

Building Audio Amplifier with integrated Power stages (Chipamps)

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Table of contents

Why using chipamps ?	2
The projects goal: Engineering is the way.... not obscure parts	4
Removing "bad" sounding components	4
Coupling capacitors	4
Relais	7
Obscure design theories	8
General thoughts about components	9
Voltage regulators	10
Inverting mode	10
Paralleling	10
GND and pseudo differential Signals	11
Deal with noise and noise vs. THD	12
Composite amplifiers/nested feedback loops	13
Layout	14
How it works	16
Power supplies:	16
Input stage and signal path:	16
Output Stage	19
DC Servos	20
Miscellaneous	21
Output Protection	21
Controlling	21
Conclusion	22
Literature:	22
Appendices:	22

Table of figures

DC Servo principle.....	5
DC Servo pseudo differential.....	6
simplified grounding sheme.....	11
Inputstage, balanced, High impedance.....	17
Inputstage, unbalanced, High impedance.....	17
Inputstage, unbalanced, Low impedance.....	18
Inputstage, unbalanced, pseudo differential, Low impedance.....	18
paralleled Ouput Stage.....	19
pseudo differential DC-Servos.....	20

Why using chipamps ?

Audiophile people (or people who think they are) normally look down one's nose at integrated amplifier.... but why ?

Let's have look into the beginning of the 70's. The first partly integrated power amplifiers were designed in this time and sounded horrible. They should be cheap and easy applicable, so no further investigation was made for quality. Right in this time, building amplifier with discrete components gives better results.... but only if you would know what you do. Transistors were slow and couldn't handle high voltages and great current, especially the integrated ones.

During the 80's, discrete parts would advance in this topics, but analog integrated circuits doesn't. Computer industries, the main driver for integration, only demands for smaller transistors which should be faster and run with lower voltage than the preceding technology. Advantages in integrated "high" voltage and high current processes remain a rare event.

Even to the end of the 80s, it wasn't that difficult to build up good sounding amplifiers with discrete parts (even Opamps for the small signal stuff were not really able to produce good sound).

All the cheap consumer stuff used integrated amplifier, like car-radios, clocks with radio, cheap stereos and cheap containing-all-the-stuff turntables for home use.

Better equipment, for home and especially broadcast, used discrete designs, so sounding many times better than the "cheap" equipment.

Out of this time, audiophiles think, integrated amplifiers sound "#*+%\$!".

Things changed in the mid of the 90's, process technology for higher voltages advanced, and now it was possible to integrate power transistors with usable voltage and current rating on chips and even the higher voltage small signal transistors advanced regarding speed and gain.

In the same time, the almost last integrated amplifier series in Class AB technology were presented: the TDA729x series and the LM3886 and it's cousins.

Also the computer technology, and with it circuit simulation, advanced further. Some specialist now began to research all the topologies of amplifiers by simulation rather than by soldering.

(D.Self is a candidate for it. He examined all types and sorts of output stages and many things more) Now, there was no trial and error engineering anymore but predictable results on how silicon devices would perform. (Means that no trial and error engineering was necessary... but doesn't mean that all designers had used the new methods)

If you know what you must do, you can even use two NPN transistors for a "complementary" output stage. And the LM3886 does. And it does it in an almost perfect way.

With this chips, you are able to use all the advantages, a integrated design would give:

- (1) almost perfect Bias voltage regulation, because both output devices are on the same Chip
- (2) integrated protection against overcurrent and even 2nd breakdown on bipolar devices
- (3) internal transistor matches perfect because they are on the same die and of same polarity
- (4) saved boardspace. This can be used for other functions to increase sound quality

Bias voltage regulation in discrete designs is a hard to do task, because of internal transistor temperature is almost unknown. Also transistors with integrated temp sensing diodes make this task not perfect. Because of different transistors and especially diodes have different temperature coefficients, even NPN/PNP pairs have different V_{be} 's and temp. Coefficients.

Many people have invented many circuits for bias voltage regulation, but there was never a solution which satisfied all demands. If using integrated amplifiers, all process parameters are known and variations always apply to all devices on the same die. So the chance to have a good temperature compensation of the bias voltage is high, if (yeah..if) the internal design of the amplifier is done right. But with modern simulation tools, there shouldn't be a problem to avoid mistakes made in the past.

The protection of the output stage transistors are another task making things worse. The problem arise with the 2nd breakdown problem on bipolar devices. If the voltage over a transistors CE increase, the allowed current decreases. But this is not linear, getting worse as higher the voltage and as higher the temperature is. Even the popular 2SC5200 is only capable of 4A at 40V and with inductive loads, it can happen that the output voltage is negative if positive output current flows and vice versa. Creating a protection scheme which takes this in account is not trivial, even if you don't want that the protection influence the signal too early. [1] and [2] have written some interesting stuff about this topic.

Using an integrated protection scheme and parallel 3 devices, you are far away from protection and on the safe side of the edge in case of short circuit or overload.

The LM3886 is build up with 2 internal NPN output devices. This is a circuit design, often seen in the 60's and 70's as PNP transistors were expensive, slow and couldn't handle much power. In integrated circuits, this design is also nowadays often used, because of integrating only (power) NPNs is much easier than it is with complementary transistors. Even the NE5532's output stage is build up out of two NPNs.

