Fall / Autumn 2006

DIY MAG

"Doing the best at this moment
puts you in the best place for the next moment. "
Oprah Winfrey

Hot News Flashes!

Products just released, soon to be released, expected to be released…or products that have been around for a while, but that I think are real cool…

Robert Bastani launches the Atlas the pinnacle of his diy line of products.

http://www.bastanis.com

See also the Bastanis Prometheus reviewed by http://6moons.com

Hot News Flashes

Not a news flash but a cry for help. Would anyone be interested to make an edition or 2 of this DIYMAG?. I find it tedious to make the magazine nowadays. And some fresh blood will surely do the magazine a world of good.

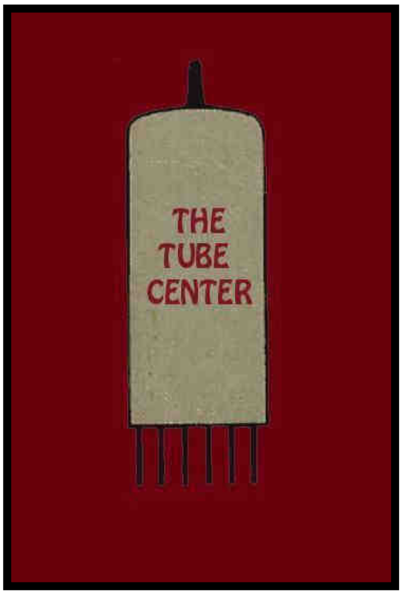
"Anyway, the new Jag..."

Check out this magazine.
http://www.makezine.com/

The first magazine devoted entirely to DIY technology projects, MAKE Magazine unites, inspires and informs a growing community of resourceful people who undertake amazing projects in their backyards, basements, and garages.

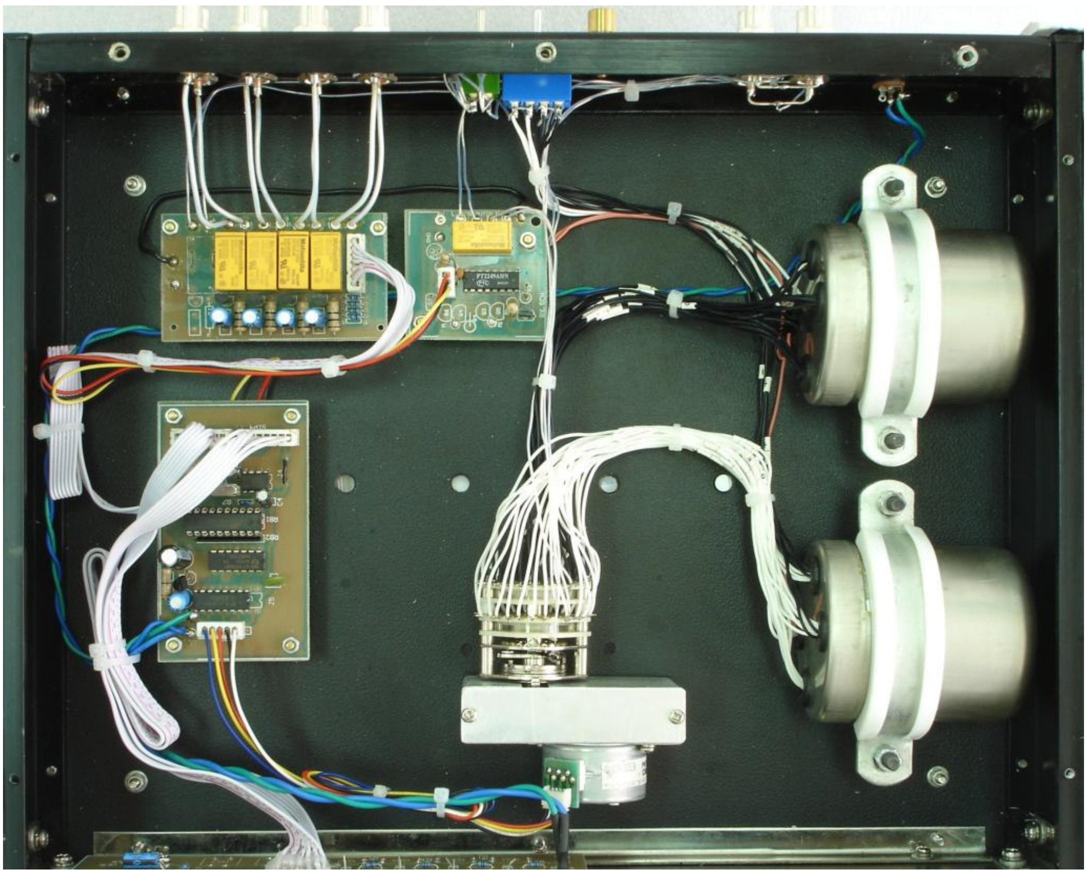
Sample from the mag

http://www.thetubecenter.com



This is meant as an ad but also as an apology. I am not a Christmas card kinda person (Just another way of saying I'm the grumpy type around the festive season). But as a tube/valve amp builder the most beautiful gift I could get for Christmas is a card from my tube vendor saying happy Christmas. This is exactly what I got… I got a Christmas card from the Tube Center!!! And I kept forgetting to answer it with a card of my own…(2 years in a row now!) So here it is! To all the friendly folk of the Tube Center, have a very Merry Christmas and a happy and prosperous new year!!!

http://www.diyhifisupply.com updates the Django, new enclosure, remote control option and more choices in magnetics. Choose between S&B, DiyHiFiSupply amorphous and now also autoformer.



Readers Forum

Hi,

While electronically 'flicking' through the pages of your latest DIY Journal, waiting for it to print out I noticed on page 37 in Part II of the PCB etching article by Sylvain Bergeron, Sylvain comments on the Dremel drill press attachment being discontinued. That was a long sentence! Anyway, Sylvain is quite correct that the particular model he is using in the photos has been discontinued, but fortunately it's been replaced by a new model as seen here:

http://www.justtools.com.au/prod1171.htm

The Dremel website isn't the quickest horse on the track and as yet hasn't updated the site (28th June). I believe their plunge routing attachment has been updated also. And lastly, a new model tool is being released very soon that uses a digital speed control circuit instead of the mechanical switch on the 395 tools. How do I know this? Simple, I was in the market for one, a chap at the store said the new model tool was out in "a couple of weeks", "I can't wait" says I. I bought one and wouldn't you know, the switch was faulty :) Such cruel irony.

Thank you for all the hard work put into the Journal from you and its contributors :)

Cheers
Kendal.

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Hi Bas. In my PCB articles I mentioned that I was using EaglePCB. I found a fantastic online training and reference manual for it at:

http://www.interq.or.jp/japan/se-inoue/e_eagle.htm

You may mention it in your next issue. It may be quite useful to other users.

Bye...

SB (Sylvain Bergeron)

Motor Cars

By John F Hope

Nearly all automobiles are propelled by an engine which runs by making explosions inside a heavy iron block and which has a thermodynamic efficiency of around forty percent. Disciples of this clumsy art feel they're doing quite well, compared even to God, because the modest thermodynamic efficiency of the human body isn't a whole load better. But unlike the human body which can be fuelled on a huge diversity of substances found all over the planet, the motorcar engine will only run on a single substance. A substance in finite, diminishing supply and which has to be extracted at great cost from deep wells in areas occupied by belligerent religious fanatics. God wins the design award yet again.

Looking at motor cars from this fundamental perspective it's difficult to work oneself into a lather of enthusiasm over the progress that's been made over the last century or so. I view progress in the automotive sector as akin to taking a cat and dressing it up as a baby; it may pass casual inspection from coo-ing ladies in the supermarket, it may look irresistibly sweet and endearing, and you may even get away with taking it on a transatlantic flight, but underneath the disguise it's still a cat and when it wets itself you are forcibly reminded of this.

In the early 1970's there was a glut of hobby projects in Practical Electronics and similar hobby mags for variable intermittent windscreen wiper controllers. Intermittent controllers appeared in production vehicles 5-10 years later, which is an understandable delay if you consider the design time of a new car model. But why did it take 30 years before an appreciable number of the auto manufacturers began fitting a pot to these things so you could vary the rate of wipe yourself? Is a pot such a high-technology device with so much attached risk and capacity to harm gurgling children and innocent animals that it needs 30 years of lab testing and clearance from the F.D.A. and the W.H.O.?

Back in1990 I was attached to a research group investigating new technologies for transport and automotive applications, and one of the topics we looked at was automatic windscreen wiper controllers: systems that would automatically detect the presence and quantity of rain and adjust the wiper rate accordingly. Our study was done on the assumption that digital processing power in the future would follow Moore's law and would not be a practical limitation. We concluded that a rain-sensitive wiper system would not work 100% properly because of the difficulty in accurately sensing the quantity of rain over a small surface area in the presence of numerous other factors. But we failed to consider the effects of consumer-softening that the IT revolution would bring about. The computer industry has proliferated so much unreliable, dodgy junk that consumers no longer expect things to work 100% properly all the time. Most people are now satisfied if something works more or less reasonably about two times out of three and lasts two years before it becomes unmaintainable through obsolescence. With this in mind the auto industry threw bucket loads of microprocessor technology onto all control topics, from dimming an interior light to rain sensitive wipers. By 2000 many cars had rain sensitive wipers but even today I know of none that work 100% properly. These wiper systems are like toddlers – they can't be left to themselves unattended for any length of time. In the model of car I drive, the rather hit and miss arrangement they've created has the spectacular oooops! that it doesn't work at night unless you're stopped under a bright streetlight. On another make of vehicle I've had cause to use a lot lately, the rain sensitive wiper controller requires constant wet nursing. It has two speeds: lethargic or hysterical, which appear to be randomly selected by an electronic coin flipper. On yet another make of vehicle, it's broadly understood that you have to change the windscreen every two years or else the rain-sensitive wiper would degrade into a rain-tolerant wiper and not operate whatsoever. Flaky stuff like this appears on the market very quickly, but the fitting of a pot takes 30 years.

I am certainly not against electronics in cars in general. My motoring days started in the 1970's around cars like the BMW 2800, Jaguar XJ6, Alfa Berlina 1750, and the original Mercedes S-shape, which all had multiple carburettors and contact breaker ignition systems with distributors. These vehicles probably represented the ultimate in mechanical hubris. The impressive boot-lid toolkit in the BMW 2800 even included a feeler gauge because the (MTBSAS) Mean Time Between having to Screw Around with Something was around the same time it took to drive out a tank of fuel. You had to get out of the car to fill up with petrol anyway, so while you were outside on your hind legs you may as well adjust the points, set the timing and check that none of the spaghetti tangle of vacuum hoses was holed or had come adrift. And get to your own wedding ceremony late and covered in black grease. The breakerless ignition systems and mechanical fuel injection systems which began infiltrating the industry in earnest in the mid 1970's – mostly as a result of mandatory emission control regulation in the USA - represented an enormous step forward. The concept of fuel injection had at least been proved in principle with diesel engines over the previous 70 years so there were several generations of engineers who what they were doing with injection systems. Electronic engine management systems evolved into those we have today, which keep the vehicle in proper tune virtually indefinitely and are usually based on a multi-dimensional lookup table (MLUT) or engine map. This is all very ticketty-boo but I cannot help but wonder: Have they covered each and every single combination of bits that can ever possibly result from different combinations of all input variables? Is there not a rogue combination with a missing code (as the ADC fraternity would call it), that results in injector time falling to zero and your engine instantly dying when you're halfway across a busy intersection with a low-loader carrying a Challenger II tank bearing down on you? Or injector time slewing to max and your engine revving uncontrollably as you approach a pedestrian crossing where a little old lady is doddering across the street in her Zimmer frame, accompanied by her great-grandchildren and their puppy dogs? It's of meagre comfort to know that there may be a 1:14 million chance of this happening when it's also true that someone wins the Lotto on similar odds nearly every week. It could be you.

Now that computerised engine management, ABS, airbags, and fuel injection are virtually universal, the motor industry seems at a loss as to what to develop next. True, there's a long list of really meaningful challenges – like windows that can de-ice themselves in 5 seconds - that they've abandoned either because they're too difficult or because consumers are too conditioned to toil, misery, and fear. Instead they've set themselves a more achievable target of building a car which consists entirely of either plastic or silicon.

