The Schlockwood Laboratory
Brea, California • www.schlockwood.com

Engineering Brief

The Schlockwood StereoScriber™

An Experimental Cutterhead for Phonograph Disc Recording

BACKGROUND

Rumor has it that vinyl phonograph records are making a comeback. Vinyl fans go so far as to claim that these plastic discs actually sound better than today's technically-perfect, all-digital methods of storing and delivering music and speech. It really doesn't matter whether the movement has foundation in fact or merely represents a brief surge of nostalgia. Some of us who have a long association with sound recording still find delight in "lathe-cutting" phonograph records… just for fun.

Phonograph disc recording equipment frequently pops up on eBay and in junk shops, and ranges in capability from the 1940s 'suitcase' recorders for home use to professional gear, once the mainstay of radio stations and recording studios. These machines date to the 'hi-fi' era or earlier and are strictly monaural.

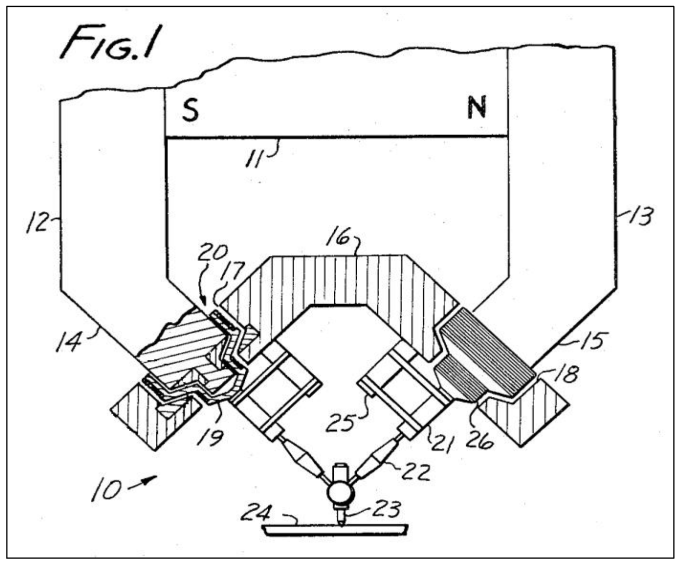
INSPIRATION

A growing number of YouTube videos on the subject, along with the Internet *Lathe Trolls* forum lathetrolls.com, testify to a continuing interest in disc recording technology. Enthusiasts' success stories range from squeaky reproduction of barely-recognizable tunes to truly spectacular audio quality. Construction techniques vary from innovative use of shop scraps to beautifully-machined and 3D-printed parts.

The electromechanical disc recording head, or 'cutterhead,' remains the weakest link and the ultimate challenge in making quality stereo phonograph records. The projects described here were undertaken with only a rudimentary understanding of mechanics and physics, and just a tinkerer's skill with home-garage tools, notably a band saw, a drill press, a belt sander and the usual assortment of files, hammers and screwdrivers.

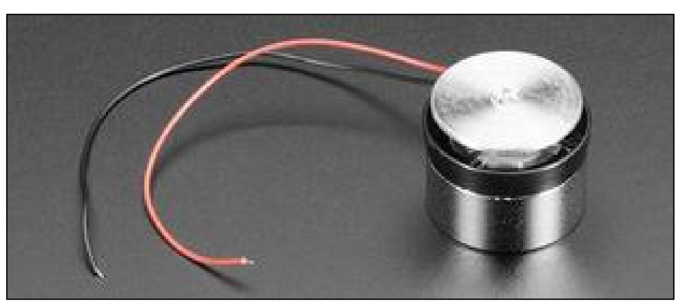
THE DESIGN

To properly credit stereophonic disc recording from its inception would require a historical treatise going back to early acoustical (non-electronic) recording methods from the turn of the last century, and would include myriad citations and credits. For the purposes of adhering to what became the world standard for stereo records, we'll simply acknowledge the '45/45 system' invented by England's Alan Blumlein in 1931 and commercialized in the US by the Westrex Corp. in the late 1950s. This drawing from Westrex patents illustrates the firm's fundamental cutterhead concept.



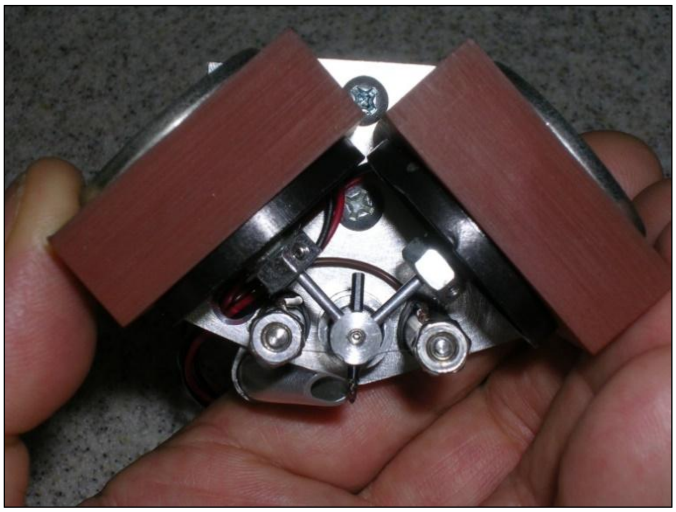
There's almost enough information here to try to copy the Westrex design directly. But with the limited skills and materials at hand, it seemed better to use simple off-the-shelf component parts wherever possible.