What was not done in the old days was the compensation diode which is elementary for low cross over distortions. With modern simulation tools, D.Self has invested some time on this special topic and found out, that it is possible to meet modern requirements when using the right design. [1] But literature about this topic is rare and real measurements also. Interesting is, that also the popular NE5532 uses two NPNs and the compensation diode in the output stage.

Looking in [3], it can be seen, that on low output currents, good THD results (~ 90dB..100dB) can be obtained. Using 3 devices in parallel will reduce output current by a factor of 3.

Unfortunately, integrated amplifier ("chipamps") are only usable for power amplifiers with output power below the 100W..200W range.

If you go further (voltage is the problem, not current), then there is no other way than making a discrete design.

But for home use, more than approx. 50W-100W are not necessary.

The projects goal: Engineering is the way... not obscure parts

The goal of this project is to make high quality sound with solid engineering work.

Using obscure parts like oil-paper electrolytics or special types of cables will often cover missdesigned hardware. The warm sound of an amplifier is often a Highpass function of the coupling capacitor paired with the harmonic distortion it produce...

Using well known circuit topologies makes designs easier, more predictable and also better.

Covering design errors and mistakes by "special" parts is common, but doesn't solve the problems.

Also, in my opinion, there exist many customer (and also many hobby designer) who relies more on curios theories (called "snake oil") than on solid engineering.

Here are some exapmles of common mistakes made with power amplifier design and how to solve them:

Removing "bad" sounding components

Coupling capacitors

The components with the most written articles about is the coupling capacitor.

In the beginning of electronic design, most amplification stages were AC coupled with a capacitor because of most singly supplys were used. It was essential to remove the DC component when amplifying AC, especially with discrete transistor design. With the upcoming of Operational amplifiers, also Dual supplys were established. Most coupling capacitors were designed out of the signal path, but two special cases remained:

- removing DC in the amplifiers input stage
- removing DC in the amplifiers power gain stage to get an alomost DC free output (speakers don't like DC)

It should not be forgotten, that Audio in general is a special case regarding harmonic distortions.

Most other applications doesn't really care about ditortions as they are below -60dB and much other applications than audio are in a frequency range, were the coupling capacitor can be much smaller (pF to nF range), so that good ceramic or foil capacitors can be used. But in Audio, the frequency range goes down to 10Hz or lower and you need high capacitance values to not influence the signal too much. So electrolytics, and in the last years ceramic capacitors with capacitance values up to 22 μ F, are necessary for signal coupling, but especially electrolytics have many drawbacks. But before discussing it's quality, try to remove it and stop discussion :-)

In many standard Application notes and cheap devices like consumer stuff, electrolytics will be used for coupling purposes in the input section for only two reasons: cost and simplicity. An electrolytic capacitor will cost < 10ct when buyed in large amounts and needs not much boardspace.

But there's an other way to remove the DC components: using DC-servos

DC servos consists of an integrator with one non-electrolytic capacitor, an opamp and one resistor.

The capacitor and especially the opamp will cost much more than the electrolytic cap, especially if you want use a "good" capacitor (NPO for SMD mounting, Foil for THT) and a good or excellent Opamp.

Not all opamps are a good choice for this job. Initial Offset voltage, noise and bias current are a concern. This combination removes many common parts out of the selection. High bias currents will create an high additional offset to the initial existing one and the opamps noise will be directly transferred into the signal. But in the input stage, normally some mV of DC wouldn't be a problem (of course with gains of >100 it matters, but there are only a few applications were a gain >100 is used in the input section) so that even a NE5532 can be used as DC Servo, if the bias currents are compensated. See fig.1 for a generic example for a DC-Servo.

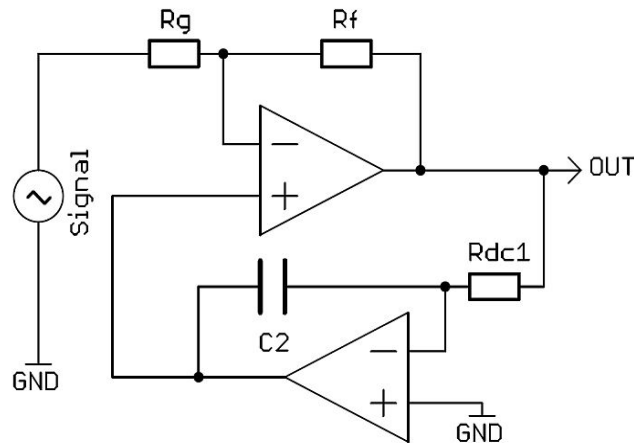


fig 1: DC Servo principle

The other place where electrolytics are most used in the signal path is the feedback divider. Also here, many literature has been written about his topic and many "experts" discuss about the newest, the best, the funniest, the most expensive and the cheapest capacitor.

