AUDIO BASICS

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Meanwhile, back at the bitstream.

It appears that we need to elaborate a bit (ha ha) more about the operation of one-bit CD players and to try and explain what is going on (and why) in terms that I (and hopefully you) can understand.

Certainly the January, 1991 issue of *Audio* magazine has a interesting article on one-bit technology entitled "Music of the Bitstream" starting on page 56, and we urge you to read it. Certainly I have a full time electrical engineer, Aado Perandi, on staff to help me interpret what the *Audio* article said (and didn't say). We also received a brochure from Philips regarding the operation of their version of the one-bit system. Now all we need to do is translate all this technical data into terminology that we can get a handle on without going back to night school.

First of all, Why bitstream?

Why would many of the CD manufacturers want to invest the money in a new CD technology? Why re-invent the wheel? The main reason, we suspect, (aside from the lower cost of many 1-bit chip sets) is that actually you cannot accurately get 16-bit resolution from most current 16-bit CD players.

The D to A converter in a 16-bit system works by incorporating a series of circuits that represent each bit of the 16-bit incoming word from the data stream. The circuits involved each represent a portion of the data starting with 1 unit (the most significant bit) and going down from there at one-half of the size of the proceeding step at a time (1, 1/2, 1/4, 1/8, 1/16, and so on). By the time you get to the tail end of the 16 bit word (the least significant bit), the data is being represented by a circuit dealing with only 1/65,536 of the information. Actually this is a bit of an over-simplification, because the system has to deal with both plus and minus signals, but we don't need to deal with the mechanics of it in more detail to grasp what is happening.

The important thing to understand is that by the time we get down to the last couple of bits, the value of the signal is apt to be badly distorted because it is of lower resolution than the tolerance to which the circuit is built. For example, the last bit at 1/65,536 of the most significant bit is only .0015% of the value of the first bit. If the circuit representing the bits were done to 1% accuracy (few are that good) then that would be 1000 times too crude to represent the last bit. The circuit would need to be built to .001% accuracy to have a chance of doing the job (none are!).

Thus 16-bit D to A converters really don't have true 16 bit accuracy. They (and their supporting circuits) simply cannot be economically mass produced with the precision necessary.

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Obviously a D to A converter requiring a trim-pot isn't up to the job as there is no way to trim the first bit to the necessary precision. If your starting point has more "slop" than the size of the small signal data you are trying to read, the uncertainty in the starting point wipes out the low level information – it kind of falls into the cracks. In addition, we can observe that no trim-pot has the necessary long term stability to keep the unit working precisely even if you got lucky and trimmed it correctly in the first place.

Perhaps the most important thing the *Audio* article had to say was that the Philips method of designing and building 16-bit D to A converters was best of all, as the internal current feedback mechanisms and the "no trim-pot needed" design yields as close to theoretically perfect performance as has been done, and gives much better results than other converters.

But even the best D to A converter will start to show deviations from perfect response as the signal level gets smaller and smaller.

With a bitstream converter it is possible to get much better measured linearity, *as long as the signal is averaged over a long enough time period.* Unfortunately, it appears that the engineers in search of better specifications but who never listen to the music have done it to us again. Remember those direct drive turntables that sounded so harsh and bright because they traded off the problem of small long term deviations in speed (an inaudible problem to those of us without perfect pitch) for horrendous harmonic bastardizing of the music by building turntables that rotated, on a long term average, at exactly 33.3333333333 RPM but did so by making continuous rapid rates of change with their overly powerful feedback loops, inducing continuous tiny and sharp changes in pitch? Remember the quest for lower harmonic distortion amplifiers achieved by great gobs of overall negative feedback - never mind the transient distortion this generated, never mind how often these amps went unstable into difficult loads and blew up, never mind how awful they sounded, they measured better and what you really wanted was .001% THD wasn't it? Essentially 1-bit 256 times oversampling CD player designs are doing much the same the same thing in spirit - trading off better measured test signal performance for poorer musical performance - all done because you can measure the test signal data (and everyone buys "better numbers" - remember the seven shinny pennies?) but you cannot measure the music.

What can a 1-bit 256 times oversampling system do that a 16-bit non-oversampling system cannot do?

Remember that last month I told you again that 1 bit data with 256 times oversampling resolves only 512 levels. Well, there is a way to get around measuring that limitation *under certain circumstances.* Essentially, if you have enough repetitive samples of information, or if you average the error over a long enough time frame, you can get measurements that look as if you are getting true 16-bit resolution, measurements even better than the best available current true 16-bit converters can do.

