped stylus can no longer trace the inner groove at maximum modulation (volume). A cautionary "speed limit" at the dawn of Hi-Fi. [1]

Perhaps too strong-sounding, *cutoff* is standard engineering terminology for the *onset of signal filtration.* "f_3 cutoff" also characterizes the mechanical vibratory responses of speakers, mics, etc. In the case of the phonograph pickup, filtration is caused by the physical limits of stylus tracing – its "groove curvature overload." Cutoff is the start of descent of a hill, not the bottom of a cliff. At f_3, frequency response is reduced 3dB, causing an audible but fleeting

[1] The LP's 1,970Hz "cutoff" (–3dB point) using a 1mil spherical [Bachman (Goldmark), Columbia 1948]. The best compromise wear v. f_3 is 8.7kHz for 0.2mil line contact [author]. The outer groove f_3 is ~double.

reduction in brightness at highest level. It's not the end of the world – there are 60+ dB to go! In the case of the ultimate inability of a stylus to trace a high frequency groove, high level peaks of brassy trumpets, crashing cymbals, triangles, guitar fuzz, etc. are instantaneous mellowed. It occurs with the innermost cut of a pop album, or the blaring climax of a symphony. Competent mastering engineers make test cutting(s) at the end of a side to ensure no deleterious effects.

In the chart, size data in the previous chart are entered in yellow boxes, the results calculated in green. ***Relative wear*** (compared to a D81, given a "1.0") combines an average coefficient of friction of vinyl, the calculated contact area at each wall and side of a stylus ***conAμm²***, record ***speed***, and the vertical tracking force ***VTFg***. The *VTFg* column are rounded values. Then for any fractional value one actually uses, such as 1¼g or 2¾g (in the book as optimal for SL & Sii styli), relative wear is easily adjusted by linear interpolation. The chart notes whether a stylus might erase a groove's HF peaks, and the doubly wearing friction effects of shellac at 78rpm.

Of the most common interest today in the chart in **braced } green** is the hi-fi "stereo range." Data show a tradeoff of high frequency response v. wear! **In blue are calculated f_3,** "worst case" inner groove at maximum level (double for the outer f_3) that have improved since the mono microgroove 1.0mil spherical stylus introduced with the LP, with its "1.97kHz cutoff" [Goldmark 1948]. Sharpening the side scanning\tracing radius from 1.0 to 0.2 mil (25 to 5μm) would improve –3dB cutoff to about ~9kHz. Still higher frequency content is then achieved by the mastering engineer backing off the cutting modulation, linearly increasing the maximum HF any tip shape can reproduce. E.g. a 6dB lower cutting modulation doubles f_3, but also lowers playback volume and worsens signal-to-noise by 6dB, a compromise to attain –3dB ~18kHz.

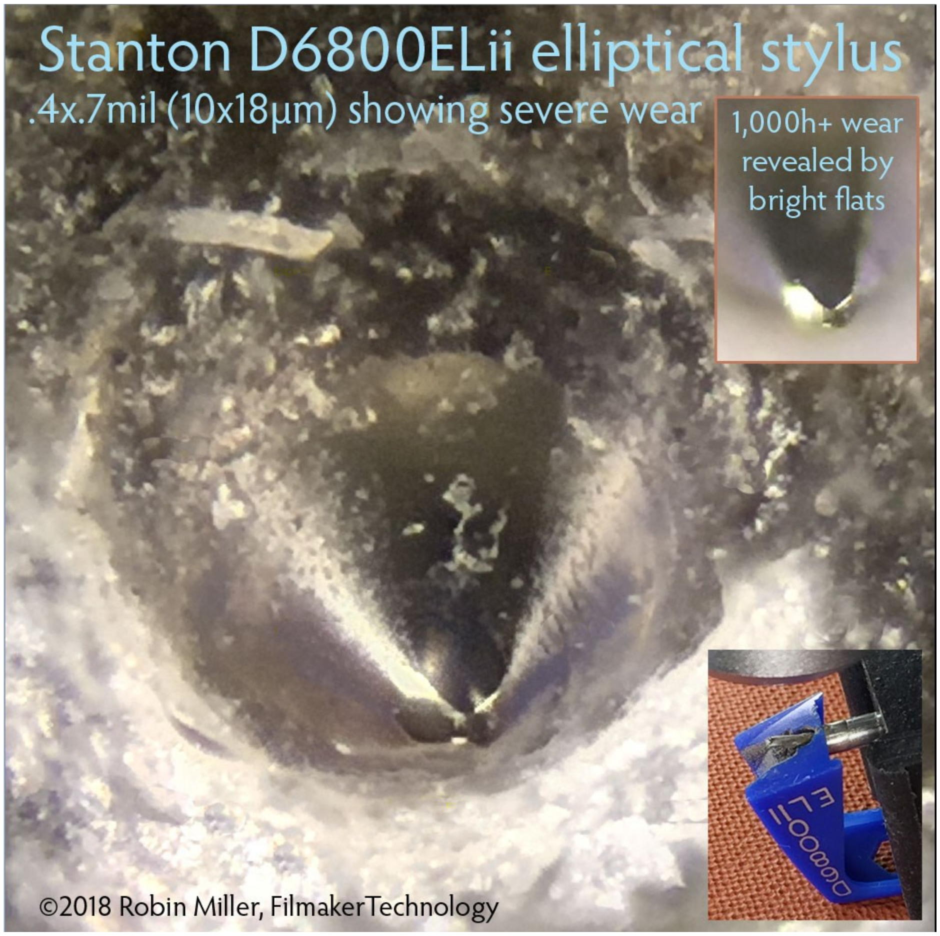
The rightmost columns, **wear in red** v. **f_3 in blue**, are the essence of the chart. As examples: *a)* A 0.3x0.7mil elliptical tracking at 2g is characterized by an onset of attenuation at peak levels of about 6kHz, but has nearly 4 times the rate of groove and stylus wear as a higher compliance line contact with the same cutoff, the D81, tracking near its minimum of 1g. *b)* Also tracking at 1g, a hyper- elliptical 0.2x0.7 that begins to mellow peaks above 8.7kHz has 3 times the rate of wear, but does no harm for rock albums and 45s if they were intentionally rolled-off above that frequency. *c)* The roundest 0.4x0.7 elliptical cuts off above 4.3kHz tracking at 2g with about 3 times the D81's rate of wear (interpolated from 4.6x, listed for 3g), but is fine for piano music that, with its ~7kHz maximum spectrum, will likely not ever be so loud as to test that limit.

Cutoff didn't matter much during the phonograph's first 70yr, with spherical needles on rock-hard shellacs tracking in ounces (1oz =28g)! Hard surface, high friction 78s, with maximum HF evolving from 5 to 7 to 10 kHz, wore out a diamond in 200 hours; popular sapphires in 20! A change of steel needle was advised for every side! But then a 78 side lasted only 3~4½min!

In 1948 at the dawn of the hi-fi microgroove LP, recorded releases were of acoustic music – orchestral, big band jazz, opera – and other "natural" sounds. Natural acoustic sounds normally diminish in energy level above ~2kHz, which suited the original LP slope beginning at 1.97kHz. So they would suffer no ill, diligent engineers made test cuts of a symphony's climax, or of the higher energy of jazz and eventually popular music, then *rode the level* accordingly for that side. Then from 1953, the RIAA standardized pre-emphasis (boosting) of highs above 2kHz, reduced on replay, along with noise, to be flat again. At about half the linear speed of the outer groove, the inner groove crowds the tight turns of high level high frequency undulations, the "curvature overload" manifesting as so-called "sibilance distortion." At frequencies above a tip's inherent f_3, its side tracing\scanning size begins to be too large to fit. There, a too-wide-contacting

 ISBN 979-8218067304 through Ingram supplied stores, or amazon.com/dp/B0BGNF1HVQ

spherical begins to *cut corners,* causing soft vinyl's HF *erasure!* Innermost content can lose brightness permanently in just one pass. The groove's midpoint becomes the playground for high level HF "pinch effect" distortion. These limitations keep mastering engineers alert. High energy outside cuts are not so affected; linear speed is about double, so f_3 is about double.

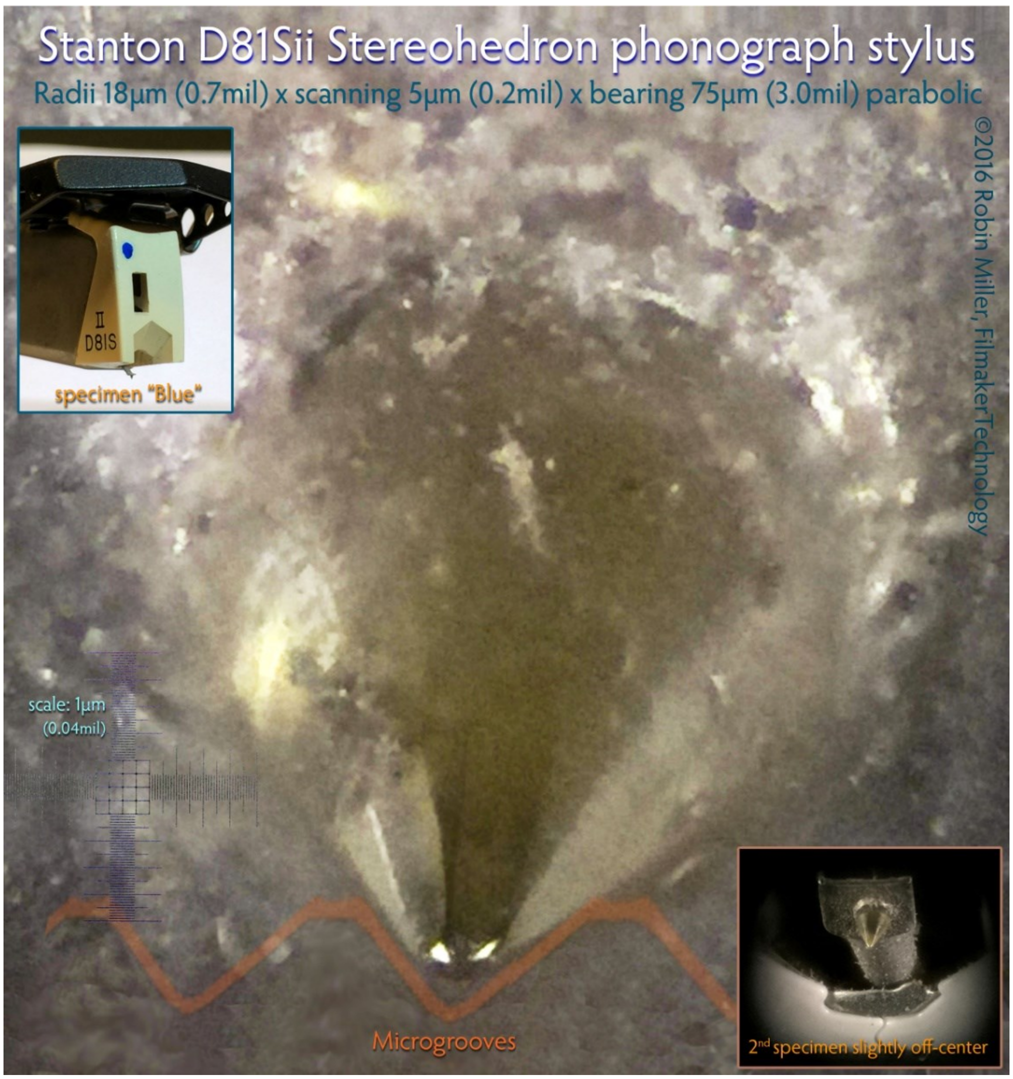


A client's overly worn (>1,000h) fat .4x.7 elliptical shows severe flats at both groove contacts. The sharpened edge around the perimeter of each flat damages records. Its f_3 when new was 4.3kHz. The much broader flat, that rode the outside wall, was due to too much anti-skating.

From the title of its 1975 Patent for an "improved Shibata," Stanton's wear-reference 0.3mil D81 line-contact Stereohedron (or predecessor Pickering D2000) was among the finest of styli, though reasonably priced. Its first was the 0.1mil D4500Q for quadraphonic surround sound that reached 50kHz. Stanton tested each Stereohedron MM 881 and sister MI 681 in order to provide a individual "calibration" sheet upon which a mastering engineer could rely. Many chose it for quality control, spot checking a trial LP being lathed. Why continue the side, wearing their sapphire cutting stylus, if a glitch had occurred? Many LPs during vinyl's heyday had quality assured by the accuracy of reproduction of these pickups. Users playing with the same pickup came closest to the sound intended by the makers of the record. But market forces caused audiofan magazines not to bless them, leading audiophiles to favor more exotic types for $$$. Now, 40+ years after their debut, NOS line-contact Stereohedrons have become scarce.

Sound of the stylus close-up p23c

Sacrificing some wear for even better HF response, Walter Stanton, who had assumed Pickering's management and renamed the company, then introduced a middle-edged 0.2mil line contact. In the Wear & Cutoff table, reducing to 0.2mil (5µm) raised the worst case f_3 high frequency cutoff in inner grooves of an LP to ~9kHz (~18kHz at –9dB). He elongated a bit the radius up each groove wall from 2.8 to 3.0mil (75µm) to make up for some lost contact area. So at the same tracking force, wear is about the same as the fattest 0.4x.7 elliptical. Illustrated in the poster below, the new Stereohedron was dubbed "mark II," or D81Sii.



The poster above, accompanying micro-photos, and frequency response poster on p26 tell a story of two D81Sii specimens acquired by the author as NOS, the slightly better made awarded a blue dot ●. A flat smooth frequency response (FR) indicates that the sound of the measured audio component will likely be accurate in tone color (timbre) – the holy grail of high fidelity. Using a CBS STR140 pink noise test disk, within its limits of 30Hz to 15kHz, the response overall of the complete turntable system with this stylus is within ±½dB, including errors of

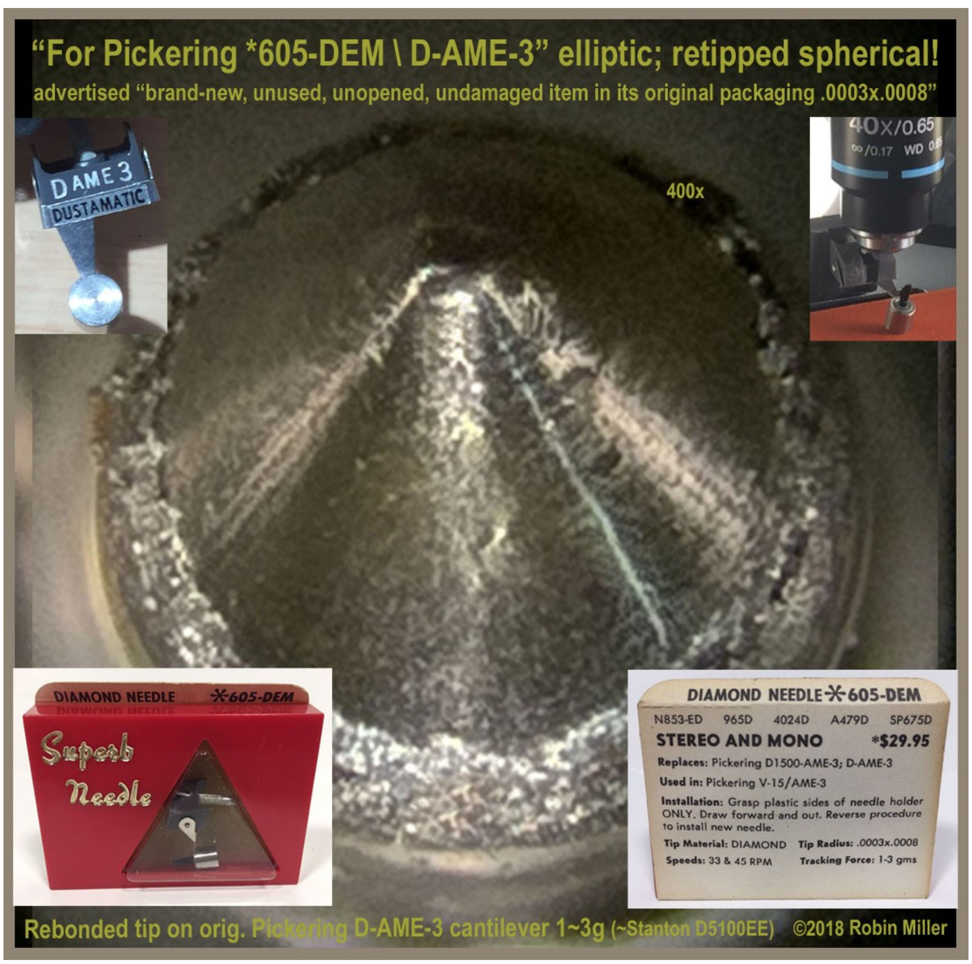
Sound of the stylus close-up […continues on p25 of the book] p24

The aftermarket for generic replacement styli; Pfanstiehl nomenclature – 1st & 2nd ed's.

Audio Technica, Shure, Ortofon, and others are making fine new cartridges and replacement styli of moving magnet (MM) & moving iron (MI) types, along with moving coil (MC) makers. The innards and stylus shapes and their audio quality and wear characteristics are explored in the book by the still pertinent examples set by Norman Pickering and Walter Stanton in their pioneering products for professional and hobbyist use. Cartridge (aka pickup) bodies generally last indefinitely, and can be had used for little money. Unfortunately it is the pricier styli that wear out and need to be replaced every few hundred hours. However highest cost line-contact shapes, which give the best audio and lowest wear, are decreasingly available. So the elliptical 0.003in (7.5μm) has become the common audiophile choice despite its high wear, while lowly sphericals, despite lowest audio quality, remain the most popular for consumers on a budget.

For professionals & audiophiles, Pickering\Stanton new old stock (NOS) today have become scarce and pricey. Just as pricey are JICO's SAS and Expert's ParaTrace line contacts for some pickups and retipping. The internet is home to reputable dealers like EsotericSound, KAB, and Voice of Music. However, many consumers fall prey to unscrupulous resellers of poser needles.

A case in point: a replacement stylus bearing the universal Pfanstiehl catalog # *605-DEM. The * is intended to indicate NOS made by the original equipment manufacturer (OEM) that, if properly stored, is likely good as when made, possibly decades ago. Add a 4, as in 4605-DEM, identifies a generic replacement, whose quality is all over the place. Many stylus replacement makers & sellers use Phanstiehl's system, though it is long out of business. But posers abuse it in bogus labeling. The poster is of a supposedly NOS stylus advertised (eBay) as a "*605-DEM in a brand-new, unused, unopened, undamaged item in its original packaging" with elliptical dimensions ".0003x.0008." Its grip is an original Pickering D-AME-3 for V-15 pickups (or Stanton 500), but the micro-photo shows the cantilever now has a re-bonded and hardly polished spherical, proven by no flats to create an elliptical. Unworn, but NOT the expected purchase!



Discussion related to *The Better Sound of the Phonograph*

There are hundreds of disk formats, phonograph pickups and stylus options, tonearms, turntables, and preamplifiers ("phono stages") to scrutinize in making the sound of the phonograph better. Disks are an unchanging historic record: 140+ years of music, speech, and natural sounds. New and re-releases are a story for later. Today, despite Shure's abandoning its cartridge business and Gibson (owner of Stanton) undergoing reorganization, high-performing turntables with integrated tonearms are being introduced in response to the "vinyl" resurgence. Solid state preamps that approach perfection are now easy to make, so are economical to buy (see the *Phonograph* book's maker project). But as said, to a large extent the "sound of the phonograph is the sound of its stylus." No book could chronicle them all, so the book *The Better Sound of the Phonograph: How come? How-to!* explores them by example. In the hi-fi stereo era, Pickering\Stanton MI & MM pickups were pioneering. Often copied (and copying), the science can be applied to styli made today. But other reproducer types are behind the needle: *ceramic* and *moving coil*…

What about ceramic & crystal phono pickups?

These lower-end solutions were not included in the book because generally they do not meet the criteria of better quality. [Those by Sonotone get some praise on internet audio forums, but this has not been verified by the author.] Though cartridges are light in weight, typically their cantilevers have high moving mass (limits HF and transient response), low compliance, need higher tracking force (that also produces stronger skating), and are not recommended for elliptical tips because of high wear. We'll explore why to put them in perspective by how they compare to magnetic pickups. First to consider is that each works a totally different way.

Ceramic transducers respond to *torque* applied by the stylus tip when shoved by the groove off its centerline. The crystalline element is bent slightly, producing a voltage. The signal varies quite linearly by *how far their position has been displaced from center*. If the groove's "waveform" would remind you of a graphed wave plot or oscilloscope display, that is also how a ceramic output signal looks. The simple ceramic pickup, as well as the ideal groove, exhibits the same amplitude at any frequency, so they are called "constant amplitude" (with frequency). And ceramic pickups have the further advantage of high sensitivity, enough output for the 300mv "line level" input of an amplifier, and, with compromised quality, no need for a preamp! They are cheap to produce, so historically they enabled mass market record players & changers.