The problem arises at the point where the amplifier stage has an input voltage offset (which is in most cases not avoidable) which is then multiplied by the gain of this stage and gain is in the $\times 10 \dots \times 30$ factor range, so a 10mV offset at the input would give up to 300mV DC Offset at the output, too much to be ignored. Putting a capacitor in the feedback path would reduce DC gain down to 1, so only offset itself will be present at the output. But this capacitor will form a highpass with the feedback resistors and the highpass frequency is increased with decreasing resistor values of the feedback divider. To get the output noise down, these resistors would be of low values (hundreds of ohms to some kOhms) making this construction a point where you must decide between more noise or a bigger capacitor if leaving the highpass frequency in a usable range (1Hz...2Hz). Most designers decide for the bigger capacitor, making it high in value (100 μ F...1000 μ F are a common range) and therefore fix the type of it to electrolytic.

It should be mentioned, that with paralleling the integrated output stages, the resulting DC at the output of every stage must be low as possible. Having a DC at each output will produce an idle current be drawn by the current sharing resistors and you know about Murphy's law: The DC offset you get from three inserted devices is the same in amplitude but with inverted polarity. At this point, only having an amplifier with +10mV DC at output (coupled with electrolytic) and the other with -10mV (also coupled with electrolytic) will give 20mV total above 200mR (if each device has 100mR output resistance) which will end up in $I=U/R=20\text{mV}/0.2\text{R}=100\text{mA}$ idle cross current, flowing out of one device into the other one! In case of $\pm 35\text{V}$ supplies this will give additional $70\text{V} \times 0.1\text{A} = 7\text{W}$ power dissipation.

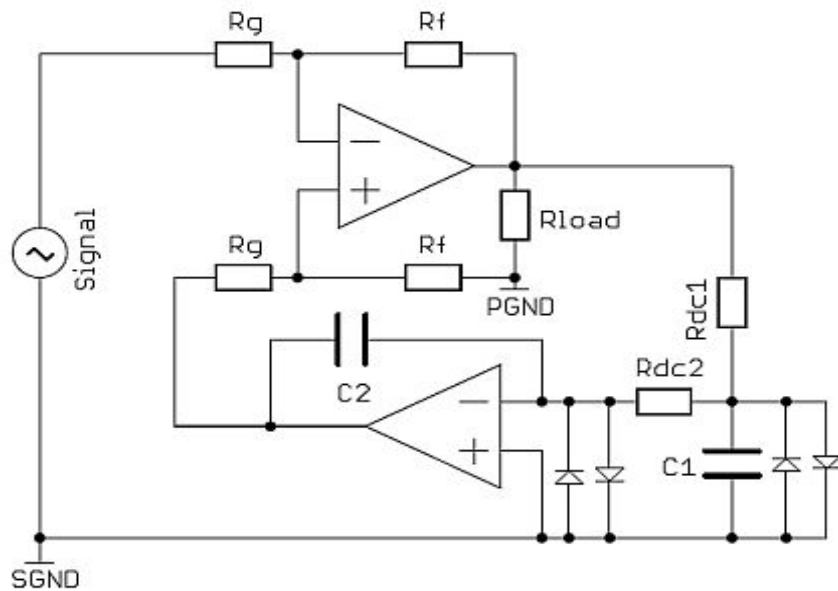


fig 2: DC Servo pseudo differential

So, also here, the capacitor can (and must !) be replaced by a DC servo. Doing things right also means that the servo must be inserted in the differential design to get highest possible Common Mode Rejection. See fig. 2 for this !

Put the whole amplifier into a big servo loop and there should be almost no DC at the output (see fig.2).... with some drawbacks at this point:

If the amplifier starts up, remaining voltage in the DC servo's integrating capacitor would create a big DC bump at the Servos output wich is then directly amplified by the amplifier... not a good idea.

Also the "MUTE" function of integrated amplifiers will make things worse....

This "MUTE" function doesn't switch off the output transistors but set the amplifier in a state with internally amplifying the GND connection by a factor of 5 or something in this range. This is ok for the amplifier, because of now the output pin is held at ~0V until the amplifier and its supplies are stable {you can test it by e.g. connecting a 10k resistor between one supply and the output and a current will flow because the amplifier tries to hold the 0V. If the transistors were open, almost no current would flow through the resistor}. Now, the DC servo loops input is connected directly to the amplifiers output wich is held almost at 0V, but remember: a minimum gain of 5 is used and the amplifier has it's internal offset voltage, so 5mV...50mV DC can be seen at the output in worst case. This voltage is now fed into an integrator wich is integrating and integrating and not stopped. After a short time, it's output would reach several Volts (dependent on time and RC values) wich are then amplified by the amp if it is turned on. The servo loop would regulate this, but it needs time, so there are some hundreds of ms with high DC at the output. This would make a very very big "plopp" in the speaker....

So, if we want to remove the electrolytics (and we want !), we must find a solution for this. There exist 2 different concepts for it.

First, the integrator is reset shortly before the amplifier comes out of "MUTE" by a JFET or a small signal relais. Second, you use a speaker "relais" to remove the speaker from the amplifier until all is safe. For this topic, see next.....

Relais

In the last paragraph, we found out, that a "relais" that remove speakers from the amplifier could be a good idea. There are also usefull in the case, the amplifer dies and it's output transistors are in a way defective that they conduct always to a power supply rail. In this case, muting the amplifier with the mute function wouldn't really help. The only way to handle this is to remove the speaker or better, to never connect it.