The secret to good measurements is to build 1-bit systems that cleverly allow the test measurement equipment to provide numbers that are much better than what is actually happening. This can be done with noise shaping circuits that average the resolution errors over multiple clock cycles and spread the noise and distortion frequencies out so that there are many frequencies of distortion but each at a low enough distortion that they can fall below the noise floor of the test equipment. All the distortion is still there, it is just that many small errors cannot be measured within the resolution of the test equipment, while the same distortion, without noise shaping techniques, would be concentrated and easily measurable. Also by allowing the test equipment to measure many repetive cycles of the signal, the equipment will show a resolution that really is not there when only signals of short duration are considered.

On a Compact Disc the highest frequency that can be recorded is 22.05 kHz. Given the 11.2896 MHz clock frequency used with current 1-bit systems, this 22.05 kHz signal is represented by only 256 samples on the positive half and 256 samples on the negative half (the maximum 512 samples available at this frequency). However, if we had 256 consecutive cycles of this 22 kHz wave, and each cycle changed by only 1-bit, then on the average we could measure 256 x 256 samples, which equals 65,536 samples, which would yield a measurement of full 16-bit resolution instead of 8-bit. Thus we can get superior measurements from a 1-bit system, even better measurements than are possible with current good 16-bit systems.

But there is a catch!

The great measured resolution will only hold up for as long as the signal is measured and averaged over at least 256 cycles of the sine wave at 22.05 kHz. As the measurement of the signal is averaged over fewer and fewer cycles, the resolution of the measurement falls apart, showing the inherent 8-bit limitation of the too slow 1-bit 256 times oversampling system. Note that the *Audio* article confirmed our discussion of the past two months that the clock speed would have to be 256 times faster for 1-bit systems to measure well without special resolution enhancing techniques, and that speed-up simply cannot be done with current technology at a price you could afford to buy.

When the signal is lower in frequency than 11.2896 MHz divided by 65,536 (172 Hz) then there are always 65,536 samples or more per cycle to average, and the resolution of the system will actually be and measure as a true 16-bits. But as the frequency in question rises, then true 16-bit resolution (defined as actual resolution of a single cycle of a waveform) becomes impossible. The measured resolution will still appear to be 16-bit if there are enough consecutive cycles of the signal to allow averaging the measurement over 65,536 clock cycles. But if the duration of the signal is short, there are fewer and fewer samples for the test equipment system to work with, and even the measured resolution will fall back to show the limitations of the uncorrected 8-bit base. Meanwhile as the frequency of the signal goes up past 172 Hz, the real resolution of the system goes down, bottoming out at 8 bits at 22 kHz.

In other words; tests good, sounds lousy!

A test signal, being completely repetitive will obviously test perfectly with a 1-bit system. You have all the consecutive samples of an identical frequency available to average and use to provide measured 16-bit resolution. But, on music, you simply don't have this consecutive data available. Real music, by its nature, is ever changing (unless you consider machine made stuff real music - if so find another newsletter to read). Thus with music, the 1-bit 256 times oversampling CD player will offer poorer and poorer musical performance as the music becomes more and more complex and at higher frequencies, because it is basically an 8-bit system no matter how good it measures under certain condtions.

Thus the cure of the 1-bit 256 times oversampling system is worse than the problem it tried to resolve (that 16-bit systems don't really make 16 bits) or shades of the direct drive turntable. Which do you think is worse, a 16-bit system that really only makes 14 or 15 bits (both measured and on music) or a 1-bit system that measures 16-bits on test signals but only makes 8-bits at high frequency on music?

For a final comparison, remember what we have told you in the past – namely that the true signal to noise ratio of good CD players is actually about 70 dB *as long as you measure all the noise.* We don't first filter out the lows and the highs before measuring (that is like first filtering and boiling the water from the Ganges and then telling you it is fit to drink). Remember too that -70 dB is very quiet, you have to crawl up and stick your ear in the woofer to tell the system is on. But, in the final analysis, the signal from the CD player starts to vanish into the unweighted noise floor at about 70 dB below full output. How many bits do we really need to support this level of real resolution? Actually only 12 real, accurate, working, full time bits. That will give us 72 dB of dynamic range - better than what we get from any other source in any event. But the Philips 16-bit times 4 oversampling system does much better than that, so we have much better resolution than we can use now, all the time, even if it doesn't measure as perfectly as 1-bit systems.

The Philips 16-bit times 4 oversampling system works, it allows us to play real music at much better resolution than the noise floor of the rest of the audio system, and it is reasonably priced and very reliable. We will keep using it. We want the best music, not the best numbers if the best numbers conflict with the best music, and with 1-bit systems, in our opinion, based both on engineering evaluations and listening, the music is compromised.

