Thermionic Valve Analogue Stages for Digital Audio

A short overview of the Subject by Thorsten Loesch

1. Introduction

In the last decade the whole issues how to interface the Silicon nastiness of CD-Players and Digital to Analogue Converters to Valve Equipment has received extensive coverage in the Audiophile and DIY Audio Press. Much polemic has issued forth. A number of companies have even introduced add on Valve Stages claiming them to be "the missing links" and the like. This brief article shall not concern itself with the polemic surrounding the issue, or with specific criticism of many implemented solutions.

Nor shall we discuss the inherent futility of attempting to "reform" or "improve" by introducing Valves, a Digital Audio Format which not only suffers from enough build in limitation as to make true "High Fidelity" inherently impossible and for which the vast majority of recordings not even make use of the available quality. Nor shall the issues of massively incompetent digital design (clock jitter, PSU decoupling, managing of RF Noise and so on) in almost all commercially available equipment be discussed here in length.

Before continuing with this Article the august reader is advised to make herself or himself comfortable with the underlying principles of Digital Audio and CD Player in special as no further consideration will be given to this topic.

Instead we shall concern ourselves only with delivering the however compromised output Signal from the Digital to Analogue Converter to the Output Jacks with the lest amount of further perceived degradation.

Considering this task, the use of Zero Feedback circuits seems almost mandatory. This is because with a DAC or CD Player we deal with a generator of massive levels of wideband RF Noise which, if used with conventional feedback based circuits will lead to severe levels of intermodulation distortion. This is easily observable with noise-load testing as much as 1% with leading High End Converters and worse for much consumer equipment.

So, having defined the almost absolute need for zero feedback circuits if our goal is to offer high fidelity without recourse to repeated design iterations and use of expensive superfast, high resolution FFT Analysers and noise-load testing, the only readily available devices with an ability to offer low enough distortion levels in zero feedback mode are thermionic valves, most notabene triodes.

Now, how do we interface triode valve circuits to the kind of DAC Chip's commonly found in consumer digital equipment? To answer this we must first understand that currently two fundamentally different DAC Architectures are being used and that the two different Architectures require a fundamentally different approach to analogue stage design. The two DAC architectures in question are multibit and timeslicing DAC's. Let's look at the technology.

2. Multibit Digital To Analogue Converters

The Multibit DAC is the classic technology used from the beginning and has a very long history outside digital audio. In most simple terms a Multibit DAC contains a number of switches and a resistor network (and some auxiliary circuitry). Each of the switches corresponds to a "Bit" and will switch in a resistor allowing a certain current to be presented on the output of the DAC Chip.

If take the classic Philips TDA 1541 Multibit DAC as example, the MSB (most significant Bit) will toggle on or off a Switch which will switch on or off a current of 2mA. With all Bit's at Zero all switches will be off and the output will be at zero. Switching the MSB on will produce -2mA current and represents basically "digital silence". Switching all other 15 Switches on will add another -2mA to the total now bringing the output to -4 mA. So switching from all bit's off to all bits will change the Output Current from 0mA to -4mA. Many more modern DAC's use a seperate offset current source (often unaccessible for the user of the chip) to bring the output at "digital silence" to 0mA, giving a +/-...mA Modulation.

As this current is normally derived from a reference voltage via precision trimmed resistors the output needs to be held close to zero volts in order to avoid introducing distortion. However, it might be noted that the distortion actually introduced is of a type similar to that of a triode and hence, if we do not specifically concern ourselves with measured performance we can more or less ignore this requirement within reason.

However, regardless how specifically we convert the output current of the DAC into a voltage (in the easiest way simply by using a small value Vishay bulkfoil resistor), with a classic Multibit DAC we generally require some form of voltage gain and quite a bit of it.

Commonly used is a op-amp chip where the output current of the DAC is applied to the inverting input of the Op-Amp, which due to it's normal function presents a virtual ground. Due to the steep slopes of the steps in the output current between changes in the bit's we require a device in this position that is not subject to limitation by it's Slew-Rate. If this is not observed severe Transient Intermodulation Distortion (TIM) will occur which while very audible as aggressiveness, clipped sibilants and similar artifacts that are relatively hard to measure, using conventional methods.

In many existing Players and DAC's the used Op-Amp's are NE5534 or NE5532 or functionally equivalent types from JRC/NJR. All these devices are incredibly ill suited for this purpose and no doubt the often less than ideal sonic character of early CD-Players can be blamed at these devices. In the practical section we will look at options for doing all this with Valves and somewhat better. However, the coverage of classic Multibit DAC's is somewhat redundant, as they have been almost entirely replaced with the other type of DAC, the one I refer to as TIMESLICING DAC.

Multibit DAC's where (and are) made chiefly by Burr Brown (PCM56, 63, 67, 69 and PCM 1700,1702, 1704) Analogue Devices (AD1862, AD1865) and Philips (TDA1541, 1543 and 1545). Others exist and any of the above Manufacturers also made custom chips for large OEM Manufacturers like Technics/Panasonic, Sony and so on). The classic Multibit DAC's often require fairly complex PSU Arrangements and are comparably expensive to make.