Output relais in audio power amplifier for home use are common since the last 40 years (nowaday more and more Class-D amplifier can be found where muting is done in other ways). But these relais have a drawback: There are designed for high currents or for low currents, but a relais capable of handling both equal good are not existent. Relais for high currents have massive contacts wich contact is the best, if a big current flows, at least for a short time. But who want's to start listening music on high volume levels ? The types for low currents, sometime called signal relais, cannot handle much current because of optimized for very small currents...

Also, a fact not known by many people, the contact resistance is nonlinear and current dependent, so it introduces distortion wich is not included and supressed by the feedback loop. Even a relais measured fine in a prototype can change it's characteristics over time and signal dependent... that's not a thing you can rely on.

But, how everytime, there's a modern solution for this topic. Since some years, the industry is able to produce MOSFETs with very low R_{dson} wich is also linear until a certain point.

With this MOSFETs you are able to build SSR (Solid State Relais) wich are suitable for Audio work. To not allow flowing current thourgh the MOSFETs body diodes, they must be connected back to back, having the advantage that now both source pins are connected together but are floating. To switch them ON, you need a voltage wich is by the V_{gs} higher then the current output signal and it exist already as positive rail voltage. So, connecting the positive rail to the paralleled gates will at minimum turning on the lower FET wich then pulls down/up its drain and source to GND. In this moment, also the upper FETs source pin is near GND and tunred on. Turning them off means to put and hold their gates below any negative output voltage that can occur. Also here, the negative rail is available. See shematic for the FET configuration and it's driver.

The only point THD can occur is, if something is nonlinear and the nonlinearities must be seen in respect to the signal. The selected MOSFETs here have an R_{dson} of 14mR, so paralleled it is 7mR. Because of you need two of them in series, the R_{dson} is 14mR in the end. Taking a common load of 4 ohms, this is a factor of ~ 285 for the static resistance. Measurements and simulation showed that the dependence of the R_{dson} vs. Current is very small and in the 1:500 range and higher.

Taking a peak output voltage of 30V and a load of 4 ohms, then a current of 7.5A_{pk} flows. This 7.5A_{pk} multiplied by the 14mR gives an error voltage of ~ 105 mV with nonlinearities become 1/500 of it. Dividing 30V by ~ 0.2 mV gives a ratio of minimum ~ 100.000 wich is -103dB. As little bonus, the nonlinearities are smaller with lower current, so that the amplifier normally creates more THD than the relais. See Appendix A for details.

The only minor drawback is the driving of the gates, but as you can see, this is not that problem, it's only not that simple like relais.

Obscure design theories

Completely removed negative feedback:

A bad idea... If there's no feedback, nothing can fight against nonlinearities of the output stage. Perhaps this came up with early transistor designs where designer tried to use large amount of feedback and hold the amplifier on the edge of oscillation with no signal. If the amplifier was then driven hard, the oscillations starts and this gives a "special" sound. So, for all audiophiles and until infinity, feedback has to be avoided !

For the well informed designer, feedback helps getting THD down and amps stable.

Special mains cables and connectors:

Not necessary until you have an amplifier delivering 3000W or more for a longer time. If you are below, and most home equipment is below, this limit, the currents drawn from the mains are in the 1A to max. 3A range. So, if special cables and connectors are necessary here, there's a problem with the device power supply. Especially Switchmode Power Supplies, nowadays widely used, should be able to run with even a very bad mains connection without problems.

Internal cabling with special wires and thick ground-bars:

If you read the next chapter, you know that this things are not necessary. But sometimes I think, some designers haven't read it. The use of twisted cables, perhaps shielded, is enough you can do. Thick ground bars are only necessary if currents reach the 100A limit wich isn't the case in home use equipment.

Expensive small signal and speaker connectors

You can buy very expensive Cinch connectors / cables and high grade banana plugs, made of massive copper and gold... you can also throw money down the drain, if you want to.

Input and signal terminals should be designed for low capacitance, usable input resistance and should NOT be directly coupled to GND. Then special cable is not necessary... Use full differential connections ("XLR") before buying expensive connectors.

(One exception is the use of gold-plated contacts. They are more reliable then the standard ones, especially after some years of use, because no oxidation occurs)

For the speaker terminals, Neutrik's "SPEAKON" are a good choice. They are equipped with 2 channels (4 wires), easy to use and all over the world, professional amplifier and speaker manufacturers use them widely since 20 years or longer. Banana plugs can be used for the domestic /home area as long as the voltage is below 25V AC.

Use special parts anywhere:

A well designed amplifier with the right designed parts in the right place is a good choice. Using ultralinear opamps for measuring DC at the output are a bit oversized and not necessary. Also oxygen free foil resistors in the Zobel network at the output are good parts in the wrong place.

Even with modest cost you can design amplifiers sounding and measureing good. While design, you must ask yourself everytime if this part is absolute necessary for the job it should do and if you choose the right one. An opamp for the DC-servo-loop could be the same type as the one for signal amplification but it must not, and perhaps it should not. The DC servo has other constraints than the signal... So it goes on with all other parts and even resistors.

General thoughts about components

SMD mounted parts are nowadays more standard than THT types. But some things must be observed when selecting it.

Resistors are available which are so small that you need a magnifier to even find it on the PCB. These resistors are useful in digital stuff like pullups. Don't use these very small parts for audio signals!