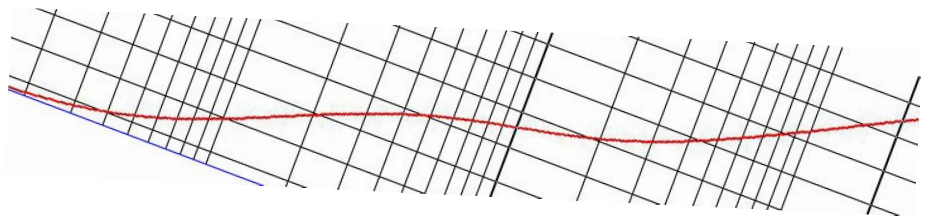
Magnetic transducers – cutter heads as well as cartridges – impart or respond only to stylus' shoving *velocity*. For the cutter, the higher the signal, the faster the motion of the chisel as it crosses the midpoint of the groove. It seems paradoxical, but as the replay stylus decelerates with the groove's swing coming to a stop at each peak, the output of the magnetic cartridge is zero! Both cutting chisel and replay stylus move faster at higher frequencies (HF) because the slopes of the wave are steeper; slower for lower frequencies (LF) because the slopes of the wave are gradual.[1] Sweeping with an electrically flat frequency response through the whole audio spectrum from 20~20,000 cycles per second, 10 octaves ($2^{10,}$ also equal to three decades 10^3), the mechanical swing of the groove while maintaining the same cross-groove speed would *decrease* with frequency by 20dB/decade, for a total reduction of 60dB, or a ratio of 1000:1. We call these devices "constant velocity" (with frequency). Sweeping at the same level, HF grooves swing narrowly, succumbing to dirt and surface noise, while LFs swing so widely they cut into their neighbor grooves! Grooved media would function more simply with "constant amplitude" transducers, like ceramics. However something extra must be done for the ubiquitous "constant velocity" magnetic transducers. The solution was to tilt oppositely the recording and replay filters – "equalization" – standardized by the RIAA 1953 and still in effect.

The RIAA curve and its many predecessor recording characteristics were intended to convert constant velocity of the cutter to some approximation of constant amplitude in the groove, and back again in the pickup. The entire cycle results in flat frequency and phase responses, and

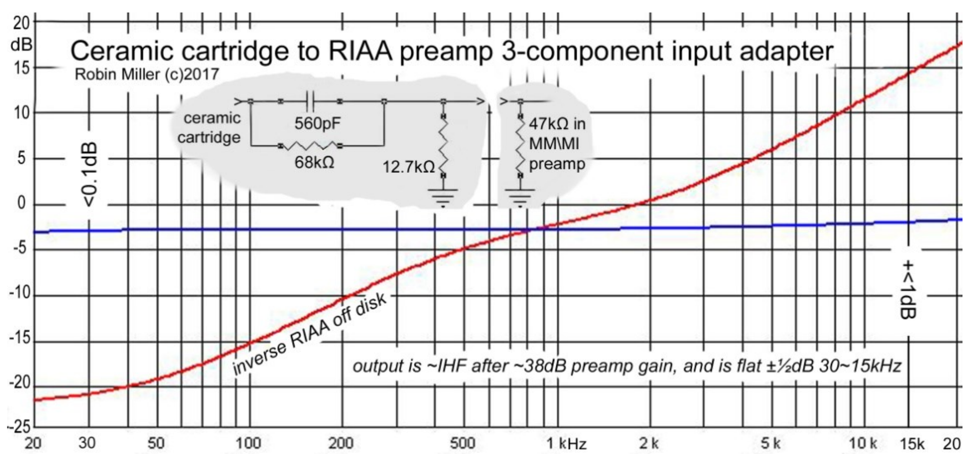
[1] The steepest (HF) slope a stylus can trace is ~45º, which is a determiner of f_3 cutoff for a spherical stylus [Eargle].

other benefits including LF grooves not so wide as to cut into the next groove, and master HF pre-emphasis followed by deemphasis on replay that also reduces scratch and dirt noise. But a 1000:1 slope (60dB) of the characteristic curve is electronically problematic. 100:1 (40dB) is doable. So the RIAA curve is given a flat portion for 2 of the 10 octaves, one octave on either side of 1kHz, that also serves for more readily setting a measurement reference level at 1kHz.

If the groove were cut with a straight line characteristic ascending 60dB, resulting in an ideal constant amplitude groove, then a constant amplitude ceramic pickup would need no playback equalization (EQ) – its natural output would be flat with frequency. But nearly all records from 1954 were cut requiring RIAA filters for replay, so ceramic play is not flat. Manufacturers of inexpensive players saved the cost of EQ filters, feeding the amplifier directly with the ceramic cartridge output, so reproduction is *tone-colored* by the resulting non-flat frequency and phase response errors. Below, tilting the RIAA groove curve enough to approximate a horizontal line, reveals the deviations from flat, audible as distorted timbre of voices & instruments. On a full-range speaker, it sounds weak in the bass, honky in low-mids, lacks presence in high-mids, and is too bright. Deemed acceptable for low-end record players, it falls short in audiophile quality.



An easily made adapter converts the ceramic's constant amplitude output to a quite good-enough approximation of constant velocity response for feeding a common magnetic RIAA preamp. Because the adapter needs to vary attenuation across the spectrum nearly 40dB, the preamp must have about that much gain, which most do. Inserting three parts per channel, the two resistors and one capacitor in the inset below, results in a nicely flat response (**blue line**) at the preamp output, down less than 1/10dB at 30Hz and up less than 1dB at 15kHz. *Voila!*



The MM\MI-recommended R_{load} of 47kΩ at the preamp is included in the circuit in order to interchange between a headshell with a MM\MI pickup and another with a ceramic cartridge with two copies of the adapter (one per channel, in the left cloud above). A mistake would be locating the 6 small parts inside a ceramic's headshell ignoring the 47kΩ load resistors in the preamp (in the right cloud), thus wrecking the adaptor's frequency response. (Without the 47kΩ in the preamp, change the 12.7kΩ resistors to 10kΩ to realize the flat response shown.)

While an interesting, the adapter solution might be impractical. A fine ceramic plus adapters cost about the same as a ubiquitous magnetic pickup. Modern ceramics do have low mass, and replaceable elliptical styli. However the lower compliance of all but the best requires greater tracking force, raising concern for higher wear of both the stylus and groove. Another issue is that if the adapter parts located in a headshell to permit swapping with MM\MI cartridges, then tonearm mass will increase, which affects resonance.

Moving coil (MC) cartridges – darlings of "high-end" stores and online groups

In my own ongoing quest in sound reproduction, I participate (with many grains of salt) in online audio forums and Facebook groups, with dubious rewards for the effort, except for occasional thanks for my help. On one usually respectable one, I read 100+ replies in answer to a troll's question: "What is better, MC or MM?" (Moving Coil v. Moving Magnet phonograph cartridges.) I pictured the original poster (OP) grinning as those practicing more *expectation bias* than science took the bait.[2] The discussion became heated, although no one offered any objective reasoning. And it is NOT because MC pickups are not generally good pickups.

About ¾ of the opinions favored MC. If any reason was given (mostly none was), it was meaningless attributes of "high-end" (interpreted by skeptics as "over-priced"). Or "more detailed" (suspected by engineers as a symptom of distortion that artificially brightens the "color" of the sound of any audio component). Or the ever-delusional "How could [MC] not be the best when mine cost me $$$!" Except for knowing the exact prices, many contributors admitted they didn't know what each technology was. Unmentioned was MI (moving iron). Most echoed purveyor-hype. One condescendingly cast aside the more accepted MM as "drech." Seriously?! It became clear that most posters didn't know what they were talking about, even while trumpeting it loudly. So what's the *real truth* about the esteemed MC?

MC was an early electrical phonograph pickup technology, developed alongside moving coil microphones & loudspeakers. Harnessing electromagnetics, you can move the coil (MC), move the magnet (MM), or simulate either by modulating the flux field (MI). Later in MI and latest in MM types, coils and magnets swapped places between the back-end of the stylus cantilever, or fixed inside the cartridge body. Largely equivalent, there are more minor advantages and disadvantages for each method. In early days of radio, coinciding with the start of the electrical era of the phonograph (mid 1920s), broadcasters had available MC made by Western Electric, Fairchild, and RCA. Later MI & MM took over for their stylus interchangeability to accommodate various record formats and user replaceability, neither of which MC permits. With four delicate wires to deal with, after 500+ hours of play time, the MC pickup must be returned to the factory. Used many hours per day, broadcast styli wore quickly. Vast libraries needed protection from damage by worn styli. So user replacement became much preferred.[3]

Priced at what one dealer proudly calls a "Herculean" $3,700~14,000 and climbing (in a race to the bottom of marginal value) are boutique MC cartridges that some audiophiles recommend with pride, if for no other reason. I've auditioned them; at this price, cost-effectiveness comes into question. And there are disadvantages. Required for MC is an additional 20dB of preamp gain, or an ultra-linear "step-up transformer" (SUT). Either can only degrade audio quality by adding noise, distortion, and cost. Aficionados of MC tout the low moving mass (moment of inertia about the elastomer) they claim traces better the HF intricacies of the groove. True when magnets weighed a ton, but invalid from 1975 with availability of tiny rare-earth rod magnets that fit in the end of the MM's cantilever tube, and are lighter even than the iron slug of an MI.

An MC's shorter cantilever reduces its compliance, requiring higher tracking force, which increases friction, skating, and wear. And an MC's moving mass is effectively greater for the springiness of the four wire leads coming off the MC's coils. With a cantilever length of 6mm (for a Dynavector 20X2), compared to 7.5mm for a typical MM or MI, an MC's stylus tracing

[2] An example of *confirmation bias* is to be convinced (without blind testing) that the three layers of candy corn taste different. Brach's ingredients are: sugar, corn syrup, confectioner's glaze, salt, dextrose, gelatin, sesame oil, artificial **flavor**, honey, Yellow 6, Yellow 5, & Red 3. Only the food die is different. Except if the big layer is brown, then to the author it tastes like chocolate (his favorite). Or does it?

[3] Broadcasters used RCA's popular "light-weight" pickups, tracking 8~12g instead of 28 (1oz), with a moving coil rotated by a bell-crank stylus. Despite requiring factory retipping, they were a mainstay.

path is more of an arc than a straight line across the modulated groove. Cutting loud passages, a groove's maximum displacement is ±1.5mil (38µm). The added tracking error of a MM\MI's 7.5mm cantilever is ±0.3º compared to ±0.4º of a MC's 6mm. 33% higher mis-tracking adds that much even & odd harmonic "sawtooth" distortion by scanning ascents of the groove waveform earlier (faster), then later (slower) turning back to the centerline. Further removed from neutral in sound, the added brightness might be a reason the MC cartridge has found favor by some "ears." But not for whom any non-neutral distortion or coloration is undesirable.

At 1/10 to 1/40 the highest prices for MCs, a "budget" ($329) MC pickup that is respected by users and well reviewed, is shown below with its complete published specifications, *aside its comparison with two popular and very good sounding (~$99) MM\MI pickups…*

Denon DL-30mkII specifications	*v. Shure 97E or Ortofon M2red .3x.7mil ellipticals*
• Output: 0.4mV	*1/10 of a MM\MI; needs "head amp" or Xformer*
• Output impedance: 33ohms	*Definite benefit: needs no special cables or* C_{load}
• Stylus: special elliptical tip	*No dimensions given; and what is "special?"*
• Frequency range: 20Hz-60kHz	*UHF implies "audibly flat;" but it only adds noise*
• Tracking force: 1.2-1.6g	*Bit higher skating & wear than quality MM\MIs*
• Compliance: 13x 10-6cm/dyne	*Low, causes higher arm resonance to be dealt with*
• Weight: 6.0g	*Not an issue, about the same as most MM\MI*

True, measured specifications intend to predict, but don't always prove sound will be good. But omitted data often hides trouble. Missing from this MC's specs are three important ones:

• Frequency response (unspecified)	*E.g. ±≤2dB 20~20kHz ("range" is often –10dB!)*
• Tip profile (unspecified)	*Ellipticals differ at sides from 0.4 to 0.3 to 0.2mil*
• Channel separation (unspecified)	*Better MM\MI units specify 25~30dB at 1kHz*

The MC's extra octave+ to 60kHz ("range" implies *–10dB*) will find no signal in the groove, including Fourier components contributing audibly to phase response, and only reproduces dust pops, vinyl imperfections, and tracing errors that increase audible IMD. User changeable styli enables playing different disk formats – vintage 78s to ETs – as well as replacing worn needles.

But get ready to open your wallets, what's even "better" than moving coil? The exclusive "static coil." These audiophile pickups are, in other words, moving iron (MI)! But unlike most MI or MM, pickups, they have low output requiring transformers or head-amps, and require factory-only tip replacement ($350~600). And as one acquaintance has suffered, they have delicate unprotected cantilevers that are prone to damage, luckily included in that re-tipping fee.

The book has much evidence to refute the provocative "drech" comment above about the sound qualities of practical MM\MI pickups for audiophiles who are not of the conspicuous consumption contingent. For me, any audiophile using the adjective "high-end" reflexively raises suspicions of bias if not fraud. In these cases, the price-value curve for audio products has been inverted. Perhaps it is *liking what you own,* especially if you paid dearly for it? Pure psychological "confirmation bias?" Or judgment strongly influenced by visual coolness?

More ordinary audiophiles, less granfalloon-ian[4] music lovers who have other compelling needs in their budgets, will take comfort in the fact that many disk mastering engineers chose particular MM or MI types for quality control while cutting. Why did *they* not embrace MC over MM\MI? Stylus field-replacement might have been a reason: the nuisance, wait-time, and expense of mailing pickups back to the factory for a re-tip. But regarding audio quality, these professional savvy & reputable engineers would certainly have practiced *no compromise*.

I do not disparage the MC – they were early pickups, and many are good – but they might not be as omnipotent as some would have us believe. Do the dollars make *sense?* It's possible (not

[4] A group sharing identity (belonging) or purpose that is meaningless – Kurt Vonnegut in *Cat's Cradle*.

recommended) to adapt the book's DIY preamp for MC, adding 20dB gain.[5] This book is intended for the less nonsense hobbyist, whether on a budget or not. This was hi-fi of the 1950s, when the author and his contemporaries were bit by the bug. When fan magazines presented honest performance data in reviews that used numbers instead of marketing rhetoric. That catered to DIY rather than what-to-buy. No high-end solution can magically make up for poor recordings, so wasted are expenditures beyond extracting what's in your best disk's groove.

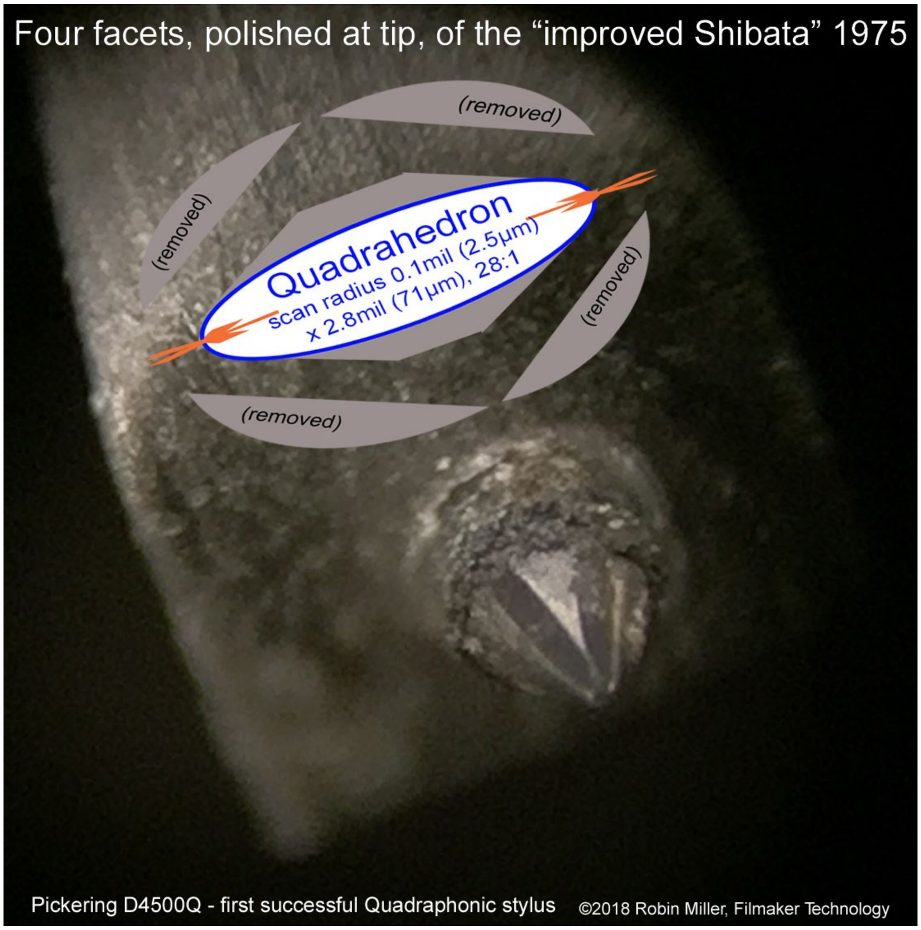
The evolution that led European audio magazines to proclaim it "the finest cartridge"

A big step in phonograph quality is illustrated by the next three micro-photo posters. It came about to solve the high frequency challenges of quadraphonic surround sound on disk, called "CD-4" although it came a decade before the compact disc. Quad required response to 50kHz for its second pair of signals, completing, like 5.1, an enveloping circle around the listener.

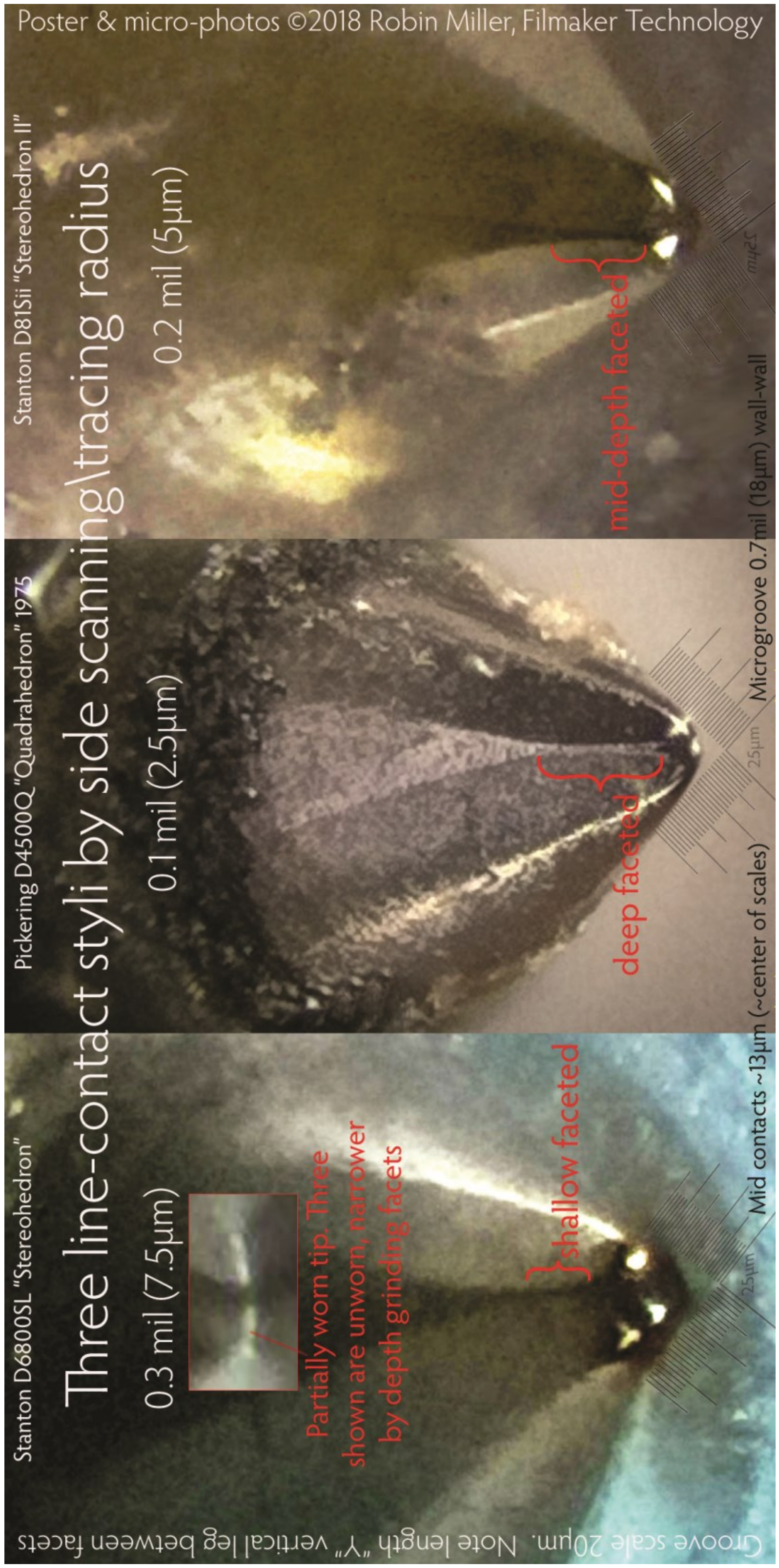
The solution was a strong thin-walled, very low mass aircraft aluminum cantilever sporting a super-narrow line contact tip at one end, and a powerful rare earth magnet at the other. Just in time for Quad's demise in the marketplace, the most successful implementation was Pickering's D4500Q "Quadrahedron" stylus in a new moving magnet body. The cantilever mass needed to be low; its tiny rod of magnetically powerful Samarium-Cobalt was harnessed for the first time to produce the high level of a moving magnet. It tracked as low as ½ gram, unlike predecessors that inflicted wear that "erased" ultrasonic nooks & crannies in Quad's groove after a few plays.