In the extreme High-End Equipment (Spectral, Mark Levinson, Krell, Audio Synthesis and Wadia) Multibit DAC's are still the order of the day. Multibit DAC Holdouts in the better budget consumer CD-Player Market are Rotel and Denon. All but their really bottom feeding range uses Multibit DAC's. The top of the line Teak and Pioneer CD-Players (Pioneer only in their latest Model) also use Multibit DAC's. Linn and Naim have been long-standing advocates of Multibit Chips though their more recent lower priced Models no longer use Multibit DAC's.

Almost all pre 1990 CD-Players are of the Multibit Type, however, before trying to hunt such a device down to modify it consider that they have seen a lot of service in the last 10 to 16 Years, spares tend to be unavailable and breakdowns are likely.

One possible option would be to buy (very cheaply) one of the earlier Philips or Marantz CD-Players and remove the SAA7220 digital filter and the TDA1541 DAC. From these one can make a stand-alone DAC, similar to the one offered by Curcio Audio (from where one could obtain suitable PCB's) or the van Alstine Valve DAC.

With budget standalone DAC's one tends not to find Multibit DAC's at all, as they became common only in the 90's. And to buy an older "High-End" DAC for modification is also not such a good Idea as their input circuitry is often jitter ridden and hence will not allow much scope for sound quality.

The best bet perhaps is to start from a Denon (the 825/835 Models and upwards) CDP, a suitable Rotel CDP or the Pioneer PD-S06. Otherwise, the above section is primarily history, for in the current days reigns supreme the Timeslicing DAC.

3. Timeslicing (or Delta Sigma) DAC

The device I call Timeslicing DAC is known by many Names. Technics calls it "MASH", Pioneer calls it "Pulseflow" and Philips calls it Bitstream and so on. These DAC's are often also referred to as Low-Bit or single DAC's. It is in many ways an ingenious Device.

Instead of attempting to make precision analogue silicon (always expensive), they circumvent the issue by using only one "on/off" switch and by toggling this switch very, very fast. The result is a squarewave of very high frequency which is however symmetrical. So once we pass this squarewave through a lowpass filter we are left (assuming Digital Silence) with a voltage of zero.

If we now start taking, let's say, every 65536's pulse out we will achieve a tiny deflection of the Voltage. Depending if we omit the positive going pulse or the negative going pulse we will get a deflection to the positive or negative side.

Now for this to really work for CD Style Digital Audio we theoretically require 2^{16} Pulses (65536) pulses for every 44.1kHz Sample. That means we need theoretically a 2890 MHz (or about 2.9 GHz) Clock to run the DAC. This is clearly not practical and a number of mathematical "tricks" (noiseshaping especially) are employed to get a reasonable performance at much lower Clock Rates (around 16.9 or 33.8 MHz).

Usually the DAC Chip switches the PSU Voltage (+5V) and in order to make the output more immune from the PSU noise all Timeslicing DAC's use a differential output. This means we normally have a positive and a negative output for each channel. These outputs are voltage based and usually carry a 2.5V DC offset and have a swing each of +/-1.4V peak or 1V RMS. This enough voltage to require no further amplification.

These two outputs however need to be combined into a single-ended output and then further lowpass filtering is needed as the signal is overlaid with large quantities of ultrasonic noise. This is actually much more problematic than the already problematic situation with Multibit DAC's. As a result many designers screwed up their analogue stages for the early Timeslicing DAC's up so badly that most DAC Manufacturers started integrating various parts of the analogue stage onto the chip to avoid this (notable exception - Nippon Precision Circuits - NPC).

Later Phillips Bitstream Chip's (like TDA1547) are famous for having been crippled sonically by some Idiot who integrated a analogue stage based on the NE5532 Op-Amp onto the silicon. Another DAC vendor who really made a sonic mess is Cirrus Logic (under the Crystal Brand). After their superb CS4303, which offered pure digital outputs (and which almost no-one could get to sound good as the analogue stages could not cope with all that RF Noise) and the CS4328 which implemented an excellent internal analogue stage, all their other DAC's (CS4327, 4329, 4390 and so on) have incorporated switched capacitor lowpass filters (sonic poison) and lousy CMOS Op-Amp output buffers.

While these measures make the Crystal chip so easy to implement that even the most incompetent and hamfisted designer cannot screw up the sound of the chip too much, they seriously massacre the inherent sound of the device. This makes (all other things being equal and a competently designed analogue stage assumed) the current generation of Crystal DAC's by far the worst sounding Timeslicing DAC's yet, closely followed by the latest generation of low-cost DAC's from Burr Brown.

Of recent Timeslicing DAC have gone further and even incorporated the rest of the analogue stage onto the chip (Philips TDA1305 and Burr Brown PCM 1710/16/28/32 come to mind). Just as it the habit of large corporations, these Op-Amp's used on chip are not your audiophile favorites. On the contrary, with Philips it is of course the equivalent of a NE5532. Burr Brown did not even spring for something like their well-respected OPA2604 or OPA2132/2134, they used a really nasty CMOS Op-Amp.

Of course, these Op-Amp's you cannot see. Nor can you bypass them anymore. They are always there and inside the Black Box you think of as "DAC".

But back to basics. The big advantage of the Timeslicing DAC is that it is an almost pure digital circuit. All and everything can be done in the digital domain, so much so the output from earlier Timeslicing DAC's is a pure pulsewidth modulated squarewave. And unlike precision analogue Chip's, even very complex digital chip's can be made dirt-cheap. Even the integrated analogue circuits are usually of the "5 pence the piece if you buy 1000" type, making not much of an impact on production cost. And let's not forget that many Multibit DAC's require four to six separate supply-voltages. The Timeslicing DAC usually requires only one and that of course costs a lot less again.