When designing the audio signal chain, use at minimum 0805 sized parts if power is less than 50mW (peak!), else use 1206 sized ones. As smaller resistors are, as more THD they produce. That is a fact and has been verified. But for all other non-signal related stuff you can also use also 0603 types (if you want). Also have an eye on the resistors power rating and the real power dissipated in it. Often these small resistors are overloaded because the designer overlooked the power dissipation.

Also, you don't get anything you want, you must always make a compromise, even with resistors. Keeping its value low will help keeping noise low, but below a certain point, THD will raise because opamps are not capable anymore to drive it right with low THD. Look at the opamps THD diagrams over load and use a value above the one giving your desired THD figure.

Capacitors for the signal chain should be very well selected. If their capacitance is below 10nF, COG (formerly NPO) is available and you should select this. Don't use X5R/X7R or Y5U ceramics for signal related capacitors if you need low THD. But be careful with the voltage (AC & DC) over it. The piezo effect changes capacitance with voltage, so there's a signal modulation. If you are unsure about it, a foil capacitor is a good choice, but not in SMD. Foil capacitors in SMD are not widely produced because of the heating while soldering will partly destroy the thin foil and kill the capacitor before its first use. As long as the capacitance is small, almost all types are usable (don't use capacitors designed for mains voltage use for signal filtering purposes!) MKS2, MKS3 (obsolete), FKP, Polypropylene, and their cousins are a good choice. If unsure, look at manufacturers websites for information about usage. As mentioned before, for capacities below 10nF and voltages < 10V, the SMD COG's can be used. Prefer wide packages over small ones (1206/0805 are better than 0603 and smaller).

Integrated circuits should be directly soldered to the PCB (if not testing on breadboards), this will avoid additional capacitances and allow better decoupling because of shorter connections.

If buying expensive components, have a look on the seller. Power transistors and integrated circuits are sometimes faked.

See the next paragraphs why things are designed as they are:

Voltage regulators

There's an extra Appendix D describing the design of the voltage regulators.

They are usefull for extra PSRR and to avoid overvoltage

Inverting mode

The paralleled output devices are used in the inverting mode as the driving opamp in the previous stage and this has a reason: Common mode voltage.

If using integrated amplifiers in the noninverting mode (like very common to get high impedance), then there's always a signal dependent voltage swing at the input, the signal itself and the feedback from the output. So both inputs are at the same level but this is not constant.

The voltage refered to GND or the supplys is called Common Mode voltage (because of it's common to both inputs) and it's signal dependent. This tends in the input stage to create higher distortion. D.Self and others have made research on this topic, see [1],[4].

So, if designing with integrated amplifiers in a way that there is (almost) no common mode voltage, then distortion is lower (that not only counts for power amplifier but also for operation amplifiers). That's the reason the inverting mode should be preffered.

In that case, all loads at the output must be considered. Take as an example the output of OP 2.2.

It has it's own feedback reistor as load (3k3) and then 3 times the input resistor of the amplifier stages (2k2). So calculating the reistance you end up in $2k2 // 2k2 // 2k2 // 3k3 = 600$ ohms. That's the lowest resistance you should load the LM4562 with to avoid distortions. Increasing the resistors would increase the noise at the output. So, this values are the best you can choice if you want low noise and low THD (and you want both).

Another disadvantage with inverting mode is, that you must have two inverting stages to get the right phase at the output, so a positive input pulse should give a positive output pulse. Because of this, the inverter around IC2.2 was inserted.

Paralleling

In this design, 3 output devices are paralleled. If you have a quick look at the datasheet, also a single device would be sufficient... but only on a quick look.

Paralleling of devices has the advantage to (almost) share the output current over many output stages. This is always a good idea because of sharing currents will also share the dissapated power between more devices and this makes cooling more relax.

The other fact is, that the integrated output stage is more linear with lower current, so you will get better THD figures, especially with the "only" quasi complementary output stage of the LM3886. In the datasheet's diagramms you see, that THD raises with lower load (8 ohms -> 4 ohms). When paralleling 3 devices, every device sees 3-times the load resistance, so a 4-ohms speaker would be seen as 12-ohms load by every device.

There's also the minor aspect of getting lower noise when paralleling devices. If you parallel n devices, the output noise would be lowered by \sqrt{n} , giving a factor of ~ 1.7 for the 3 devices used here, making the total output noise figure better by 4.8dB. Have a look in the "old" datasheets of OPA27 devices [6] for examples used widely in the past. And why not use nowadays circuit arrangements wich were good in the past ?

GND and pseudo differential Signals

Most designers starts with a "GND" plane in their design, hoping that all points on this plane have the same potential. This method is widely used in small signal electronics and especially in digital designs. It's so easy to draw GND symbols on the schematic, creating a polygon on the PCB and all work is done, because we now have a "reference" point with a known potential.....

It seems too easy to be true and it isn' true....

Don't intermix signal and other reference levels in your design ! Creating grounding shemes is essential in audio design and at this point you can decide wether to have a sloppy design with much hum and noise or a very good design. The only additional cost is some hour's of work with your brain (and some resistors) but no special parts, as costly as they could be, can save your work if this point is not done well. See appendix C for it.

In amplifier design, your input stage should (if necessary) receive differential signals with respect to PE (protection earth) wich are then translated into a pseudo-differential signal, related to the Speakers GND connection.