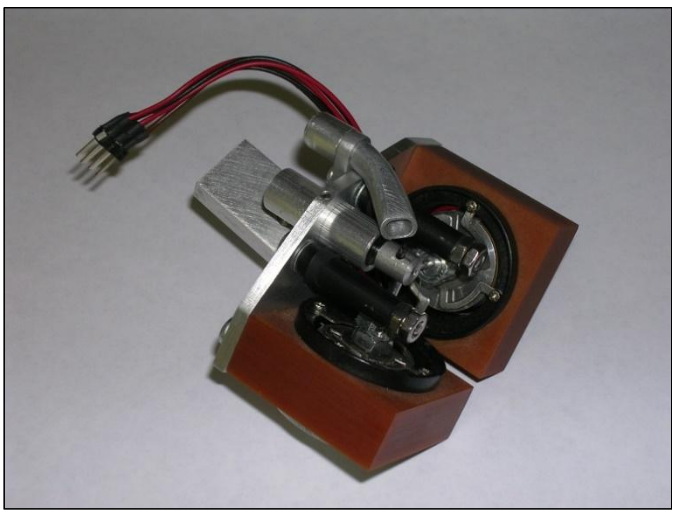
"Audio exciters," or "transducers" sold on the Web (example below) are essentially loudspeakers without cones. They are intended to "…turn your coffee table into a giant loudspeaker," undoubtedly giving questionable listening satisfaction.



Nevertheless, these exciters are well-made and rugged, and capable of handling about 3 watts of audio without complaining.

Quickly jumping through numerous fits and starts to the finished cutterhead, here is a series of pictures showing how the exciters were mounted and coupled to the recording stylus. Some notes follow.





CONSTRUCTION NOTES

1. The exciters are held in blocks cut from 1/2" phenolic sheet stock. The blocks were drilled with a 30mm Forstner bit and slotted on the mounting edge to pinch the exciter and hold it firmly in place. Screws through the backplate into tapped holes in the phenolic secure the blocks.

2. The exciters have lightweight aluminum voice coil bobbins that terminate in a 5mm threaded stud. A 5mm nut was spun-onto the stud, which was then drilled axially to accept the pushrods from the stylus holder. A setscrew through the side of the nut secures the pushrods, which are 1/16" hard aluminum alloy welding rod.

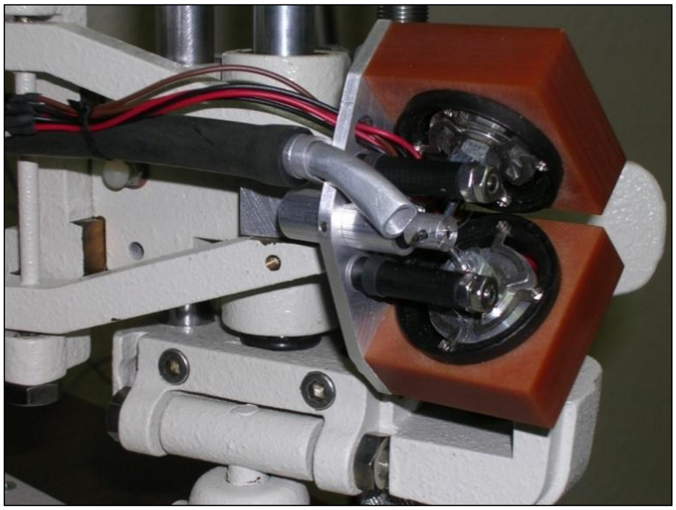
3. The stylus holder is a short piece of 1/4" aluminum rod. The stylus hole was drilled 18° off perpendicular and the cutterhead is mounted with a complementary 18° forward tilt to return the stylus to a 90° relationship with the disc surface. 18° fits the range of the industry-designated 'vertical tracking angle' spec. A setscrew locks the stylus in place.

4. A 1/16" stainless rod is pressed into a hole at the back of the stylus holder and runs through the center of a rubber damper line, a length of Buna-N rubber O-ring material squeezed into an aluminum tube. The tube is secured to the backplate with a setscrew. The damper allows the stylus to move in any direction, except fore and aft, without a 'twang'… well, sort-of, as we'll see later.

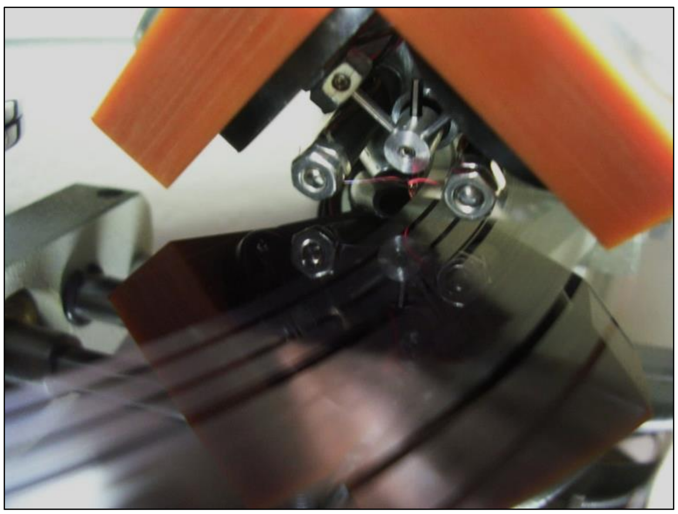
5. The rest of the design is as-required: a couple of fiberglass standoffs to secure the stylus heater wires and a vacuum nozzle positioned as close to the stylus as it can get.

Mounted on a 1950's-vintage Rek-O-Kut lathe, the recording test setup looks like this.

Rek-O-Kut lets you up-end the overhead mechanism for easier access to the stylus. This shot, taken in that position, presents a bottom view that shows the electrical and vacuum connections.