And the production cost is exactly the reason for the introduction of Timeslicing DAC's. Any claims for superior performance or any better technology and such, can be safely relegated into the realm of fairytales spun by marketing futzies. Neither sonically or measurable are there ANY advantages for the current Timeslicing DAC's over classic Multibit Designs. On the contrary, they are technologically much less capable of high fidelity reproduction than Multibit DAC's and as they operate on the basis of clock cycle modulation any instabilities of the master clock (jitter) have a much larger impact on sound as with the old Multibit Designs.

However, the presence or absence of Multibit DAC's or Timeslicing DAC's should not be seen as an absolute indicator of quality. As usual, much depends upon the implementation. This is shown by the fact that several Digital Processors or CD-Players using Timeslicing DAC's sound very good (Timbre DAC, Acoustic Precision EIKOS CD Player) and certainly a lot better than the worst Multibit DAC sporting Machines.

I did feel that the discussion of the different technologies was necessary, as too often even experienced Designers tend to view DAC's as a "Black Box" with little concern about how they really work. Together with mistaken assumptions, based on the datasheet of the DAC issued primarily not as design-guide but as thinly disguised marketing material, one often sees a lot of effort and care being lavished on designs based on Chip's that should be buried quietly in a dark spot around midnight and not used in quality audio equipment.

With all this bleak view I have of the CD digital format, it's recordings and the competence of the designers of much of the equipment used to play CD's one is tempted to ask: "Why bother?". And I must say that I have almost taken this view. With a huge (and still fast growing) record collection and excellent analogue replay equipment I personally can do (and mostly do) without CD's.

However even I as founder (and currently only Member) of "The Holy Order of the Latter Day Luddites" own a CD-Player (a DVD Player actually) and own a number of CD's. Indeed, as I type this I'm listening to Phil Collins from CD. Others have committed themselves entirety to the CD Medium. I doubt that ANY current audiophile or ultra-fi Audio-System is without a CD Player.

Love them, hate them, we have to make do with them and then extracting the last bit of quality becomes essential.

4. Analogue Stage Requirements

After having discussed the different types of DAC we can now sit down and formulate a few basic requirements for analogue stages (solid state or thermionic valve based) to be used with the respective DAC architectures.

An analogue stage for current output (Multibit) DAC's requires a low input impedance (the lower the better - about 100 to 200 Ohm appears to be a sensible maximum), a fairly high voltage gain and while it must withstand some notable RF Noise, no extreme lowpass filtering is required.

This suggests a resistor for I/V conversion, followed by a suitable gainstage with plenty of gain, to present a reasonable output Signal. Schemes for this have been long available and what is presented below for Multibit converters is mostly a rehash of other peoples work with the odd little touch from me here and there.

An analogue stage for timeslicing (Bitstream/Delta Sigma) DAC's requires a very low voltage gain, notable RF Filtering to protect following equipment from high levels of ultrasonic noise. On the surface it also seems to require a form of balanced to single-ended conversion. It is the latter requirement that has thrown many people off track.

I have seen one solution (by Chris Found) which uses differential SRPP valve stages but ends up with by far too much gain, excessive output voltage and a substantially unnecessary complexity. Another rather ingenious solution (Sephano Perugini) was to use a transformer, which of course works great, offers galvanic isolation form all the noise in the DAC and allows bandwidth limiting (to eliminate ultrasonic noise) in the most simple and direct way.

While each solution has it's merits, one is too complex and has too many glowing bottles while the other is really ingenious but has no glowing bottles and relies upon a very high specification 40k : 40k Transformer, neither a readily nor cheaply available part. A pair of good quality, wideband 40k line-level transformer can easily exceed £ 60 and the best from Jensen cost several hundred dollars each.

So I rejected either approach and looked again at the issue. I decided to try the Microsoft approach* and decided to "just keep driving because no-one will notice anyway". So the output was taken from only side of the balanced pair. Some form of lowpass filtering is needed together with a buffer stage. An LC Filter followed by a cathode follower seems to answer the purpose admirably. My own original solution omitted the cathode follower entirely and later introduced a J-Fet based buffer. This works and sounds very good indeed.

All this said, under certain conditions (amplifiers or preamplifiers with an input impedance of 47kOhm or higher) we can completely omit ANY analogue stage for Timeslicing DAC's and use only a simple passive circuit, avoiding any additional active components. This is especially true for recent devices from Burr Brown (PCM 1710,1716,1728,1732 Series), Analogue Devices (insert list of devices) and Philips (TDA1305).

Also for Multibit DAC's it is possible to completely avoid active analogue stages if the following amplification chain has enough spare gain available (usually one would expect to see about 25 to 100mV RMS Input sensitivity for clipping).

The cases where one can afford to operate without external analogue stage are comparably rare though. I might discuss the "DAC-Direct" Output Topology I have developed for both types of DAC's another time. For the time being, lets get practical.