The following posters show the historic Quadrahedron stylus that Pickering debuted in 1975, acquired by the author in 2018 as NOS. It is unworn. The very narrow contact area 0.8x8.4µm approximates a line 10+ times as long as it is wide. Tracking at 1g (range ½~1½g), its relative wear compares to a stereo 0.7 x 0.7mil spherical, hyper-elliptical, or DJ Stereohedron. However its ultimate f_3 is 17kHz, 5 times the spherical's. If the stylus is used for base-band stereo, this results in ~1/5 the tracing forms of distortion at HF by the inner groove. This stylus is long out of production, and now quite scarce. Its use for playing records is limited to pristine archiving. But the Quadrahedron would lead to adaption for stereo in 0.3 and then 0.2mil Stereohedrons.

[5] Decrease to 1/10x four resistors 1L, 1R, 2L, & 2R. Increase to 10x capacitors CGL, CGR and parallel film Cs. Opamp OPA2134PA will be too noisy, so substitute a LM4562NA or equivalent LME49720.


Four facets, polished at tip, of the “improved Shibata” 1975
(removed)
(removed)
Quadrahedron
scan radius 0.1mil (2.5µm)
x 2.8mil (71µm), 28:1
(removed)
(removed)
Pickering D4500Q - first successful Quadraphonic stylus
©2018 Robin Miller, Filmaker Technology

After Quad's demise, with less aggressive tip grinding, the Quadrahedron's wonders improved stereo reproduction in the successful Pickering & Stanton Stereohedrons, explored in the book. The following poster illustrates this evolution: Shibata at JVC had invented the line-contact shape (1971) that more closely emulates that of the cutting chisel in order to implement quadraphonic surround sound on a phonograph disk. However the tracking force to keep the stylus in the ultrasonic groove caused high wear, and HF erasure. So in 1975, Walter Stanton hired a consultant to develop the "improve Shibata" that significantly reduced VTF, simplified fabrication, and reduced costs. Pickering dubbed it a *Quadrahedron,* the author's NOS D4500Q specimen in the middle. Its deeply ground facets, indicated by the vertical leg of the "Y" shape between facets, result in very narrow side contact areas that extend f_3 to 17kHz, and the –10dB response to 50kHz (worst cases). Although this 0.1mil (2.5µm)-tracing D4500Q debuted just as Quad's short life was ending, it was repurposed for stereo the following year as a Stereohedron 0.3mil (7.5µm), shown at left, that is a touchstone of the book. Later, at right, came the in-between Stereohedron II tracing 0.2mil (5µm) along the groove. Others' line-contact variants followed. The relative wear and f_3 "cutoff" frequencies are in the updated table p5 (book p23).



Depending on the HF energy of a disk's content, very narrow shapes deliver flattest response and lowest "sibilance" distortion (a price can be more clicks & pops). Perhaps not needed for a soft piano album, line-contact styli are unsurpassed tracing raucus guitars or violins. Together with the tiny rare-earth magnet at the opposite end of a light cantilever, this achievement in disk sound drew praise in Europe beginning 1976 for Pickering\Stanton models XSV-3000~7500, 880\1S~980, WOS-100\CS-100. But marketing through discounters caused backlash in the US.

As has been lamented, these are no longer manufactured, and as NOS are scarce. However the legacy exists in Expert Stylus' Paratrace, new or as a re-tipping service, in the JICO SAS, in the Audio-Technica Micro-Line, and in others, although the author has not tried them all. Most new cartridge suppliers today compromise challenging sounds by using a less costly elliptical.

A brief history of sound reproduction

Nearly a century ago in the early 1920s, with the application of the triode vacuum tube as an amplifier, "active electronics" offering signal voltage gain dramatically changed sound reproduction. The prior era had been mechanical and "passive" electrical reproduction – the telephone of Alexander Graham Bell, the phonograph of Thomas Alva Edison, an improved carbon microphone for the telephone by Edison, etc. These pioneer methods involved only *transducers,* converting one form of energy into another, with no energy gain, only losses in the process. So public address (PA), cinema sound, and radio broadcasting were not yet possible.

Mostly these inventors were independent tinkerers rather than academically trained engineers. The former were driven, if not by sheer curiosity, by anticipating needs – useful applications – for their inventions. They were more marketers than scientists. Looking for ways to fill needs by trial & error. Harnessing electronics in the mid 1920s, new audio inventors were engineers, most working for large organizations: Bell Laboratories, GE, Western Electric, or RCA. A few independents bridged the gap: Edwin Armstrong and Nikola Tesla, whose inventions were purloined by predatory corporations with more resources and legal beagles. In this transition, applying elbow grease evolved into *applied science,* the very definition of engineering.

Consumer recorded entertainment from the late 1800s to the mid 1920s had consisted of the player piano, the acoustic phonograph, and the kinetoscope – all *mechanical* sound or motion picture devices (not yet both). They foretold what the public would demand in succeeding decades: sound and motion picture entertainment with faster delivery and higher quality. First, about 1922, was Radio – wireless *broadcasting,* one transmitter to many receivers, of music & speech. Content was adapted from prior media: News reportage written for the ear. "Theater of the imagination" requiring only dialogue and sound effects, the listening audience supplying the "visuals." The comedic skits and live audience laughs of vaudeville. Music programs by orchestras, big bands, and popular jazz singers were broadcast live, as recording was not yet broadcast quality. Expanding the sizes of auditoriums and sports stadiums was aided by public address systems. Unknown by the public using the single, visible, unchanging instrument in their homes was how extensively the telephone was improving, hidden behind the scenes, and enabling its long distance communication, by active electronic vacuum tube (valve) technology.

In Depression-proof industries (Bell Telephone, Western Electric, RCA, GE), the 1930s was a great decade for the development of audio technology. Furthering the discoveries in psycho-acoustics – the study of human auditory perception begun in the 19th Century by Lord Rayleigh in England and Helmholtz in Germany. From then into the 1950s, audio hardware was essentially the same for telephony, radio, recording, and sound motion pictures. In these four worlds, manufacturers' R&D units developed the same amplifiers, microphones, mixing consoles, loudspeakers, phonograph pickups, etc. A few devices were application-specific: transmitters for radio; optical sound recorders and scanners (players) for film. Early "talkies" borrowed from radio the same "transcription" phonograph equipment, playing 16in disks in sync with 35mm film projectors. There was little need for differentiation in audio genre until its performance could surpass telephone quality. Followed quickly by huge public taste for it.

Western Electric telephone-like components and 16in transcription turntables were cobbled together by a club for a 1923 radio station, featured in the author's book "American Radio Then & Now: Stores of Local Radio from The Golden Age."

What separated these audio genre and their supporting industries into specialties was a result of evolving demand for higher fidelity that in time differed profoundly for each. Limited to the frequency range 250 to 3kHz, early telephones were capable of 90% *understanding of speech*, termed "articulation" [Fletcher, Steinberg 1929]. Limited as this technology was, it was "good enough" for speech communication. However adding long wires for long distance, it couldn't muster good-enough quality without installing along the way expensive cascading repeater amplifiers and equalizers, demanded (and paid for) by the radio networks to get decent sound to their affiliate stations. Today, cellular telephone *fidelity* is unchanged, and its utility is worse by the savings of "full duplex" operation, where conversations can include natural interruptions. More awkward for users today, who nearly need to say "over" to hand-off to the other party!

Having no wire-mile limitations, voice public address (PA) systems (not yet "needing" modern rock concert's excruciatingly loud capabilities) achieved 98% articulation by broadening the frequency range by an octave to 7kHz, mostly still limited by the transducers. For PA and the cinema, amplifiers could generate 25w or more! And together with much more efficient and therefore much louder loudspeakers, performed well beyond what was needed at home, for broadcast monitors, or for telephony. PA mixers required the same frequency range and low distortion (~1%) as broadcast, but did not need radio's *build quality* for 24hr operation, nor to be maintainable on short notice. The imprimatur "broadcast quality" was extended to turntables and other devices for the high reliability & maintainability needed even more than high fidelity by radio networks and stations, and later for television. For consumers, reliability & maintainability were not critical, so consumer-grade equipment became cheap and disposable.

Designed for *zero-defects* in long-term (20+yr!) operation, PA and broadcast mixing desks only needed to be the "switch & fade" type, akin to telephone switchgear. Usually they had no in-board filters, equalizers, or compressors. Small radio stations had almost all their audio electronics contained in a 135lb steam locker-shaped control "board," with only an external limiter "patched" in on the way to the transmitter, until marketing in the 1970s of the Harris 110 hybrid broadcast\recording console made by Auditronics of Memphis.[6] Beginning in the 1960s, mixers for records and movies had evolved to build these audio processing devices into each input, and to have state-of-the-art performance. A dozen cascading stages of tube amplifiers and effects needed to perform to 0.1% distortion to add up to 1.0%. Today they are an order of magnitude (10 times) better still or more. More complex, often experimental and custom built,

[6] Welton Jetton made consoles to record Elvis, and hybrid model 310 for larger stations and John Candy's home studio!

more maintenance was expected, but it could be attended to more leisurely than for broadcast. In the hands of competent operators, records made from 1960~80 exhibited highest quality, exceeding the fidelity of broadcasts they were played over. Played properly, vinyl from this era often sounds better than today's technically superior digital media due largely to recording engineers whose achievements were made in spite of remaining artifacts of their "vintage" gear.

Chasing *verisimilitude,* and despite the old-fashioned mechanics still involved, phonograph records bested FM and pre-recorded tape in delivering to consumers' ears the quality sounds from professional quality microphones, mixing consoles, monitor speakers, and tape machines. Master disk cutting lathes gained variable groove pitch & depth, heated styli, then 2-channel stereo in a single groove. For 34 years from 1948 to 1982, the LP reigned. With the "hi-fi" hobby movement begun in the 1950s, millions of consumers sought life-like quality listening to music on disks. At first optimizing monophonic replay quality, then stereo from 1958, mid 20th century "audiofans" were more quality conscious DIYers than well-healed showoffs. They built HeathKits and others from EICO, Knight, and DynaCo who offered DIY preamps to amplifiers to tuners to tape decks to test equipment. Their labor saved ¼~⅓ the purchase cost. The book's instructions for two DIY projects, and vinyl's resurgence itself, harken back to those times.

Beyond raw sound reproduction quality, psychoacoustic verisimilitude required the more lifelike *spatiality* of stereophonic reproduction. Spatiality conveys to the listener perceptible "cues" for *localization* of auditory images, including moving sources. Stereophonic ("solid sound") utilizes a minimum of two channels developed by Blumlein at EMI in 1931, three channels championed by Bell Labs also in the '30s, four channels for Disney's "Fantasia (I)" in 1939 and matrix surround in cinemas into the 1990s, and in the 21st century, five to seven main channels plus Low Frequency Enhancement (LFE) 0.1 channel that introduced the "subwoofer" for both cinemas and home theaters. All these are binaural (2-eared) based systems. For 2-channel stereo, speakers share a horizontal equilateral triangle with the listener; for surround, a horizontal circle around and behind a listener. All mentioned so far are spatially 2D: left-right, near-far, plus for surround, front-back. The best 2D illusion the author has experienced is wave-field synthesis (WFS) with 40~800+ channels! [de Vries, Delft University]. However owing to our outer ears (pinna), human hearing is also able to detect up-down – a full *3D sphere* of auditory space – implying speakers above and below the listener as well as around and behind. This audible *virtual reality* requires at least 8 speakers for Ambisonics (lacks interaural time cues, ITD due to mic coincidence), or the 10, 14, or 26 speakers for Ambisonics+Ambiophonics (includes ITD), a hybrid system championed by Ralph Glasgal, Angelo Farina, and the author (Patented as High Sonic Definition 3D, HSD-3D), as demonstrated in the author's Bethlehem studio\lab and the House of Music at the University of Parma, Italy. Other systems are the hemisphere of Dolby's Atmos, 22.2 of Hamasaki in Japan, and IMAX with its single speaker above the screen, which are all in fact "vestigial" 3D, aka "2½D" despite any marketing hype.

Calling conventional 2-speaker stereo "3D" is bogus. Any up-down *immersive* sounds heard are only reflections supplied by, and the signature of, the *listening space.* They're not recorded, and do not represent acoustical contributions to the performance. For acoustic music, reflected-reverberant energy exceeds direct sound, especially for the 3D sphere of sound. If captured live, recorded early reflections & reverberation were extensions of each performer's instrument, of how s\he created tone color in conjunction with that space, even the tempo chosen.

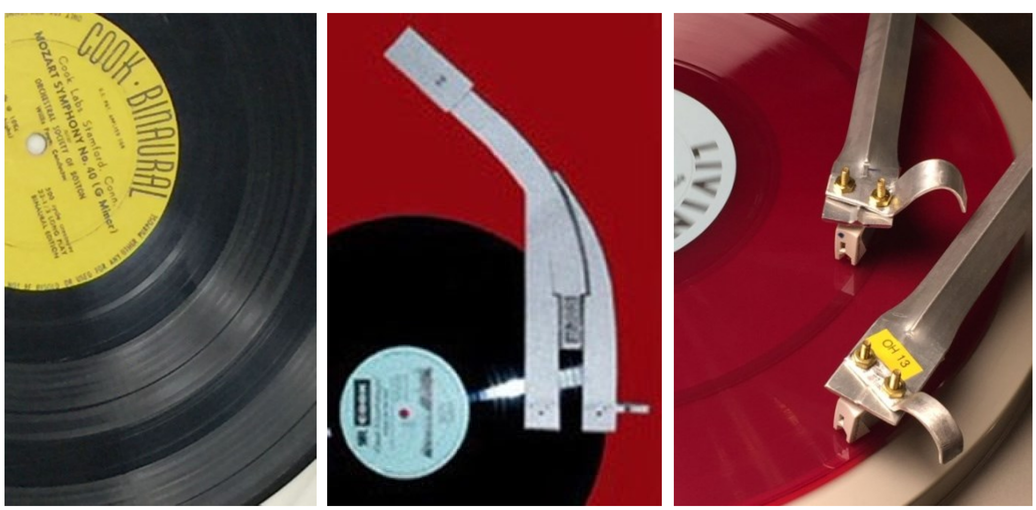
On the other hand, listening room acoustic cues are unchanging for all recordings, imparting a "they are here" intimacy to close-, multi-mic'd popular music. Acoustic classical and jazz sonically transport listeners from their homes and cars to the recording space, as though "you are there." The acoustic signatures of large concert spaces easily trump one's small listening space. As in natural hearing, height is detected by the highly individual pinna of each listener. In replay, either listeners' own pinna for HSD-3D, or simulated for a single listener using a custom dummy head mic with prosthetics molded from his\her own [Choueiri 2013]. Like unique fingerprints or retinal scans, individualized 3D cues work for no one else. Attempts to generalize pinna effects work for only about 25% of the population (i.e. fail for 75% of us).

Still largely in the experimental stages, next-generation 3D audio reproduction systems are not yet available to consumers, and have only been heard by few, mostly audio engineers, although the author's work has involved statistical studies with musicians, audiophiles, and students of virtual reality and gaming (Lehigh University's Integrated Business\Engineering program, NYU grad students using the author's PanAmbiophone). VR requires capturing (or simulating) and replaying both the direct sounds of a *seen* source AND, from any direction in 3-space, the unseen reflections of the recording space (grand concert hall, spooky cave, etc) that contribute more to perceived timbre (tone color). HSD-3D test subjects comparing live sounds to 3D reproduction universally declare it as much more realistic than anything they'd heard previously. But this is not the same as the "wow-factor" experience of Edison's audience in 1915, that pitted live performers v. a phonograph replay of them. Edison's trick was to hire professional musicians to *imitate* the tinny, nasal, volume-limited sound of the machine! It remains unknown whether the audience was taken in, applauding the miraculous machine, or got the con, and was acknowledging the talent of the singer to duplicate its sound so precisely.

Edison's marketing "tone tests" were monophonic (1D near-far, but not very!). But even if in 1915, and for a decade of demonstrations in many towns, Edison had exceeded expectations for sound reproduction, neither he nor HSD-3D proves recordings could ever be indistinguishable from live. And after all, Audiophiles generally are content with the flat pie shape of 60º stereo, which imparts minimal listener envelopment (LEV) compared to 2D Ambiophonics, the envelopment of 5.1~7.1 surround, or the immersion of full-sphere 3D – methods that increasingly preserve the colors of sonic arrivals from all around let alone above\ below. Their only "3D" is the sameness of their (acoustically untreated?) listening room. But recorded music nowadays is about creating unreal sounds *that never were performed live.* As in Edison's demos, today's performers in concert imitate their recordings, not the other way around.

The very first stereo records 1952, before stereo in a single groove

Lest we forget, the first stereo records did not wait for the stereo groove introduced in 1958. Emory Cook new it was coming, after all it had been invented and Patented by Alan Blumlein of EMI in the 1931! But in 1952, Cook saw an opportunity to distribute stereo recordings on LP disks with dual monophonic grooves made by strapping two cutter heads together. "Cook Binaural" releases – about 49 in all – were distributed on his own label, on Atlantic records, and on the Livingston label of the manufacturer of the twin-headed tonearm needed to play them.



The Cook "Binaural" LP with major tracks (2) for the stereo channels – the outer is L, the inner R. Cook partnered with Livingston to make the twin-head tonearm that was finicky, but worked. The author uses two 12in transcription tonearms, DIY instructions for which are in the book.

These were finicky to play, as they required precise 1-11/16in separation of the styli, and just as precise relative adjustment of the styli front to back, so the tracks would play in synchronism or there would be a time difference between the channels – a *phase error* audibly shifting the stereo image. A test disk to make these adjustments used clicks to match L-to-R timing. Furthermore, Cook's 2-channel preamp was needed because he chose differing recording characteristics. Its limited success was due to these inconveniences, which were all solved by the 45\45 stereo groove. Nevertheless for six years the Livingston arm was sold, and Cook disks have a place in history as the first publicly available stereo in the hi-fi LP format.

When The Golden Age of Hi-Fi ended – and will there be a 2nd?

In the mid-1960s, music recording entered a paradigm shift away from studios owned by the big labels to independent ownership. Those owners (about 800 USA-wide) were mostly recording engineers. Previously unsung and uncredited by the record labels, they were known mainly to each other, often helping each other behind the label executive's back, as problem-solving engineers in any field often do. But now as entrepreneurs, they relied on autonomy, keeping their innovations secret, and seeking credit for their work for needed self-promotion. They remortgaged their homes to buy studio equipment packages, including mixing consoles and tape machines. In the fickle minds of uninformed clients – label executives and garage bands studying only the trade magazines – hits became associated with the specific makes & models of equipment used, which became perceived as magical – still true today. In reality studio time is equipment rental, so in order to sell clients recording services, particular gear went viral to become *de facto*. A new trend was spending lots more time in the studio, editing, & mixing that fed the till that repaid the bank, maybe with a little something left over for the studio owner. [Author interviews with pioneer studio owner-engineer Clair Dwight Krepps]

Most of these indie owners were more technical engineers who relied on lab instruments to calibrate recording equipment for highest audio quality. But about this same time, the role of "recording engineer" spun off the function of "music mixer" ("tonmeister," console operator) who used his\her ears to adjust the sound. Now that seat's occupant was the producer's (recording director's) favorite freelance mixer – sometimes him\herself, or a volunteer from the band. It was and is human nature to value your own contribution more than others'. And as fewer knew or appreciated what the engineer did behind the scenes, the mixer visibly twiddling the knobs became regarded as a *wunderkind.* However, working in a different control room every day, largely lost was the ability to *hear through* the acoustic idiosyncrasies of the room, errors familiar to and therefore able to be compensated for by the owner-engineer. Monitor speakers of the day were also far from neutral, so the mixer's decisions, now baked in the mix, were tainted by colorations (distortions) unique to each control environment. Add adjustments during disk mastering and errors in pressing, anomalies went on to be set in groove "concrete."

Separately, an engineer and mixer not only have different personas, each approaches the music differently. Engineers were inclined to capture as well (neutrally) as possible the music, speech, and other sounds as they sounded live. High fidelity implied replay would be accurate, the recording *transparent.* Using test tapes and instruments, engineers aligned tape machines for flattest response, set their bias for lowest distortion, tended to grounding for lowest hum, and positioned microphones by individual directional and frequency characteristics. They poured over measured performance data, and evaluated gear in rigorous "shoot-outs" before committing to buy. Tops at this were acoustic music and film sound engineers. They prepared the studio and control room ahead of the client's arrival, maximizing everyone's productivity, including the studio's. Suddenly the trend was to reduce productivity by expenditure of lots more time. A mixer showed up just in time, so setup was on the clock. Players' performances were captured, often individually in many "takes" accumulated on 24-track tape machines (or 48 synchronizing two). Editing evolved from labor intensive physical splicing to semi-automated punching in & out of individual tracks, rinse & repeat by trial & error, to correct every little fluff of an instrument on its own. Performances by isolated instrumentalists became sterile, less musical. Mixdowns were complicated, racking up more clock time than sessions. When digital audio workstations (DAW) arrived, more was automated, but savings were turned into even more tracks, takes, and mixing. The loudest sound in the studio became the cash register's cha-ching!