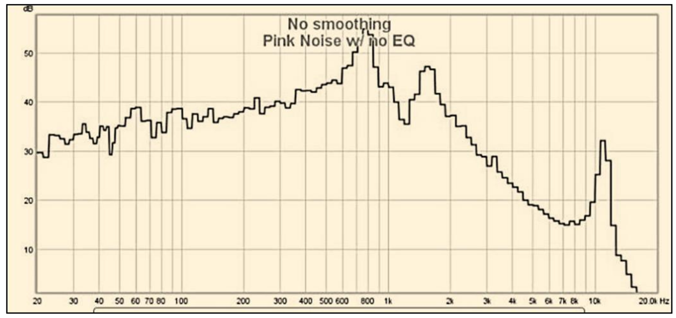
Here's the cutterhead in action making a test recording on a lacquer blank...



...and the test recording in playback.

INITIAL RESULTS

As one 'practiced in the art' might well have predicted, the test recording was only barely recognizable as music. But assuming the optimistic attitude that malfunctioning devices can be forced to operate properly, the first step in the head's redemption involved recording wideband pink noise, followed by RIAA playback into a PC-based RTA (Real Time Analyzer). Here's the raw overall response.



That's at least a ±20dB amplitude response error! Can that really be fixed?

BFEQ (...?)

From the early days of hi-fi disc recording, professional cutterheads have utilized motional feedback to tame mechanical resonances and help secure flat frequency response. But this cutterhead doesn't lend itself well to the additional coils and magnet assemblies needed for this exercise (see the Westrex drawing). The alternative? BFEQ. What does that stand for? Brute Force EQualization!

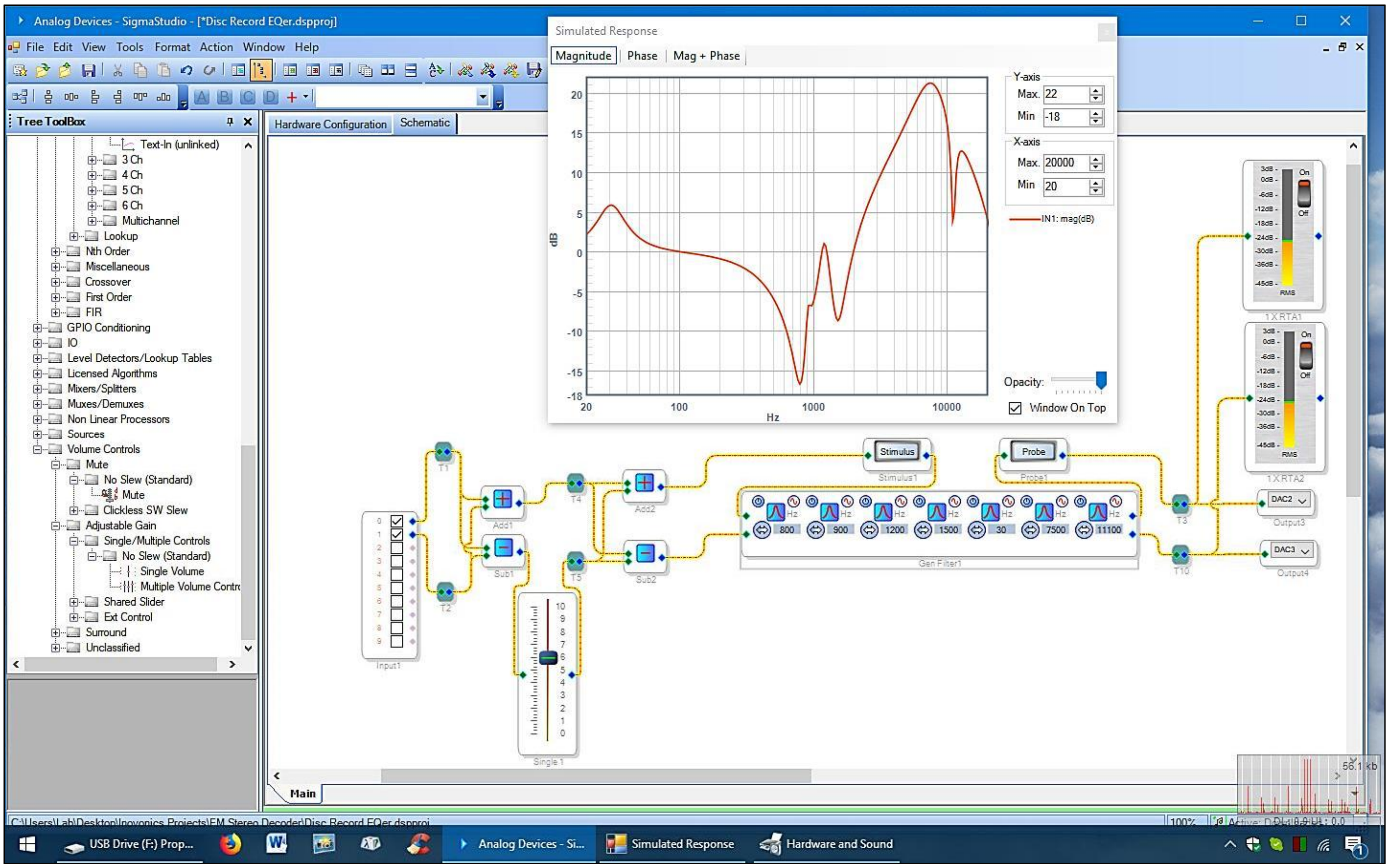
There are pronounced, sharp resonance peaks at 800Hz, 1.6kHz and 11.1kHz; a dip at 1.2kHz, plus more gentle rolloffs at both the bottom and the top. Trying to correct these deviations with graphic or other analog equalizers would probably have been a frustrating failure. But modern DSP (Digital Signal Processing) techniques allow quick, precise and repeatable response corrections that may easily be fine-tuned in real time.