The silicon target is approached by applying computers to do the simplest task imaginable. Most medium to high market cars produced today have a computer to control winding the windows up and down, moving the seats back and forward, turning the interior light on and off, controlling the radio, and adjusting the wing mirrors. I recently had a rental in which some part of this microprocessor Tower of Babel had croaked and I was forced to adjust the wing mirror manually by blind estimation. And that before I got in the car because once the ignition was on the driver's window had a quaint habit of rolling up and down completely unprovoked, and I didn't fancy having my arm squashed in it.

Someone at a household-name German electrical consortium - which I shall call Bonk so that they can't sue me - thought it would be a major coup to fetter automatic transmissions with full microcomputer control, notwithstanding that mechanically-controlled automatics have been providing reliable service for years. This decision was probably influenced by the success that Rodney Bonk GmbH had enjoyed with their Chinese-made electronically controlled toasters and washing machines, and they thought they could pull a neat hat trick with a Seriously Computerised gearbox. And so they gave the world the Bonk SillyTronic transmission, considered best contender for the worst automatic transmission ever made.

To add insult to injury, the SillyTronic is probably sold by Bonk GmbH as being very efficient on power throughput, so that you'll find these disgusting things coupled to puny 1200 cc engines in tiny European hatchbacks and French quirk vehicles unsure of their status as car or tram or ice cream van or dog kennel. In essence, driving a car fitted with the Bonk SillyTronic is rather like driving a very gutless manual car with a broken clutch cable. When I sniff around a piece of work like the Bonk SillyTronic I smell stuff like Windows CE operating system . . I smell Object Oriented Somethings and Shells . . . I smell programs written in high-level language. . . I smell things that have been simulated on PCs rather than tested on the road. . .and I smell Computer Science rather than Engineering. Most of all I smell trouble.

Let's look at how this clever BonkWare copes with overtaking a big truck on a hill. At first there's nothing coming towards you, so you put your indicator on and floor the gas pedal, expecting an immediate kickdown and a surge of useful, reassuring power. The transmission responds thus:

Your SillyTronic transmission has been requested to compute the optimum gear ratio for the new conditions you have imposed. But before tasking your request it will first check for updates from Bonk GmbH. Please wait . . .

At some point during the please waiting the SillyTronic will switch you out of the gear you were in, and hold you in a kind of gearbox limbo, with all power removed from the drive wheels for several seconds, so you can relish the feeling of empty weightlessness you feel in a rapidly descending elevator. Meanwhile a big silver Oshkosh truck has appeared on the horizon and is headed towards where you are poised like a stupid pheasant in the road, next to the big truck you were trying to overtake and which is now slowly pulling away from you. And while you are fingering your lucky rabbit's paw and praying fervently, the Bonk control unit makes a decision:

The optimum gear ratio for the new conditions imposed by you is 3.1416 to 1, which is interpreted as first gear in this instantiation of the self-adaptive object-oriented gearbox algorithm. Would you like me to send these new results to Bonk? Or to the Engine Control Unit? Or to Madonna? OK then, I'll apply the transformation in a few moments, after I've updated my adaptive algorithm and shelled some more monkey nuts. Please wait . . .

The SillyTronic then slams the transmission into first or second gear with such violence that the engine is all but ripped from its mountings. It screams into redline and you get instant whiplash and throw up your lunch. And your status is changed. You are poised like a stupid pheasant in the road, making a lot of engine noise, next to the big truck you were trying to overtake and which is now slowly pulling away from you faster than it was . This change in status is what the Bonk SillyTronic understands as kickdown.

I once had a small mishap while driving a rented hatchback fitted with the Bonk EasyTronic, when parked in a tiny Spanish village in the Alpujarras. A metre in front of the car, and on a slight downhill slope, was a decrepit fence separating me from a grand Alpujarran abyss I'd just photographed. Reasoning it would be a bad thing to drive forwards, I carefully selected reverse gear and applied a sensible amount of va-va-voom. But nothing happened, and to my dismay the car started to slowly roll forwards towards the fence and my inexorable doom in the abyss. In panic I floored the va-va-voom pedal and the dinky engine roared cheerfully into redline. The SillyTronic eventually woke up, savagely whacked the transmission into reverse, and the tiny car went careening uncontrollably backwards until, with a distressing crunch and a cloud of dust, it was brought to a halt by a stout tree.

One thing all electronically controlled transmissions have is a Wounded Bird Mode. This is intended as a get-you-home option if the computer decides the gearbox is kaput. Whether or not the gearbox is really kaput, or it's simply one of a multitude of prima donna sensors that doesn't like the cold or the heat or the bumps or the sand and grit or your driving style, is not the issue. The computer sets your transmission into Wounded Bird Mode and you get to crawl along in frustration at 10 km/hr with a little orange picture of a toothed gear flashing hysterically in the instrument binnacle, while pensioners on mobility chairs pass you and scowl.

Someone at the motor manufacturers has a really cool job designing all these universal graphic symbols. The little pictures of engines, gearwheels, open doors, lights, open boot lid etc are done with artistic flair and imagination. When a friend of mine somehow managed to drive his wife's Audi TT onto a traffic island, completely ruining the entire undercarriage and a lot else, the first thing he noticed when the dust settled was the appearance on the dash of a little flashing orange picture of a radiator with a fountain spurting from it. Sometimes it's worth wrecking your car just to see all the nice symbols that are normally blacked out come alive and wink at you.

Let's see how maintenance of motor cars has changed over the last three decades.

Suppose it is 1976 and your car has a problem: it's an automatic and occasionally when you pull off from a stop the engine hesitates and hiccups and chokes and all but dies. You take your car to your Friendly Service Agent where your problem is dumped on one of the factory trained mechanics called Bob. Bob wears a filthy blue overall with spanners and screwdrivers in his pockets, he reeks of garlic and beer, and he hasn't shaved in three days. After a cursory inspection and a wee drive to his girlfriend's house for a few beers and a quickie in the sack, Bob gets back to you and says there's nothing wrong with your car. You then go back to the Friendly Service Agent where you emote and create a big scene in the service reception in the presence of several worried-looking customers. Exhorted by his manager's threats of violence, Bob reluctantly agrees to examine your case further. The next day he starts at the front of the car and systematically begins replacing parts until he thinks the problem was solved. You are presented with a crippling bill for £250, covering the cost of 47 parts Bob replaced which didn't cure the problem, (including horn relay, front bumper, radiator cap, hi-beam relay etc.), and the one part (accelerator pump diaphragm) which he believes did. Two days later you are enraged when you find the problem is still alive and kicking, but decide in the interests of your cardiac health to forget about the whole sad experience and move on. Some months later you read somewhere that your car's problem is widespread and is caused by ethyl alcohol the oil companies have been surreptitiously adding to the fuel.

Now let's suppose it is 2006 and your car has the same problem: it's an automatic and occasionally when you pull off from a stop the engine hesitates and hiccups and chokes and all but dies. You take your car to your nearest Vehicle Service Clinic and your case file is assigned to one of the qualified master technicians called Jason. Jason wears a starched white dustcoat with a hand-held oscilloscope in the pocket, he's clean shaven, smells discretely of Brut 99 and sports an earring. After a cursory inspection and a wee drive to his boyfriend's house for a glass of Chardonnay and a quickie in the sack, Jason gets back to you and says there's nothing wrong with your car. You then go back to the Vehicle Service Clinic where you emote and create a big scene in the service reception in the presence of several worried-looking customers. Motivated by his manager's polite suggestions, Jason reluctantly agrees to

examine your case further. But unlike Bob, Jason won't just guess at what's wrong. The next day he connects all the on-board computers in your car to PC-based Service Equipment, which milks them of their entire fault history logged since the car was born, and offers up fault codes and helpful suggestions of parts to replace.
Fault Code 101: Flat tyre at 12h06 on 02Jan2004: Info Code 101: .flat tyre inflated 07Aug2004 08h30. . . Fault Code 18 - Hesitation and misfire: 20Feb2004 14h30: 14h32 : 14h33. . . . .15h30 23Feb2004: 15h30: 15h31. . . .7410 other occurrences of Fault Code 18 - Hesitation and misfire. Print log? Y/N . Suggested actions: Take the whites of three eggs and gently fluff them in a bowl. Then add two teaspoons cinnamon powder and whisk . . .
Jason trusts his equipment. He starts at the top of the PC-generated list and systematically begins replacing all the suggested parts until he thinks the issue has been fully addressed. You are presented with a crippling bill for £2500, covering the cost of 147 parts Jason replaced which didn't cure the problem, (including horn relay, tacho sensor, temperature sensor, humidity sensor, fuel pressure sensor, seat-belt sensor, fuel pump, fuel filter, sunroof motor driver, airbag sensor, engine management computer), and the one part (set of six injectors for your four cylinder engine), which he believes did. Two days later you are enraged when you find the problem is still alive and kicking, but decide in the interests of your cardiac health to forget about the whole sad experience and move on. Some months later you read somewhere that your car's problem is widespread and is caused by government environmental legislation which mandates oil companies add 30% rapeseed oil to the fuel.

One area which stands out as a shining beacon of improvement over the years is the car audio system. In the 1970's there were very few pre-installed car audio systems and you usually had to go buy one of the following and install it yourself:

A Compact Cassette Player
This sad-ass device was rather like the magnetic medium equivalent of a shredder, whose main function was to destroy tapes. After Watergate in 1972 destroying tapes was all the vogue and everyone wanted a slice of the action. Your Compact Cassette Player helped you do this in 3 easy to remember stages:

- Your music suddenly starts speeding up as tape begins to wrap itself around the capstan, then with a final gurgle it stalls the motor and stops altogether.
- You attempt to remove the tape from the player. But you find it is hooked on something in the player's innards by means of a long umbilicus of crumpled, twisted, wrinkled tape. While steering with one hand you gingerly extract the cassette and try winkle free the umbilicus using a pencil and without breaking the tape - or your winkling pencil - and leaving some of either inside the player.
- Having freed the umbilicus you make a half-hearted attempt to wind it back into the cassette, using the same winkling pencil. After a while the futility of this operation becomes painfully obvious and you throw the mangled cassette – and sometimes also the broken winkling pencil - out the window.

An 8-track Cartridge Player
This even sadder-assed device was intended as a robust music medium for motor cars and an attempt – albeit a pathetic one – was made to address the performance shortcomings of the compact cassette. The 8-track cartridge had an endless loop of ¼" tape running at 3¾ ips and housed in a great big plastic casing about the size of a book. Then to completely nullify any advantage this may have offered, they employed a tape head with only two sections, which was – wait for it - mechanically repositioned relative to the tape in order to play the 4 different pairs of tracks. If I remember correctly half the tracks on this tape ran in one direction and half in the other, making it a true half-assed, half-baked system. The key characteristics of the 8-track cartridge system with this no-win design were:

- At least one out of 4 track pairs had azimuth misalignment and your music was reproduced soggy and muffled.
- But this didn't really matter because the vertical misalignment of the heads caused two or more of the other tracks to play backwards at similar volume to the track you intended playing.
- And you couldn't hear diddlysquot anyway over the spark noise and alternator whine.

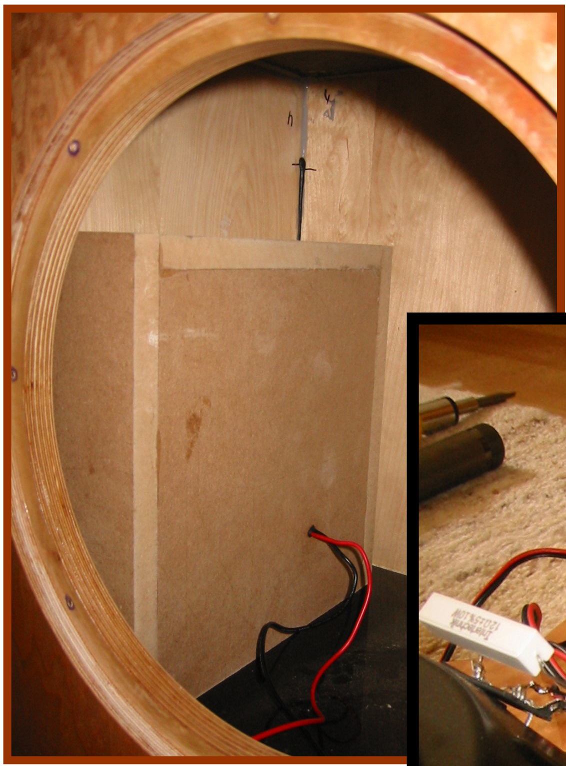
Unbelievably, both types of player were very attractive to thieves. You could buy anti-theft brackets, which enabled you to remove the player and carry it away with you in your conspicuously bulging briefcase through a gauntlet of muggers when you parked your car it in an insalubrious area. Anti theft brackets also enabled you to seize the player from under the ubiquitous passenger side parcel shelf, and heft it out the car window onto the freeway when it jammed once too often and mounged up The Gypsy by Uriah Heap.