Never, but really never, use GND planes without asking for it's usability.

See fig. 3 to see how things can go right:

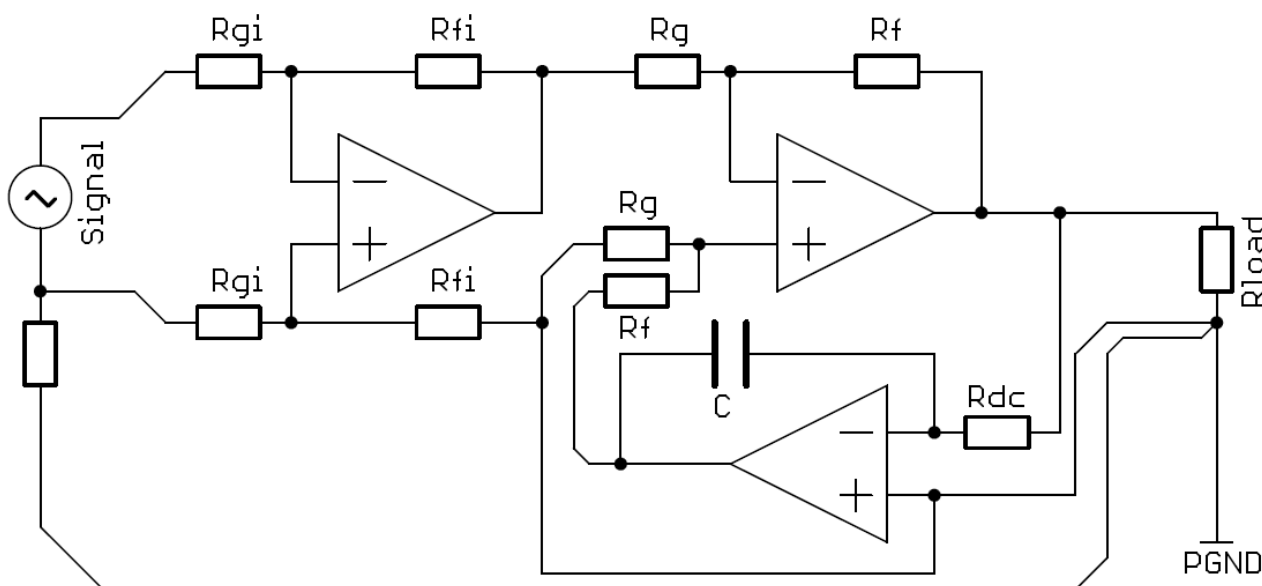


fig 3: simplified grounding sheme

Deal with noise and noise vs. THD

In this paragraph the term "noise" is related to thermal noise only, not distortions by currents or anything else.

When designing in audio, there are two main goals: Low distortions and low noise.

The (thermal) noise of a resistor is given by it's value, the (thermal) noise of an amplifier is (roughly) given by it's internal input noise multiplied by the gain it's set to. See [5] to go deeper into detail, it's worth understanding it !

Like described earlier, there's always a tradeoff between the highest resistor value tolerable regarding noise and the lowest resistor value tolerable regarding THD. Nowadays opamps are capable of driving 600R loads without increased THD wich led to decrease resistance values down to this value. Also you can take into account that you can go even lower, because in most datasheet diagramms the figures are taken at output voltages higher than you need. But normally, it's not necessary to go far below the 600R because things wouldn't change much.

If starting a design, you will first look, wich stage or device will create the most noise and then, how to avoid it or keep it low.

In this design, the LM3886 is the "noisiest" part with initial $2\mu\text{V}$ (typical and $10\mu\text{V}$ max.) reffered to the input at a gain of 20. This datasheet value is a bit unconventional, but that's not a problem.

A bandwidth of 22kHz is used (A-filter) with some minor filtering (we want to get an idea how big the noise will be, so +/- 3dB are don't care at the moment) and $\sqrt{22\text{kHz}}$ is $\sim 148 \sqrt{\text{Hz}}$. Dividing the $2000\text{nV}/148 \sqrt{\text{Hz}}$ you will end up with $13.5\text{nV}/\sqrt{\text{Hz}}$. That's a rather high value (the NE5532 has $\sim 4.5\text{nV}/\sqrt{\text{Hz}}$). We know that an amplifiers output noise is the input noise multiplied by it's gain, so next is to check it's minimum gain without instability wich will be 10.

Using the $2\mu\text{V}$ (typ) multiplied with a gain of 13 (choosen here) the best result we can get is $26\mu\text{V}$ at the output, the worst one is $130\mu\text{V}$ with the max. value used.

So, our last joker is using 3 devices in parallel, where the noise figure drops by $\sqrt{3}=1.7$ (because of the noise of the devices is not correlated to each other, see [6]), getting $15\mu\text{V}$ respective $75\mu\text{V}$.

At this point, there's no chance to make noise lower than this numbers.

So taking in account the feedback divider network (30k and 2k2) and using $10\mu\text{V}$ input noise as worst case you will get $\sim 90\mu\text{V}$ at the output, if using 3 devices in parallel. This means if you can deliver 22Vrms at the output you will get a ratio of $22\text{V}/90\mu\text{V} = 107.8 \text{ dB}$ (in worst case having $10\mu\text{V}$ noise).