The Analog Devices Corp. has a "DSP for Dummies" utility they call SigmaStudio. You can pull digital building blocks from a menu to create an analog-like schematic diagram on your computer screen. Sigma features a host of audio processing functions that allow you to generate wild EQ curves.

The Sigma development system consists of a small circuit board cabled to the USB port of your computer. The DSP chip resident on the board does all the A-to-D and D-to-A functions internally, and the board accepts TRS mini phone plugs for stereo analog audio-in and audio-out.

You could probably use the development platform for everyday operation, but compiling the project to code and blasting that into memory on your own dedicated board should the ultimate goal. The Sigma Studio concept is versatile, powerful, and well worth getting to know for all manner of audio signal processing requirements. The DSP chip, itself, is relatively inexpensive, but obviously would require additional component parts and a thoughtful circuit board layout to create a convenient working system that could be embedded in any worthwhile audio construction project. More about this later.

The computer screenshot below shows how SigmaStudio runs on the PC. This is the circuit uploaded to the development board for determining the equalization values.



This circuit has seven(!) cascaded sections of parametric equalization. The center frequency, the 'Q' (shape) and the gain at the peak or dip may be set for each section independently as you listen and tweak in real time. The equalizer was initially placed in the playback path and EQ parameters carefully adjusted to flatten response of the recorded pink noise while watching the RTA screen. Sigma's Simulated Response graph (inset) essentially mirrors the raw pink noise playback curve pictured earlier.

When this equalizer was finally placed in the recording signal path, the cutterhead response irregularities had indeed been compensated as so fervently hoped. Playback was then within 2dB of flat from 30Hz to 12kHz. Music sounded much better, but there were still some sonic issues.

OTHER CONSIDERATIONS

Using huge amounts of EQ to flatten response works only to a certain point. The mechanical resonances are still there, causing a certain amount of ringing or 'holdover' at the resonant points. This makes music components at precisely those frequencies sound a bit 'smeary' despite a relatively flat overall frequency response.

Another notable shortcoming was that stereo separation didn't seem quite up to snuff. The disc did not sound as 'wide' as the digital source recording, nor was the stereo image as well defined. This is undoubtedly due to the rigidity of the stylus holder and its pushrods. Note the little flexible coupling linkages (#22) that Westrex built into their cutterhead.

Sound sources central to the stereo image are strictly monaural program components, pop music vocals are a good example. Mono sources impart only lateral groove modulation. Unlike stereo loudspeakers, the cutterhead exciters are purposely driven out of phase. As one exciter is pushing, the other is pulling. This causes the stylus to twist back and forth to modulate the groove laterally.

Out-of-phase program components, the stereo directional and ambience elements, cause the exciters to push and pull in unison, again the opposite of loudspeaker practice. And because the coupling mechanism is quite rigid, the voice coils have to flex somewhat to move in and out simultaneously and modulate the groove vertically. Because these transducers are stiff, this undoubtedly accounts for a certain amount of distortion in the vertical movement, which would manifest mainly in left-only or right-only soundstage locations, and in room reverberation and other ambience effects.

A stereo matrix/de-matrix circuit ahead of the equalizer (see schematic) translates the left and right channels into L+R 'sum' and L–R 'difference' signals, and then directly back again to left and right. When about 6dB additional gain is introduced in the L–R difference domain, the stereo image is improved dramatically. Measured separation is frequency-dependent and collapses to a somewhat pitiful 10dB at some frequencies, but with this difference-signal enhancement music sounds more as it should.

AND HOW DOES IT SOUND?

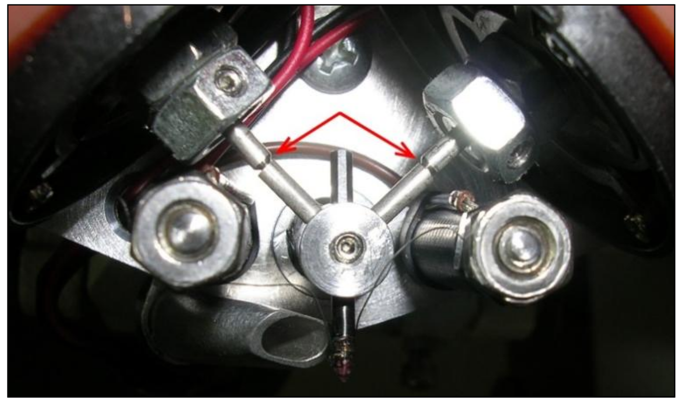
Here's a link to an MP3 file from the very first cutterhead test:
https://www.dropbox.com/s/jaut25lg9z8upbt/Music%20Test.mp3?dl=0

This short clip from a classical piece was digitally recorded by a friend at an orchestra rehearsal. The whine you hear in the background is from AC-mains ground loops, as the unbalanced audio is patched from the mixer to the test bench, and then back to the lathe.

AFTERTHOUGHTS AND MODS

A few weeks following its initial tests, the cutterhead received these few modifications.

1. Tiny ball-and-socket-like joints were added to the pushrods that couple the drivers to the stylus holder. These give each driver freedom to move along its axis without the other driver having to twist from side to side. Here's a photo showing these 'articulated' pushrods.