An Upgrade for the B&W DM640 Crossover

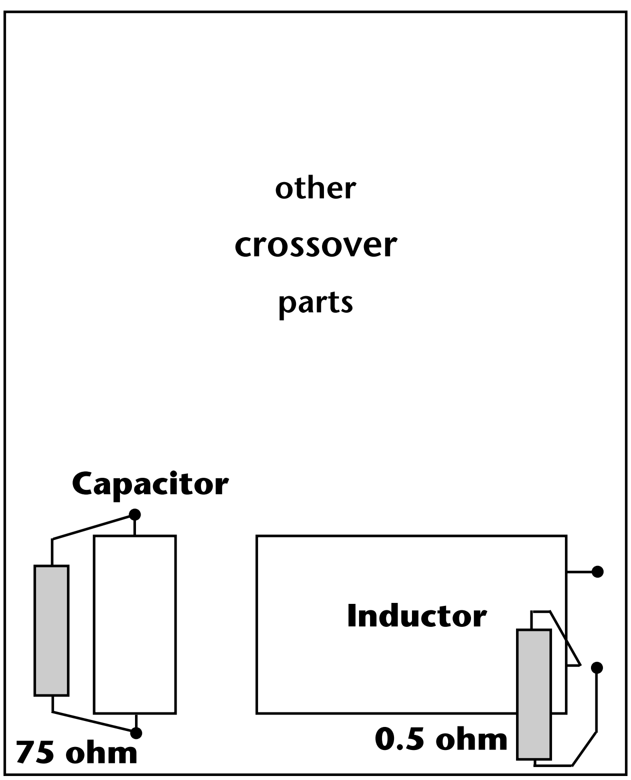
The new B&W DM640 loudspeaker has the potential for being a truly magnificent value. It essentially has all the driver components of the $4500/pair Matrix 802 – the metal dome tweeter, the Kevlar mid-range, and two Cobex diaphragm woofers mounted in a lower cost but much larger volume cabinet at one-third the price of the 802, just $1500/pair list. Only the free air mounting of the tweeter and the fancy real wood finish are missing. Unfortunately, B&W made a compromised design judgement in our opinion regarding the overall sonic balance of the speaker. While we have never liked the Matrix 802 because it had too little bass response (too small a woofer cabinet for its price range), the DM640 has the opposite problem - too much bass.

Although the speaker comes with multiple ports (ala Acoustitune) so that the bass response can be toned down, we found that efforts to reduce the bass with the longer port or the port plug simply made the speaker sound compressed and start to generate a "box" sound. Unfortunately even though box coloration went away completely when the shortest port is used (the box is very stiff and well made), the resulting too fat bass started to overpower the rest of the speaker. Although we are sure the rock and roll crowd will love it, we wanted higher bass definition and better balance than the stock configuration provided.

Fortunately, too much bass is a much easier problem to solve than not enough bass. *We simply pulled the crossover card, made a schematic of it, ran a quick circuit analysis, and fixed it.* The speaker, with our crossover refinements, is so well balanced and musical and so high a definition overall that it easily beats the Matrix 802 overall on an absolute basis, and is the best $1500 pair loudspeaker I have yet heard - certainly better than any version of the 801 prior to the current Matrix series.

To upgrade the crossover yourself all you need is one pair of B&W DM640 speakers, a good stubby Phillips screwdriver, two 0.5 ohm 5 watt resistors, two 75 ohm 5 watt resistors, and a solder pencil.

You need to remove the grill and the lowest woofer (6 Phillips screws hidden under the removable grey trim ring surrounding the woofer cone). Carefully lift out the woofer and unclip the two color coded wires (keeping track of which wire is adjacent to the red dot on the frame – a reversed hookup will remove all of the bass). Now you can see the crossover board. It is held in place by four screws; two at the bottom corners of the board that you can see and two more at the top corners you will have to find by feel (or remove the top woofer too for better access).



It is possible to make the circuit changes without actual further disconnection of the crossover as long as you can turn it enough to access the bottom. Look at our sketch for the parts that are affected.

First unsolder and lift one lead of the large inductor that resides at the bottom end of the PC card. Install one end of the 0.5 ohm resistor into the hole the lead was removed from (solder) and solder the free inductor lead to the free resistor lead, thus putting the 0.5 ohm resistor in series with this inductor. This keeps the woofer from going to DC and generating out of band low frequency distortion below its linear range. Now the crossover can be refastened to its mounting supports.