This also means to have less noise at the input of the 3 devices, at minimum 3dB less, to make things not more worse as they are.

The values of 30k and 2k2 were choosen so, that the opamp in front of the paralleled devices don't see less than 600R. This left only one value to fit in: $3\text{k}3 (2\text{k}2//2\text{k}2//2\text{k}2//3\text{k}3 = 600\text{R})$.

Now, have a look at the gain structure:

The last stage should give $22V_{rms}$ at its outputs and has a gain of $30k/2k2 = 13.6$. This gives an input voltage of $22V/13.6 = 1.6V_{rms}$ to get fullscale output voltage. Looking for an input voltage in the 1V range, a ratio of $R_f=3k3$ (given above) and $R_{in}=2k2$ will fit and gives an input level of $1.6V_{rms} / (3k3/2k2) = 1.066V_{rms}$ for fullscale. The differential input stage has a gain of 0.5, so that you must put in a voltage of $\sim 2V_{rms}$ to drive the amplifier at its voltage limits.

Taking the opamps and resistor noises into account, you end up with a total SNR of $\sim 104dB$ for the whole amplifier in worst case, which gives a noise level of $22V / 158000 = 138\mu V_{rms}$ at the output.

The "high" resistor values (10k and 4k7) in the differential input are used regarding the input impedance seen by the previous stage, but if using a gain of 0.5, also the noise would be halved and you wouldn't gain anything by using much lower resistor values.

Composite amplifiers/nested feedback loops

An composite amplifier consist of two, or more, cascaded amplifiers and one global feedback loop from the last devices output to the feedback input of the first stage. It's called nested when there's an additional feedback around every stage or between stages. That's a fine thing to use high-speed buffers with gain a little bit lower than one and operational amplifier.

But doing that, means you must keep an eye on stability. If the first stage is faster then the second one, the first stage will hopelessly try to drive the output stage where it should be, but the output stage is too slow to follow correctly. This game end ups in oscillation.

It could be worth a try to use the LM3886 in such a scheme, making total feedback much greater than it is. But, the LM3886 is no high speed device and it is internally compensated for gains > 10 .

The LM3886 has a relative low GainBandwidthProduct and much phase shift, which seems to come from internal nested or two-pole compensation. Testing the composite principle in simulation showed that it's very hard to compensate the composite pair. Many poles and zeros had to be inserted in the feedback loops to get it stable and then not many benefit remains. A few people has tried it and made their schematics public, but I believe that there is not much benefit regarding THD or the circuits are only partly stable.

Layout

In front of this paragraph, a warning should be given !

Do not use breadboards if you want to have a reliable circuit for a long time. Also don't use breadboards if you don't want to add stray capacitances and inductance and even don't use them if you are dealing with high currents and voltages. Don't underestimate the voltage of +/-40V respective +/-35V DC which could be lethal (80V DC !). Making PCBs isn't that difficult and even with single sided ones (which are very cheap and in worst case can be etched at home) you can make useful layouts.

In the last chapters and paragraphs we found many useful design rules (I hope you do :-)) to create a layout which supports the active parts to do their work at it's best.

Always keep in mind, that the layout is like one part of your circuit, if not the most important one !

The most critical things are covered by the differential signal paragraph. Now, it's time to use the know-how. Don't make a ground-plane covering the whole PCB, better is to create more but smaller ones and connect them that they don't create a loop. Connect them in a way, that they all start at the negative Speaker terminal of your PCB *1. This will be our "central" GND point on the PCB.

First start with making the wide GND areas and tracks to the electrolytics. The GND connector to the supply, the GND potentials of the big electrolytics, and in general, the GND potentials of all parts drawing huge current pulses should be connected by wide traces or copper areas ending at the central GND point.

For the low power supply (+/-12V), an extra GND trace/area should be used, connected (only once !) to the central GND. Don't connect anything else to this track than the low power supplies, it's electrolytics, it's regulators and the opamp blocking caps.

Now, you can design your SGND (Signal GND). There are two of them. One is the "GND" connection of the input differential amplifier, which must be directly connected to the central GND by a 100R resistor and second the SGND we use for signal handling. This signal handling ground must also be only once (!) connected to the central GND point.

So, I repeat an advice which is very important: Do not create loops !

If you have differential lines or pseudo differential ones, you must layout them differential. Keep both lines as near as possible, but don't create loops. Impedance matching isn't a concern in analog audio, because of the frequency's are too low.

As we have seen previous, PCB copper thickness isn't that big, so if you are dealing with high currents, and $> 1A$ is high current !, you must use very wide traces. This can be also handled in another way, (but only by the hobbiest where time doesn't matter) by soldering copper wires with e.g. $1,5mm^2$ over the tracks where necessary. This gives very low impedance connections. The candidates here are GND, the both supply lines and the output from the power devices to the speaker terminal. This not only keeps resistance low but also inductance, which is a concern regarding the power supply connections to the LM3886.