2. The rubber-damped rod securing the stylus holder to the backplate was modified to reduce stiffness and allow freer movement. This greatly improved stereo separation and also helped flatten the uncorrected frequency response. Only five equalizer sections were then required to make a flat recording.

3. A fast, unobtrusive wideband peak limiter was incorporated in the drive electronics, along with an independent high frequency 'acceleration' limiter. The wideband limiter protects against overcutting into adjacent grooves on bass peaks, and the HF limiter controls top-end energy.

ACCELLERATION LIMITING

Treble pre-emphasis is used in disc recording, FM broadcasting, cassette and reel tape recorders, and in certain other analog audio systems. When complementary listening de-emphasis is then applied, system noise is reduced. This does, however, create a frequency-dependent headroom situation, as pre-emphasis raises the level of high frequencies over the 'full modulation' level of lower ones.

A distinctive frequency-selective limiter can quickly reduce just the treble range for high-end energy like vocal sibilant and snare drum peaks, and then just as quickly restore full response. This mitigates the headroom shortcoming without appreciably dulling the sound, and also helps protect the cutterhead from overload and damage.

Frequency-selective limiters are universal in FM broadcasting; the CBS Labs *Volumax* was a noted early broadcast product. The Fairchild *Conax* (constant acceleration) limiter was used early-on in monaural LP disc recording, and mastering lathes from Neumann and Ortofon incorporated acceleration limiting as a part of their built-in signal path processing.

A SECOND CUTTERHEAD

Adding the little 'ball and socket' joints to the driver pushrods proved that each driver requires some compliance (flexibility) beyond its own axial push/pull motion. But these little joints require substantial loading to keep the ball in its socket as the driver shoves and yanks. Such loading is provided by the springiness of the stylus holder's rubber mount, but the force required to keep the ball firmly in the socket puts the equivalent of DC bias on the driver. This compromises its actuation symmetry and leads to even-order harmonic distortion.

The drivers initially chosen for the project are not only very stiff, but the voice coil is sprung (supported) by a means that resists movement in any plane other than along its axis. What is really needed is a driver with a voice coil held in a manner that allows it to angle somewhat with respect to its axis.

These popular and inexpensive 'coin exciters' are available from multiple Web sources.

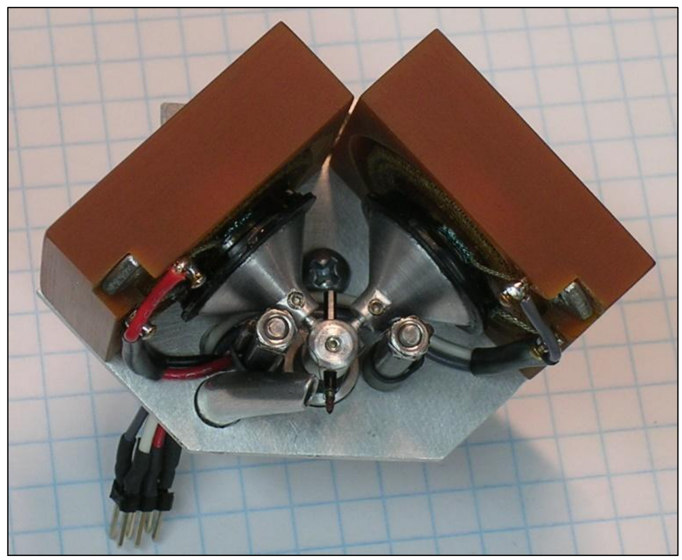
Not only are these exciters very compliant along their axes, but the voice coil plane may also be tilted by a considerable amount without the coil rubbing against the magnet structure.

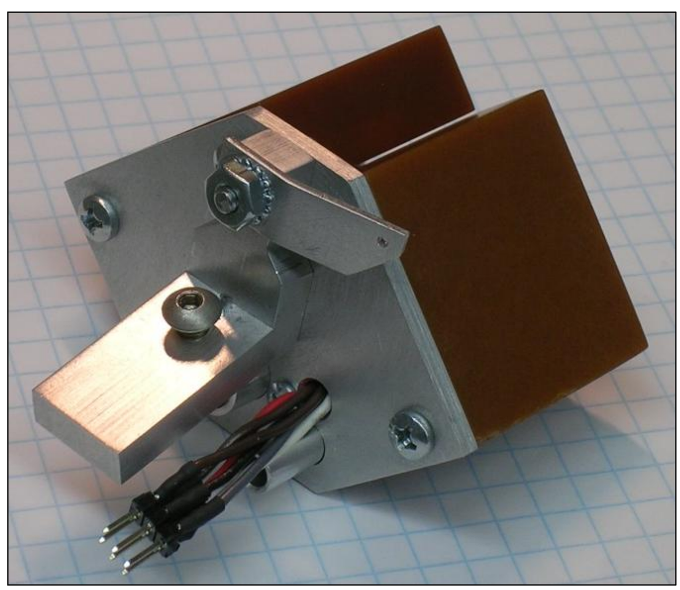
Like its predecessor, this exciter lends itself to being embedded in a block of easily-worked phenolic, using a urethane adhesive to hold it firmly in place.

The top photo in the next column details modification of the exciters and shows how they are seated in the phenolic blocks.