**Three Microsoft VP's, notably a Marketing Manger, a Helpline Manager and a Software Engineer drive down the Autobahn in a BMW. The Car has a flat Tire. The marketing futzi immediately proclaims "We need a new Car!". The helpline guy say: "Just a minute, let me ring up the BMW Helpline, maybe they can ship us a new Wheel?". Finally the Software Engineer says: "What's all the fuss about, guy's? Just get in and keep on driving, no-one will notice!".*

Practical Implementations

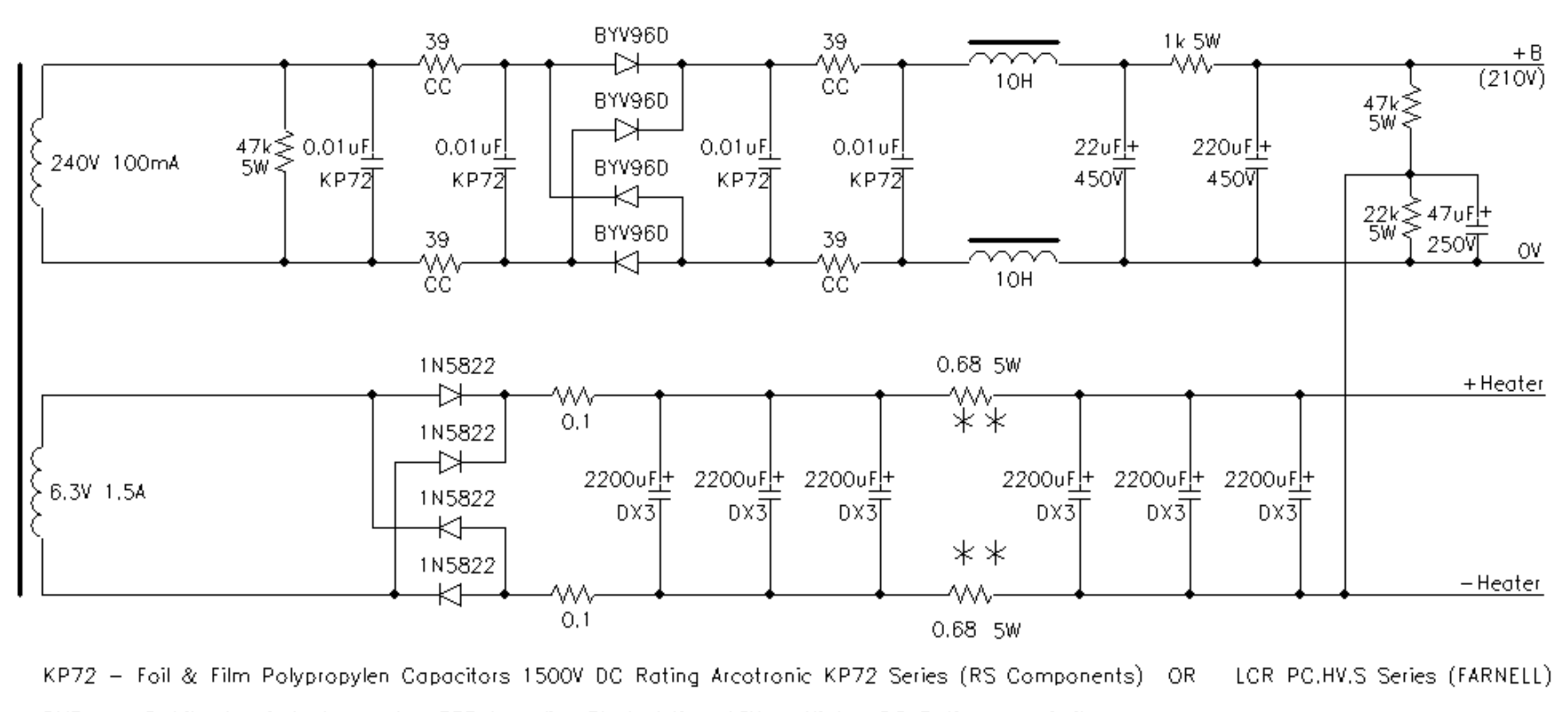
1. I've got the Power - a powersupply digression

Prior to launching into a lengthy discussion of the specific example circuits for analogue stages let's briefly cover a simple, fairly inexpensive but high performance powersupply for these circuits. For simplicity of procurement it primarily uses components from Maplin.

Maplin have an inexpensive Valve Mains Transformer in their program, as well as inexpensive but adequate chokes. Used for this PSU is one XP27E Mains Transformer (around £14.00) and two ST28F Filter Chokes (around £ 7.00 each). Other sources exist, but I seriously doubt that a 50mA 20H Choke and a suitable mains Transformer can be obtained for under £ 30.00 the set from any other source. The quality is more than adequate.

Most other components are standard industrial quality ones. The electrolytic capacitors in the HT Supply should be of high quality but it is not necessary to spring for Elna Cerafine's or Black Gate's. The lowest cost suggested devices are Nichicon VX Series axial types, available from good electronic stockists.

The Circuit for the Supply is given below.



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DX3 – Dubiler low inductance, low ESR long live Electrolytics, 16V or Higher DC Rating or similar

CC – Carbon Composite Non-Inductive Resistors 2W (Allen Bradly, Morganite, Rikken or similar)

** adjust these resistors in circuit under load for correct heater tension of 6.3V

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I believe one or two unusual features in this supply need explaining. First looking at the HT Supply there are a number of unusual networks around the very standard rectifier bridge. These are snagged from the late John "Buddha" Camille and have two purposes. First of all the CRC pi-networks ahead and after the rectifier bridge help to reduce any noise. The use of carbon composite resistors in this position is mandatory to avoid resonance effects.