Installing either genre of these ridiculous tape machines was usually a labour of love, which is why young men used to do it for their girlfriends in the hopes that it would expedite them getting laid. 1970's cars seldom had factory fitted speakers. Worse still, there was often little or no provision for speakers to be fitted. It must have been really weird working at the BMW factory and having to keep up this charade that nobody ever listened to music in their 3.0CS.
'Of course music is not necessary because der engine sound is so sveet . . .'

‘Anything you say, Herr Boss. Ein biskit mit your coffee you would like, yes? ’
‘Ja, danke. It vill take several so I can giff to mein friends Alice und der Hatter.’

Inevitably there would be a sturdy metal plate where you wanted to cut holes for speakers on the rear parcel shelf. Sometimes the automaker would have bloody-mindedly cut a puny 100mm hole for a mono radio speaker in the centre of the shelf just to tease you. The angled rear window glass served to prevent you getting any tool to bear from above, forcing you to lie on your back in the boot and cut big oval holes in the metal plate with your Black & Decker jigsaw. While shards of burning hot metal and grit fell on your face and in your eyes and the jigsaw jumped around in your hands like a crazy bronco and half-cut, half-melted a ragged hole. In the boot, the cacophony it made was deafening: Braaaaaakkkkkkkkkkk...kakkaka kkakka. . .
Fitting speakers in the doors was always a tempting option because the inner door panels were made from some kind of compressed Weetabix board which cut relatively easily, and you could at least work right way up in some degree of comfort. But there is nothing so sad as the devastated expression on the face of a man who has just cut big holes in his door panels, fitted his big speakers, and discovered they prevent the windows rolling down.
Today’s car audio systems are usually integrated into the vehicle in such a way that they’re almost unstealable. They no longer play Crummy Cassettes or Half-Track Cartridges or CD’s from a sixpack buried under the boot lining, but from a nondescript slot in the dash. It’s like spoonfeeding a baby; you keep feeding CDs into the slot and the machine keeps slurping them up and ingesting them into it’s mysterious interior: slirrup-klikka-whirrrrook. . . slirrup-klikka-whirrrook. . . slirrup-klikka-whirrrrook. . . Then when it is full it signals you to stop feeding it and refuses to slurp up more discs, but unlike a baby it doesn’t throw up all over you. Very clever stuff. Car audio has come of age at last.
Slirrup-klikka-whirrrrook. . . slirrup-klikka-whirrrr. . .slirrup-klikka-brakka-brakka-brakka . . . brakka. . .glunk.
What the hell. . . .??!

Mamal's Prometheus, a pictorial



Analogue audio magnetic recording

(one of MJ's handouts when he was teaching)

by Morgan Jones

Why do I need to know about analogue recording?

Now that most audio and video recording is done digitally, why is analogue audio recording relevant? Surprisingly, some music recording studios make a feature of providing pure analogue recording facilities of superlative quality (in one case, half-track 1" stereo at 30ips). Even though analogue studios are in the minority, all the recording labels have large back catalogues of priceless recordings made on analogue tape. Unfortunately, many recordings made in the 70s were made on tape with an unstable binder (the binder is the material that binds the magnetic particles to the polyester backing). These tapes ooze stickiness and clog the replay machine, or seize solid on the supply reel. Although the tapes can be temporarily recovered by baking, this means that the recording must immediately be transferred to a more stable format, usually 24 bit digital sampled at 96 kHz.

Similarly, the broadcasters have a large archive that needs to be transferred to a smaller format simply because the original recordings take up so much room. Once transferred, the original recording is destroyed...

Record labels and broadcasters need to be able to replay analogue recordings with utmost quality.

Many of the concepts in analogue audio recording form the basis of analogue videotape recording. With the addition of a little external processing (such as the Sony PCM F1 or 1610) an analogue video transport can record digital audio, and if the recorded data rate can be increased by a factor of 100 (requiring a more stable transport), uncompressed digital video can be recorded.

Recording requirements:

? When replayed, the recording should be indistinguishable from the original.
? Off-tape monitoring is needed to establish that a recording is being made, and that it is of adequate quality.
? The recording must be interchangeable between studios.
? We must be able to erase previous recordings.

Analogue audio recording - basics:

A tape machine uses an electromagnet known as a head to record bar magnets on plastic tape coated with fine magnetic particles that is drawn past the head at a constant speed. The problems of recording can therefore be resolved into magnetic considerations of the head and tape, and the mechanical problems of drawing the tape past the head.

Magnetic considerations:

Magnetic considerations split between the BH loop of the head and tape magnetic materials that define dynamic range, and frequency response problems.

BH loops:

Any magnetic material can be considered be composed of many randomly aligned magnetic dipoles that can be forced to align themselves with the external magnetic field. Once all the dipoles have been aligned, the material is said to be saturated. If the material is magnetically hard, when the magnetising force is removed, most of the dipoles remain aligned (remanence.) In order to demagnetise the material, an opposite magnetising force must be applied. The force required to demagnetise the magnet is known as the coercivity. Recording tape needs to have high remanence and coercivity, resulting in a large, square BH curve. Recording heads require low remanence and coercivity, resulting in a thin, nearly vertical BH curve.

The significance of saturation is that it sets the upper limit of dynamic range on a magnetic recording. On a well-designed tape machine, the tape saturates before the recording head, so tape saturation is the significant factor. Tape saturation is described by Maximum Operating Level (MOL), which is defined as the recorded flux in nWb/m2 that causes 3% THD at 315Hz, although 400Hz and 1 kHz are also occasionally used. Although 3% THD seems a remarkably large figure compared to the 0.1% that might be specified for electronics, this figure is tolerable for two reasons:

? We try to reduce generations of recording - certainly to less than four, whereas the signal has to pass through many pieces of electronic equipment.

? The distortion produced by tape saturation is comparatively benign because it is low order - drummers often use tape saturation for soft limiting because it produces a nice sound.

Relating MOL to electrical peak level:

Audio tape is produced by various manufacturers, and each manufacturer produces different formulations of magnetic materials with different compromises. Recording studios and broadcasters thus choose a particular tape and use it for all recordings. Since the tapes are all magnetically different, their MOLs are also different.

Broadcasters standardise on +10dBu as the maximum level expected to leave a desk (PPM6, +2dB to allow for calibration errors.) Music recording studios do not have such tight control of recording levels due to their use of VU meters.

The Media Technology local standard is that we use Ampex Grand Master ¼” tape at 15ips and define MOL as 1000nWb/m2, and we expect this magnetic flux to be achieved by a level of +10dBu. Thus, if we play an alignment tape specified to have a recorded flux of 160nWb/m2, we can calculate the replay level we would expect:

dB~=~20`log``left ({160} over {1000} right)~=~-15.9dB

In other words, 160nWb/m2 is 15.9dB below MOL (which corresponds to an electrical level of +10dBu), so we expect an electrical level of -5.9dBu. We can now adjust the replay level of our tape machine to achieve -5.9dBu and know that if the record level has been set to produce 0dBu on replay when 0dBu is presented to the machine, then +10dBu will achieve the 1000nWb/m2 MOL of the tape. Highest level peaks (+10dBu) will suffer 3% THD, but if the PPMs on the desk are carefully calibrated, we can reasonably expect only to produce peaks of +8dBu (PPM6), which we have found produces 1% THD, hence our local choice of maximum level.

If we make a recording preceded by some 0dBu line-up tone, another studio can replay our tape knowing that peak levels will be 8dB higher than that line-up tone, and they can adjust the replay level of their machine to suit their local standard if necessary. Thus, interchangeability has been achieved.

Dynamic range:

Good quality domestic amplifier/loudspeaker combinations typically achieve a maximum Sound Pressure Level (SPL) of 100dB(A). Typical ambient noise in suburbia due to traffic noise etc. is ?30dB(A), so the reproducible domestic dynamic range is ?70dB.

The noise floor of magnetic tape recording is determined by the thermal noise generated by the record and replay electronics, and by the tape itself. Using our local recording standard, our Studer A80 tape machines achieve an unweighted noise floor of ?-55dBu, resulting in a dynamic range of ?63dB. This means that the noise of the recording will be audible if the listener listens at full volume.

Broadcasters take the view that it is unusual for their listeners to listen to the “Archers” at 100dB(A), and that 63dB dynamic range is perfectly acceptable. Music recording studios expect their recordings to be played at higher levels, possibly on headphones (which attenuate ambient noise and can easily achieve 100dB(A) with low distortion). Music recording studios therefore consider 63dB dynamic range to be inadequate and usually add analogue noise reduction such as Dolby A (?10dB improvement), Dolby SR (?30dB improvement), or DBX (?20dB improvement).

Head gaps: sensitivity and extinction frequency:

The tape head is formed of coils wound on a circular magnetic core with a gap. The tape is in intimate contact with the gap, and the flux balloons out of the gap to pass through the magnetic material of the tape. If the gap is made narrower, more flux will jump directly from one pole piece to the other, and less flux balloons out of the gap, reducing sensitivity. By reciprocity, a narrow gap reduces replay head sensitivity.

Considering replay, when one cycle of recorded flux on tape is the same dimension as the gap, there is no change in flux as the magnets pass the head, and this is known as the extinction frequency. The extinction frequency can be raised arbitrarily by reducing gap width, but this reduces sensitivity. Below extinction frequency, the output of the replay head falls at 6dB/oct due to d?/dt, causing low frequencies to be replayed at a low level. Equalisation corrects the 6dB/oct slope, but also amplifies the thermal noise of the replay electronics.

The factors of extinction frequency and d?/dt combine to mean that magnetic tape can only record ten octaves of bandwidth, although these ten octaves can be moved anywhere between audio frequencies and an upper limit of ?30MHz.

The record head needs good sensitivity otherwise there is a danger of saturating the head before the tape, so the record head has a wider gap than the replay head, further justifying the need for a three head machine (erase, record, replay). The inevitable HF loss caused by widening the gap can be overcome by additional record equalisation to suit the head design.

Erasure and bias:

Previous recordings need to be erased. The previous recording can be erased by saturating the tape both positively and negatively and then taking the tape through progressively smaller BH loops until there is no resulting remanence. Progressively reducing flux is achieved by applying a constant magnetic field from the erase head and allowing the tape motion to cause the decaying magnetic field. To take the tape to both positive and negative flux limits, the erase head is fed by a sinusoidal erase oscillator. To ensure that the tape passes through many hysteresis loops before leaving the erase head's field, the sine wave must be at a high frequency. 300kHz is typical.

Early magnetic recording attempted to record the signal directly, and this caused gross distortion, but it was quickly found that distortion was reduced if the recording signal was mixed with a small amount of bias from the erase oscillator. The precise level of the bias is critical. As bias increases from zero, tape sensitivity increases until the bias begins to act as erase current, and thus reduces sensitivity. Fortuitously, at maximum sensitivity, distortion is almost at a minimum, and modulation noise is also at a minimum.

High frequencies are more easily erased, and are therefore more sensitive to bias erasure, so a good test signal frequency for finding the optimum bias point is 10kHz. In order that the test signal should not be near saturation, a level of -20dB ref MOL is used.

As an example, our Studer A80s have an optimum bias point at 15ips when 3dB of overbias is used. This is achieved by recording a 10kHz signal at -10dBu and simultaneously replaying it whilst monitoring the replay level. Bias is adjusted to give maximum output (which is noted) and then further increased until the output falls by 3dB. If a different type of tape is used, it will require different bias...