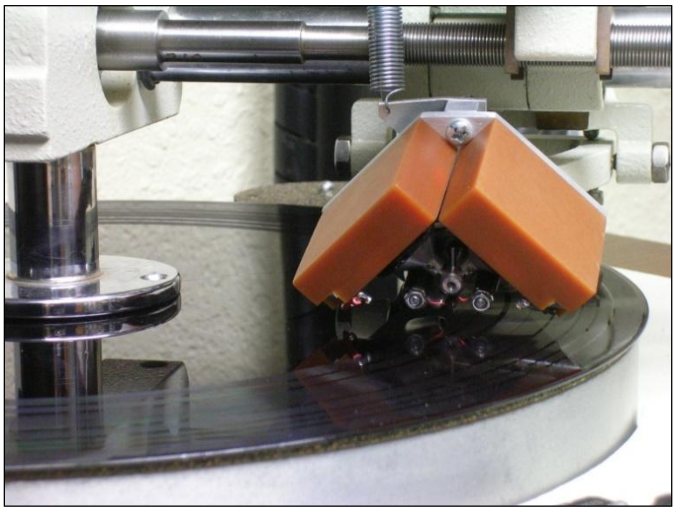
Mouths of the little aluminum 'perfume funnels' were ground down to match the voice coil diameter, and are fastened to it with cyanoacrylate adhesive. The funnel necks are plugged with aluminum rods, which are then drilled to accept the stylus holder pushrods. Otherwise, the balance of this design closely parallels the first cutterhead version.

Here are front-and-back shots of the finished head.



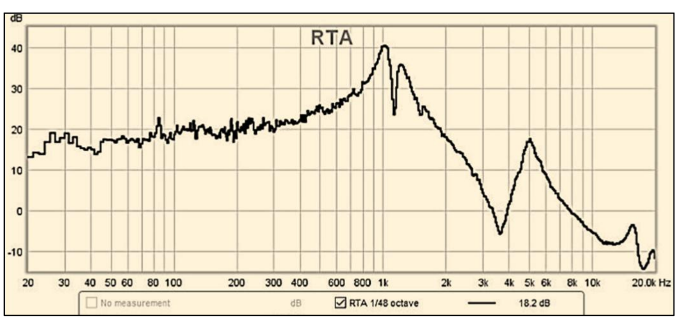


Here are shots of the second cutterhead in operation on the Rek-O-Kut lathe.



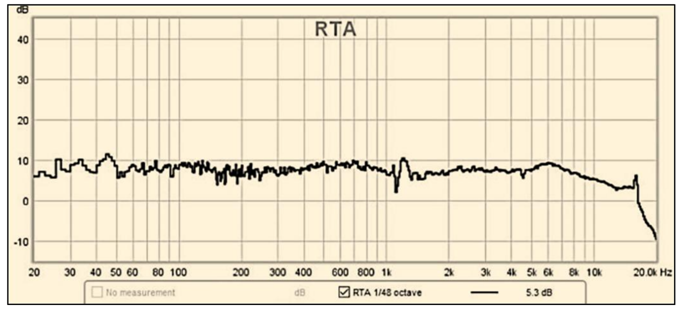
AND THE RESULTS?

It came as no real surprise that an initial RIAA playback of flat-recorded pink noise was as ghastly as the first version of the head. A bit worse, actually, although the top end appeared to hold more promise. Here's playback as displayed by the RTA software.



Amplitude response error is nearly ±25dB! Again we need to turn to aggressive, brute-force DSP equalization to correct this mess.

Pink noise playback was routed through the 7-stage parametric equalizer, which was tediously adjusted and readjusted to bring the amplitude response to optimum flatness. This turned out to be about ±3dB most of the way to 15kHz. A sag between 8kHz and 15kHz could have been picked up with an eighth section of parametric EQ, but as top-end performance in disc recording is somewhat fleeting at best, this overall response was judged to be quite good enough.



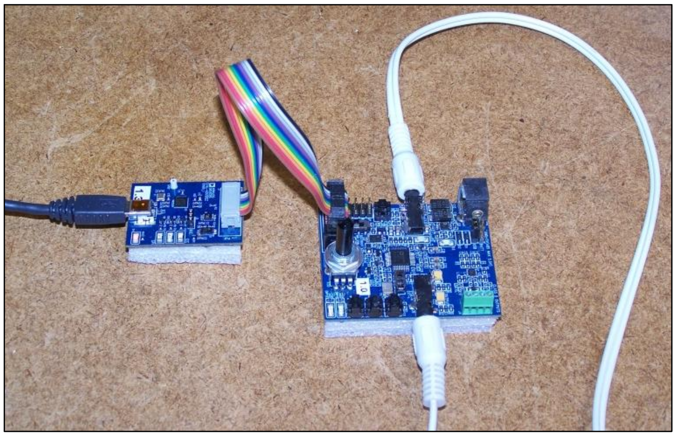
Sigma's simulated plot below depicts the amplitude response of the equalizer. Again, it looks pretty close to the inverse of the raw, unequalized playback picture.

Once the equalizer was returned to its rightful place in the recording path, subsequent pink noise recordings exhibited substantially the same flat response as predicted by initial equalization in playback. The acceleration limiter was then fine-tuned for this cutterhead and its own, very specific EQ. The full SigmaStudio DSP schematic diagram is shown on Page 9 with the circuitry detailed on Page 10.