However, the chosen resistor values are unusually high, for a reason. They simulate the anode impedance of an EZ80 in full-wave mode. The resulting rectifier (with added networks) behaves very much like a valve rectifier, removing the often-notable unpleasant sharpness when solid state rectifiers are used. The diodes used should be soft recovery types but are relatively uncritical. Without having tested all available diodes I recommend the Telefunken BYV and BYW Series as well as the Motorola MUR4XX Series. All these cost less and sound as good as the much-vaunted "Hexfreds".

Furthermore, the HT Supply circuit operates in choke input mode. For load currents of about 12 to 15mA the Maplin chokes can be used for choke input supplies. It is however necessary to place a piece of rubber between chassis and core to securely and strongly clamp the choke's core in place as otherwise the core will buzz. I have myself used these chokes for up to 40mA choke input supplies in the described manner and fashion with no ill effects. Using one choke in each "leg" offers improved common mode rejection and doubles the Inductance.

The rest of the circuit is bog standard, though the noise fanatics in the audience might want to replace the 1k filter resistor preceding the 220uF main filter capacitor with another Maplin choke. In my books the 1K filter resistor is just fine. Not shown here is the final filter stage that for each individual channel will be shown in the schematic for the analogue stage. In any case these will be a 22uF Ansar Supersound (or Solen) polypropylene capacitor combined with a 4k7 Resistor.

It should be noted that the above PSU Design is hardly ultimate or in any way maxed out. Significant improvements can be applied. However, the deciding criteria's here where simplicity and reasonably low cost to get our project underway. More elaborate designs are open to anyone caring to implement them. Lastly, the very same supply is obviously suitable for any Circuit needing about 210V +B with 12mA (or less) current-draw and 6.3V Heaters with 0.6 to 0.7A current-draw. For the heaters about 1.2A can be accommodated if one of the 0.1 Ohm Resistors and one of the 0.68 Ohm resistors is bridged out.

2. Thermionic Valve Analogue Circuits for Multibit Digital to Analogue Converters

The circuits discussed in the following are suitable generally for any DAC Chip having around +/- 1mA output current. This includes Analogue Devices AD1862 and AD1865 as well as Burr Brown PCM56, PCM67, PCM69, PCM1700, PCM1702 and PCM1704. The Burr Brown PCM63 and Philips TDA1541 will require somewhat different circuit values.

Due to a number of factors the circuit proposed is based around the SRPP circuit. This tends to be somewhat colored sonically, however the specific coloration's seem to blend well with digital audio's failings, producing a generally subjectively pleasing sound in this context. In order to allow easy procurements of valves and reasonable gain as well as reasonably low output impedance the circuit is proposed for use with the ECC88/6922.

However, a 6CG7/6FQ7 can be used without changes and even a 6SN7 can be employed if the higher heater current and the 8-Pin socket is accommodated. I know we have a few people in the circle who think the 6SN7 a good sounding and useful valve, even if I don't. If desired the ECC83 or ECC82 can be used. In this case the cathode resistors should be increased to 3.3kOhm.

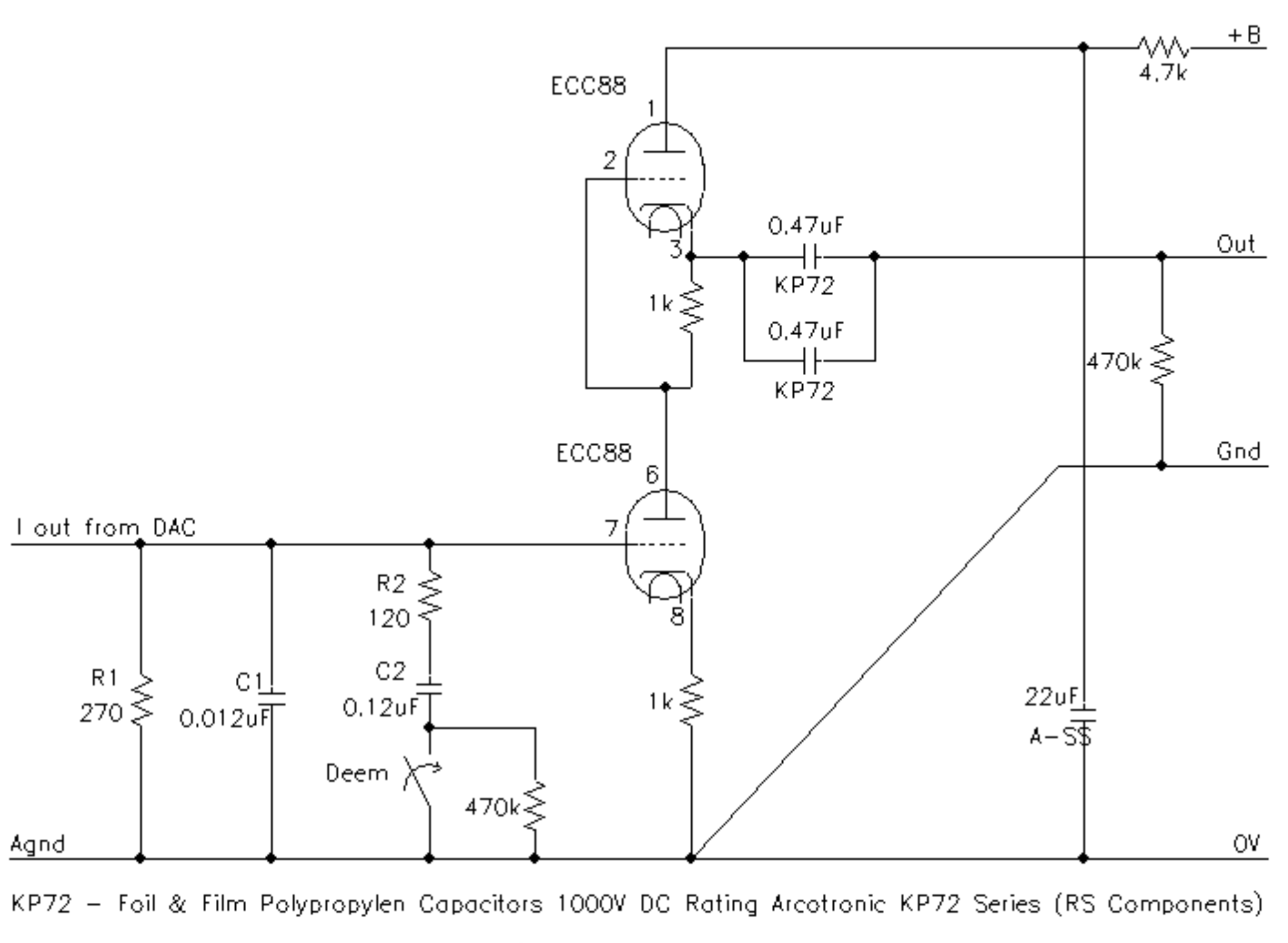
Depending upon Valve used the output voltage of the circuit will range from a nominal 2V RMS for digital full scale with the 6CG7, ECC82/12AU7 and 6SN7 in circuit, 3V RMS if the ECC88/6DJ8/6922 is employed and 8V RMS if the ECC83 is used. As a special case, the use of the 12AY7/6072A will result in about 3.5V RMS out.