Head wear and equalisation:

The recording is made by placing the tape in intimate contact with the head and drawing it past at a constant speed. Although the magnetic surface of the tape and the head are polished, friction between the tape and the head wears the head and the tape. Tape wear, sometimes known as oxide shedding, is the reason why tape machines must be regularly cleaned - otherwise oxide build-up on the face of the head prevents the tape from being in intimate contact with the head, resulting in spacing loss, which causes HF loss.
The gap in the head is not made by simply cutting a slit in the magnetic core, but by cutting a wedge - this forces the flux into the tape. However, because the cut is a wedge, as the head wears, the gap widens, since fext. = v/?, this lowers the extinction frequency, and causes increased HF loss. Head wear is inevitable, so user-adjustable HF equalisation is provided for both the record and replay heads to compensate and maintain a flat frequency response.

As the head wears, the profile at the front of the head changes from a curve to a flat. At low frequencies, the recorded wavelengths on tape are so long that the amplitude is affected by the head profile. If the profile changes over the life of the head, the LF response of the machine also changes, so an LF equaliser is provided on replay. Record LF equalisation is not provided because the increased record level could cause the tape to saturate. A 3mm wide flat usually indicates that head replacement is due.

Tape wear:

In general, audio tapes are used once only because a recording is made and then edited by physically cutting out unwanted sections with a razorblade and then joining the tape back together again with splicing tape. The result is a weakened tape, so it is never re-used, only replayed as necessary. Occasionally, a tape may be used for odd short recordings and when this tape is full, it may be erased and re-used - this tape is often known as a "dump tape".

Video tapes are electronically edited, so they are re-used. When video tape rushes have been used for a story and edited down to the master tape, they are kept for some time, and then bulk erased, but a log is kept on the tape of how often it has been bulk erased. Beta SP tapes tend to be used ten times before disposal, but digital formats such as D5 are only used five times.

The one tape that is replayed many times is the alignment tape. Alignment tapes are made on new tape stock. As the tape is repeatedly used, the surface of the tape becomes smoother due to the lapping action of the heads. Because the tape is smoother, it now makes more intimate contact with the replay head, and the reduced spacing loss increases the replay level of all but the lowest frequencies. In order to achieve a flat response, LF replay equalisation is adjusted. On record/replay alignment with a new tape, the record head to tape contact and replay head to tape contact are the same, and spacing loss does not change, so bass frequencies are replayed at their correct level vis-a-vis other frequencies, and the incorrectly boosted bass replay equalisation becomes apparent.

Moral: Worn alignment tapes cause a bump in the LF response. Even though alignment tapes are expensive, they should be replaced frequently.

Because of the danger of burnishing the tape, alignment tapes should not be used any more than is absolutely necessary and should certainly not be repeatedly spooled backwards and forwards across the heads as this causes unnecessary wear. It is common practice for alignment tapes to be kept in a locked cupboard with the key held by only one engineer so this engineer knows the state of the tapes.

Mechanical considerations:

Mechanical considerations of tape recording split between the problems of causing the tape to travel past the head at constant speed, and maintaining the correct alignment of the head to the tape.

Correct speed:

If a recording is made and replayed at a different speed, then the duration of the recording will be incorrect.

For a broadcaster, correct speed is critical - a programme will be recorded and edited to be 30 minutes long to fill a 30 minute slot precisely. One second error is too much - the error should be less than a second, requiring a speed error <0.05%.

Correct pitch:

If a 1kHz tone is recorded, we could crudely say that 2000 bar magnets are recorded in one second. If replay speed is fast, then more bar magnets will pass the replay head in one second, and the result is that the frequency has been increased. A semitone is 6% change in frequency, and people with perfect pitch can detect errors of ?1%, so the correct pitch requirement is much looser than the correct duration requirement.

Music recording studios are not especially worried about exact duration, and used to use tape machines whose speed was locked to mains frequency. Mains frequency is guaranteed to keep a clock to correct time, but short term frequency variations of ±2Hz from 50Hz are permitted, which equates to 4%. Thus a recording could be made when mains frequency was 4% low, but replayed for mastering to another format when mains frequency was 4% high. An example of this occurred in 1959 when the most famous jazz recording of all time was mastered by CBS. “Kind of Blue” by Miles Davis was mastered at the wrong speed. Worse, nothing was done to correct this error, even when the original recording was mastered for CD - it was only in the last few years when the recording was re-mastered for CD that the pitch error was corrected!

The capstan servo:

The tape is drawn past the head by trapping it between a capstan rotating at constant speed and a freely rotating pinch roller. In order to achieve the requirement of correct duration ±0.05%, the capstan must rotate at correct speed to this accuracy.

If the capstan motor has a means of producing a known number of pulses per revolution, then the frequency of these pulses will be directly related to rotational speed. The ways of producing these pulses vary from optical methods involving an LED and a phototransistor with the optical path interrupted by a rotating Maltese cross or spoked wheel to magnetic methods using the leakage field of the motor, or even a rotating drum fitted with six individual permanent magnets which pass over a coil. The pulses are either known as tacho (short for tachometer) pulses, or Frequency Governing (FG) pulses, frequency is typically >100Hz square wave.

Quartz is a piezoelectric material. This means that if a voltage is applied across opposite faces of a quartz crystal, a mechanical stress is produced within the material and it bends. Changing the polarity of the voltage causes the crystal to bend in the opposite direction, so the crystal is an electro-mechanical transducer. Just like a ruler twanged on a desk, the crystal has a frequency at which it is mechanically resonant. The significance of the resonance is that its frequency can be finely adjusted by trimming the dimensions of the crystal. Because the crystal is a transducer, the mechanical resonance is seen by any surrounding electrical circuit as an electrical resonance, so an oscillator can easily be made. Quartz crystals can be bought cheaply, and a 1MHz oscillator with an error of <±30ppm can be made for <£2. With a little care, the errors can be reduced to ±1ppm, which is why quartz crystals are the basis of modern watches.

If the precision 1MHz quartz master oscillator is followed by a divider made of D-types or JK flip-flops, any required lower frequency can be produced with equal accuracy. Thus, the master oscillator can be divided down to the frequency that the tacho pulses would be if the capstan were rotating at the correct speed. If the frequency of the two pulse trains is compared, an error correction signal can be generated which is fed to the capstan motor.

In practice, it is easier to compare phase than frequency, but correct phase implies correct frequency. Because one of the signals is derived from the motor shaft (the tacho pulses), and another (the correction signal) is applied to the motor, a loop is formed. Phase is compared, allowing the tacho pulse frequency to be locked to the divided down master oscillator, so the complete servo system is known as a Phase Locked Loop (PLL).

Wow and flutter:

Not only do we need the tape to be transported past the heads at constant speed to ensure correct duration, but we also need to ensure that there are no short term deviations from correct speed that would cause short term frequency errors.

If the tape speed was momentarily fast, it would cause an increase in replayed frequency, if momentarily slow, a decrease in frequency. Short term speed errors frequency modulate the replayed signal.

The term "wow" is used to describe speed variations that change replay frequency at a rate of <5Hz. This means that there could be a variation of speed that changes from fast to slow at a rate of up to 5Hz. Wow makes a piano sound as if it is drunk! Flutter refers to variations in speed >5Hz, and makes a piano sound rough. The distinction between wow and flutter is broad, and different authorities may deem the crossover frequency to be different.

Because wow is a slow variation, the PLL of the capstan servo is quite effective at reducing it, so although music recording studios do not need the absolute speed accuracy of broadcasters, they obtain it by default. Flutter is combatted by adding a flywheel to the capstan whose rotational inertia resists the fast acceleration and deceleration required to generate flutter.
Tape tension and spool servos:

As the capstan draws tape past the heads, the tape must be collected by the take-up spool. Equally, the supply spool must release tape in a controlled fashion. The take-up spool therefore has a motor that rotates to collect the tape, and the supply spool has a motor that attempts to rotate in the opposite direction to provide back-tension.

If tape motion is to be totally under the control of the capstan servo, the supply and take-up tensions must be equal and opposite - this would allow the pinch roller to be withdrawn and the tape would remain stationary. Unfortunately, when the recording begins, the supply spool is full and the take-up spool is empty, but at the end of the recording, the situation is reversed. The changing effective diameter of each spool means that the spool motors cannot operate at constant speed, and that tape tension must physically be measured.

If a spring arm is pressed against the tape, its deflection will be proportional to tape tension (Hooke's Law). Arm deflection can be measured by mounting the spring arm on a potentiometer having a constant voltage across its ends. Wiper voltage is thus proportional to tape tension and can be compared by a high gain amplifier to a reference voltage. The output of the high gain amplifier is proportional to the tension error, and this error signal can be sent to a motor to control tension. Thus, each spool motor has a tension sensor allowing supply and take-up tension to be accurately controlled by feedback.

Tape tension and spacing loss:

Correct tape tension is important because it is the tape tension that provides the force to press the tape in intimate contact with the heads.

Spacing~loss~(dB)^{(1)}~=~{5 5d} over lambda d = distance, ? = recorded wavelength

Since imperfect contact causes spacing loss, non-constant tape tension causes amplitude instability at HF. Spool servos can be made more elaborate to control tension, but one alternative is to add a second capstan and associated servo system. The traditional capstan is after the heads, but the second capstan precedes the heads and attempts to run fractionally slow, producing an extremely controllable tape tension. This dual capstan technique is expensive and thus rarely used (Nagra T).

Azimuth alignment:

Azimuth is the angle of the head gap (viewed from the front) to the tape. If record azimuth is different to replay azimuth, effective head gap is wider, extinction frequency falls, and HF loss occurs. Additionally, because one section of the head has been moved along the tape with respect to the other, inter-channel phase errors are created.

Since incorrect azimuth causes HF loss, correct azimuth is found by replaying the 14.5kHz segment on an alignment tape and mechanically adjusting head alignment for maximum output. Fine azimuth adjustment then minimises inter-channel phase errors.

Zenith alignment:

Zenith is the angle of the front of the head surface to the tape as viewed along the tape. If zenith is incorrect, part of the head will no longer be in intimate contact with the tape, and spacing loss will cause HF loss on one channel compared to the other.

Assuming that 1kHz replay levels are identical, the HF equalisers on each channel can be set to minimum and correct zenith found by replaying the 14.5kHz segment and adjusting for equal levels on the two channels.

Head height alignment:

Height refers to the vertical positioning of the head with respect to the tape, so that the head lays down correctly positioned tracks. Incorrect adjustment causes a loss of level at all frequencies, and increased crosstalk as one section of the head reads information from another track.

Correct height can be found by playing a 1kHz alignment tape, and adjusting head height to first maximise level on each channel, and then to minimise crosstalk.

Head adjustments: Broadcast practice versus music recording practice:

Broadcasters take the view that tape heads are fitted once, correctly, and that they are only adjusted when the heads are replaced. All their machines have perfect azimuth.

Music recording studios may master a tape from another studio whose technical competence is unknown, so they match replay azimuth to record azimuth. Repeated adjustments wear the azimuth screw, so replay azimuth has to be adjusted on every session because it is no longer stable. The engineers see this as further evidence of dodgy record azimuth on other machines and the practice is self-justifying.

Moral: Unless you have good reason to doubt azimuth (or any other mechanical head adjustment) leave them well alone.

Chassis stability and alignments:

Head to tape alignment errors can also be caused by guiding the tape incorrectly due to chassis flexure or guide movement. Well-designed tape machines have precision machined rotating guides with bearings having minimal play, mounted on a rigid chassis. The more sensitive a record format is to alignment errors, the more massive the chassis, so digital videotape machines require an extremely rigid chassis.

Reference:

(1). "Some factors governing frequency response in magnetic recording" R L Wallace Jr. Acoustical Society of America. Meeting, November 1948. Also, Bell Laboratory. 1951

Morgan Jones Nov2001
(Revised Feb2002)

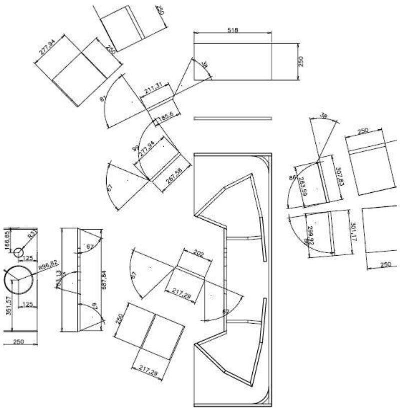
***

How Onur became a DIY'er

(Or how the most boring sections on his website (according to Onur) became one of my most exciting internet reads in a long time)

by ONUR I. ILKORUR (Ph.D.)