Various music tracks were cut and evaluated for audio quality. Despite the second cutterhead requiring a bit more of the drastic EQ, its performance was subjectively judged as superior to the first version of the head. You can hear the second cutterhead here: https://www.dropbox.com/s/l50j5nqtz01r37e/Cutterhead%202.mp3?dl=0

DSP ELECTRONICS UPDATE

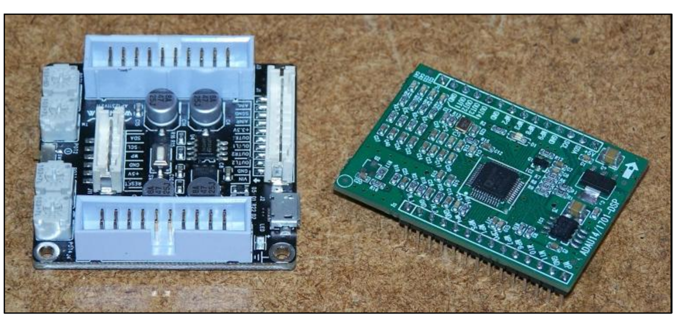
The initial discussion of the DSP-based equalizer and associated signal processing circuitry was based on using the Analog Devices development platform, a small evaluation circuit board that runs off a USB port on any Windows computer. Here's a snapshot of that very board in use with this project.



This is the firm's 'Mini' evaluation platform, EVAL-ADAU1701MINIZ. The 'Mini' is scaled-down from their 'Full Capability Engineering Evaluation Board,' but still has plenty of DSP horsepower for projects like this and many others. The cost of this board is about $200, a third the price of its big brother, and it still requires a USB Interface Board (the smaller board off to the left), which is another $85.

Digital signal processing does add significant expense to the project, especially if this evaluation board is going to be embedded as an integral and permanent part of the cutterhead driver system. In that case the board could not conveniently serve its intended function as a versatile lab-bench tool for the multitude of interesting projects that this DSP family can support.

Fortunately, enterprising "offshore" suppliers have come up with a couple of compatible alternatives priced in the $20/$40 range, a fraction of the cost of the 'real thing.' These two offerings are currently listed on eBay and can be found searching: ADAU1701.



Like the Analog Devices board, these two alternatives include both DSP and RAM chips. This means that once your DSP design has been finalized on your PC screen, you can upload it to the onboard RAM. After that the board will automatically boot to your project and run with no computer required.

Several GPIOs (logic connections) come out on pins to let you drive indicator LEDs for display of circuit action. (See notes at the bottom of the DSP schematic on Page 9.) The board's I/Os accept analog audio or I^{2}S digital audio, but you still need the $85 USB interface board to connect to your computer.

YOU CAN DO IT!

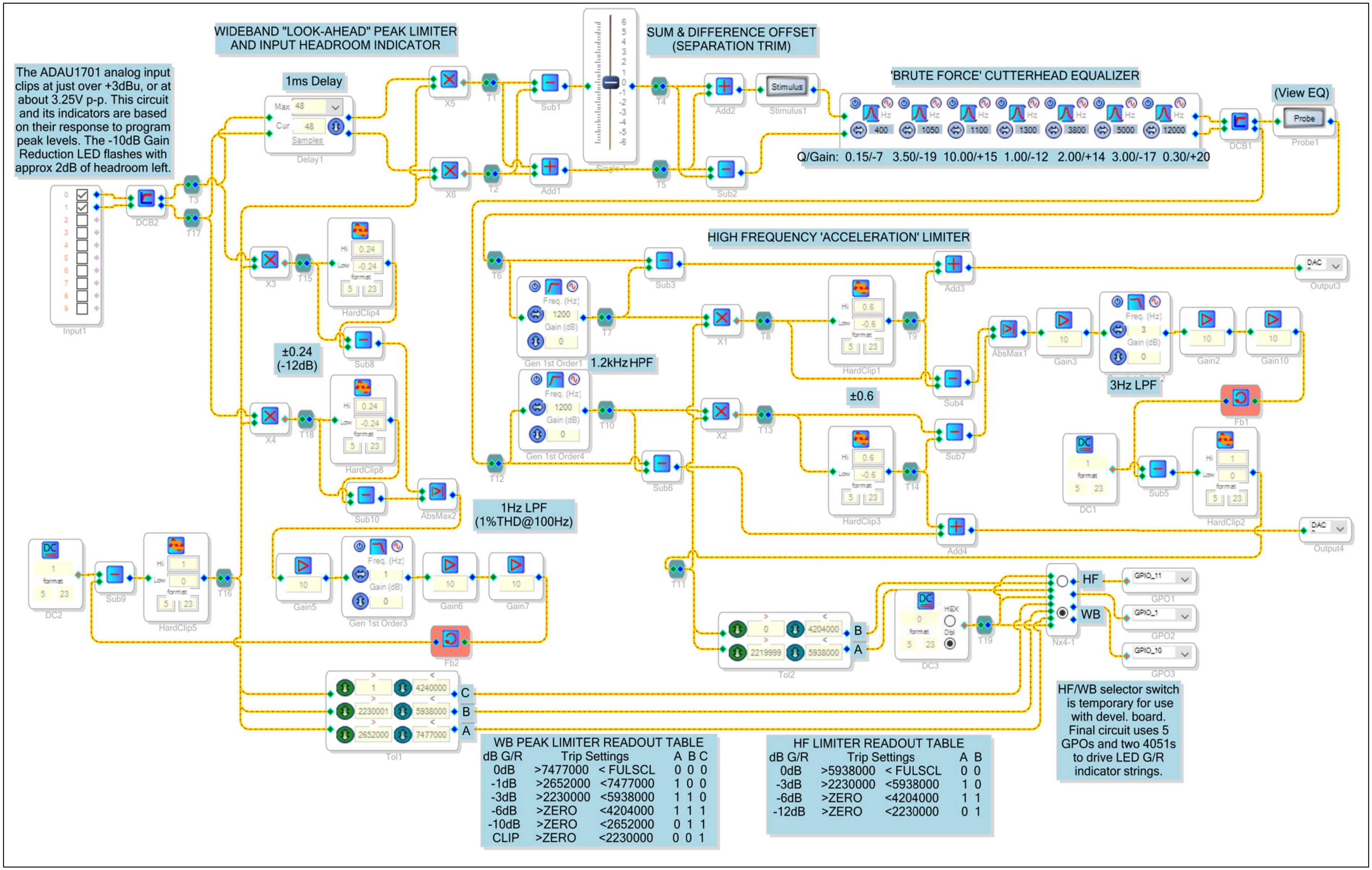
Chances are that you already have many of the raw materials for this cutterhead in your shop. But even if you buy the mechanical parts at the start, you'd spend only about $50, including the two exciters. Be forewarned that crafting this mechanical design can take a lot of hours, and sonic performance correlates directly with your attention to build quality. Chances are you'll make mistakes and have to redo things a few times. The second head in this project took about three days to complete. But again, the only fancy tools needed were a drill press, a band saw and a belt sander.