This escalating process was under the baton of the mixer\tonmeister. He\she might know as much as the producer and musicians about technical matters – often nothing! His\her choices were based solely on his\her ears – how it sounded to him\her. Matching a mic with its non-ideal sonic coloration to an instrument that seemed *better* with it. Not by study of the mic's characteristics, but by trial, error, i.e. learning the hard way. Treating abnormal sound not as trouble to be fixed, but as a novel sound that might make this record stand out to become the next hit. Sitting at the console, often with little understanding what the controls did, tweaking them randomly or from memory, until said new sound emerged. The time was justified to the accountants as *the creative process*. At length, some of this turned out to be musically good. But even if only outrageous, the mixer had made "art," or even better – a saleable product.

By the late 1960s few producers possessed the talents of George Martin (The Beatles), a classically trained composer\arranger. Among many lesser imitators, what occurred more often was ricocheting off the studio walls (fueled by mind-expanding substances & mind-numbing liquids) that finally resulted in over-processed, over-the-top sound that the label executives noticed wide-eyed, like they envisioned record buyers would feel in stores. Sound not like live; sound by accident, unique to a recording studio. Accuracy\neutrality\transparency was out the window in the new era of high *infidelity.* Artificial though it might be, if it was new, or loud, it might get attention, and sell. Eventually instead of records imitating the excitement and musical energy of a live concert, fans began to expect concerts to sound like recordings. With ever more capable audio processing technologies used both in the studio and on tour, live concerts *could* sound like studio recordings. PA consoles today emulate recording ones, with duplicate effects for instruments and voices. While the band plays along live to click-tracks in their in-ear monitors, should a vocalist forget words or not wish to risk a high note, he\she simply mouths it without emitting a sound, and the PA system automatically switches to a recorded track. Adoring fans are none the wiser, and are satisfied in the value received for the price of a ticket.

A stadium of thousands see stacks of prop guitar amps, but all they hear is the PA. They may have paid $100 mostly to hear a recording played loud.

The sound on the recording, and now heard live in concert, exemplifies the current state of audio, one where both studio monitor and PA peak sound pressure levels up to 115dB-SPL are louder than prior studio standards of ≤105dB-SPL, and way higher than typical at home ≤95dB-SPL. However, mixing decisions based on abnormally loud monitoring levels lead to weak-sounding bass when played at much lower levels at home [Fletcher-Munson 1933]. Concert sound is compromised by non-linearity of air in the throats of horn drivers, comb filtering from individual singers and instruments heard over multiple loudspeakers, over-loud bass, and brittle high frequencies boosted by near-deaf operators. Savvy road crew and audience members wear ear plugs to reduce the inordinate level to normal, sparing a day of ringing in the ears, if not permanent hearing damage, while enabling understanding the lyrics. In a workplace, this SPL would trigger an OSHA fine; here it has set a *new normal* for wedding DJs, boom-boxes, and ear buds. In cities feeding the crescendo of noise competing with truck\bus air-brake farts.

Will hi-fi generally, or the phonograph specifically, have a 2nd Golden Age? I *can* imagine it, possibly as more re-discover good sound, especially new music-lovers who don't yet require *deconditioning.* Necessities foster invention – think enjoyable music fostering good sound – tend to stick. But Needs posing as Wants born as Fads come and go. So we still need buggy whips, just not nearly as many. With the exception of 9ft concert grands, piano sales that topped consumer entertainment a century ago have dropped precipitously. Electric guitars sell like hotcakes. (Ukuleles are enjoying a comeback!) Radio is in its long tail of demise, ruined by sameness, loss of *localism,* and its own audio over-processing; but television is enjoying a 2nd Golden Age, empowered by wide-screen high definition and positive creative competition. Most people I've met don't (yet?) feel the need for better sound, even if they have felt the need a bigger, better TV picture. But in individually-minded, civilized societies, the majority doesn't rule what the rest like or do. Even if its resurgence does not trigger a mass 2nd Golden Age of Records, the author would have it foster better sound in any medium for those who appreciate it.

Reference-quality sound, like a car, starts in neutral

Components of any audio system are each links in a chain, broken at the weakest. Many audiophiles approach strengthening an audio chain backwards. Continually dissatisfied with their system, they, without controlling any variables, randomly swap components by trial & error, and only by listening (i.e. little research and no measurement). This practice – less about science and more about shopping – has redefined the hobby. The inclination is not "How do I fix it?" but "What can I *buy* to fix it?" Advice is heeded from others naively doing the same thing rather than seeking proven experts. The book reveals a common mistake by audiophiles and magazine reviewers who declare a cartridge "too bright" or "too dull" when the capacitive loading of the cartridge has been ignored – a user responsibility. If the tail-chasing has led to more pleasurable sound, it might be because a change just happened to introduce an error that compensated for the opposite error in another link in the chain. [7]

In a typical domestic listening room, a speaker with narrow dispersion might have good sound for anyone listening along its axis. However the sound is dull everywhere else in the room. Move a bit aside stereo's "sweet spot" the sound from one speaker is bright, the other dull, the "soundstage" shifted, unfocused. Change to wider dispersion speakers and the quality improves for every one. But then, close proximity to reflective walls inserts an acoustic "graphic equalizer" of *comb-filtering* that alters tone colors, and causes fatigue. Finally, even as anomalies are compensated for in the brain to some degree once it's learned the space's "acoustic signature," judgment of sound might inexplicably change with each recording.

Recorded low frequencies below the *Schroeder frequency* of a room (typically 250Hz, about middle C!) roughen the frequency response one hears due to cancellations & reinforcements of standing waves, varying with the positions with respect to reflective surfaces of loudspeakers and a listener. Bumps and dips change with small re-positionings of either, even the constant tiny head movements we employ unconsciously to sharpen perceived localization, whether listening live or recorded. Effects are greatest at very low frequencies (VLF <100Hz) – a good argument for subwoofers in positions that are more ideal than where the main speaker woofers wind up. Correcting these moving targets by readjusting other components can make for a hobbyist's fun pass-time – or frustrating tail-chasing better spent enjoying your music!

But the next ill-researched component change might start the process over from the beginning, casting aspersions on a formerly acceptable component that still is not to blame. After a researched plan, the recommended order for optimizing phonograph replay is as follows:

7 Haphazardly boosting frequencies in one component to offset a dip in another can escalate, as when a client blew out his $$$ amplifier cranking the treble after his speakers blew ¢¢ tweeter protection fuses.

1. A decent new or used turntable (low rumble, wow & flutter) & tonearm [$30 DIY project in the book] with a good cartridge and best needle (0.3mil elliptical, ideally a line-contact), and properly setup\aligned, as detailed in "The Better Sound of the Phonograph" book;

2. Flat RIAA "phono stage" with low noise, calculated C-load selection, adjusted channel balance, and proper mixing for lowest distortion mono. [$30 DIY project in the book];

3. Flexible but not "feature-laden" (difficult to operate) control preamplifier; integrated with power amplifiers as a receiver, or with separate power amplifiers, typically 100+ watts per channel if solid state, both channels driven (requiring a "stiff" power supply), flat response 20~20kHz, <0.1% THD distortion across that range, and <0.1% IMD;

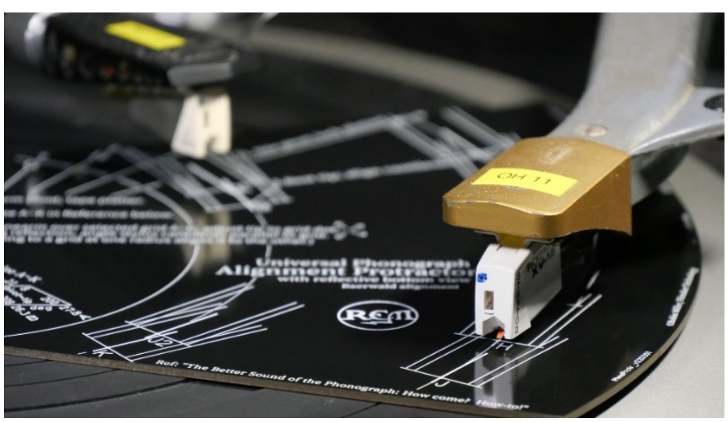
4. Loudspeakers at least 2- if not 3-way, with good phase through the crossover network, honest response in-room within ±3dB 50~12,500Hz (may require a subwoofer), low distortion, non-ringing impulse response, 60+deg dispersion horizontally over the full frequency range for good power response in-room. *Position each speaker no closer than 3ft (1m) from any reflective wall to avoid harsh sounding & fatiguing comb-filtering;*

5. Listening room acoustics treated with absorption, diffusion, bass traps, etc. A subject well beyond the scope of this paper; generally the *deader* and more *diffuse* reflections the better.

Don't forget tonearm, stylus, & cantilever alignment

"Alignment" means setting up a turntable for minimum tracking distortion. This is different from tracing\scanning distortion (groove "curvature overload") caused by the stylus tip not fitting the tight undulations of the groove, especially the inner grove at high level high frequencies. With mis-tracking, the stylus tip is not perpendicular to the modulation in the groove walls, causing more distortion, smearing and of transients and phasing between channels (that mix poorly in mono), and wear of both stylus and records it plays. Unlike linear tracking turntables\arms, like the lathe that cut the master, mis-tracking applies to pivoted tonearms that cannot be perfectly tangent to the groove spiral (except at two points across a side).

Several protocols have been devised to minimize pivoted arm mis-tracking and the distortion this causes: Lofgren (aka Baerwald), Lofgren B, Stevenson, and miscellaneous setups including a few by the author. Usually alignment is facilitated by the user\installer using a *protractor* (see below) to set up the two points where the stylus\cantilever is precisely tangent, and distortion across the side is optimal. Based on geometry by disk size, these points are not at the beginning and end of the side, but at two points within. The charts next calculate for you the mounting dimensions for the arm pivot, "offset" angle of the headshell holding the cartridge, and the "overhang" past the spindle that results. The charts graph the distortion in red – zero at the two alignment points – and minimized everywhere else in most cases except Stevenson, which slightly compromises the entire side in order to optimize distortion at the end of the side, where distorting a loud climax in the music is of greater concern.



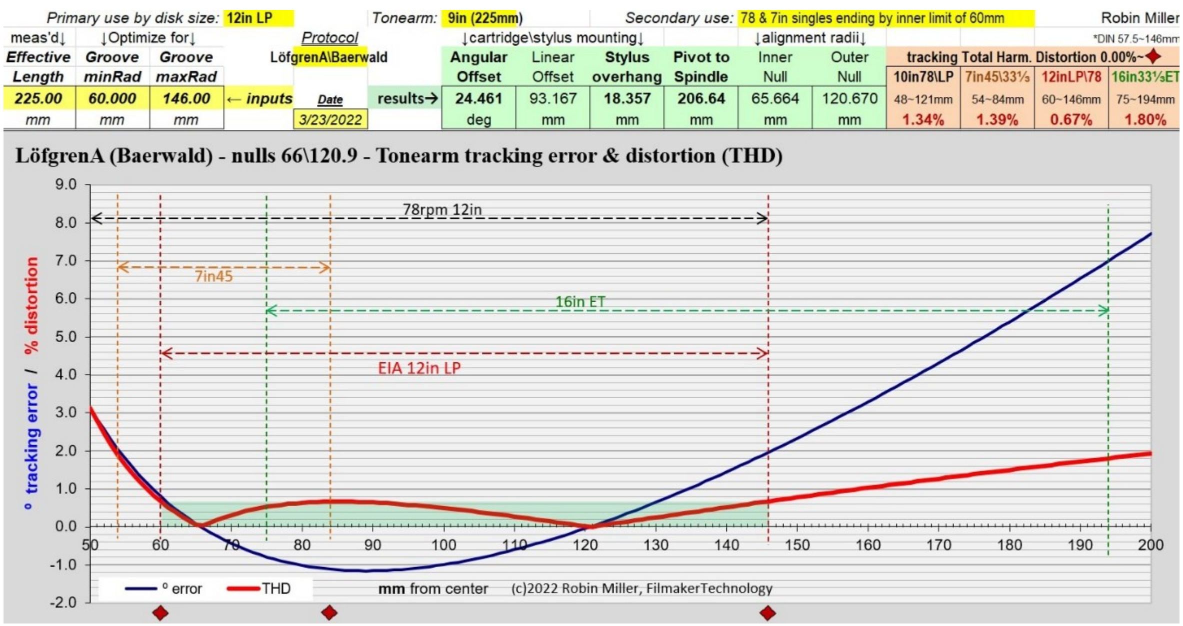
The author's Universal Phonograph Alignment Protractor (UPAP) with mirrored surface to view the cantilever from underneath the cartridge for aiming precisely.

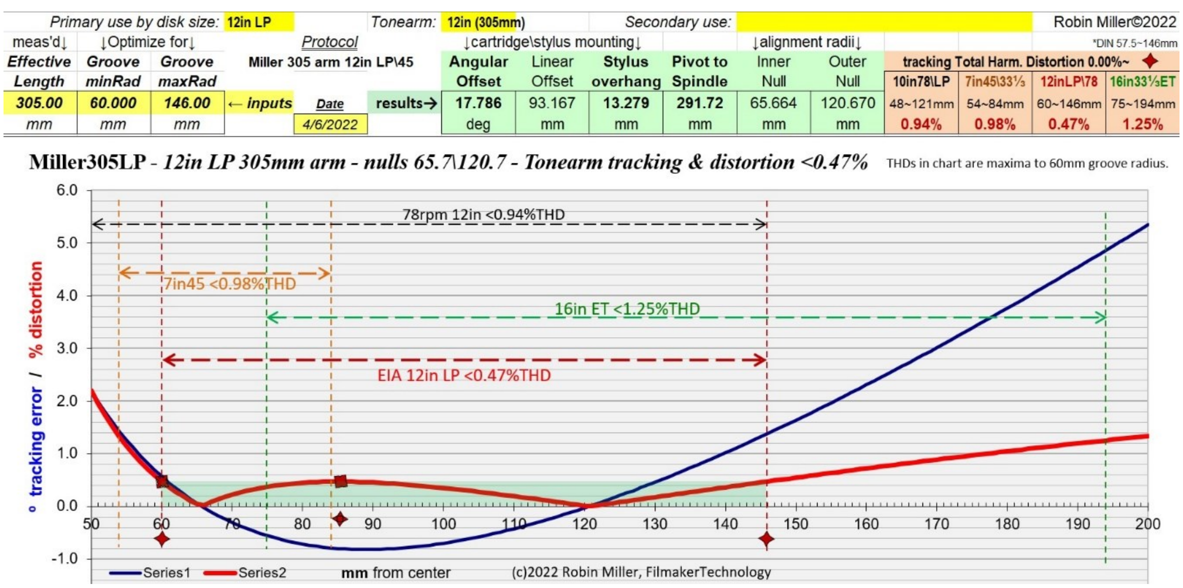
 ISBN 979-8218067304 through Ingram supplied stores, or amazon.com/dp/B0BGNF1HVQ

The charts next will not be as daunting after understanding the first one. Desired operating conditions have been input in the yellow boxes, including the "effective length" (pivot-to-stylus tip) of the pivoted tonearm and the groove's inner & outer extremes of radius. Dimensions are calculated in green boxes, where bold values are used for installation of the tonearm & cartridge. The maximum distortion due to mis-tracking by disk size are in red boxes, with the selected disk size plotted in the red curve, with the two alignment radii at 0% distortion, and the corresponding three maxima are highlighted by red diamonds. A second (or third?) tonearm – or removable headshell – may be setup for a 2nd (or 3rd) alignment protocol, and may once aligned may be freely exchanged, perhaps with a quick alignment check using a protractor.

The four charts that follow are (as titled within the chart area):

1. The typical 12in LP played with a 9in (225mm) tonearm aligned per Lofgren\Baerwald;
2. 12in LP played per Lofgren\Baerwald;with a 12in (305mm) transcription-length tonearm;
3. 12in LP aligned per Stevenson for long sides (classical movement or pop album);
4. Miller alignment for 7in 45rpm or 10in 78rpm with a 9in (225mm) tonearm;
5. Miller alignment for a 16in ET played with a 12in (305mm) transcription-length tonearm.

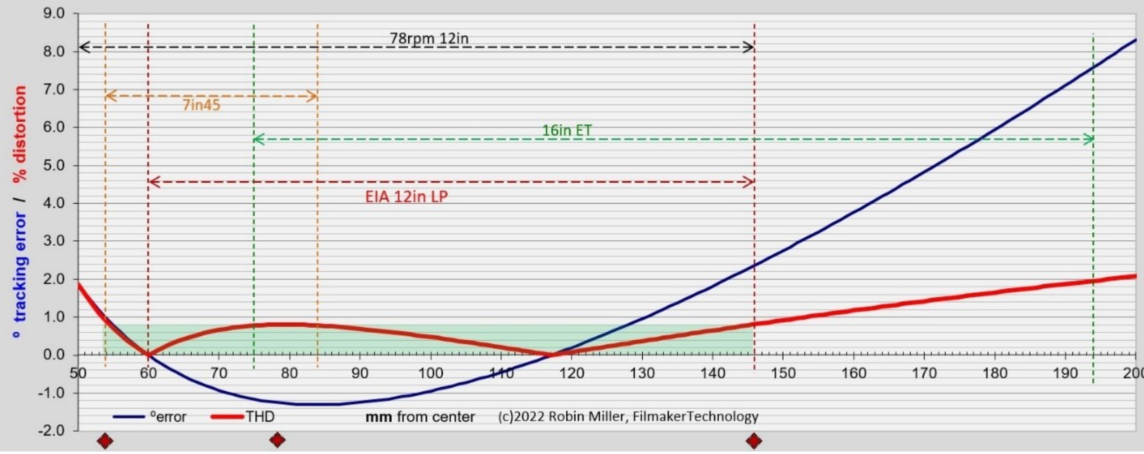




Primary use by disk size: 12in LP — Tonearm: 9in (225mm) — Secondary use: 12in LP, 78, 10in & 7in singles ending by inner limit of 54.5mm Robin Miller

meas'd↓	↓Optimize for↓			Protocol		↓cartridge\stylus mounting↓				↓alignment radii↓					*DIN 57.5~146mm
Effective	*Groove*	*Groove*		Stevenson		**Angular**	Linear	**Stylus**	**Pivot to**	Inner	Outer	tracking Total Harm. Distortion 0.00%~◆			
Length	*minRad*	*maxRad*				**Offset**	Offset	**overhang**	**Spindle**	Null	Null	**10in78\LP**	**7in45\33⅓**	**12inLP\78**	**16in33⅓ET**
225.00	***54.500***	***146.00***	← *inputs*	**Date**	**results→**	**23.189**	88.597	**16.211**	**208.79**	60.007	117.187	48~121mm	54~84mm	60~146mm	75~194mm
mm	*mm*	*mm*		3/23/2022		deg	mm	mm	mm	mm	mm	**0.80%**	**0.67%**	**0.81%**	**1.96%**

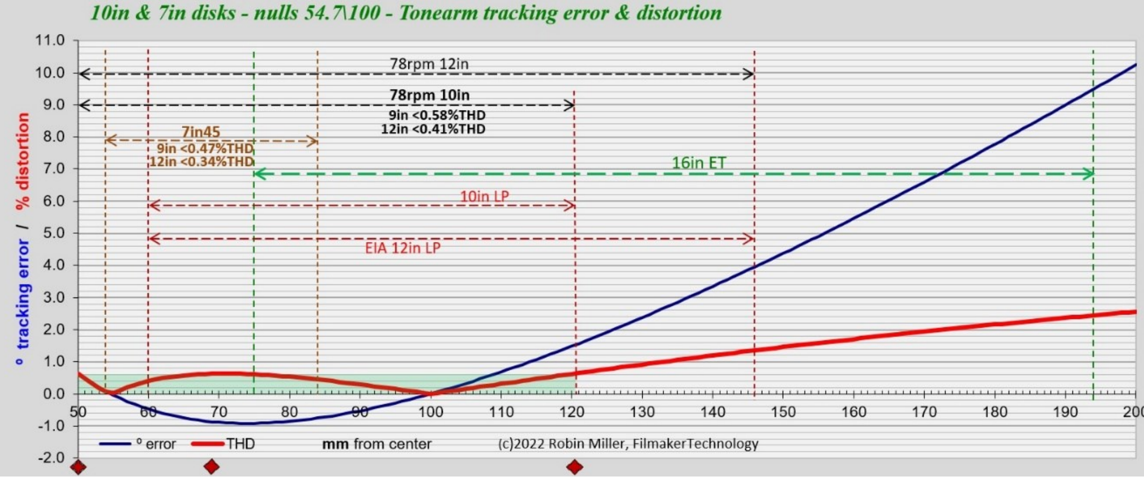
Stevenson - nulls 60\117.2 - Tonearm tracking error & harmonic distortion (THD)