I can remember perfectly that it was a cold February Sunday morning.



I got up early and visited the bazaar where second-hand equipment was sold. Just seconds after I entered the bazaar I saw this "thing". It had two handles on the sides and it had a look no different then a welding transformer. I examined it closely and I noticed the writings on it; "Right, Left, 4, 8, 16 ohm". I figured out right away that it was an "old fashioned tube amplifier". With a sour impression on my face I asked the salesman its' price. I don't quite remember the answer but it was 5 or 7 USD. I tried to lift it in order to examine the screws under it but it was very heavy. Then, I gave it a second thought weather to buy it or not, since it was very heavy and there were no **transistors** at all! Being a technology lover, CD addictive kid, it didn't impress me at all. Besides I was very comfortable with my beloved 1.5 KW mono-blocks and 1 meter-cube 18" front horn loaded subs with 2" front horn loaded driver units. What combination of audio equipment could be more satisfying then those?

Anyway, I offered the salesman 4 USD and he granted it. With the same sour impression on my face I carried it to my car and then drove back home.

I cleaned the **mud** and polished the chrome handles before I plugged it to mains. After the thick mud cover has gone away the firms' label appeared on it. It was a **Radford STA-25**. When I learned that the "25" was standing for 25 watts I thought I did make a mistake by buying it but it was too late. At least it was **stereo** I thought to myself miserably! More search about it over the net yield more results. Well, after all it was a Class AB1 tube amplifier with Ultra-Liner output transformers, which can deliver *only* 25 watts.

Every time I turned on the mains switch, it was blowing up the fuse. I realized that one of the Diodes in the AC/DC converting circuit at a fault. I changed it and the tubes started to heat up. Then I removed the 1.5 KW mono-blocks from my system and I plugged Radford in their place. I turned the volume of my pre-amplifier well up since I wasn't expecting any sound from a 25-watt amplifier. After waiting 10 or 15 seconds the sound emerged the room.

The sound was beyond my imagination. My speakers were dancing in their places. Radford seemed to have endless power, since every time I turned the volume on a little it could deliver more power to the speakers. At that moment I realized that some things weren't developing parallel to the technology.

Although they were big and powerful, the sound stage and the sound character of my loudspeakers weren't that good. Also they were occupying a lot of place in my room. So, I've decided to build smaller enclosures with the same low frequency response and dynamism.

This is how exactly I became a Hi-Fi hobbyist.

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The Story

(Of My Search for the Perfect Ella)

by David Baron

The Story of the digital Bias control

It all started when I finished building Ella, a 40W push-pull class AB tube kit from DIY HiFi Supply. I adjusted the bias voltages for all four tubes, put my favorite piece on the CD player (The Moldau, Smetana), and started to listen to my newly created amp. A week later, just to make sure that it is Okay, I checked the bias voltage, and found that it had changed. I adjusted it again and made a mental note to myself that I needed to keep an eye on it. And so I did. It kept changing. I then checked the 120VAC and found that it changes, I then thought to myself that the changes in the wall outlet must be the reason for the bias voltage fluctuations. Like a true DIY'er, I was on a mission to fix it.

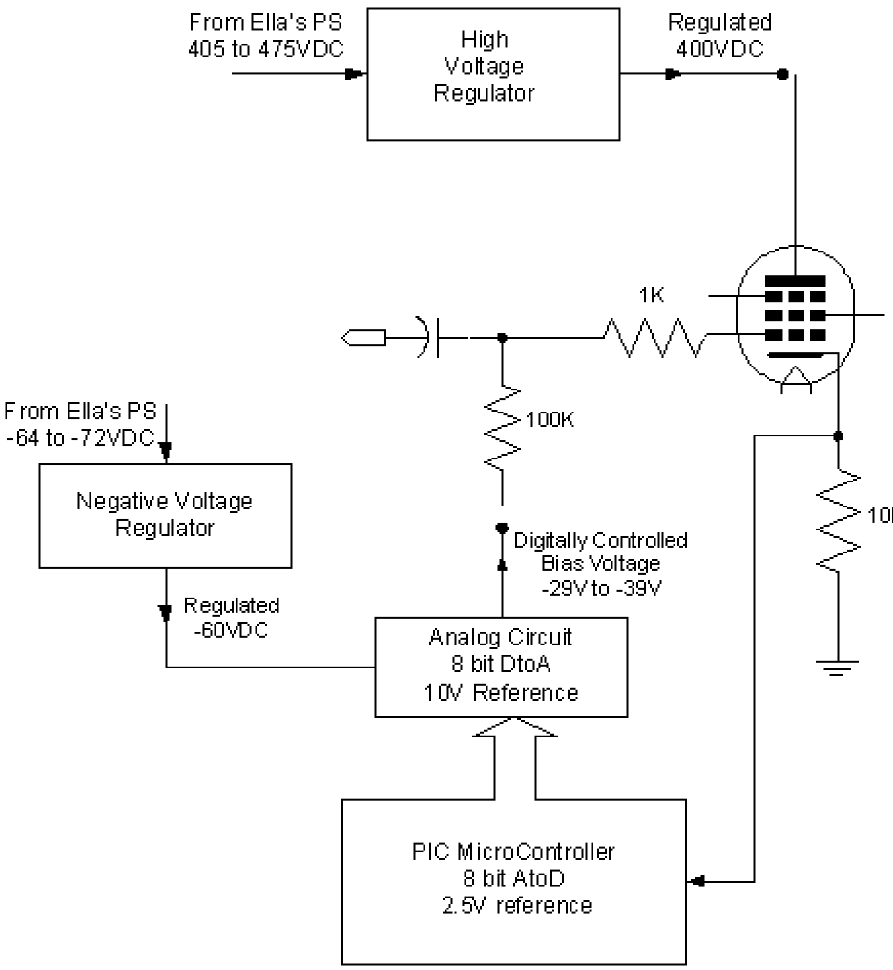
The first thing that came to mind was to regulate the B+. I couldn't find any kit that I liked so I decided to design it myself. After a few smoked parts, mostly from heat runaway, I had a B+ regulated power supply. To my surprise, the bias voltage still fluctuated. It continued to fluctuate as before. Scratching my head I came to the conclusion that I also needed to regulate the negative bias voltage, so I added another regulator. This one is low voltage and very simple to build. To my amazement, the bias voltage continued to fluctuate as before. Scratching my head harder, I started to research tube theory and concluded that thermal instabilities may contribute to bias fluctuations, the solution for that would be an automatic bias adjustment circuit.

I couldn't find any circuit examples for automatic bias adjustment and in addition, if there was an analog solution for self adjusting bias it would not work well for push-pull class AB. Since each tube has half of the signal and it does not average to zero as in class A.

At that stage I was quite frustrated, and told the Ella story to a co-worker at my day job. He is a down-to-earth hardware engineer and he replied "David, you are an embedded software engineer, you can control the bias with a microchip". Then it hit me. That is a great idea. At my day job as software engineer, I write code for 32 bit microprocessors, and deal with time constraints of microseconds. Bias control is in the range of tens of milliseconds, piece of cake! I started to look around and found a very cheap 8-bit microcontroller (PIC by Microchip Technology) with all the features I need; my co-worker helped me out with some of the analog circuitry and that's how my Ella became the perfect singer with a stable bias voltage.

2. Overview of digital bias control

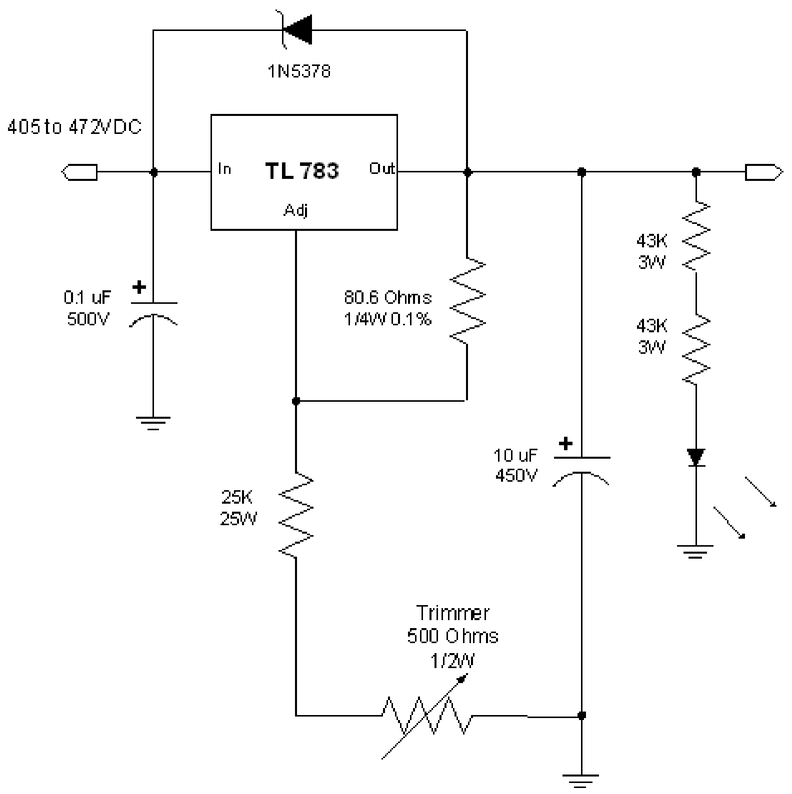
The system overview of Ella's digital bias control is displayed in the Figure below:



The high voltage regulator supplies a steady plate voltage. Similarly, the negative voltage regulator supplies steady voltage to the analog circuit that is controlled by the PIC microcontroller. The analog to digital converter (ADC) is integrated into the microcontroller, and acquires the voltage over the 10R viewing resistor. The code that is embedded on an integrated flash memory of the PIC, checks for differences between the acquired voltage and the desired voltage and adjusts the output of the digital to analog converter (DAC), accordingly. The changes In the DAC output result in changes of the negative bias voltage that controls the flow of current in the tube. The variations in the current flow change the voltage over the 10R, the code gets that voltage from the ADC and changes the DAC output as needed to achieve the desired voltage of the 10R, and hence, a stable current flow and that's how the bias control works.
Simple control loop isn't it?

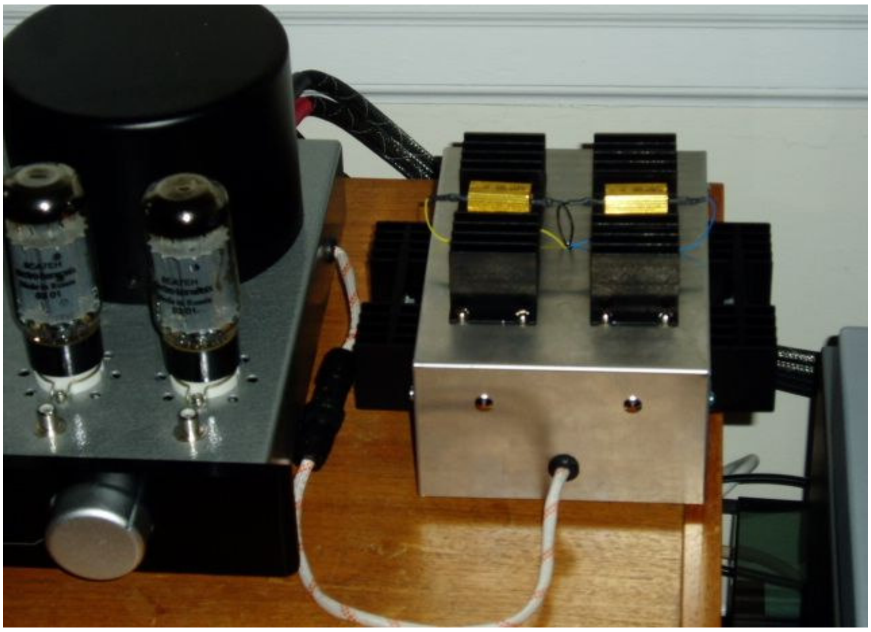
3. The High Voltage Regulator

This regulator is based on Texas Instruments, TL783, with regulated output adjusted to 400V as shown for one channel below