The circuit includes a lowpass filter and (I believe uniquely - at least I'm not aware of any other design that does) the deemphasis correction. The lowpass will produce a mild (0.7db) attenuation at 20kHz, which should be innocuous. CD's using deemphasis are rare, but they exist. Many older CD-Players and DAC's use analogue circuitry to carry out deemphasis, so I believe the function must be provided where needed in the analogue stage. In this case R2 and C2 provide it together with the switch. The switch is ideally a high quality relay (mercury wetted reed relays are best) and needs to be controlled from the Players/DAC's internal logic.

The two parallel output capacitors are used to achieve a suitable load compatibility. The capacitors used are very high quality industrial Foil & Film types from Arcotronics or LCR and sold by respectively RS Components or Farnell in multiples of 5 and for a sane price. Their performance is broadly in line with Hoveland Musicaps. The resulting 0.94uF output coupling capacitance will allow loads of 20kOhm or higher to be driven without audible bass rolloff.

While it is possible to further increase the capacitance, I would not recommend to do it in order to driver lower load-impedance's. For ECC88, 6FQ7 and 6SN7 the 20kOhm load possible in this configuration are the lower limit, for ECC82, ECC83 and 12AY7/6072 the load impedance should not fall below about 50kOhm and hence in this case even a single 0.47uF coupling capacitor will suffice.

All of this gives a basic circuit now almost a decade old and well worn, but it still delivers the good's. Here is what it looks like:



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C1,2 – Polystyren Capacitors

A–SS – Ansar Supersound metalised polypropylen Capacitor, 400V DC Rating or equivalent

All Resistors Holco H4, Vishay EE10 or better

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For anyone looking for something a little freakier, I can suggest the use of the Euridice preamp circuit presented here in issue 34. Simply drop the Euridice circuit in substituting the lower ECC88 Grid and loosing the rest. This will then offer balanced or single-ended outputs, a very low output impedance and the ability to operate with pro-audio 600 Ohm loads. It will also cost a lot more.

Now to the PCM63 and TDA1541. The TDA 1541 is still one of the absolute best sounding DAC Chip's ever made (followed closely by the PCM63). Both Chips are also fundamentally different from most other DAC's of their kind by delivering 0 to 4mA output current. In order to keep the error due to the non-zero impedance of the I/V conversion node limited it is essential to reduce the resistors in the circuit accordingly (halve their values) and to increase the capacitors (double their value). Yet we are still not there.

The PCM 63 is unique in that it allows us to disconnect the output from the pull-down current source. No such thing exists in the TDA1541. In order to offer a 0mA output at digital silence (that is NOT all bit's set to zero by the way) almost all Multibit DAC's use a pull-down current source offering a - ½ full scale current offset, zeroing the output at digital silence.