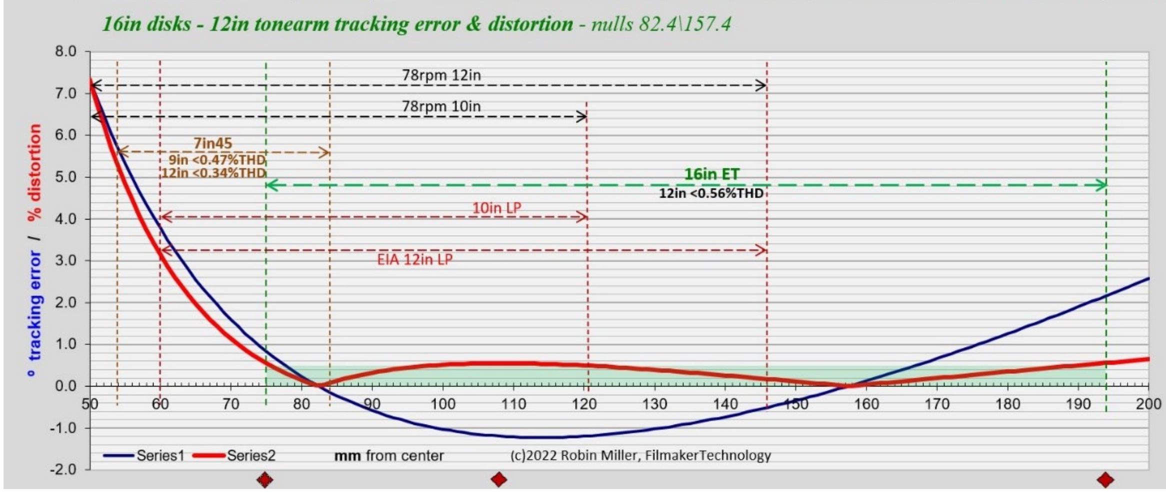
Tonearms 9in & 12in - aligned tracking distortion — Optimized use: 10in disk (any speed) — Robin Miller ©2022

meas'd↓	↓Optimization↓							Secondary use:	7in disk (any speed)						*DIN 57.5~146mm
Effective	*Groove*	*Groove*		**Tonearm Alignment**		**Angular**	Linear	**Stylus**	**Pivot to**	Inner	Outer	**max tracking Total Harm. Distortion**			
Length	*minRad*	*maxRad*		Löfgren-B2		**Offset**	Offset	**overhang**	**Spindle**	Null	Null	**10in 78**	**7in 45**	**12in LP***	**16in ET**
225.00	***50.000***	***120.65***	←*inputs*	**Date**	**results→**	**20.101**	77.327	**12.496**	**212.50**	54.690	99.964	48~121mm	54~84mm	60~146mm	75~194mm
mm	*mm*	*mm*		4/8/2022		deg	mm	mm	mm	mm	mm	**0.58%**	**0.47%**	**1.36%**	**2.44%**

10in & 7in disks - nulls 54.7\100 - Tonearm tracking error & distortion



Primary use by disk size:			16in ET		Tonearm:	12in (305mm)			Secondary use:	16in Vitaphone "talkie"				Robin Miller ©2022	
meas'd↓	↓Optimize for↓			Protocol		↓cartridge\stylus mounting↓				↓alignment radii↓				*DIN 57.5~146mm	
Effective Length	Groove minRad	Groove maxRad		Miller 305 arm 16in ET		Angular Offset	Linear Offset	Stylus overhang	Pivot to Spindle	Inner Null	Outer Null	tracking Total Harm. Distortion 0.00%~✦			
												10in78\LP	7in45\33⅓	12inLP\78	16in33⅓ET
305.00	75.000	194.00	←inputs	Date	results→	23.151	119.912	22.063	282.94	82.402	157.421	48~121mm	54~84mm	60~146mm	75~194mm
mm	mm	mm		4/8/2022		deg	mm	mm	mm	mm	mm	3.13%	3.90%	3.14%	0.56%



What is behind claims of "3D sound?"

You hear 3D in everyday living. It's the full sphere of sounds you are immersed within naturally. In addition to sounds you can see in front of you, or 5~7.1 surround sound encircling you from around and behind (flat 2D), 3D adds sounds and reflections from above and below – the sphere of sounds humans are capable of detecting and the brain perceiving as "real." The main difference in our perception of height is that it is a result not of binaural ITD or ILD, but of comb filtering by our pinna (outer ears) and so is not as accurate as our ability to localize sounds horizontally [Trapeau & Schönwiesner 2018, Journal of Neuroscience 3/5/2018]. Furthermore, pinna are as individual as fingerprints, so 3D fails for most if generalized algorithmically.

In a room, sounds from above and below are usually reflections. In a concert hall with musicians on a stage, the paths of direct sounds are in the approx. horizontal plane as listeners. However more sound energy arrives at listeners ears from all around, behind, above, and even below than from the instruments in front! More reflected sound than direct sound. While the microphones pick up all sounds in the original sphere, all directions are rendered in 2D in the final 2 or 5~7.1 channels. Then in replay at home, height sounds do not localize as originally, but are squashed into a flat 2D pie-shape or circle. If you're intentions are for a "you are there" experience, your listening room reflections are bogus with respect to the recording. They are not deleterious to that experience so long as they are much weaker than the recorded reflections of the original recording space. And bogus height might be perceived from strong, never changing listening room reflections, but no real 3D reproduction is occurring.

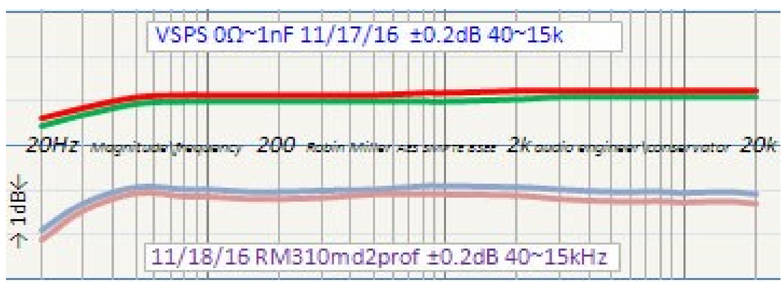
Actual 3D reproduction requires separately mic'ing, recording, and replaying the height sounds. This means, acting as a surrogate for a human head in the best seat in the concert hall, using at least eight microphones with polar patterns that dissect the sphere of sound, recording all the mics precisely, and speakers in the acoustically-controlled listening space to play them all, with some speakers above and below the listener(s). There are such systems in engineering labs and exhibition spaces, but none in distribution to consumers. The first was Michael Gerzon's Ambisonics using 8~16 speakers. Imax uses a single speaker at the top of the giant screen. Dolby ®ATMOS and other incomplete spheres are "vestigial 2½-D." The author's full sphere High Sonic Definition 3D (HSD-3D, U.S.Patent 7,558,393) using 10 speakers (plus subs) is focused for six viewers in an immersive home theater, or for a single gamer or VR subject.[8]

What about Richard Murdey's VSPS?

As many control preamps + amplifiers and receiver combinations nowadays do not have a "phono stage," they've become a cottage industry. Audiophiles wax poetic over their "sound," when their job is to be totally neutral. Others sport fancy enclosures and high prices, while lacking essential controls, such as capacitive load selection, cartridge channel balancing that ensures good stereo staging, and cancellation of vertical artifact distortion in mono, even the monophonic switch itself. (With all these controls, this book's DIY phono stage costs ~$30.)

Especially substituting premium parts (dipped mica capacitors), the Very Simple Phono Stage (VSPS) by Richard Murdey is a fine low-cost DIY preamp on a printed circuit board (PCB) for implementation by the user. His and the author's DIY preamp in the book are variants of the work of Lipshitz & Jung in 1979, differing in minor ways – the jury remains out about which might be "better," if only picking a nit. These have low noise & distortion and accurate RIAA frequency & phase responses that are predictors of accurate timbre (tone color) and behavior toward transients. Whether built-in or separate, these measurable qualities are all that a phono stage needs to be. The VSPS has a following with hobbyists, and so deserves mention here.

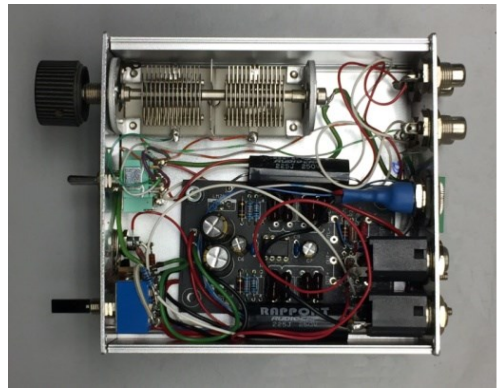
Below are measurements of my (purchased) VSPS as I built it. I recommend it. I very well may find others to recommend, if life were not so short. However, be advised that some high-end phono stages are audio rubbish in comparison – you deserve better! And not to be taken.



Above (each channel pair separated 0.2dB for clarity), the VSPS's frequency response at top (**red, green**) is smooth ≤±¼dB40~20kHz. The author's maker project is shown in **blue, orange** (instructions in the book, polypropylene capacitors used) is also ≤±¼dB 40~20kHz, differing mainly from the VSPS in steeper roll-off <40Hz for less rumble. Any phono stage measuring within ±¼dB implies top notch frequency and resulting phase responses that, along with a properly loaded cartridge, audibly approaches the sound intended by the record's producers.

Shown next before final labeling and wiring are the back panel and inside the author's encased VSPS, adding 3-input selection, variable C-load, channel balancing, and switching for mono, the last three of which the book's DIY project provides, but the VSPS board lacks. Although we can *hear through* many control- and listening- room acoustic effects, others do influence both mixing decisions and listener perceptions. In the author's studio, via top-tier 3-way monitors (JBL LSR32\6332), both preamps sound very good, if even distinguishable. Either would be a strong link in an audio system. [Ref - http://phonoclone.com/pcb.html].

[8] Demonstrations of HSD-3D by appointment at FilmakerTechnology in Bethlehem PA or the House of Music of the University of Parma, Italy. Scientific papers with illustrations of the author's work are at www.filmaker.com/papers.htm.

Electro-magnetic sound recording – the other *analog medium*

In the book "The Better Sound of the Phonograph: How come? How-to!" we have explored the evolution of the gramophone once it evolved in the mid 1920s beginning its electrical era. Written 35 years after popularization of digital audio, the book is meant to help old hobbyists as well as new in today's resurgence playing "vinyl." But it skirts an intervening technology that prevailed during most of the author's audio engineering career – magnetic tape.

Sound recording as technologies combine three facets: *transport, modulation,* and a *medium.* Occurring over time, the transport must "move" some medium at a steady rate in time. Then after *transducing* acoustical vibrations into either mechanical or electrical signals, a modulation facilitates the capture of a sound record. Only then can the physical recording medium accept, permanently store, and allow retrieving these signals to reproduce them again as sounds. All audio reproduction media – phonograph, mag tape, & digital – possess these three elements.

In 1877 Edison yelled "Mary had a little lamb" into a horn with a needle embossing wax paper on a spinning cylinder, and the phonograph was born. It was not the first sound *recording,* but it was the first to be *played back* again as sound. Eventually copies were molded and sold to the public. The *transport* was a lathe riding a spinning mandrel; *modulation* was mechanized analog sound vibrations; the *medium* was a groove in a revolving cylinder.

By 1898, Poulsen substituted a steel wire for the groove around the cylinder, and a telephone for the acoustic horn. It was the first electro-magnetic recorder. The transport was again a lathe riding a mandrel; the modulation now was *electrified* analog of sound vibrations; the medium was a magnetic wire. In 30yr, it finally would lead to the magnetic *tape* recorder. The transport would be a motor driven capstan plus coordinated turntables for the tape supply and take-up reels; the modulation is an analog audio signal biased beyond regions of magnetic hysteresis (distortion); the medium became iron oxide coated paper, later plastic tape.

Taking longer than we might expect, it was the late 1920s, well into the phonograph's electrical era, when German engineer Fritz Pfleumer in Berlin made and demonstrated to AEG-Telefunken (a pre-WWII partner of GE) a recording machine using iron particles lacquered onto a strip of paper. It led to the 1936 "Magnetophon." In 1945 at the end of WWII, U.S. signal corps Major Jack Mullin shipped to California two machines and 50 rolls of erasable and re-recordable tape, playable many times without degradation. He went about promoting magnetic tape recording with industrial\educational filmmaker and audio engineer Bill Palmer. Demonstrations to other engineers led to a key association with Harold Lindsay, chief engineer of Alex M. Paniotoff's little electric motor manufacturing company Ampex. While they were still developing what would be the first professional tape recorder, they demonstrated it to the fledgling ABC Radio Network and its main star Bing Crosby, who after securing deals for 20 machines and investing $50,000 became Ampex's west coast distributor.

Like the stories of many entertainment technologies, including the LP and stereo (both conceived in the early 1930s but only entering the mainstream in the 1950s), one reason this story takes decades was the intrusions of world wars. Another was the long road of electronic sophistication, especially compared to today. Vacuum tubes made electronics possible. But it took innovative circuit design combining simple triode amplifier with other components was decades in development. Decades to realize the tubes' full usefulness, then to transition to solid state. Decades to solve problems, such as magnetic recording's severe *hysteresis* distortion…

A case of luck is the accidental discovery – simultaneously in the US and Germany – of adding *bias* to the audio signal on the way to the record head: from no bias to DC bias to ultrasonic AC bias. Each step significantly lowered hysteresis distortion by elevating the signal to the region where tape magnetics no longer lag behind the audio before catching up. Now better at re-recording (for editing and mixing) than 78 or 33⅓ transcription disks and optical sound recording, it sounded to radio listeners and sponsors indistinguishable from live.

Leaving NBC for the Justice Dept-mandated spinoff ABC, Bing Crosby negotiated for new technology to record his popular half-hour radio show. Ratings were down due to the degraded audio quality from multiple disk generations, needed to edit, plus time-shift for the west coast rather than doing the show live twice three hours apart. Just in time came the Ampex 200 with a 14in reel of tape running 30 inches per second which could record & play uninterrupted for 20min. By the introduction of the model 300, running a less unwieldy 10in reel at 15ips for 30min with the same quality but at lower cost, plus the use of sprocketed magnetic film for synchronism with moving picture film, the future of audio was cast for live-sounding radio, television, sound motion pictures, and phonograph disk mastering…

…Until entry in the early 1980s of much more complicated digital audio. Now the transport was a stream of binary bits from periodic sampling the audio waveform; the modulation was word lengths (bit depths) per sample of up to 24 (16,777,216 levels); the medium was magnetic tape, hard drives, optical discs, and solid state memory. Computer digital audio workstations (DAW) and video editors have taken over in broadcasting, movies, and music recording, replacing magnetic tape in the resurgence of phonograph mastering. What might be next?

Related reading – too short a list of seminal and lesser known works on sound

From Tin Foil to Stereo: Evolution of the Phonograph – Oliver Read, Walter L. Welch, ISBN 9780672212062, chronicles the invention and advance of phonograph recording & reproduction.

Chasing Sound: Technology, Culture, and the Art of Studio Recording from Edison to the LP – Susan Horning, ISBN 9781421410234. An academically researched history of recording.

Perfect Audio Forever - Greg Milner beautifully articulates the art & science of audio recording and psycho-acoustics. And reveals the generational lag in acceptance of new technology, whereby some audiophiles favor analog over digital in the same way Edison decried electrical recording over acoustical [likely out of financial self-interest].

The Revenge of Analog - David Sax on musical richness returning to his life after decades of digital downloading, now that used vinyl stores are cropping up like Starbucks.

Handbook for Sound Engineers: The New Audio Cyclopedia – Ballou, Howard W Sams & Co.

Handbook of Recording Engineering – John Eargle, Van Norstrand-Reinhold.

The Handbook for Stanton and Pickering Phonograph Cartridges and Styli – Richard Steinfeld, self published, rsteinfilt@sonic.net.

The Theory of Sound – John William Strutt, 3rd Baron Rayleigh, OM, PC, PRS 1877 (coincident with Edison's phonograph); 2nd ed. 1894, ISBN 9780486602929. [Lord Rayleigh was a member of the House of Lords, and Chancellor of Cambridge University, he discovered Argon, and invented the smoke shelf that solved the poor draft of fireplaces to expel exhaust.]

On the Sensations of Tone - Hermann von Helmholtz, who like Lord Rayleigh, investigated sound long before electronics or recording, in laboratories with little more than mechanical sirens and switching oscillators! His *resonator* is a staple of studios and performance spaces.

Spatial Hearing – Jens Blauert peels back the onion of the physics and psychoacoustics of sound & perception through a logical sequence of empirical and ear-opening experiments.

Sound Reproduction: Loudspeakers and Rooms – Floyd Toole shares JBL's evaluating speakers of quality, taken within their most influential extension, the listening acoustics.

Surround Sound: Up and Running – THX inventor Tomlinson Holman gives both audio engineers and home theater owners the skinny on 5.1~7.1 multi-channel surround [2D].

The Audio Expert (Second Edition) – Ethan Winer's truly expert review of "all things audio" (but with scant little on record playing, so augment with The Better Sound of the Phonograph).

…with more in future updates. Add scientific papers by many audio engineers & acousticians, e.g. Stanley Lipschitz, Leo Beranek, David Griesinger, the author's fellow presenters to Audio Engineering Society (AES, incl. student chapters), Acoustical Soc. of America (ASA), Canadian Acoustical Assn (CAA), Australian Acoustical Society (AAS), German Tonmeisters (VDT), Society of Motion Picture and Television Engineers (SMPTE), Boston Audio Society, etc. I've read these, and quite a few others, some more than once, studying and re-studying them, always discovering something new. Also novels & histories about life's comforts & conflicts, from music to war. May I recommend U S Grant's "Memoirs" along with "Grant & Twain" and "Huck Finn." History in Chernow's "Hamilton" and "Founding Brothers." Biology in "Seven Daughters of Eve." Statistics in "Here's Looking at Euclid." Our universe in "Astrophysics for People in a Hurry" by Neil de Grasse Tyson….. So much knowledge, so little time!

An audiophile's preference for coloration, and "soundstage"

Some audio-centric online forums and FaceBook groups can't agree what an audiophile is. They even criticize their own as more interested in audio gear than music. Pink Floyd's Dark Side of the Moon engineer Alan Parsons, has said: "Audiophiles don't use their equipment to listen to our music; Audiophiles use our music to listen to their equipment." I've written above that, unlike most fledgling cost-conscious science-oriented Hi-Fidelity hobbyists of the 1950s, today many who speak the loudest are clearly less interested in science than in shopping.

True, some audiophiles focus, more than listening to music, on shopping, purchasing, installing, and endlessly (and expensively) swapping audio components and cables by trial & error. This activity has redefined the hobby today. Is the attention inequality significant? A long time filmmaker, I can attest that a crew spends more time setting up a shot than shooting it. Then spends considerably more time in editing and other post-production activities than the final length of the film. Despite many more man-hours, enlightened filmmakers accept that these behind-the-scenes efforts are subservient to the creative process and its artistic results.

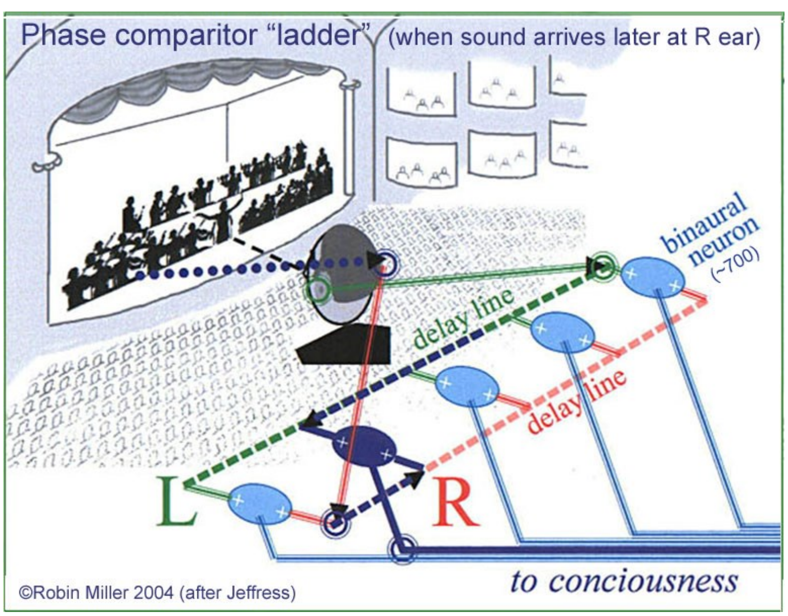
To an audio engineer, this seeming imbalance of time & effort is an occupational hazard. As a profession, we design, prototype, construct, calibrate, document, optimize, and maintain technical systems apart from listening, and over many more hours. Also true for television, film, or any technological art-form. An engineer's purpose is to implement systems for others to enjoy with little such effort. A few audiophiles too know the hours they spend tweaking are for the purpose of better sounding music – *subjectively better,* if only to conditioned taste.