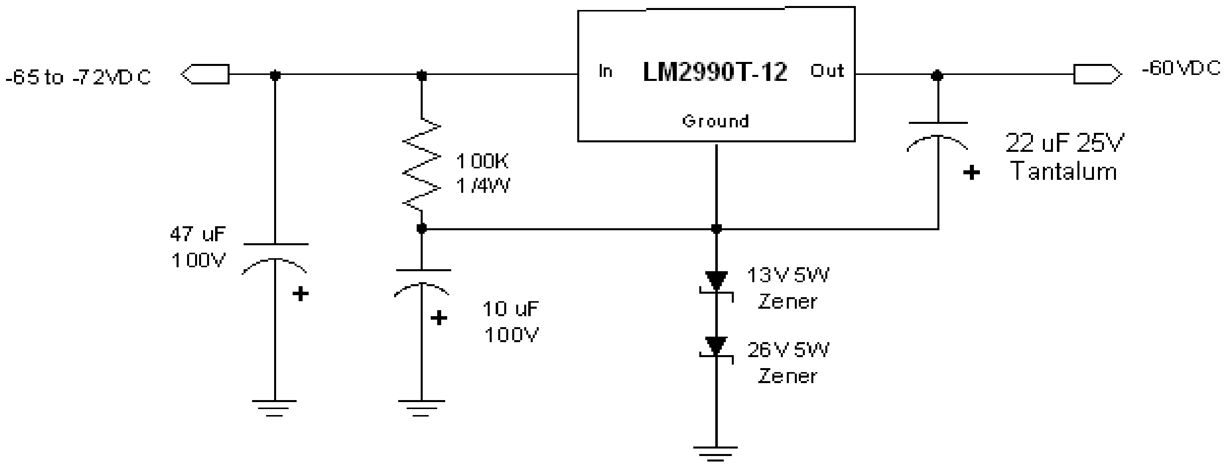
There is nothing complicated about this circuit except for the selection of the resistors and heat sinks. I underestimated the heat that was generated and ended up with quite a few burnt parts. The 100V Zener diode protects the TL783 from power surge. It turned out that one 100V 5W Zener burns easily I therefore replaced it with six 16V Zener Diodes in series and had no problem with this circuit ever after. The trimmer is used for fine-tuning of the regulated output voltage level.

The top heat sinks are for the two 25K wire wound resistors and the two TL783 are mounted on the side heat sinks. The cable has five wires, high voltage from Ella power supply, ground, chassis, and two output lines of the regulated B+ for the two stereo channels. All the parts you see are available at Mouser.com.



4. The Low Negative Voltage Regulator

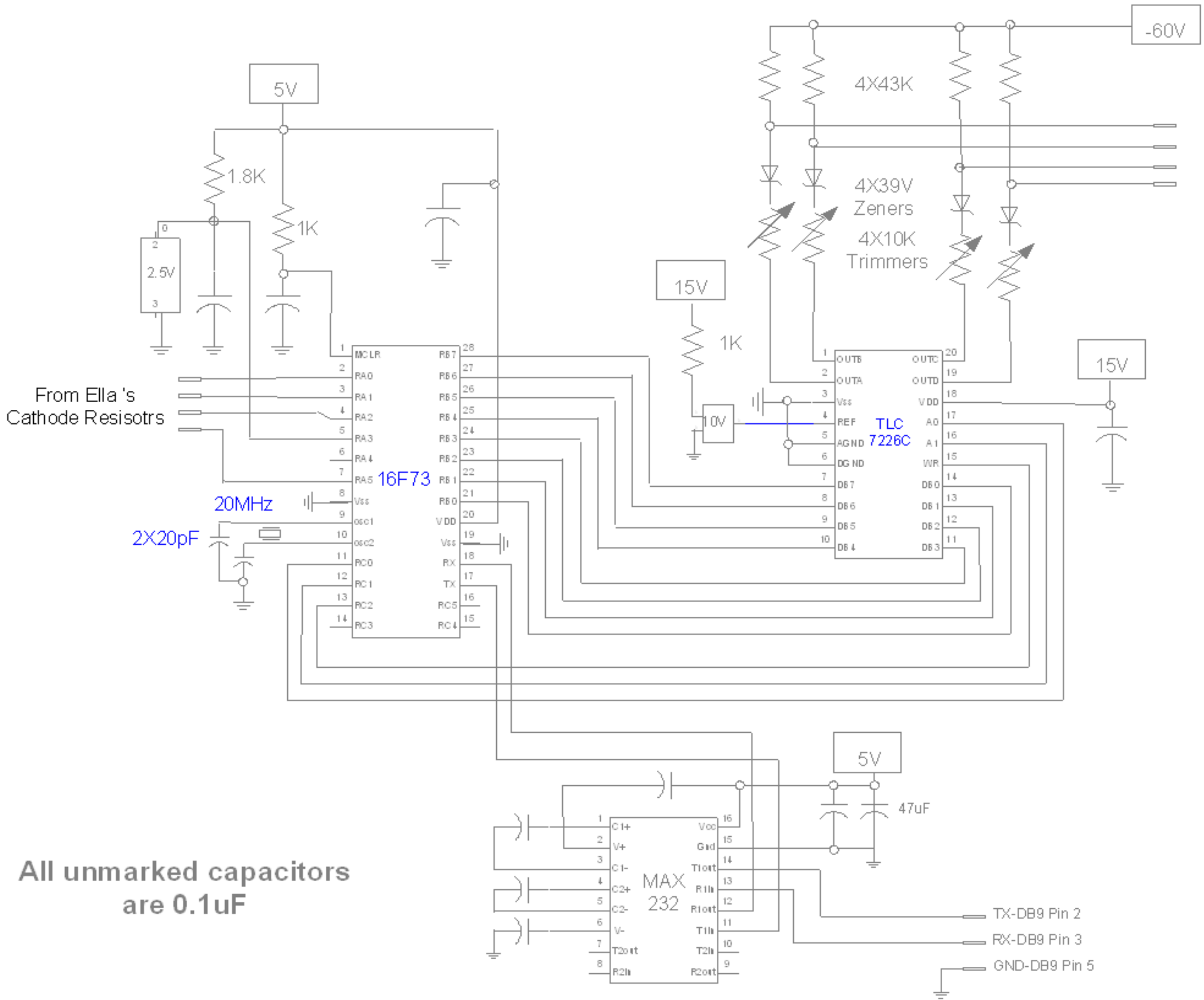
This is a simple circuit where the ground of the 12V negative voltage regulator is anchored by 39V combination of Zener diodes, as shown:



5. The Digital Circuit

The digital circuit is made of hardware and software. I'll start with the hardware and then explain how the software controls it. The heart of the digital circuit is the PIC microcontroller. These chips are very versatile, inexpensive, and widely used in the automotive industry, and home appliances. Mouser electronics carries large selection of them in the range of $2 to $5. I selected two microcontrollers, the 16F73, and 16F876A. They are very similar with 28 pins, ADC, RS232 (UART), 22 I/O lines, on-chip flash memory for code, and on-chip memory for data. The main difference between the two is that the former has 8-bit ADC and the latter has 10-bit ADC. In fact, there is larger selection of microcontrollers with 10-bit ADC than 8-bit ADC. As it turned out, the 8-bit worked better, and I'll explain that later in more detail.

The other main component in the circuit is the DAC. It is a quadruple 8-bit digital to analog converter, made by Texas Instruments (TLC7226C), also readily available at Mouser. It is connected to the PIC through 11 lines, 8 data lines, and 3 control lines. These lines are TTL output lines from the PIC to the DAC. The circuit is built on a PC board made for PIC microcontrollers by DH Microsystems (dhmicro.com) for DIYers. I got the bare board with DB9 connector option. It is called Rapid Development Board Ri28XL ($12 from apogeekits.com).



In addition to these main components, there is a standard RS232 driver and analog parts as shown in the circuit schematics shown above:

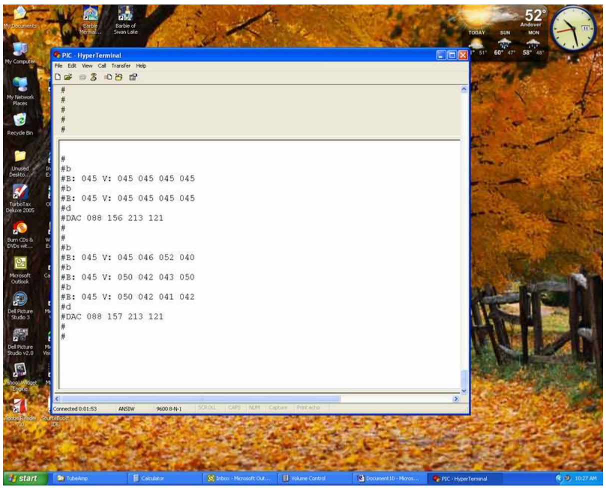
The DAC outputs anchor the 39V Zener diodes with adjustment potentiometers. When the DAC output voltage is zero the negative voltage is about -39V. Upon increase in the DAC output voltage, the negative bias voltage decreases up to maximum of 10V below the Zener voltage, i.e., -29V. The MAX 232 is connected to the DB9 connector and through pin-to-pin cable that goes to a standard serial port on any PC.

The photograph below shows the PC board built with the parts of the above drawing, along with the power supply and other components (switch, LED, etc.).

The shielded cable carries the four low signal wires from the 10R resistors, and the other cable contains the four negative bias voltages and ground. The connector underneath the PC board is used for downloading code into the flash memory of the PIC. Once the code is working satisfactory, there is no use for this connector and the DB9, and both stay away from the outside world when the box is closed with the cover (not shown).

The software is the most challenging part of the design. First one needs to write, compile, and download the code into the flash of the PIC. Then, there must be a way to test and debug the code. To start with, I found a C language compiler free for non-commercial DIY use from SourceBoost Technologies (sourceboost.com). Next, was the USB Flash PIC programmer from alltronics.com. This is a $32 device that downloads code into the PIC flash memory. And last, but not least, is 12 feet cable that I made to connect the RS232 output, the DB9 female connector in the picture above, to a serial port of my PC. On the PC I ran a standard hyper terminal connection that displays anything that comes from the PIC microcontroller, and sends what I type in to the PIC. The hyper terminal is the test and debugging tool that I used to view the parameters such as acquired ADC values, maximum differences of those values, DAC outputs, and other parameters.

With software development environment ready, it was time to design, write, and test the code. The design is based on a 25ms internal timer ticks that initiate data processing. At each timer tick the ADC acquires the voltage of one input channel. Since there are four input channels (one for each of the four tubes), each channel is sampled at a rate of 100ms. When the ADC data is ready, it is passed to the processing code where the value of the data is stored in a history array and then compared to the other values that were stored previously. A maximum difference search is performed on the array and if the result is a difference of 1 count or less it is considered a quiet time. It means that music is not playing, or playing so low that a no change larger then 0.0097V (less than 1mA change) was detected on the 10R. Under this condition, the desired value of the 10R voltage is compared to the acquired voltage, and if there is a difference, the DAC output is adjusted. After another 100ms, if the same condition still exists, the new ADC value is compared again to the desired voltage and DAC output is adjusted if needed. Each of the four ADC input channels is adjusted independently with its own history array. The length of the array defines the maximum quiet time. For instance, array of length 15, is 1.5 seconds of history samples. I experimented with history lengths of 1 to 3 seconds, and settled on 1.4 seconds.



Below is a screen capture of the hyper terminal showing the ADC and DAC voltages. The first two lines show the reading of the 10R voltages during quiet time. The first value is the desired value followed by last four ADC acquired values. The third line displays the DAC output values. Following couple keystrokes, there are three lines that show the 10R voltages during music playtime. The last line depicts the four DAC outputs during a playtime where no much change occurred as expected.

The display is in counts of 0 to 255 for a scale of 2.5V for the acquired ADC voltages and 10V for the DAC output voltages. The first displayed value is the desired (programmed) voltage on the 10R, (45+1)/256*2.5 = 0.45V followed by the four channel input values. Likewise, the displayed value of 52 counts presents 10R voltage of 0.5175V. The DAC voltages are displayed next, and as you see, each tube requires different negative voltage to achieve the desired current flow. For instance, the first DAC voltage is (88+1)/256*10=3.475V which translates into –39 + 3.475 = -35.525V negative bias voltage at the grid, similarly, the third value of 213 is translated into –30.64V. What a difference!