Here's a rough DIY cost breakdown based on actual expenses incurred in completing the second head project described here.

Misc. metal stock and hardware (eBay & similar sources)	25.00
Phenolic Block ($45 buys enough to make 15 heads)	3.00
Perfume Funnels (2)	2.00
Transducer/Exciters (2)	15.00
2X 50W Class D Amplifier (Typical 'offshore'/DIY with power supply)	50.00
DSP Board ('Offshore' option from eBay)	30.00
USB Interface Board (From Analog Devices – others at your own risk!)	85.00
5VDC "Wall Wart" for DSP board	5.00
GRAND TOTAL:	$215.00

These figures are conservative as bargains can always be found, but you're on your own for packaging-up the electronics. The Analog Devices SigmaStudio software is a free download from their Website, and the DSP design shown on the last page will gladly be shared by email for the asking.

Have fun and good luck!



Analog Devices SigmaStudio DSP Schematic

Wideband Limiter • Separation Optimizer • Parametric Equalization
HF 'Acceleration' Limiter • WB/HF Gain Reduction Indicator Logic

SigmaStudio DSP Circuit Description
for the Stereo Cutterhead Signal Processor

Stereo audio is applied to the analog input pins of the DSP kernel board. Note the clipping level restrictions mentioned on the schematic. The DC-Blocks used in this design guard against inevitable DC-offset buildup.

The L/R input is split, going both to the input level sensing 'clippers,' and to the input of a 1 millisecond delay block.

These 'clippers' are not actually in the signal path, but instead serve to determine when the input exceeds a level 12dB below the analog input clipping point. The clipper outputs are subtracted from their inputs and the recovered 'clippings' are the basis for wideband gain reduction.

'Clippings' go into an OR-gate function to sense the higher of the two channels. This determines gain reduction for both channels to maintain stereo image. 'Clippings' undergo gain and low-pass filtering to derive a gain-control basis. Because this is a feedback limiter, attack is relatively fast, <1ms. Limiter release is also quick, calculated to give a maximum 'signal self-modulation' distortion figure of <1% at 100Hz. This is an acceptable figure for a peak limiter intended to give only program peak protection and not to change the perceived dynamics of the program.

The amplified and filtered 'clippings' are subtracted from a nominal unity-gain value of '1' and fed to the four multipliers in the signal paths. The multiplier in the clipper input path closes the limiting-action loop, and the multipliers at the output of the delay block, driven by the same G/R value, follow to reduce the program signal level. This 1ms delay turns the limiter into a 'lookahead' configuration, whereby the gain is actually reduced before the signal arrives. This avoids overshoots during the first millisecond of limiting.

As mentioned earlier, the Left and Right channels are matrixed into L+R and L–R, and then right back to L and R. But variable gain in the L–R path gives a 'separation trim' adjustment to compensate for crosstalk between channels from mechanical coupling.

Next is the 7-section parametric equalizer to flatten cutterhead response. These equalizer sections may also be changed to shelving, peaking or other characteristics, and the SigmaStudio Stimulus/Probe function will display the overall EQ plot on your computer screen. Take plenty of time to equalize your cutterhead. EQ sections close in frequency will have interaction, so experiment with this a lot.

The audio is next split into two frequency bands with a gentle crossover at 1.2kHz, sort of a 'bass and treble' split. The filters are actually only high-pass elements, the low-pass outputs are derived by subtracting the filter outputs from their inputs. This assures that the signals will add back up to remain flat in frequency and phase response.

The high-frequency 'acceleration' limiter works pretty much like the broadband limiter already described, except that the clippers that determine the limiting threshold remain in the signal path to clip the first half-cycle (or so) of the limited 'treble' component. Attack and release of this limiter are both much faster than the broadband one, helping make this a very unobtrusive function.

The limited high-frequencies are summed with their respective low frequency components and delivered to the analog outputs.

Both limiters include threshold detector banks that output a digital '1' at predetermined degrees of limiting action. This logic drives GPIO pins high at these discrete values of limiting so that you can drive a string of LEDs to show how much limiting is going on in real time. Traditionally G/R (gain reduction) metering will have 0dB at the top of the scale and give a downward-reading display.

The wideband limiter serves here as a 'protective indicator of signal headroom rather than a signal dynamics processor. A simple limiter like this one works (and sounds) best when reducing only small-value signal peaks. –1dB G/R tells you that your source material is at the proper ballpark level, –3dB G/R on peaks is still okay, but keep in mind that you are out of steam when the CLIP LED comes on at –12dB G/R.

The HF limiter LED string will tell you how 'bright' your source material is. If the –12dB G/R LED lights a lot, you may have the highs jacked-up too far in your mix. Don't forget that you can't get as much high frequency energy off vinyl as you can a CD!