And *music* isn't all that audio reproduction is used for. There's *speech,* such as aural history, standup comedy, and film dialog. And natural "sound effects" such as bird chirps and battle sounds, if this last can be called "natural." It could be argued that speech and natural sounds warrant the most natural, neutral reproduction. Distraction-free support of the story-telling is what film sound recordists & mixers strive for as much as cinematographers. However music since the introduction of the 45 pop single and the digital CD rides the coattails of ever-novel, fictitious, unnatural, bigger-than-life sound. High-*Infidelity*! Experienced generations often feel music has devolved into musical pretenders and marketing shysters. Curmudgeonly (if not correctly) audiophiles who feel the best music is at least 40 years old. Music found mostly on analog records, played on hi-fidelity systems, not on digital CD re-releases, or via audibly lossy streaming to cheap earbuds and tiny speakers. High fidelity reproduction as a science obeys irrefutable laws of physics. Sometimes best implemented by the restoring of vintage equipment.

Another big change from the early hobbyists of the 1950s and today is the gulf separating audio amateurs from audio professionals. 1950s fan magazines were filled with lab tests and expert opinions, with less of the hype or undefined adjectives in popular use since. Audiophile lingo today departs from scientific terms; unknown long-established terms sport new meanings entirely. The term "soundstage" to one practicing professionally in entertainment media means a television or film studio, with controlled physical properties for recording both sound and picture. To an audiophile, I gather (in context) it means the auditory "image" spanning the two stereo speakers. In audio engineering parlance, this is more precisely several qualities termed spatiality, localization, and directional coloration. Audiophiles post how "soundstage" changes by swapping expensive cables etc., where in fact any nuances are likely inaudible, acoustically masked, or psychologically biased by visual cues or price. Auditory events (sounds) are biologically encoded, then consciously interpreted (perceived). It's amazing how this works…

Our binaural hearing is pretty good at horizontal localization, accurate in front to within about one degree. Our pinna (outer ears), their convolutions coloring sounds according to the direction of each sound's arrival, confirm horizontal directions, and can localize up-down accurately within ~10º. (Owls can nail a rabbit in the dark using their much better up-down localization, owing to the vertically offset bony outer ears behind each eye socket.)

Along the road from ears to conscious perception of auditory events, positioning sounds in 2-speaker stereo reproduction is the work of reptilian structures in the base of the brain that discriminate both how loud or soft and how early or late is each sonic arrival – direct sounds and reflected ones. Above the crossover frequency of the human head, about 700Hz, level differences are compared. Below 700Hz, time of arrival is measured by a "ladder" where each "stile" is a neural "delay line" fed from opposing end by its respective ear [as modeled by Jeffress 1948]. And along which about 700 dual-input neurons form the "rungs." Illustrated below, a sound left of center arrives first at the left ear, its signal traveling farther down its respective delay line (green) until meeting the later signal from the right ear (red). This coincidence fires neural rungs in the vicinity, statistically summed in conscious perception to home in on a sound's position. Another comparator deciphers level differences above 700Hz. In nature, timing & level cues usually agree. If not, we perceive an unfocussed, smeared "soundstage." It happens in recordings, for example when improperly panning spot mics (only level differences) conflict with a binaural main mic (providing mainly timing differences).



In the base of the brain, a "ladder" with "rungs" of 2-input neurons between neural delay line "stiles" permits comparing time-of-arrival of ear signals. A sound left of center arrives earlier at the left ear, traveling farther down its delay line (green) until it meets the later-arriving signal from the right ear, sending the sound's horizontal position to our consciousness.

The brain tries to fill in any blanks as best it can, but sometimes is confused by artificial stimuli. For example when using headphones, all sounds are at the extreme sides, but do not change in pinna coding when we rotate our head even slightly. The brain can only conclude the auditory space is fixed inside our head! Another case is when the channels contain signals that are 180º out-of-phase. This can happen even if the speakers are wired correctly in-phase. The book explores the cause of "pinch effect" distortion, worst when using a spherical stylus unable to negotiate high frequencies in the inner groove. Pinched up and dropping down twice for every cycle of signal produces 2nd harmonic distortion, adding false brightness, and changing the tone of instruments that produce only odd harmonics, such as the flute, clarinet, and trumpet. *This 2nd harmonic distortion is out-of-phase between the stereo channels.* If signals arrive at the ears predominantly out-of-phase, our comparator ladder is driven to extremes; the brain interprets the sounds as coming from the sides, well beyond the arc of the two speakers. A solo trumpet becomes disembodied, its natural odd-order tone is in front, but the added even-order harmonics flash around the room. Novel, even exciting at first, so only a temporary effect.

Many audiophiles engage in swapping components until they find a sound they like better. That these candidates sound different one from the other implies they impart different tone colorations. In other words, specific *distortions.* One might initially like an unnaturally even-harmonic trumpet or flute sound, but when the brain tires trying to make sense of the disembodiment, listener fatigue has set in. Eventually it can lead to irritability, and the music is stopped. This explains why some audiophiles in time discover they prefer the clean sound of monophonic recordings, especially played with a monophonic cartridge that more perfectly rejects vertical artifacts such as pinch effect. (Mono is best played over a single speaker to avoid comb filtering by identical signals arriving at different times from displaced speakers. [The author's archiving preamplifier outputs monophonic audio only from the left connector.]

Seasoned audiophiles in time come to appreciate uncolored, undistorted sound as more real, more transparent, and less fatiguing. They populate links in the audio chain with components that are neutral and color-free. Whether to realize the intent of the record producers, or to reproduce correctly the instruments as the listener knows them from experience listening live. This is the goal of the book – to aid in your making the sound of vinyl "better" than otherwise.

What about the future of audiophilia? If conversation on the Internet is any indicator, many audiophiles are faith-based believers in whatever has been most whispered down the alley of fellow hobbyists, or in advertiser-supported magazines. Much of this is deluded balderdash. Unsubstantiated pseudo-science. Specifications that are meaningless marketing, intended not to quantify performance, but induce salivation. Swallowed whole by many who mindlessly follow in unquestioning agreement rather than who seek answers open-mindedly. They pay little attention to the most important and least understood aspect: listening room acoustics. Knowledge is power, held either by disciples of sellers, or by aware buyers. Which are you?

Humans are more visual than aural, so it shouldn't be surprising that much of the focus of audiophiles is how cool components look, effectively biasing how they are perceived to sound. To judge subjective quality, it is necessary to do *blind testing,* with "devices under test" hidden behind a curtain, and presented to listening subjects at precisely the same volume and using short, non-sentimental clips of music and other recognizable live sounds. Audio science researchers do it this way; audiophiles do not. *Confirmation bias* claimed to be valid testing.

Audiophilia at its ugliest is condescending to newbies. It would have them feel they are not good enough, not rich enough, or haven't the ears for it. This is the worst balderdash of all. Appreciation of music is a primal instinct for humans. Almost any budget can find a high-fidelity solution. Even a person who is deaf in one ear can develop *an ear for good sound,* maybe better than some audiophile's two. Many audiophiles online are generously helpful to newbies, but some patrons I've met at high-end stores are haughty. Even with professional education & experience, my discussing audio with them falls on deaf ears. So the book is for the more curious if not already technically-astute reader, regardless of age, ability, or means.

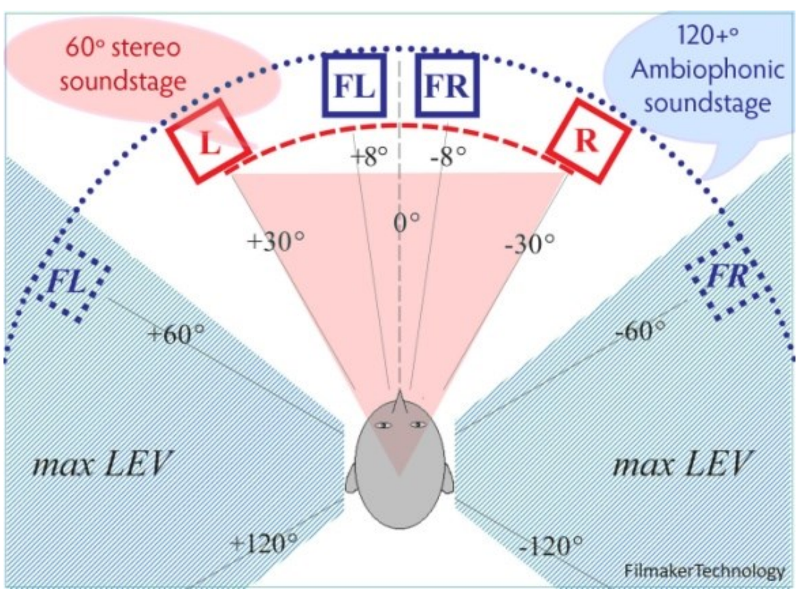
The web is home to trolls, divisive jokesters who live to bait and to foment ill feeling. And pretentious, self-proclaimed "futurists" who safely dispense quackery, as prognostications take years to prove\disprove. Ones I've met have a hidden agenda. Some are paid lobbyists. I find off-putting the title futurist itself, but many audiophiles believe and often repeat the prophecies. Beware those adding "Trust me!" or "No question!" to their discussion-stopping *non sequiturs*.

No one can reliably predict the future, and guesswork comes up heads only half the time. A less pretentious, more accessible moniker might be Director of Humble Beginnings, DHB. A subtitle might read "noticing the small things to discover what *might [but is not sure to]* become the next big thing." A DHB is a discussion starter: "What is it really? Why do it?" Rather than shutting it down, rational input is nurtured. The DHB distills it, and then reports it unbiased. A small example: When in 2014 two grandmothers tell me they're buying vinyl for grandsons, I get that the phonograph hobby may be due an updated bible. So I began to write this one, published in 2017. And its acceptance suggests that that coin has come up heads.

How tomorrow might stereo evolve?

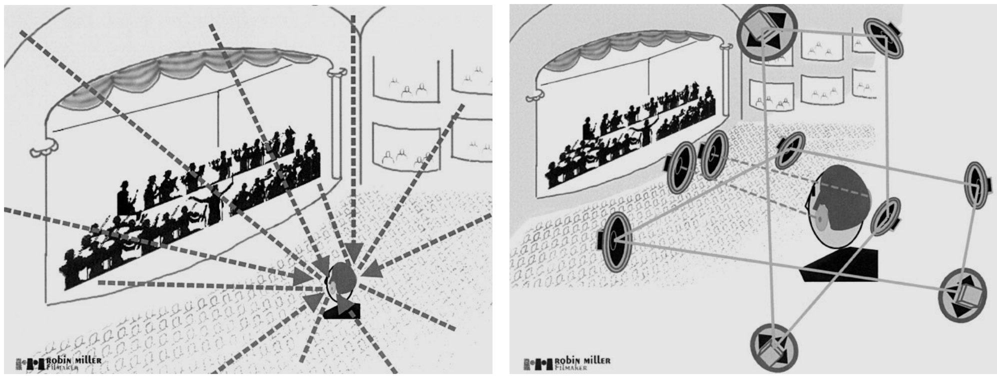
Another online poster began an active discussion with: "I went to a concert hall to listen to a symphony a year ago. I have had little luck trying to replicate that experience." There were many replies lamenting, no matter what the equipment or expense, it was "a fool's errand."

My contributions were: 2-speaker stereo is a long way from rendering a "you are there" concert experience. Stereo speakers separated the conventional 60º (±30º), illustrated below **in red,** do not induce *listener envelopment* (LEV) that is maximum for sounds arriving from the **regions in hashed blue** beyond ±55º at each side of a listener's head. 5.1~7.1 surround is a listener-enveloping circle, but is still flat 2D, with no up-down (3D) cues for life-like sound. Yet most audiophiles reject any more than two speakers, thus giving up the realism possible with the 2D envelopment and 3D immersion afforded by multi-channel reproduction.



Listener *envelopment* (LEV), from the author's papers at www.filmaker.com/papers.htm or www.ambiophonics.org. **Stereo's stage (red) spans speakers L & R** v. **Ambiophonics' 120+º (blue) spanning virtual speakers FL & FR,** although the actual speakers are close together, which maximizes LEV. Use the plug-in RACE (recursive ambiophonic crosstalk eliminator, developed by the author for the Ambiophonics Institute) at http://www.filmaker.com/products.htm

The *verisimilitude* the original poster seeks requires the listener's *immersion* in a sphere of sounds that preserves the directions of concert hall reflections from above & below, as well as in front, around & behind – illustrated below left. (Your listening room reflections are *not* the acoustic signature of that concert hall). In reproduction, these directed reflections are mapped by your individual pinna (outer ears), just as in live hearing, to recreate life-like timbre (tone color), the holy grail of high-fidelity audio. I demonstrate this virtual reality in sound in the full-sphere prototype High Sonic Definition 3D (HSD-3D) – www.filmaker.com/papers.htm . The demos compare full sphere 3D with conventional stereo's 60º triangle and the 360º of 5.1 surround (a flat 2D circle). Listeners agree that the custom 3D recordings, played over the 10 speakers of HSD-3D (shown below right), come closest so far to a real concert experience.



L, Listening live, *reflected* energy arrives at the listener unseen from above & below, as well as from all around. Original tone color (timbre) results from our individual pinna's (outer ears') combined response to each sound's individual direction, arriving over time. R, the author's Patented High Sonic Definition 3D (HSD-3D) over 10 speakers preserves the directions and the sum of tone colors of the full-sphere of direct & reflected sounds. Playing back custom HSD-3D music recordings, movies, or games – back-compatible with stereo or 5.1 over conventional media – completes an immersive life-like experience.

Its popularity notwithstanding, 2-speaker stereo embodies certain drawbacks: 1) Crosstalk at each ear (from the speaker on the other side) creates comb-filtered tone colors for center voices, and pinna conflicts for all "images;" 2) Reproduced "soundstage" is limited to between the speakers, usually 60º apart (half the typical 120º recording angle), and lacks LEV. The key to unlock many recordings is crosstalk cancellation – Ambiophonics, illustrated above. Grammy nominated mastering engineer Kenny Mixx says: "The plug-in I wouldn't work without is [the author's] Ambiophonic DSP." – info & purchasing ($10) at - www.filmaker.com/products.htm .

New-released records & reissues

What is holding back the resurgence in vinyl? Largely it is economics. A technical book, "The Better Sound of the Phonograph: How-come? How-to!" says little about marketing vinyl. But the author scours journals that do. And finds gems to paraphrase, plus his 2¢. How can those who market vinyl today explain its stalling? From a boom in 2014, the trend for *new release* vinyl sales slipped in 2015 & '16, per the Recording Industries Assoc. of America (RIAA) [The Guardian]. Why are new record sales falling, when used record sales continue to rise? How come the difference? On the heels of CD's downturn, you'd think the labels would want to know why. It's no coincidence – the two downturns may be related.

"Behind the resurgence of 'vinyl' records in recent years, the quality of new LPs often stinks... loud and harsh-sounding, optimized for ear-buds, not living rooms." [Wall Street Journal 7/22/17] Not so for every new release, but for many the quality "stinks" for reasons that are no secret on the pages of this book. Producers of popular songs in digital form on CD and streaming media, delusional in their engaging in Volume Wars, have sacrificed natural dynamics in order to sell releases impulsively by making them sound louder. Over-producing is too easy, mouse-ing around the wonderful processing tools of a digital audio workstation (DAW). Tools were more primitive in the analog era, and required savvy engineers to work what was possible. Then in 1982 when the digital CD signaled the sunset of the phonograph, would-be apprentices slunk away. And with the passing of their mentors, mastering engineers of The Golden Age, the best know-how, skill, and restraint were gone. It shows baldly in new releases & reissues. On being recognized by the Audio Engineering Society and turning 99, the last, Grammy winning Clair Krepps, observed that many today don't know what they are doing. The last of the great mastering engineers of the phonograph era died a month later. [9]

Today shoppers in stores or online comparing samples of new releases, without touching the volume control, are struck by the loudest sounding stuff. Although after they take it home, they find in time the dried-up presentations tiresome, the processing artifacts irritating. They could have had BOTH a fresh sound AND have it as loud as they want simply by turning up the volume. Of course the record label would have had to refrain from making over-produced, compromised product in the first place. But the labels are repeating CD history – taking the same failing approach to new record releases! This applies mainly to popular music, not acoustic classical and jazz which seek to evoke a "You are there" live performance, but are not money juggernauts, nor sold by compromising quality, whether in analog or digital form.

As of this writing, 80% of new records are mastered from digital sources, a percentage that inevitably will rise toward 100. Digital mastering is not inherently poor in quality – in fact it is by design superior technically to tape. Except that to save cost, the vinyl master is often the same over-produced file used for CDs! Oftentimes the lacquer master for a reissue is cut *from a distribution CD!* Vinyl has no magic power to reverse over-processing baked into the CD, so the vinyl sounds as bad – worse, adding vinyl's own replay artifacts, as are explored (seeking to minimized them) in this book. Deciders at the label have also raised retail prices of new records $4~6 in the two years that saw slowed sales growth, as if to milk vinyl's last hurrah.

However used records continue their boom times, likely in part *because of their better sound.* LPs in the hi-fi era were produced uncompromised, skillfully made to higher sonic standards than releases today. Not artificially juiced for impulse buying, at a cost to lasting satisfaction. Don't these label bosses get it? Probably they do. But what they get more, after bonuses have been paid, is that they can move on with a (fleeting) "success" on their resumes. Their successors then rinse & repeat! Music lovers lose. And it's unrealistic the labels will change.

If only dirty, used records are a safer bet. They are reviewed on many web forums and FaceBook groups devoted to vintage recorded music. For reissues, it would be helpful to savvy buyers – older sentimentalists as well as newer converts to good sound – to know the source of mastering. Was it the original master tape? [10] Perhaps a perfectly acceptable 1:1 digital transfer of the master tape, especially if partially deteriorated, and a true *restoration,* not re-mastered according to contemporary abuses. As an example, The Beatles "Sgt. Pepper's Lonely Hearts Club Band" reissue is sterling, either in monophonic as originally recorded, or remixed in stereo. Let the buyer beware, more than ever. } ;<{)# [meme of the author winking]

What to do if capacitive loading is too high? Too low?? Compensating with EQ???

We've explored why it is important to phonograph sound to match the electrical capacitance "seen" as a "load" (C_{load}) by a high impedance cartridge (moving magnet or moving iron\variable reluctance, MM & MI). It is the sum of the capacitor inside the preamp (C_{preamp}) plus the tonearm-to-preamp interconnect (C_{cable}). It is easier to add C than to reduce it. But what if the total C_{load} is greater than that specified for the cartridge? Many makers chose 275pF (pico Farads of capacitance) as a good value – not so high that many users would have to add much more than typical cabling plus the capacitor inside the preamp; not so low that more effort or expense was needed to reduce it.

However, if you do have too much, here's whether it matters, and what to do about it. The chart's blue curves show the estimated effect on the sounds of Stanton's 880\881 (MM) and 680\681 (MI) pickups with just 20% too much C-load, 330pF. This happens if coax cables are not of at least average quality (have cable capacitance greater than 25pF/ft), or are longer than 5ft, for a C_{cable} >125pF. Adding the typical 150pF built into many preamps results in total capacitive load that may

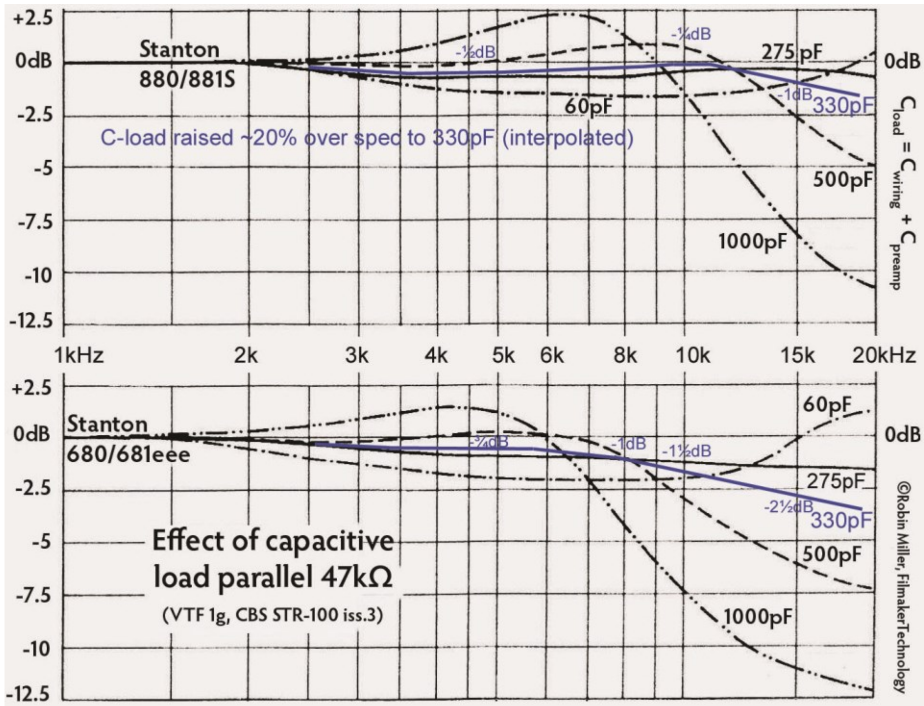
[9] Clair Dwight Krepps, interviewed for the book into his 100th year, passed on 12/15/17 – see Afterword.

[10] E.g. not the 1/4in tape "dub" (copy) for Everest reissues, but the original 1/2in 3-track of old Everest?

ISBN 979-8218067304 through Ingram supplied stores, or amazon.com/dp/B0BGNF1HVQ

be wrong for many cartridges, as the resonance with the cartridge's inductance peaks at a lower frequency and begins to fall off early in HR response.