The PCM63 allows us to eliminate this current-source (a bunch of transistors, resistors and Jah knows what else), simply by disconnecting pin 5 (offset) form pin 6 (output). Having done that we are left with a DAC that will swing now from 2mA for digital silence to 0 and 4mA peaks for negative or positive full-scale signals, respectively. It is in this mode that the highest degree of sound quality from the PCM63 is realised, at the expense of a ½ full-scale offset. In many circuits the measures required to eliminate this offset would be worse than the problem, so this is rarely done. In our own little circuits the offset matters little and hence we can safely employ this scheme. The highest value of I/V conversion resistor usable before significant distortion becomes notable is 100 Ohm in this case for the PCM63. Similar Limitations apply to the TDA1541

We hence need to substitute the following values:

R1 = 100 Ohm

R2 = 43 Ohm (44 Ohm accurately)

C1 = 33nF

C2 = 330nF (320nF accurately)

By the way, I'm looking at making a DAC using the PCM63 kindly donated by Geoff Mead which will use the above with a Euridice style amplification stage and NO digital filter. *(added later - this developed into the TDA1541 based "Adagio" DAC found elsewhere on this website)*

The circuitwise similar Audio-Note UK made DAC5 (at a whopping £18,500 retail) was just noted by Martin Colloms in HiFi-News as "best from CD so far" and uses actually a much inferior chip, the AD1865 a dual 18-bit Multibit DAC. Martin noted a subjective quality approaching that of experimental new high-resolution digital systems (96kHz/24bit/SACD). Perhaps it's time to go out and buy the last remaining PCM63 and TDA1541 before they walk in order to make THE DEFINITIVE CD format DAC.

3. Thermionic Valve Analogue Circuits for Timeslicing Digital to Analogue Converters

As noted before, the timeslicing or Delta Sigma DAC doesn't really need any analogue stage in order to provide enough voltage output. However, due to it's ill conceived modus operandi we require lowpass filtering and (if we insist on being boring and conservative) a balanced to SE converter in case of most high performance devices.

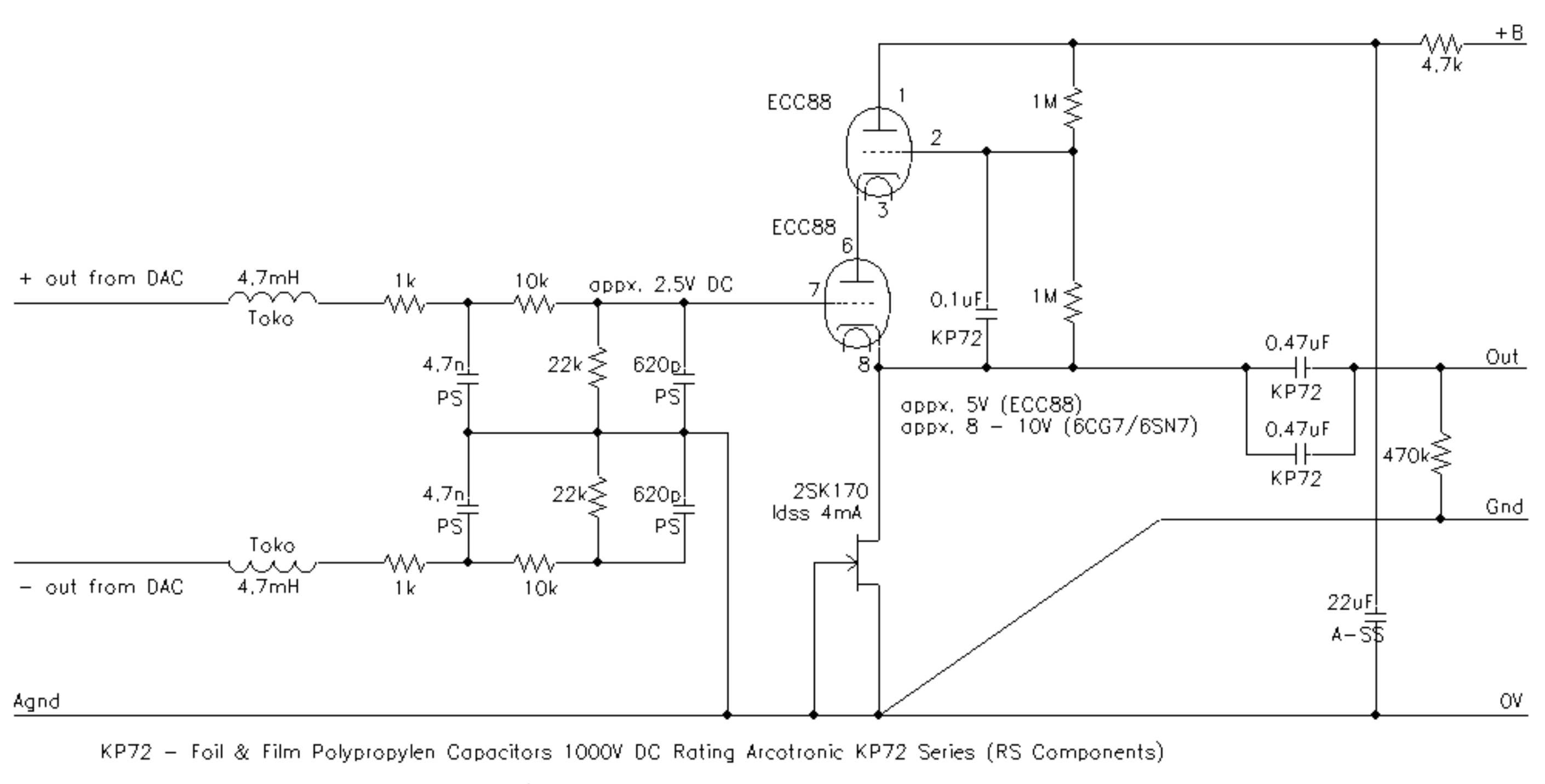
Let's first split the range of timeslicing DAC's into three groups. The latest, most recently introduced group uses extensive on-chip analogue stages, sounds by far worst of the lot and for some reason receives in it's implementations consistently rave reviews*. These are as mentioned the Burr Brown PCM1710/16/28/32, Philips TDA1305, Cirrus Logic CS4327 and Analogue Devices (insert list).