Next solder the 75 ohm resistor in parallel with the capacitor that is located beside the inductor. You can do this from the component side of the board, attaching a resistor lead to each capacitor lead. This resistor will actually be located in series with the two woofers, and will effective cut the power to them by about 0.5 dB - enough to balance them with the mid-range much more perfectly and to remove all the mud and boom from the system. Reinstall the woofer(s) and enjoy the results.

Used Equipment List

We have some exceptional and highly desirable used equipment this month – things that you have been calling us about regularly and that vanish just as fast as we tell you about them. So if you see something here that you really want, don't wait, call us now. Tomorrow it may be gone!

One pair **new B&W CM1 Mini-Matrix Loudspeakers** at $600/pair, delivered in the continental USA. Your choice of black or white. We had a special order cancel for the white ones leaving us with an extra set. The ones you don't take go into my own video system, but you get first choice. List price is $800.00 so this is a special one-off value.

Super Pas Three Omega Preamplifier built in a very nice used chassis. The owner liked it so much he decided that our Fet-Valve preamp must really be even more wonderful (it is) and traded it back in. You can have this one for $595.00 plus shipping (the price of the unit without the Omega buffers) instead of the new price of $745.00 and get our 2 year new product warranty too.

Mos-Fet 150C Power Amplifier built in a very nice Dyna St-150 chassis complete with perfect walnut side panels. This is an especially nice unit I personally built for my parents a few years ago and replaced for them this Christmas. It has only been used to play classical and choral music at low listening levels - never moved or abused since I installed it in their system. It can be yours for $345.00 plus shipping with a 6 month warranty.

Brand New Omega Pat-4 Preamplifier. We just custom built this unit for a customer, tested it, burned it in, and were ready to ship for a Christmas delivery, when the customer's check bounced. I guess that is one way of canceling an order. Oh well, we would like to get our money out of it so this brand new unit with a brand new 2 year warranty and our Omega phono, line, and tone control circuits is yours for the price of the circuits and new selector switch alone ($295 plus $75 = $370.00. The never used and previously unassembled Pat-4 chassis we will toss in for free. Add $8.00 for shipping in the USA.

Upgraded Super Fet Pat-4 Preamplifier with precision volume and balance controls. This is a clean unit that we dramatically upgraded with our latest IC chips for performance at the Omega level. It is a straight line unit (no tone controls) and it plays musically and without rough edges. Only $195.00 for modern Audio by Van Alstine performance and a 90 day warranty.

Stock Dyna Pas-2 preamp. We found yet another of these famous little vacuum tube preamps, cleaned all the controls and switches, upgraded the heater supply, and tested it for proper operation. Its not as pretty as the past few, but its only $75.00 with a 90 day warranty. If you want it pretty, for $20 more we will install a nice Pas-3 gold faceplate and knob set. Buy our Super Pas Three rebuild kit at the same time and get both preamp and kit for $250 ($270 with the gold faceplate).

Save $100 on new Omega 50 or Omega 150 integrated control amp circuits. We have one very nice used Dyna SCA-50 chassis and we will build an Omega 50 for $495 or an Omega 150 for $695 in this chassis ($100 less than brand new price) and give you a complete 2 year warranty too.

One More Chance at our Patent Granted Celebration Sale. We will give you until January 15, 1991 to get those checks in for Fet-Valve equipment at 10% off and B&W speakers at 20% off of list price. We know you have been "thinking about it" (many of you have already taken advantage of this one time special) and we want to give you the opportunity to get those after Christmas checks in to us. Do it now!

Last Reminder for those "90xx" Mailing Codes.

If the four digit code after your name on your *Audio Basics* mailing label begins in "90" then this is your last issue. Thanks for your support in 1990. We certainly would like to keep you as a subscriber. Get that renewal check ($16 USA, $20 Canada, $24 Foreign) in to us now to continue. And for the many of you that have renewed already, we say thank you for your support and for the many kind notes. We need your feedback to continue to produce *Audio Basics*. It isn't written for profit, it is done because I think there are things that need saying in the audio business that nobody else seems willing to talk about. But we can only write if it appears there remain people willing to listen and to think. Too many phone calls from young people these days start out with "well yah uh…" We have lost those people before we start because they never did learn how to think or listen. They do remain good customers for magic and mysticism though. "Good sounding parts" are a lot easier to contemplate than good engineering. We want you to know better than that. Stay with us and we will try and learn more about high fidelity in 1991. Thank you.

Frank and Darlene Van Alstine