If the total C_{load} were only 20% above spec, 330pF, the sound of the 880\881 will be nearly as flat as its 275pF spec, with only a 1dB reduction at 15kHz – probably unnoticeable. However with the 680\681, highs will be down 2½dB, and sound noticeably dull. For the 680 and other similar cartridges, 10% over spec is about all that can be ignored. (Professionals using balanced twisted pair wiring of 35pF/ft typically should take particular note, and adjust any capacitor inside the preamp).[11]



Imagining it even worse, such as the 500pF curve from thin, cheap single-conductor shielded audio cable can run up to 80pF/ft. No problem at high line levels and low impedances, but a 5ft phono cable would be 200~400pF. Then add the preamp's 150 for a total C-load of up to 550pF! The 500pF curves on the chart show what happens to an LP sound's frequency response, wrecking tone color, and phase response as well that wrecks transient response. With its lower impedance, the 880\881 sound is unnaturally "present," and the highs are muffled. A higher impedance 680\681 resonance is shifted lower by about 2/3 octave, and the sound above 6kHz is decidedly muffled.

For most audiophiles using conventional coaxial interconnects of average pF/ft, an audibly dull effect is almost intuitively addressed by cranking the treble. Or simply pronouncing the cartridge "dull" and buying another, with possibly the same outcome. Even magazine reviewers do this, and criticize the cartridge rather than own up to their installing it improperly. The better solution is to: 1) Use shorter cables, or cables of lower pF/ft, as exemplified in the math above; or 2) Remove\adjust the capacitance inside the preamp. It helps to buy a low cost capacitance meter to measure cables.

The opposite problem is too low a C_{load}. For example if, as is often the case, there is no capacitor inside the preamp, then typical 125pF cabling will be the only capacitive load for the cartridge. In this case, the 680 pickup will lose presence (–1½dB at 6kHz), and dust crackles will be emphasized (+1½dB at 15kHz – a 3dB swing in about an octave). The 880 will be less affected. These effects with these or any number of MM\MI cartridges with similar characteristics are among the most important determiners of the sound of a phonograph, up there with the choice of stylus and preamp RIAA EQ accuracy. The solution is to add capacitance either inside the preamp, at its inputs using available plug-on (2) capacitor accessories, or lengthen the interconnects, such as with extension

[11] Recall that with low impedance moving coil pickups (MC), these matters of capacitance loading do not apply.

cable(s) chosen by their length for contributing so many pF/ft. Remember if you change cartridges to one specifying a different C-load, the cabling or preamp capacitance will need to be readjusted.

You ask "Can I compensate a C-load mismatch using an equalizer, or tweak the preamp's RIAA equalization?" You could if it reciprocally undoes the erroneous curve. The caveat is that inside the cartridge are coils – inductors – with which the C_{load} forms a resonance peak that can be narrow or broad, varied in level, and shifted in center frequency. A treble control is usually not peaking (not like a resonance), but shelving (a non-resonant filter), which also describes the three filters of RIAA. Resonant peaking and shelving filter curves do not match precisely. It's best to make adjustments individually: proper C_{load}, accurate RIAA, and avoid correcting by ear using an equalizer.

Then the elephant in the room is the acoustics of the room. Room "modes" are also resonances, so are not accurately corrected by shelving EQ. Again, it's best to deal individually with any issues, such as taming room resonances using acoustic absorption, resonators, diffusion, even altering the shape and dimensions of the room – least expensively done on blueprints before the room is built. Otherwise, better sound will be a matter of guesswork and luck, trying one thing after another until an error is found that just happens to correct another error of an opposite nature. This practice nearly redefines the hobby, and is fine if an enjoyable pastime. Then something changes, and it starts over.

Where to find the best new-old-stock (NOS) Pickering\Stanton stuff for reasonable $?

Good as most new MM\MI phonograph transducers (pickups\cartridges\styli), Pickering\Stanton cartridges do not degrade whether sitting on a shelf or well worn cosmetically. Unless a MI parked next to a powerful loudspeaker magnet or tape degauser (demagnetizer), in which case the magnet might have been weakened, easily determined by a significantly (>30%) lower than specified output. So a used or NOS cartridge without stylus is still a possible bargain at auction. Styli are another story, now gobbled up by the cognoscenti, and become as scarce as hens' teeth. When found, these rarities should be bought only if NOS, or if claimed to be NOS, carefully evaluated under a true 400x microscope. Several desirable models are still findable for relatively low cost: e.g. for lighter arms, original ("genuine") Stanton D680 and stylus assemblies, replacements for Stanton 680\681 and Pickering XV-15 pickups, that are 0.3x0.7mil (7.5x18μ) elliptical with mid-to-high compliance, tracking at ~1¼g. For heavier arms, a Stanton "DJ" tracks 2~4g, and also a 0.3x0.7mil (7.5x18μ) elliptical. In either pickup, and with proper loading as harped upon in this book, these will perform well, and be hard to beat today at any price. Avoid products made after Stanton moved from Plainview NY, and Pfanstiehl listings in the 4,000 series of reproductions (* and three digits indicates originals). Another resource of auctions missed by many is misspelled listings on eBay - https://typobargains.com/stanton_680eee.html . With diligence, and using eBay's search for the model(s) shown in the book and bid for smartly, on increasingly rare occasions gems pop up.

Of course this book is necessarily limited to examples of the science presented, and cannot possibly cover all makes & models of cartridges and styli. Excellent products are available from Audio Technica, Nagoako, Ortofon, JICO, and other reputable makers. Their specifications can be compared to examples in the book. But we leave the book's rigorous analysis of them to others.

Subjectivists v. objectivists?

Might as well tackle this tar baby. Much like our real world, the audiophile world has become divided along the ideological lines of emotion and science. We are sentimental animals, and in school not everyone took to math & science. But now instead of respect for individual skills, we decry them. Long ago tribalism overtook religion and politics; now it's the scourge of our hobby.

Ultimately, audio conveys the subjective experience of the music, spoken word, or movie content it reproduces. However audio is a technical conveyance, devised, made, installed, and maintained by technicians, or engineers such as myself. Our goal usually is to make something useful, practical, and ergonomic (easy to use) – if we're successful, so easy to use that for sentimental-types we and our work is *out of sight, out of mind.*[12] But that does not mean it isn't the iceberg below the surface.

I – a musician turned engineer – appreciate both sides. Which means I recognize both the pluses and minuses of both purely subjective and purely objective approaches. The pure subjectivist having little working understanding of applied science says: "I trust my ears;" the pure objectivist having suppressed human sentimentality: "I trust my measurements." A balanced view recognizes value in both. But in the case of a technical medium conveying emotional content, there are trade-offs.

Having individual perceptions and auditory health, subjectivists more often than not *disagree.* So their opinions may not be helpful to another subjectivist. If the only voices they pay attention to are each other – often echoing magazine biased ad copy and dug-in subjectivist reviews – they lemming those notions. OTOH objective data must be repeatable independently using similar methods and instrumentation, so objectivists more often than not will *agree.* More unbiased and with heads out of the sand, experts' reports are not sentimental opinions, so need not be taken with a grain of salt.

Beyond taking this difference heart, thus keeping in mind respect for its scientifically verifiable technical performance, we look beyond the phonograph to the science of the rest of the audio chain.

Besides the record player[13], what about the other links in the audio chain?

As audio reproduction is a chain that can sound only as good as its weakest link, we might explore these, even in a book that is mostly about the phonograph. Of course the following paragraphs cannot replace the large library of books required to do their topic justice, but they're a start. Let's explore just a bit aspects of listening acoustics, loudspeaker choices, calculating required amplifier power, use of subwoofers, and wiring it all together. Putting together a total system needs attention paid to all these. In the end it will be about the music, but getting there is a technical journey.

We've talked about "You Are There" (YAT) v. "They Are Here" (TAH) modes of sound reproduction, which are determined largely by the respective musical genres you wish to play. It could be said that TAH came first, from Alex Bell's telephoning for the first time "Come here Watson, I need you" to Edison's road shows, where hired singers imitated the sound of the phonograph to "prove" how faithful its reproduction! Today, pop\electric music is prized for the illusion that the musicians are right inside your home or car. Either listening space may not be acoustically controlled, but you've adapted to it, acoustic warts and all, and probably could recognize which room you're in blindfolded, only by its acoustic signature. TAH lessens the demands for good acoustics, and to an extent the recording quality, where any errors are masked by heavy effects processing or slammin' new groove that compels you to move. Since we were not in the studio when it was recorded or mixed, we can have no remembered reference for how the music sounded live, therefore we cannot judge the fidelity of reproduction. Tailored to the consumer tastes we grew up with, and purposely limited in frequency range and dynamics, TAH reproduction is designed to be forgiving regarding acoustics, speaker quality, and amplifier power in our world of personal players and boom box stereos. Although this book is about playing grooved records, the limited predecessor to the higher performing digital compact disc, TAH content can be satisfyingly conveyed by lesser quality data-compressed streaming.

On the other hand, You Are There (YAT) reproduction – like acoustic classical and jazz – must transport listeners beyond their listening room to the acoustics of the recording venue, be it a concert hall or jazz club. The music itself, being mostly based on acoustic instruments, has reference sound most of us remember from experiencing live a piano, acoustic guitar, the human voice, etc. Based on these references, we are able to judge the fidelity of reproduction. If we've experienced a fine concert or recital hall or club venue, we also have a reference for acoustics that are superior to an untreated home listening room, so treatment becomes necessary for the optimal YAT verisimilitude.

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[12] An exception is the too-fast moving world of the computer tech, where (except for Steve Jobs' philosophy embedded in more successful Apple products) ergonomics along with QC largely are skipped by manufacturers, and left to befuddle most users.

[13] Astute readers will have noticed I've said little about the spinning turntable itself; even used 40yr-old direct-drives since the Technics SP series and its imitators achieved performance that is as good as it needs to be in inaudibly low rumble noise and speed flutter.

Compared to the highly processed electric guitar and drum track music for TAH, acoustic music for YAT is usually very lightly processed. Containing full scale sound both in frequency range and dynamics, the demands on speakers and amplifiers can be much higher, even if typically it is played no louder. Along with a suitably powerful amplifier driving quality speakers driving an acoustically controlled room, YAT content took its first step toward verisimilitude with the LP. Then, designed to fit Beethoven's 9th on a single-sided CD, the compact disc upped the ante considerably, and for most people is still about as good as 2-channel stereo delivery needs to be. (Never mind that, as already discussed, the CD has been hijacked, and its capabilities largely wasted, for over-processed popular music. And that the worst new LPs are mastered from those over-processed CDs!)

Acoustic treatment is well beyond the scope of this book. Suffice to say that for rooms with hard walls, bass traps are needed for control below the room's Schroeder frequency (typically 250Hz), absorption is needed on side walls in the regions producing first reflections for clarity, and diffusion on the front and back walls reduces comb filtering and distributes energy in a more pleasing way. Possible to introduce here are speakers for that room, and amplifiers for those speakers. The process is hierarchical, where subsequent decisions depend on those made prior. We begin at the bottom…

Audiophiles, subwoofers, and 5.1 surround

Half of so-called audiophiles I encounter today reject real audio science in favor of the science fiction of misleading marketing, magazine ads & reviews, internet scammers' claims, and endless uneducated, close-minded gossip amongst themselves in FaceBook groups. The open-minded half learn what is real, and seek cost-effective audio systems. Either side may be "right" per individual taste, but those on either side defy comparison. Presumably you are reading this book not for 100 online pages echoing undirected and argumentative conversation, but because a researcher-writer took a lot of time (8yr and counting) to distill good information to its essence. You are not among audiophiles who care little and know less about acoustics, the physics of speakers, and meaningful performance features of electronics. How did recorded music enthusiasts devolve into *audiophiles?*

The term "audiophile" is new since the audio hobby began in the 1950s, after introduction of the LP – the era the author came to know as a teenage audio enthusiast. Then, I and fellow enthusiasts listened mainly to acoustic music – classical, jazz, folk, chorale, pipe organ – which we'd heard many times live, and had a remembered *reference* for verisimilitude. Fan magazines then published components' performance measured by independent Hersch-Houck Labs, and we readers correlated the data with our system before making any change, done in order to enjoy our music to the fullest.

After much audio equipment achieved today's higher level of performance, and after pop music introduced distorted guitars and unlifelike sounds that are so variable in timbre (tone color) that there can be no reference for live, rocker audio fans resorted to talking about subjective preference, and became *audiophiles.* In time manufacturers substituted eye candy for componentry, and consumers began bragging not about audio quality, but about the more and more outrageous money they paid.

This shift to the sentimental was seized by marketers, swallowed whole by reviewers, and sworn to by consumers who find unobjective BS more compelling. And who are often well heeled, even if *innumerate* (like illiterate only lacking in the math of audio science). Subjectivists coin meaningless terms like "focus," "soundstage" (localization and spatialization are the established terms), "detail" and "clarity" (actually super-sized brightness due to added "euphonious distortion"), with additions to the lexicon published monthly. With equipment improperly setup and endlessly swapped by trial & error, uncontrolled listening acoustics, "visual coolness" bias, and whimsical changes of heart with time, the constant disagreement in online posts implies that, unlike the ultimate consensus among objectivists, subjectivists' perceptions and opinions are fleeting, and useless to others. The more arrogant trumpet they "trust their ears," actually their *brain's perception,* thus relying on *lying sacks of shit.* Totally objective measurements of sound quality might not tell the whole story either, however if a component link in the audio chain measures bad, undoubtedly it will sound bad.

Furthermore, in the 21st century era of weaponized information, belief in "alternative facts," and practice "cancel culture," many audio fans (fanatics) also reject the expertise of actual audio engineers (even degreed & published, like the author) whenever it differs from psychologically confirming their biases, or have knee-jerk reactions to intended real help, causing them to dig-in (cognitive dissonance.) Those so afflicted with ostrich syndrome prove the Dunning-Kruger effect[14] – *they can't know what they don't know.* My countless attempts on internet forums and social media to reach out to audiophiles are often rebuffed. "This book is not for audiophiles," said one reviewer of this book, giving it one star. I accept that, for the moment, audiophiles such as he are beyond help. So this book is one expert's attempt to help those new to the hobby – or audio professionals new to grooved media – to learn and practice its science, and to enjoy 140+ years of recorded history mostly only available on records.

Another anomaly among many audiophiles is decrying *multi-channel (more than two) surround sound* for other than movies, claiming "it's not for music," nor is the related use of subwoofer(s). Always depending on recording quality, both can enhance almost any form of reproduction. 5.1 or 7.1 (5 or 7 main channels, plus a 0.1 channel for infrequent low frequency effects, LFE) complete the 2D horizontal 360º circle of directional sonic arrivals that human perception integrates into more lifelike spatialization and tone color (timbre), the holy grail of high fidelity. Full sphere (true 3D) microphone pickup and speaker channels would further verisimilitude. What cannot compare, for a "you are there" impression, is limiting reproduction to two speakers in a narrow 60º pie shape only in front. Nor does the unchanging sphere of uncontrolled listening room reflections compare to real acoustic signatures captured in 2D surround 5.1. Just as monophonic recordings sound best with a single speaker, and 2-speaker stereo (today's most popular form of audio entertainment) requires two speakers, only natively recorded 5.1 benefits from five. Decrying 2D 5.1\7.1 reproduction may just be another typical audiophile practice of bragging about what one paid for two speakers which more easily impresses others of one's conspicuous consumption than what one paid each for five or seven!

Then there is the highly misunderstood *subwoofer.* It reproduces the lowest two or three octaves of low bass that two or five or seven main speakers cannot without distortion, or without the poor interaction of very low frequencies (VLF) with typically uncontrolled room acoustics where main speakers must go. Subwoofer(s) can be placed in more advantageous acoustical location(s) in the room. Optimized for low bass, a subwoofer first reproduces very low frequency signal components redirected from the main speakers, leaving them to deal better with higher frequencies in their range, and whose direction can be localized. Additionally the subwoofer alone gets a ".1" *low frequency enhancement* signal (LFE – often in error called the "subwoofer channel"), a separate path <120Hz intended for occasional use in movies, such as for explosions. Useless for a kick drum, the only music I can think of that warrants use of the 0.1 LFE channel is cannon fire in Tchaikovsky's 1812.

The consumer specification "frequency range" usually is its –10dB point, but for subwoofers this spec is far less useful than for higher ranges. For these main speakers, the consumer-acceptable half-loudness perception at –10dB for frequency components above 200Hz becomes unacceptable below that, as the perceived attenuation will be much greater. [Fletcher-Munson, updated in ISO:226] As the music pitches lower, its power rapidly loses the bottom octaves of a bass guitar (from 41Hz, 29 for a 5-string), an orchestral bass drum or viol (31Hz), grand piano (27.5Hz), pipe organ (16.4Hz), or an action movie (packing a wallop down to 20Hz). The –3dB "f_3 response knee" professionals use is more useful for any woofer because at the very lowest frequencies –3dB is perceived as half volume!

Many audiophiles endlessly tinker by trial & error, and claim to have matched components to their favorite music genre. Loudspeakers have no *inherent* musical bias; they function only to reproduce sound within their range. For subwoofers, that is signal components below their crossover frequency (40~160Hz). "Faster" signals (higher tones and harmonics & transients [Fourier]), are handled off to the driver in the next higher range, woofer to squawker to tweeter. Only huge electrostatic speakers can be full range, but acoustic summation of two or more wavelength-sized electro-dynamic drivers integrates to full range in the air between loudspeaker & listener. Use only one driver per crossover range: off-axis two playing the same signal cause harsh comb-filtering and rapid falloff (beaming). Any perceived match between a musical genre and a speaker is just filtering by design shortcomings or uncontrolled room acoustics that emphasize or deemphasize a song's tonality (key\pitch), spectral

[14] Study that showed that those with the least knowledge on any subject are the most confident of their "knowledge."

makeup of the composition, individual track EQ in mixing, or overall EQ in mastering. Your best course? If not a better recording, then a neutral *monitor*-grade speaker in an acoustically good space.

The subject of multi-channel audio and subwoofers warrants far more discussion than fits here. The author's scientific papers on subwoofers at www.filmaker.com/papers.htm illuminate how to optimize reproduction of very low frequencies (VLF), especially for acoustic music content. Most pop music relies on *perceptual synthesis,* where non-reproduced low frequency fundamentals are inferred from harmonics emphasized by electric instruments. So pop music satisfies even on too tiny computer, boombox, and smart-speakers. However for low frequency replay approaching the limit of human hearing, the most cost-effective home implementation uses two 10~18in subwoofers positioned mid side walls. And bass redirection to spare you satellite-size speakers' LF distortion. Moreover in presentations of the paper to AES, ASA, and CAA, the author refuted Dr. Bose' claim that no binaural cues are audible below 90Hz, demonstrating audible spatial perception an octave lower to 45Hz. Vibrations that make trousers tingle your leg hair at a live concert – an experience only a subwoofer or two (or more) can reproduce at home, whether in stereo or surround.[15]

Choice of loudspeakers, and the amplifier power they need

I've owned quite a few different speakers over my 60+ years in the audio hobby and profession, ranging from 3in "full range" to very large commercial horns. However for home and studio use, I've not owned "floor stander" models (excepting a Leslie for my Hammond). My best speakers currently are largish, moderately expensive 3-way "bookshelves" (JBL LSR-32\6332 monitors).

The problem generally with more typical bookshelf speakers, with smallish woofers and cabinets, is low bass capability without distortion at full scale due to non-linear excursion when trying to move enough air. But the need for VLF (<100Hz) varies with content, ranging within musical and movie genre. An orchestra bass drum typically is "tuned" to C1 at 31Hz, heard\recorded together with reinforcing reflective resonances of the concert hall. Bookshelf speakers typically are able to reproduce a bass drum from only an octave above, therefore sounding like a dull thud. Pop\electric music may not seem to suffer as much because content is produced for small speakers. For example the practices of muffling low frequencies for a kick drum, or introducing harmonics in bass guitars as stimulus for phantom fundamentals in our perception, but that are not actually reproduced.

Additionally bookshelf speakers are usually 2-way: a woofer for low frequencies and a tweeter for high frequencies, crossing over about 2kHz, presenting errors where the ear is most sensitive. So most of the important vocal and speech, having energy mostly between 300~3kHz, is reproduced by the woofer, with dispersion narrowing, missing listeners off-axis, and not distributing energy around the room for reinforcement. The solution? 3-way speaker systems with a 3rd mid-range driver called a "squawker." Or a horn-loaded HF driver crossing over at 500~800Hz. My monitors mentioned have 12in woofers crossed over to the 5in mids at 250Hz to avoid "beaming" dispersion and IM distortion, but also (optionally) at 80Hz to two subwoofers flat to 30Hz. To reduce the somewhat troublesome crossover network itself, this monitor also can be driven by two power amplifiers, separately for the low and mid\high drivers. Self-powered speakers can have this "bi-amplification" built in, matching amplifier to speaker better, and reducing to inches the lengths of speaker cabling.