The letter b that I typed is sent to the PIC and the receiving code processes it according to a simple single character protocol ID, and sends back the four recent values of the ADC with the line #B... The same protocol works for the DAC by using the letter d. The most recent four values of the DAC are sent back with the line #DAC... The prompt # is returned by the PIC code if I hit the Enter key on the keyboard with no letter following it; just to make sure that the link is alive. I have the habit of tapping on the Enter key a few times when I bring up the hyper terminal screen and I like to see the responds, otherwise, I get paranoid and suspect that the PIC is dead. I can also change the desired 10R voltages manually from the keyboard for testing, as well as the DAC outputs, history length, and other parameters.

As I mentioned earlier, I tested the 10-bit ADC and I couldn’t stabilize the control loop that adjusts the DAC. It appears that 10-bit is too sensitive and the ADC reads some noise on the way. I did not give up yet and have some plans to test it further, but for now I have great results with the 8-bit ADC. The voltages over the 10R resistors vary less than 1% across the four tubes, and I am very (very) happy with what I hear!

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Very Elementary PSU theory by Bas Horneman

Apart from my hilarious fake readers letters I have not written anything for DIYMAG (And the fake readers letters were part plagiarism).

However I have been pondering about writing an article since the inception of DIYMAG. The main reason for not writing an article has been that I wanted to write something epic....some kind of magnum opus that would really impress the readers (I am after all the Editor) Browsing through my memory banks did not generate sufficient material for such an article. The other reason of course is that I'm lazy. But I digress.

A couple of posts of mine on PSU building over at DIYAUDIO were received with enthusiasm so I've decided to write about PSU basics for valve amp builders. Not again I hear you crying!...Yes again... I say.

Being a relative newby helps when you want to explain something to other noobs. Sometimes when you try to explain something you realize that you don't actually have the theory to explain something, but that is perhaps a different article. The reason why sometimes it takes a newbie to teach a newbie is because you are on the same level and you can't resort to this or that complex formula, simply because you don't understand the formula yourself.

But before I start with my article a little anecdote. (I love anecdotes). When I built my very first psu (for my 6n1p preamp) I decided to test it without the tube connected...in case I would make the tube explode or something. What puzzled me after firing up the psu was that the voltage before the voltage dropping resistor was the same as after it??? I then posted over at the Audio Asylum and asked why. Someone answered "Get me a rope". It took me about a minute to realize what he meant... "Let’s hang this guy" is what he meant. (O by the way...that person if you are reading this...please don't download this magazine anymore. Your subscription has been cancelled) Just kidding. I don't mind that remark. I did however...to be honest...blush for a brief moment. But it was a pity that the guy did not explain to me what I had been doing wrong.

However every cloud has a silver lining. Because shortly afterwards I got an email from Steven Oda giving me a virtual pat the back and whispering ...there there...don’t cry...;-). And en passant gave me a little nugget of information about the psu. One of his mentors had told him a psu is like a stream. The more rocks (resistors/chokes/capacitors) that are in the stream the more turbulent it will become. Now obviously we should not take that literally. Because our stream is turbulent already (sort of AC) and we want to make it flow “smoothly” or “flatly”. But the messages is that you can’t *necessarily* make a good PSU by adding many LC or RC sections. Or adding huge amounts of capacitance.

Back to the voltage dropping resistors....For those of you wondering why the voltage does not drop through the resistor. The answer is that a resistor does not drop voltage unless a current flows through it. You could say then that a resistor is a current to voltage converter, because as soon as current flows through a resistor, voltage can be measured across it. The more current that flows through a given resistor, the more voltage will be measured across it (And the hotter it will become).

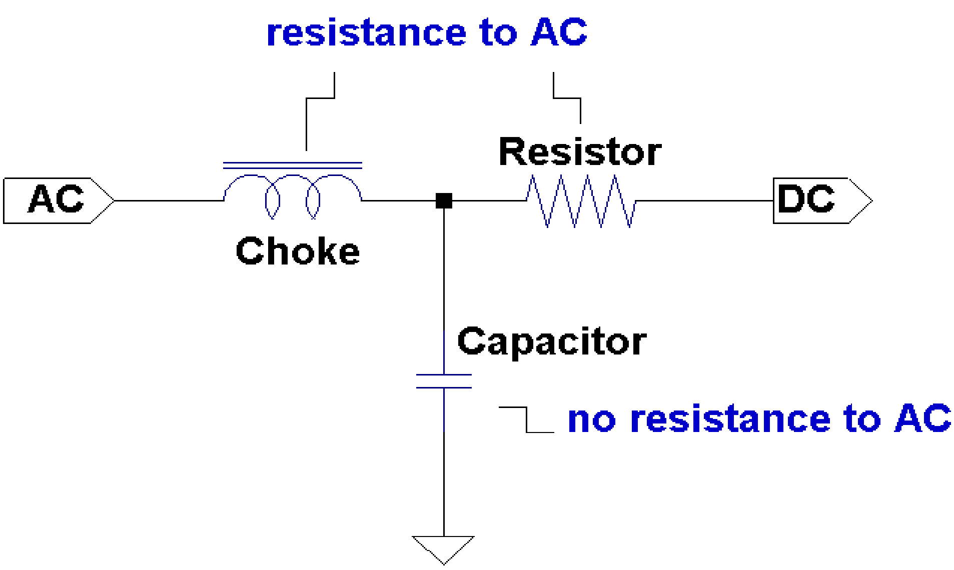
Enough anecdotes… on to the real stuff. The elements in a PSU apart from the transformer, solder, wire and rectifiers fall into 2 categories. Category one: Elements with high resistance to "AC" and elements with low resistance to "AC". The elements with high resistance to AC are resistors and chokes. Those with low resistance are capacitors. But mostly you'll hear the term impedance. Impedance is a fancy word for frequency dependant resistance. For now that is all we need to know.

Our main requirement for a PSU is to supply DC (Or at least that is the way I see it).

So all we need to do is to put the "elements" that provide resistance to AC "in line" with our B+. (Chokes and/or resistors)

And to place the "elements" that have low resistance to AC "parallel" in our circuit to provide an attractive path from B+ to earth. Another way it is often described is: "We have to provide a low impedance path to earth" See the drawing below.

So if AC is our enemy, the chokes and resistors are our knights and footmen who try to impede the progress to our castle (circuit) and the capacitors are the Sirens that try to draw the enemy away from our castle.



*** *End of the first part.*

An interview with one of the friendliest, most knowledgeable and helpful guys in the "business"

Paul "Braniac" Joppa

Hi Paul,

I realize that you are a fairly private person who does not like to be put in the spotlight...So thank you very much in advance for allowing us a peak into your life.

Joppa is not a very common surname? Do you know what part of the world that name originated?

My great-great-grandfather brought the name from Holland, around 1850 I think. It was spelled Joppe in Europe. It came from Germany a generation or two before that.

Where were you born? Where did you go to high school? Where did you study?

Oh, I'm a Seattle boy entirely, including the University of Washington for a BS in mathematics and later an MSE combining several engineering departments to focus on acoustics.

In fact, I've always lived here except for a couple years as a child in Princeton, New Jersey, and 18 months when first married in Athens, Greece. We lived there through four different governments - there were some scary times, but we did get to witness the return of democracy to its birthplace!

I believe you were an acoustic(al) engineer for Boeing? Has that helped you with some of the audio stuff you've done in your life?

I originally got the job in part because of my hobby interest in audio and acoustics. It was the only engineering field where I could work with a degree in mathematics - the engineering degree came several years later. So I would say there has been a significant cross-fertilization between the job and the hobby.

I worked mostly in the research group, about equally split between predicting the community noise of airplanes not yet built, and dealing with rotor noise inside the engines. It involved some interesting complex sound fields and acoustical modes, so there are similarities to room acoustics, horns, and speaker cabinets.

Can you tell us a little more on the work you did with Mike of MQ on the FS100PF?

I'm afraid I can't go into much detail there, it's really Mike's proprietary design. We were playing around with layer and section insulations, trading off leakage inductance and capacitance effects. Mike had shared with me some details of the old Peerless 20-20 Plus transformer designs, and we worked together to adapt some of those concepts to a single-ended design. I must say, those guys in the Fifties were incredibly knowledgeable and sophisticated designers! I think we were able to incorporate some things that I've never seen in any books.

Mike has also come up with some very trick ways to improve on the old manufacturing processes, in ways that will enhance the sonics and at the same time reduce the labor involved. I know he's started to incorporate some of them in other designs already, especially our Bottlehead custom designs, and I expect there will be more to come.

Are you also going to make an 845 based amp with your FS100PF? (Assuming you got one)

I didn't get one, but Dan (Doc Bottlehead) got two pair for us to experiment with. They will be aimed at our work with studios; the voltages are just too high to consider kits. And we have a few ideas to try out - I'd love to do a direct-coupled 845 amp, for instance ...

How did you and Mike actually communicate on the FS100PF? Since you guys are on opposite coasts. Via telephone? E-mail? Or a combination thereof?

A combination. We'll brainstorm and get ideas, maybe scope them out, over the phone. Then I'll work out details, run computer models, and the like, and email the results. Mike would often search his files off line, for instance. Not infrequently, what seemed good ideas on the phone turn out worthless, and/or much better ideas develop while working out the details. But we'd neither of us have tried to work them out if we hadn't started where we did. It seemed to work out.

But of course, if we could have done it on cocktail napkins instead of the telephone, it would probably have worked even better :^)

What is your favourite food?

Tough question; I like variety a lot. Game birds come pretty high on the list, especially with a nice Cote de Nuits to wash it down. I learned to cook from Julia Child's books and TV show, so French cooking (with an overlay of Greek) is my basic food orientation.

These days I try to eat closer to a Paleolithic diet - no grains, lean meat and lots of vegetables. Cheese is my great weakness - it's not a Paleolithic staple, due to the difficulty of milking wild animals. :^)

I'm also a home winemaker, and wine is an essential part of dinner at our house. For wines, I can be more positive - I like red wines!

What kind of music do you like?

It took me a long time to understand that my musical center is purely classical. I enjoy a lot of rock and folk - hey, I'm an old hippie after all - and I'm learning to appreciate good jazz since we started working with Jacqui Naylor and Art Khu. But it's really Mozart through Shostakovitch, with Mahler and the late romatics at the top of the list.

Seen a good movie lately or read a good book?

I'm currently working my way through the dozen volumes of Tolkien's early drafts and unpublished works. Let's see ...recent reads were "Confessions of an Economic Hit Man" and "The Mummies of Urumchi"; I'm a big fan of Miyazaki's movies too.

Would you care to describe your sound system at home?

Now that would be embarrassing! The fact is, it's mostly odds and ends, prototypes, etc. The good stuff is usually either on loan or in use elsewhere...Right now, it's an old BIC turntable/M97 cartridge, while my SOTA Comet awaits re-wiring of the tone arm. (For CDs I have a cheap Toshiba from Costco - but then, I don't like CDs all that much.) Phono preamp is a prototype of Mike Paschetto's, since I designed the circuit. Preamp is usually a Bottlehead Foreplay III (upgraded), in fact it's the one pictured in the manual. Just now, it's on loan to a friend and I have his stock version. Power amps are the first proof-of-concept prototypes for the SEX amp - 6DN7 parafeed monoblocks. Once the production prototype Paramours are upgraded with new iron (and the pictures are taken for the manual) I'll try to get them into my system. Speakers are JBL components - alnico 2220's and 2440's with the short horn/slant plate lens, homebrew crossover, in a vintage Barzilay box from the thrift store. Augmented by a pair of the cheapest Parts Express subwoofers - the ones we got for the SEXy Speaker design. Pretty funny really, a 15 inch midbass with a 10 inch subwoofer! I have parts for 6 or 8 more pairs of speakers in the basement, but never enough time to make boxes for them.