I suggest taking the hint from the manufacturer. Just take the output pin of the DAC Chip directly to the output jack with only a good quality coupling capacitor in-between. Believe me, it truly does sound best. The on-chip filtering is usually sufficiently good to not cause any problems even with solid state equipment. A valve buffer (cathode follower) can be employed, just as per the circuit below, but again, I do not feel, that it improves anything.

I must also note that I personally find the sound coming from these new devices to be exceedingly poor, when compared to really competently implemented Multibit DAC's or earlier generation Timeslicing DAC's.

The generation before the latest designs is a bit more heterogeneous. We find many DAC's with differential output but various different schemes of on chip filtering and buffering and the occasional DAC with a single-ended output and really good on chip analogue silicon. In the latter case (CS4328, certain OEM Products) the same as above holds true. For all other differential output DAC's the circuit shown further down should/could be used and should work under any conditions, offering good filtering and ensuring an easy load for the DAC Chip, keeping distortion low.

Finally, the original first-generation timeslicing DAC's have no analogue stages. Some of these are early Philips chip's as used in the QED "Digit" external DAC, also the NPC SM5872 (Marantz CD-53/57/63/67 among others) and NPC SM5864 (Arcam Alpha - lower Models) and surprisingly many OEM chips even in current use. All these will require some buffering and due to the lack of on chip filtering a direct connection without any additional lowpass filtering a direct connection even to high impedance inputs is not advisable. Again, the circuit shown below will work just fine.



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PS – Polystyren Capacitors

A–SS – Ansar Supersound metalised polypropylen Capacitor, 400V DC Rating or equivalent

All Resistors Holco H4, Vishay EE10 or better

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Thorsten Loesch

The circuit seems very novel, but it is not at all so. The differential filter at the input of the circuit is used in order to present an identical load to either output of the DAC and to implement a well-damped 3rd order lowpass filter. Again a small loss (about 0.8db) is allowed at 20kHz. The positive output of this filter is then buffered by a ECC88 cathode follower.

This cathode follower has been subjected to some improvements over the classic version. By bootstrapping (or cascoding) the anode of the follower valve the circuit is linearised. The use of a J-Fet connected as current source in the cathode of the follower valve serves the same purpose. More on this circuit can be found in Allen Wrights "Tube Preamp Cook Book" and hence it will be not discussed much further.

The usual 2.5V DC offset on the DAC Output is put to good use in order to provide additional bias to the Grid of the follower valve. This allows about 4 to 5V to develop on the cathode, more than enough voltage headroom to allow the about 1V RMS output to be buffered with low measured (and perceived) distortion.

It is possible to use a simple "classic" cathode follower in the above circuit. This will certainly result in a much more "tubey" and possibly subjectively preferable sound. In this case the ECC82, 6CG7/6FQ7 or 6SN7 seem ideally suited with a 2.2kOhm resistor in the cathode. An ECC83 might be employed with a 3.3kOhm cathode resistor.

The sound will be very different to the circuit shown above. Mostly transparency will be substantially compromised, however the subjective result might be considered pleasing. Perhaps it is worthwhile trying designs, the classic and the "Super Linear Cathode Follower".

**Either reviewers are blinded (or deafened) by the "24-Bit" claims for those Chip's or the fact that finally the last bit of control over analogue performance has been taken away from the CD-Player Designer stop's them from screwing up the Sound of potentially better sounding devices up. Whichever way, I'm just not getting it.*

Conclusion

Armed with the designs shown above and the background information presented it should be possible to convert any available digital audio replay device to a state with a more transparent, less subjectively compromised design. Most CD-Players and DAC's will have sufficient internal space to fit both the valve circuit and the PSU parts.

The perhaps best mechanical solution for most CD-Players would be to obtain a suitable sized aluminum box/chassis from Maplin (the 150mm X 114mm X 76mm one looks right). This would be attached at the rear of the Player or DAC and one would run some flying leads from the Player/DAC circuitry trough some holes in the rear panel (how about the ones previously containing the output jacks) into the valve stage.

All other components and the PSU could be placed safely in this separate chassis and again, the mains power could be send through another suitably drilled hole, making the whole solution neat, tidy and professional looking. If done well enough it would even allow the Player/DAC to be returned to stock condition with no-major cosmetic blemishes.

Now I suspect that some people would think it having been convenient had I listed in detail the relevant Pin's on all noted DAC Chip's and provided more practical notes on the way these are accessed. This was not done for a number of reasons. Primarily I feel that anyone should get a service manual for their piece of equipment before starting to work on it. This will contain a schematic that has the relevant information on it. If such a schematic, even if present would not be able to convey the needed information, perhaps it would be useful to take someone reasonably experienced with digital audio circuits and their modification as "mentor" for the specific project (please DON'T bother e-mailing me - I don't have time for that kind of mentoring.... sorry).