Distortion (% of bogus harmonics and sum & difference intermodulation products that "color" sounds) needs to be low, and frequency response needs to be as even (flat within a couple dB) as possible to preserve the tone color (timbre) of voices, instruments, and ambient sound effects. The next concern is *sensitivity:* how much amplifier power is needed for the sound pressure level (SPL) needed for your room. That measure is dB SPL at 1 meter when the speaker is driven at 2.83 volts RMS, which for a nominal 8 ohm speaker is 1 watt. The SPL falls off (gets softer) at the inverse square of distance until it reaches the acoustical *critical radius* C_R of any reverberant room, after which SPL remains constant. In a typical domestic space, C_R is a couple meters (6ft) or less. You have to estimate this to purchase the right speakers for your room, and to calculate amplifier power.

[15] In phonograph replay, typically there will be no binaural\stereo LF as disks are mastered monophonically below 150~250Hz.

Math alert (bear with me – it's easier than if first appears): You can calculate the amplifier power required for lifelike volume with distortion inaudible below ambient noise using...

P = LD^2 x 2^((SPL-SS)/3)

...where P is amplifier power per channel in watts, LD^2 is listening distance or C_R (whichever is less) in meters squared, SPL is the peak SPL per channel in dB (suggested for you in the examples below), and SS is the speaker manufacturer-specified sensitivity in dBSPL\1w\m. The math will be made clear in the following two typical examples – a small listening space, and a medium sized one:

Example A: For a speaker-to-ears distance LD of ~2m (6ft, less than C_R), at moderate listening peaks of 96dB, and compact speaker SS of 87SPL/w/1m, the peak power required in watts (w) is…

2^2 x 2^((96-87)/3) = 4 x 2^(9/3) = 4 x 2^3 = 4 x 8 = 32w per channel.

A prudent safety margin of 2 suggests a solid state amplifier rated at least 60w per channel.

Example B: For a medium size acoustically treated home theater with viewers seated an average LD of ~3m (10ft, more than its acoustically controlled C_R of 2m), SMPTE standard full-scale SPL of 105dB, and efficient speaker SS of 90SPL/w/1m, the power required per speaker channel is…

2^2 x 2^(105-90) = 2^2 x 2^(15/3) = 4 x 2^5 = 4 x 32 = 128w (use 250).

The difference might seem a bit surprising, but in *Example B)* for movies and music, the peak volume requirement is double, and the room is larger, although using speakers with double the efficiency. Your case might be somewhere in the middle.[16] In any case, less power is needed in spaces that are acoustically live, though uncontrolled reflected modes and comb filtering may cause altered tone color and your irritability. And at greater than a room's critical radius, which in a residential space (typically "undampened" using acoustical control appliances) can be a couple meters (6ft) or less, the falloff of sound level is room reinforced to be held constant, not the 6dB/dd *inverse square law* in anechoic spaces, or outdoors.

SPL can be checked using a sound level meter, which in addition to ranging will have two settings: "A-weighted" only applies to noise measurements at 40SPL or so, not higher; B-weighting at 60~80SPL, C-weighting (~flat) at 90+SPL. These adjust the meter reading to roughly correspond to human hearing perception, which falls off in sensitivity to low frequencies faster than mid and high frequencies [Fletcher-Munson, revised in ISO:226]. Most people listening to home music equate "life-like" SPL to about 10dB less than actual, and with dynamic peaks compressed in level a further 4~10dB or more above average levels, thus maxing ~90SPL. However movies may maintain their maximum 20dB headroom, reaching sound levels 10dB higher that requires10 times the power. And that the speakers chosen must be capable of the calculated power at all assigned frequencies.

I myself am a professional audio engineer since 1958 (my first paying gig after graduating RCA Institutes, and since with a BSEE, long-time member of AES and SMPTE, researcher\presenter globally at ASA\CAA and Deutsche Verein, author of two books – you get the picture). Therefore I may have a somewhat advanced notion regarding some of audio and acoustic measurements that I nevertheless think may be useful for those true audiofans. For you, a few practical "Gedanken" (thought) experiments whose only arithmetic is adding or subtracting decibels and watts.

Let's work backwards from the maximum sound level requirements that were calculated above for focused listening to life-like content, concert music or movies. Following the SMPTE cinema standard of 85dB SPL reference, 20dB below full scale (FS), which is the maximum peak sound from any individual speaker, which adds to undistorted individual channel peaks of 105SPL. Five surround channels sounding full throttle total uncorrelated 108SPL. Usually we adjust these power and sound pressure levels downward, because home listeners' opinion of "life-like" is about 10dB lower (75SPL) for "concert level" acoustic classical, jazz, or movie sound (there is no such reference SPL for pop\rock, as even hearing damage is not the limit). For movies (not unregulated trailers), the 20dB between "reference level" and "FS" is intended as "headroom" for preserving the dynamics

[16] Up to 3dB gain applies for correlated low frequencies from a nearby second speaker, such as records monauralized below ~150Hz.

that occur in natural music, speech, and nature sounds. Our system needs to deliver 75+20 or 95SPL peak power without "clipping" (distortion that can cause speaker damaging and instant migraines).

Now in your mind install in an acoustically controlled space a good quality home loudspeaker with a typical sensitivity of 89SPL of sound for 1w of power while listening at 1m (3ft) or measured with an SPL meter. In practice that scales to ~2w at 1.4m (4½ ft) "near-field" listening, or ~5w at 2.5m (8 ft). To deliver the 95SPL peaks at each of these three distances, each power amplifier channel would need to be capable of 6dB more power: 8w at 4½ ft, or 20w at 8 ft. A less efficient speaker of 86dB sensitivity (75SPL reference) would need 3dB, or double the power; a more efficient speaker of 92dB sensitivity requires only half. In all cases, less power will be needed in a typical acoustically uncontrolled room where reverberation reinforces the sound. But full SMPTE standard 85SPL, with peaks of 105SPL, multiplies all speaker and amplifier power limits by 10!

Next we explore a metric called *SINAD* – signal to total noise & distortion, or sometimes *THD+N* – that is an important quantifier of fidelity. It's simply the ratio between desired sounds and undesired ones. Like flowers v. weeds. The subjective criterion is whether SINAD artifacts are audible. If inaudible, all is well, and its causes can be ignored. Inaudibility will be determined by the environmental noise of the listening space, which we hope will mask inevitable distortion and noise artifacts in the audio. My control room and studio noise are both below 30SPL unweighted, or ~25 A-weighted, but ISO:226 (Fletcher-Munson) speaks mostly to desired auditory events below 200Hz, so we'll use unweighted figures in this example. 30SPL is an unusually quiet house. 30SPL noise is 55 dB below the 85SPL reference, or 45dB below the nominal home listening level of 75SPL. We can discern signals 15~20dB below noise, so to be inaudible, total SINAD in a quiet home must be 65dB below 1w, 2w, or 5w by their respective listening distances. –65dB is about 0.05%. So if amplifier SINAD is <0.05%, the amplifier will not be perceived to be adding bogus colorations to the sound. If a listening environment has typical noise of 40SPL, then amplification SINAD <0.15% is likely inaudible, etc. (We could go further and factor in whether other SINAD components generated in the audio chain accumulate to audibility, or mask other links' artifacts. (Loudspeakers and acoustic "ringing" can emphasize or deemphasize upstream SINAD artifacts.)

Amplifiers are specified like this: "A-weighted noise -90dB below full power." But residual noise is a fixed voltage, so 20dB lower at 1w – the volume at which average content occurs most of the time – that noise level is -70dB, still 5dB lower than can be masked in our quiet room above. However the amplifier might also be advertised as having "Harmonic Distortion 0.1% at 1kHz at full power." The slight-of-hand is that the distortion also does not diminish with power, but can be a significantly higher % at 1w normal listening level, and below. Say at 1w it's 1% at 1kHz, 40dB below signal. Now when distortion is included with noise, the combination has SINAD mostly comprised of distortion at -40dB; in a room for SMPTE movie viewing and with 40SPL noise, this SINAD is audible, adding bogus coloration to music and affecting suspension of disbelief watching a movie. Then to boot, the constant low-level crossover distortion of a mediocre AB amplifier is more clearly audible than they are amidst loud sounds, where fixed distortion is now a much lower percentage, drowned-out, or masked. We're not done, as distortion, comprising most of SINAD, is often considerably worse at lower and higher frequencies than 1kHz, where it advertises its best.

Measurements are not intended for their own sake, but for the sake of modeling our perception of sound, usually intended to point to what to fix if it's not good. Ultimately all that what matters is whether the artifacts such as noise & distortion can be heard. SINAD or THD+N (and especially non-harmonically-related IM) is a "perceptually correlated measure." Otherwise why measure it? Finally some common ground for audio subjectivists v. objectivists, not just "engineering excellence" but good sound is confirmed by good measurement & v.v. In my measuring + trained listening over the decades, as long as evaluations are done "blind" to obviate subjective bias, two amplifiers that measure in every way the same also sound the same – any differences are too low to be audible. Note that loudspeaker sensitivity is not a "marketing feature," but a critical engineering criterion that, varied by acoustics, is the sound level we can expect from a watt of amplifier power.

Interconnects (cables) and loudspeaker wiring

Three categories of audio cables beg understanding by audio enthusiasts: analog Phono pickup leads, line-level interconnects, and speaker cables. A fourth category is digital cables for SP\dif, HDMI, USB etc. connections are easy: either they work or they don't (no qualitative arguments here) depending on whether you've exceed the length at which digital bits are still decipherable. To measure these, you'd need to view eye patterns on a digital o'scope. So just try the cheapest first.

Analog cables are less robust. Due to capacitive loading effects, phono leads are critical, so are extensively covered in the book. Interconnecting solid-state players, preamps, and power amplifiers are easily done with the cheapest RCA cables salvaged in your spares box, where low impedance solid state electronics means little damage can be done by these coaxial cables, even 33ft (10m) long with their relatively high capacitance per foot and less than 100% shielding. But that long they may result in hum, electromagnetically induced from some external source, or caused by a ground loop.

To reduce induced hum using unbalanced coaxial interconnects (RCA cables), and no worse than any $400 line level "interconnect" (audio cable), make your own cables for <$10 a stereo pair using, instead of coax, STP (shielded twisted pair) to preserve the cartridges natural balanced wiring that cancels induced hum & RF noise. Shielded cat5 and up LAN cable works great. For turntable use, ground the tonearm and cartridge not through the shield, but only using a separate conductor outside and insulated from the shield, and leaving the shield disconnected at one end, usually the receiving end, and label the cable with an arrow pointing that direction. Except for this case, any "directional cable" hype is totally bogus, as audio signals travel through wires forward as much as reverse.

Full protection from either induced or ground loop hum requires "balanced" wiring, practiced by Ma Bell over three centuries as well as professional studios and broadcasters. Twisted pairs of wires are just as susceptible individually to induced hum, but it is cancelled at the destination by common mode rejection (CMR). To work, each leg of the cable pair must be the same in source, wire, and destination impedances. (Quasi-balancing a single-ended source can be done by installing in the negative terminal a resistor the same value as the positive's output resistor most output stages have to protect it from capacitive load instability.) Also fully balanced signal levels are 12dB higher (4 times the voltage) than consumer unbalanced wiring, so noise is naturally 12dB lower. But balanced output\cabling\input implementations are more expensive, therefore mostly used for large audio installations with long wire lengths, such as microphone lines in concert arenas of 250ft (75m).

No audio flows in the shield with either balanced or STP unbalanced methods, so both induced and ground loop hum are less likely to reach ears. The suggestion to ground the cable's shield at one end is to avoid a hum loop, and at the source end instead of the destination to avoid "pin1 problem" hum that too many consumer devices exhibit at their inputs; no noise in the audible range flows in the shield to be induced in the audio path. (In a high RF environment such as a broadcast station, ground the shield at the destination through a small capacitor.) Note that no "electron directionality" is implied. Hmmm, audiophile-priced cables labeled as to directionality? Do you suppose they're legit, made per my method above? I'd assumed they were BS like directional fuses, but I have no plan to experience them, so it's possible, I don't know. You could verify it with a continuity tester.

Contrary to what some marketers would have you buy, home system loudspeaker wiring isn't rocket surgery. At lengths up to 16ft (5m) for use with flat 8ohm impedance speakers, there is no truth to needing more elaborate or expensive cable than #16 AWG (wire gauge) zip cord. Avoiding audible differences is addressed simply by sizing the cable gauge by wire length and speaker impedance, in Ohms (Ω). In a medium-large home theater, 14AWG conductors work quite well enough for 50+ft lines to fine loudspeakers with a relatively flat nominal 8Ω impedance, or for 4ohm 11AWG (or two 14AWG conductors paralleled for so-called "bi-wiring"). Other than resistance over the length of the cable run – AC the same as DC – there is no applicable "cable impedance."

Best practice is somewhat larger gauge. Speakers with rocky impedance curves having minimums lower than labeled, so increase wire gauge in proportion to the minimum impedance compared to its advertised impedance. If a nominally 8Ω speaker's minimum impedance is 6ohms, a 25% decrease (and a typical deviation), increase by 25% the wire gauge (easier to calculate in circular mils), e.g.

from #16 to larger #14AWG. Otherwise the frequency response reaching your ears will have taken on the inverse of the speaker's impedance curve, thereby suffering sound with damaged tone color.

The future of audio reproduction: WaveField Synthesis? BACCH? Ambiophonics?

Invented by Blumlein in the 1930s but not available to consumers until Emory Cook cut two separate grooves in an LV, stereo adds to monophonic audio an imprecise frontal "soundstage" and partial spatiality subtending a 60deg pie-shape in front to reproduce sounds from any direction. While the de facto standard for entertainment audio, it suffers from a wider gap for sounds near the middle than at the edges, harsh comb-filtering for phantom-centered soloists, and confusion between where separated speakers present auditory events and where the brain expects cues from our pinna (outer ears). It is for these reasons that development is ongoing to improve upon stereo's quality.

To more completely satisfy our perception of natural full-sphere-3D sound, mathematician Michael Gerzon invented Ambisonics, initially using an array of four microphones and at least twice as many speakers on two or more horizontal planes spaced vertically. The system supersedes stereo in realism, but as the microphones are coincident (at a single point) in space, it relies on inter-aural level differences (ILD), but not also the time differences (ITD) of natural hearing. 1st order WXYZ recordings need four discrete channels, however higher-order Ambisonics (HOA) have 8 to 64 mics.

Capturing both ILD and ITD, binaural sound is what we hear naturally with two ears, one on either side of an approximate 650us diameter sphere – an average head size at an average speed of sound. Then our own individual "head-related transfer function" (HRTF) – as unique as our finger prints – acoustically "encodes" arriving sonic signals so our brain perceives not only an enveloping circle of 5.1 surround, but immersion in the full sphere of natural hearing, with up & down to complete left-to-right. A very realistic impression with headphones is implemented mathematically in BACCH software; Edgar Choueiri, a Princeton rocket scientist (no, really) is its auteur.

The reason for Ambiophonics (crosstalk cancellation) is to extract from ordinary stereo music and 5.1 surround movies any true binaural audio cues for "virtual headphones" over loudspeakers, reproducing more realistic sound using the listener's own individual head-related transfer function (HRTF). The result with ordinary 2-channel recordings can reproduce a full 120+deg recording angle, with accurate localization ("soundstage"), no "hole-in-the-middle," and no comb-filtering for important soloists in the center. Implementing a second pair of close-spaced speakers in back and a second instance of a software decoder, such as AmbiophonicDSP, enables 5.1-compatible surround sound with enhanced envelopment for movies & games. See http://www.filmaker.com/products

Another highly enveloping technology is Wave Field Synthesis (WFS) developed by engineers at Delft University. It uses a long line of speakers along the walls to synthesize in 2D (not 3D – yet) the sound pressures everywhere in the room, so you can walk around it, yet performers remain in position, as though they are in the room with you live. Like 2½D Dolby Atmos (a hemisphere of sound), its cost for installation & content production is justifiable for museum and theater exhibition.

I've given hundreds of demonstrations that re-immerse listeners in full-sphere 3D recordings using a hybrid Ambisonic-Ambiophonic microphone and 10-speaker layout (plus subwoofers). My High Sonic Definition 3D (HSD-3D) is a stereo- & surround-compatible leap toward verisimilitude.

Which is better: digital or analog?

If you've been with me so far, you already know the most important question is enjoying the recorded content (music, spoken word, movies, nature sounds). And that subservient to that are any technical issues for getting the best sound. Reducing electronic and acoustic artifacts is like cleaning a window in order to view the beautiful vista beyond. A century of recorded history is available only in its original medium: analog wax and bakelite cylinders; magnetic wire; SoundScriber dictating embossings; 78rpm shellacs; mag tape; mag or optical sprocketed film; and "vinyl." From the early 1980s, practical analog-to-digital and digital-to-analog converters were developed along with digital processors, tape, and hard-drives that could record audio better, and today pristinely.

From this book's first publishing in 2017, its back cover and several times inside have said vinyl is *not* "better" than digital – I've practiced a long time in both and am fully aware that it is not. (On p1 of this Update I beg readers to interpret the book's title as "The Better Sound of the Phonograph [than you are probably hearing from it now]: How come? How-to!") Those who don't get my intended meaning are likely hyper-sensitized by the online and in-print battle that was waging prior, with polarized sides unwilling to concede: *You don't have to choose – you can have both.*

Digital has the *potential* to be our very best technology for audio recording, storage, distribution, & reproduction. The caveat is that it is also has the highest potential for abuse, and unfortunately it has been. Though the CD has about 93dB signal-to-noise (dithered) and 15~20dB more perceivable dynamic range (because we can hear sounds below noise), the Volume Wars that, within its limits, infected radio in the 1970s are trivial to wage with digital signal processing (DSP) in digital audio workstations (DAW). Instead of allowing "headroom" for natural dynamics of 16dB for CD music and 20dB for movies, DSP is used to crush sounds to the ceiling, reducing CD's "useful" bits from 16 to less than the 12~13 that is *no better than analog tape or vinyl.* The result might be suitable for listening to pop music on earbuds amidst city noise, but not for high-fidelity listening in quiet homes and on good equipment, the goal of the audio hobby the 1950s, when quality was hard to come by.

Digital is a superior format, but many are habituated to its sounding worse. Initially energetic and exciting, soon it is harsh and fatiguing. On the other hand, "vinyl" is often described as "warm" – implying more natural and listenable. From the era of vacuum tubes and lathes, it could not, like digital, be "normalized" to the ceiling, then raised even higher until 4% of samples are *clipped,* as some digital labels practice, because neither a cutting nor a replay stylus can execute the right-angle turn of a topped waveform. Furthermore, an LP today produced from the same over-processed digital master – often just a distribution CD itself – sounds just as poor. Except for acoustic music such as classical or jazz that are usually not over-processed so as to sound accurate and life-like, the problem with popular music in digital form is less the medium, but much more the overdone content.

Why is the situation not the same for acoustic music, such as classical or jazz? It may be because both its producers and consumers have more conservative tastes. But it's mostly because they've heard orchestras live in concert, without any intervening sound reinforcement (PA) between instrument and ears. Therefore it is a market having a remembered *reference for fidelity*. Electric pop seeks novel sounds we haven't heard before live, so we have little reference for or even regard for reproduction accuracy. A few listeners might have appreciation for multiple genres, but they have cultivated different expectations for quality for each. More typically they are two entirely different worlds of sound. For new releases, uncompressed (level normalizing or data reduced) digital is designed and proven superior. But 70+yr of recorded history is preserved in records.

[reserved for future update]

More…

Errata in book v8/3/17 (corrected in subsequent editions)

Typo on p60 – "include<u>s</u>" should be singular.

Typo p62 – change "fix" to "per<u>fect</u>."

Typo p63 – change to "<u>a</u> few of the hurdles."

p107 – caption should read (do not insulate cartridge from arm but remove any ground strap)

In versions prior to 8/3/17, the drill size "7/16in" in the captions on pages 108 & 109 are typos – the correct sizes are in the text.

Remembering Grammy-winning mastering engineer Clair Dwight Krepps 1918~2017

photo Robin Miller

I interviewed recording & mastering engineer Clair Krepps at his home 8/31/17 for *The Better Sound of the Phonograph*. A Navy radio\radar technician, he became a legendary recording engineer at Capitol Records, MGM, Atlantic, and in 1965 his own Mayfair Studios. On the back wall above can be seen the RIAA award for engineering Roberta Flack's *Killing Me Softly;* not visible around the corner, a Grammy for Nat King Cole's *A Christmas Song* by Mel Tormé. On the floor is one of eight active "channel strips" of the console he designed & built with brother Edgar for his industry-first 8-track recording studio. He was colleagues with Les Paul, Emory Cook, Normal Pickering, Walter Stanton, and other pioneers in the high-fidelity era of the 1950s. He served on Bergenfield NJ's Board of Education, and taught disk mastering aspirants enrolled at the Jon Miller School of Recording Arts & Sciences [the author's brother]. One of his "tricks" he taught was to mix to mono stereo sounds below 250Hz (others used 150Hz) for better reproduction at home and on juke boxes of bass without groove-hopping. "Not a *heavy compressor* guy," still his 1964 mastering of the hit single "Do Wah Diddy" by Manfred Mann was widely regarded as the loudest 45 ever recorded! And he was criticized by colleagues for disrupting their "code of reasonable volume," portending what has indeed come to pass. Shortly after my interview, he turned 99, and received a life honor by the Audio Engineering Society (AES), which in 1948 he co-founded. In another month, this master of microgroove was gone, but most deservedly not forgotten.