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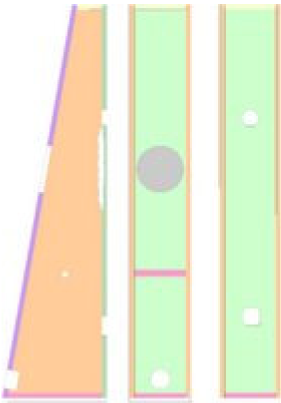
Love potion # 9

By Hudson Miller

The Tale

This tale begins with my Bottlehead S.E.X. amp and a friend commenting that now I've built an amp maybe I should build some speakers. That one quip has led me down the trail of building these speakers and I hope you get some insight on my project and encourage you to build your own.
I spent countless hours cruising web pages in search for a project that had suitable specs. for my Single Ended eXperimenters (S.E.X.) amp, a speaker design that seemed like I could fit them into my living room without damaging the WAF (Wife Acceptance Factor) too much. Then my search led me to the Fostex website – studying the response curves of their drivers, their excellent assortment of speaker plans etc. Finally, for whatever reason, I chose the Fostex 207E as the driver. In hind sight I suppose the choice was the relatively flat response, the entrance into the beloved 40Hz range, and the electromagnetic shielding that did it for me. After investing in a driver I didn't want to end up with speakers that put my cordless phone and television on the fritz. With the driver chosen I did a search on the internet for existing plans in hopes of finding a fellow DIY'er that has figured everything out for me.
The speakers and the design began with the information found at http://www.pi.alegriaaudio.com. Buried in the pages was a Fostex design using the FE207E and the Fostex recommended super tweeter. The design used a pair of book end matched speakers with the driver on the largest side. I was preferential to the look of the Cain-Cain Abby and choose to put the driver on the slanted face instead.
A good friend of mine had me go demo a pair of Von Schweikert speakers which had a rear firing tweeter. I was captivated with the air it gave the room and thought it would be a good idea to try this approach here. After digesting all the web chatter about the bright response from the Fostex drivers – I figured that the rear firing tweeter would help soften the highs from the whizzer cones too.
I attempted to run the totally overwhelming math cad spreadsheets from Martin King's quarter wave web site and put in my own figures for this design. All in all it was a good lesson in how little I know and how smart others are. Basically the numbers and response curve didn't seem to suggest that I was doing anything too terrible and off to the drawing board I went.
Not having any real drawing software I ended up using a copy of Autodesk Actrix (their version of Visio) to draw the plans. (p2). TAKE NOTICE – something's off with the plans as the original version I thought I was doing had the face of the baffle to the floor – in the as built version it's cut short of the base. As I recall it was a bit short of making the base so instead of scrapping the otherwise well fit front baffle I trimmed it short. I like the detail but I didn't correct the plans – or maybe the plans are right on the money for this effect and I measured/cut wrong. Who knows – they are what they are. Costs and Parts Lists are available in excel if you ask.



I have a VERY small work area and even smaller tool storage ability. In all my work area was about 5'x8' so the notion of saw horses, dust free paint shops, and bunches of clamps etc. was out. To this end I ended up having the MDF cut by a pro and I stuck to assembly. I countersunk and tapped before screwing the panels together, gluing the joints when I was happy with the fit up. Even with that precaution there were of course mistakes. Fear not – BONDO saved the day. One of the rear panels went on about a 1/8" off center – leaving a ridge on one side and a gap on the other. A wood rasp to knock down the big stuff and some bondo and everything was "true enough" to veneer.

In an attempt to increase the WAF I involved my sweetie in picking the veneer. She chose Lacewood – I was pleased. In hind sight I would have been better served to learn on something other than Lacewood. The stuff is brittle, expensive, and itchy. But boy does it finish nice.

In the end the project cost a pretty penny by DIY standards – mostly due to the Lacewood but they turned out well. They sound good – to me. There's enough choice in tuning them that you should be able to make them sound good to you too. They sounded better when I heard them with a preamp on my system. This hobby is subjective – there is no perfect. If you like an open, crystal clear sharp high end, with a mid that's emotional but not overbearing then these speakers are for you. If you like to thump your way through your high school and college years they aren't for you. These speakers are for Jazz and in my opinion they do a superb job with it. Regardless, they appear to be very sensitive to tweaks so you should be able to find a happy place for your own preferences. Between stuffing, port tube length, and baffle step correction circuits you should be able to hit the sweet spot. I've ended up naming the speakers LP #9's – because what could go better with S.E.X. than some love potion?? Thanks to Martin King, Paul Joppa, Terry Cain, and Tim Forman for helping answer my questions. You all are pros.

The following section describes some lessons learned that I hope you find helpful.

Lessons Learned

Screwing – I do recommend screwing - the box together. Fit the case, drill, countersink, screw, unscrew, glue, screw instead of fussing with a thousand clamps. I glued generous amounts of Elmer's wood glue to both sides of the joint, looking for glue oozing out the sides of the box. After a minute or two to dry you can putty knife off the excess when it's begun to turn plastic and you're as clean as a whistle.

Wood Choice – I do recommend using MDF for the first run. The stuff is dusty, chips, cracks, and absorbs water. It sure is inexpensive though and if you're happy – build the next ones out of plywood. It will also convince you to hone your wood working skills so you never have to use it again.

Baffle Step Correction (BSC) – I ended up not including one – principally because I only have a 2 W amp and the line level correction just stole too much of the ambiance I was striving for. Some day when I get a more powerful tube amp I'll fiddle around with the speaker level BSC circuit recommended on Martin King's site but for now I'm pleased without it.

Stuffing – I found the driver to like an empty box. I ended up taking out all the recommend stuffing and leaving only a small piece of polyester (blanket or chair batting) directly behind the driver (about 14" long and the width of the box) and just using a staple gun to secure it.

Room Positioning – I found the speakers to sound the best to me with a toe in so the intersection was about a yard or so in front of my couch. This seemed to take care of the BSC, brightness, and all the other offences that the Fostex drivers are accused of having.

Speaker Vent Port Tube – I used the 3" precision port. In hindsight I would put it rear firing so there is some coupling of the bass that is produced with the rear walls. I also fiddled around A LOT with the length of the port tube (both in the math cad software and in the actual box) and found that NO extension sounded best to me. When I added any length to the port tube at all the driver lost nearly all the mid/bass and with it out completely the driver sounded sloppy. So I ended up keeping just the front throat from the precision port as the port tube.

Cutting Driver Holes – Well, nobody taught me how to cut holes with a router. So I drilled a pilot hole, stuck in the cutting bit and proceeded to drag the router around the circle I had drawn. This worked for about 4 inches until the bit began to smoke. I switched to my dremel with about the same effect. Finally I ended up hacking the hole out with my jig saw. I shared my toils with friends who are still laughing at me not using a circle cutter for my router and cutting the hole a little bit at a time working my way through the material. I'll try that next time.

Cutting Driver Holes Too Wide – Well – so much for precision (see section above). I had chosen the hurricane nuts to use on the back side of the panel for putting the mounting machine screws into. I ended up having to bend one side of the hurricane nut up 90 degrees to get the driver into the hole. This worked like a charm and would definitely use the hurricane nut/machine screw combo again.

Internal wire – I selected some Streetwire 14 gauge stranded OFC tinned wire. I think it sounds good. No, this is not an invitation for speaker wire debate. I recommend using the female disconnects you are supposed to. When you are done with your assembly, finishing, and are putting the speakers together for the last time – solder the female plug to the speaker – it makes me feel good that it's on there nice and tight.

Speaker Cup – Initially I thought that it would be smart to be able to take the super tweeter out of the circuit so I got a speaker cup for bi-wiring. I would not do it again as the L-Pad accomplishes the same thing and it isn't as chincy. Who really uses bi-wiring anyway.
Spikes – I like the look – although I suspect there would be more bass if the speaker was sitting on the floor. You choose.

Box Stiffening – I ended up only using one dowel to stiffen the "middle" of the sides below the driver. I'm not sure this is worthwhile and may take it out if I do a plywood version some day.

Gauze for Retaining Stuffing – I used the eyelet screws and tied first-aid gauze to retain the stuffing while I was "tuning". I think I like the idea of an acoustically dead material being used here instead of string. It's a bigger surface area for the stuffing to have to leak past and is easy to stretch out of the way if you want to put more/less stuffing into the box. I ended up with no additional stuffing other than a small sheet behind the driver – again you choose with your earbuds.

Caulk – Clearly, you must caulk the seams to be sure they are air tight – problem is that I couldn't see the top inside of the box once assembled, I couldn't steer my caulk gun from the main driver hole.....I'm open to suggestions here. I ended up cutting a tip off a Ziploc baggie filled with caulk and hand mashing the caulk as high up as I could reach. My guess is there's a glob of caulk stuck to the top of the box stalactite style. It's a trade secret and I'll call it's the Half Assed Touch of Magic (HATM) that must make these speakers sound so good to me.

Box Material – Like I mentioned before I used MDF. I think next time I'll use plywood. No, I will use plywood because MDF is made by the devil and sucks more than the Sahara in summer. Also, speakers like Audio Note and Cain-Cain aren't made out of MDF – I figure they may have some clue about this that I don't have.

Veneering – I enjoyed veneering. I used water cleanup contact cement called Weldwood and got a foam roller to put it on. I used an excess block of wood with the edge sanded a little bit to round it off as the scraping tool and it seemed to work fine – until I got to the top/bottom edge where I lost track of where the underlying box stopped but the oversized sheet of veneer continued and – CRUNCH – I rolled the veneer off the top and folded it over – leaving a splintered wreck in it's wake. LESSON LEARNED – full scale practice on cheaper veneer. Next time I will go ahead and press like a son-of–a-bitch and leave the last 4 inches or so just pressed down with just my fingers then ...
Invest in a hand held router with a veneer trim bit (thanks to Steve). This tool ROCKS. You simply drag it around the edge and bingo – perfect edge just waiting for a quick sand. Same goes for finding the holes for the drivers after you are done veneering. Just tap with your fingers until you find the void, cut an X with your utility Knife stick the router bit in the hole, turn it on, drag it to the side and around you go. Presto perfect! To find the holes for the machine screws you may want to make a template of the layout before you veneer so you can use a hand drill to re-drill through the veneer.

Bondo - If MDF is made by satan – Bondo is God's answer. Use it instead of wood filler – it's faster, smoother, can be sanded out if you screw it up and re-done. Thanks Steve L. for the bondo suggestion.

Finishing – I used tung oil but I think Lacewood wants a high gloss – with less than perfect sanding to start due to the grain and lots of coats with sanding between. Next time – more gloss, more sanding.

Dowels – I didn't want to spend another penny on more dowel just for veneering so I improvised a set of "tent cards" fabricated out of photo enlargement sleeves. I think any glossy cardboard could be used instead. This worked perfectly. Once the contact cement has dried for the recommended 40 minutes or so just lay them out and then begin to place the sheet of veneer over the work and then pull them out one at a time.

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http://hudlyaudio.blogspot.com/ for details

Acknowledgements: John Hope, Morgan Jones, Onur Ilkorur, "Mammal" Niazmand, Hudson Miller, Paul Joppa, David Baron.

Special thanks go out to John C Walton of http://www.tech-diy.com for his donation. (Check out his site. Paradise for the DIY crowd.)

Quotable quote:

"I like to break things with my speakers." – Cal Weldon

How to spot a DIY'er with issues:

"He's the guy who jokingly pretends he can't lift his tube amplifier at DIY meets: Mumbling…aaah…it's too heavy but has a big smile on his face when he says it."

"He's the guy who installs a security system because he's convinced someone is out to steal his Russian paper in oil capacitor stash."

"He's the guy who when you tell him his amp sounds great, keeps a straight face and says…it's ok..but it needs a little tweaking still"

"He's the guy who keeps his most prized transformers next to his bed so he can look at it - when he wakes up - and before he goes to sleep."

If you enjoy this magazine and would like to show your appreciation I would be great if you can in any way help Terry Cain. (No he's not a little dipshit like Ferris of Ferris Beuler's Day Off) But someone who for some reason grabbed my attention a couple of years ago with his speakers. B E A utiful and then I read often about what a wonderful person he is. One could sense the man's friendliness even through the web.

http://6moons.com/showcase/terrycain/terrycain.html

Let's face it, even if you don't normally do this kind of thing. This would be one hell of an excuse to buy audio gear!

For those of you who really don't need audio gear or don't live in the USA but do want to help…you may pledge at

http://www.fundable.org/ and search for " Friends of Terry Cain"
which is run by user mercedes3 who frequents the bottlehead section at the AA.