There are many advices to make a good layout and many literature exist covering the wide world of layouting. Let me add some advices, especially for hobbieist. Don't treat a via as a track. The resistance of vias only depends on drill size, because with higher drill size, the area inside the drill gets bigger. If a track would carry high currents, use many vias in parallel. If possible, choose a drill size so that you can add a piece of wire wich can be soldered from Top to bottom. This helps very much regarding the resistance concern. If your board isn't manufactured but parts are placed by yourself, SMD parts can be placed on both sides (this would cost much more, if done for machine placement) and SMDs can also be placed below other components. A SMD resistor below an electrolytic wouldn't care for hand assembly, but it's impossible for machine mounting. Using SMD parts will drastically decrease size and saves space because of less drills are needed. And everywhere a drill is placed, no track can be drawn (on both sides). Regarding the librarys you use, double and triple check the mechanical dimensions and drill sizes. Nothing is more frustrating than seeing a component wouldn't fit in reality because of it's dimensions are wrong in the library.

*1 In this design, the negative speaker terminal is connected via a SSR to GND. The central ground point here is where the SSR GND is connected to the local GND.

How it works

Power supplies:

Directly after the connection to the outside of the PCB, the high voltage/high current regulators are placed. They regulate the incoming voltage down to +/-35V which are the amplifiers rail voltages.

The 3 output Amplifier stages, the SSR driver and the input of the low power regulators are connected here.

To have a good stabilized, low noise power supply for the opamps, the +/-35V are regulated down to +/-12V by a conventional LM317/LM337 pair with post filtering by $R=2R2$ and $200\mu F$.

The power for the micro is connected by a semiconductor fuse to the positive incoming voltage. On the micro board, this fused voltage will be linearly regulated down to 3.7V and then converted to 5V by a Step Down Converter. Taking into account the several mA which could be drawn from this 5V, a linear regulation would dissipate roughly 1.5W..2W which would be too much.

Input stage and signal path:

The input stage is designed in a way, that it can be used as either High impedance unbalanced or balanced input or as low impedance unbalanced input if there is another input stage in front of the amplifier module (e.g. for pots or filters) with low output impedance.

In all cases, the inversion of the output signal must be taken into account and the inputs must be connected accordingly.

Using the Low Impedance structure, the gain can be set in a wide range, as long as the previous stage can drive the low load resistance. If a gain of 0.5 is used, the input voltage for fullscale is $3V_{rms}$ and the output noise is 113dB vs. fullscale with the values given in the schematic and best case amplifiers internal noise ($2\mu V$).

The only drawback of both High Impedance stages is, that the input voltage for fullscale output drops down to $0.75V_{rms}$, making it more difficult for the previous stage to get the noise down. Even a single NE5532 would create such a noise that the output noise increases to -112dB. This is a drawback of the balanced to unbalanced converter with only 2 opamps, which has a minimum gain of 2. Therefore, the use of a separate input stage, outside the amplifier module, is recommended.

See the next pictures for clarity:

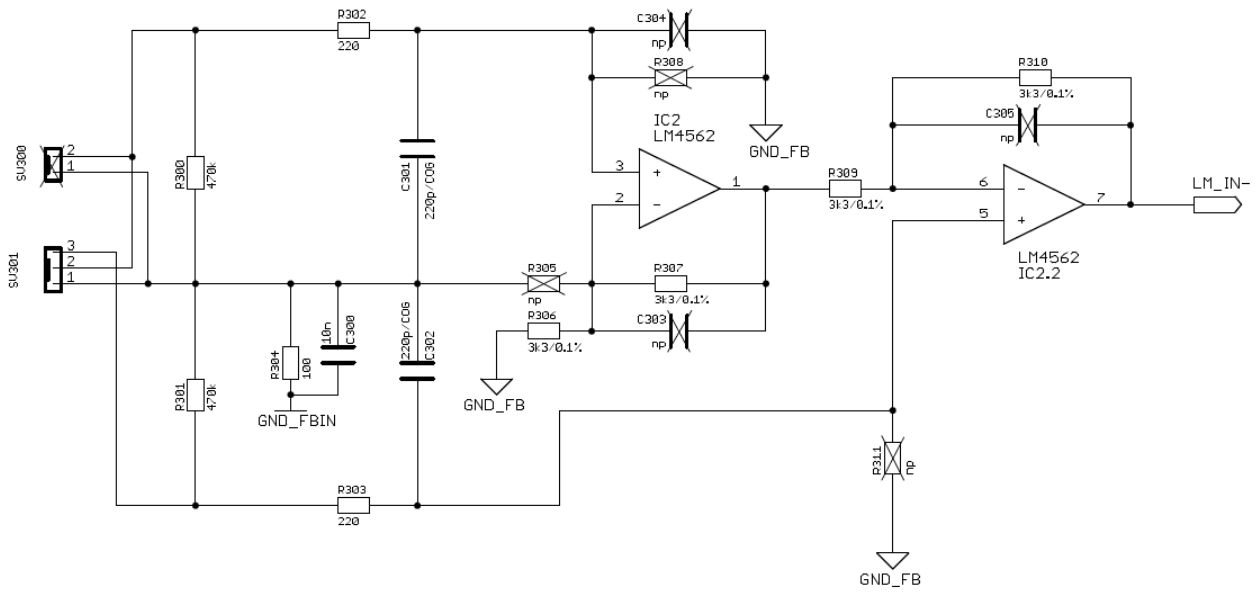


fig 4: Inputstage, balanced, High impedance

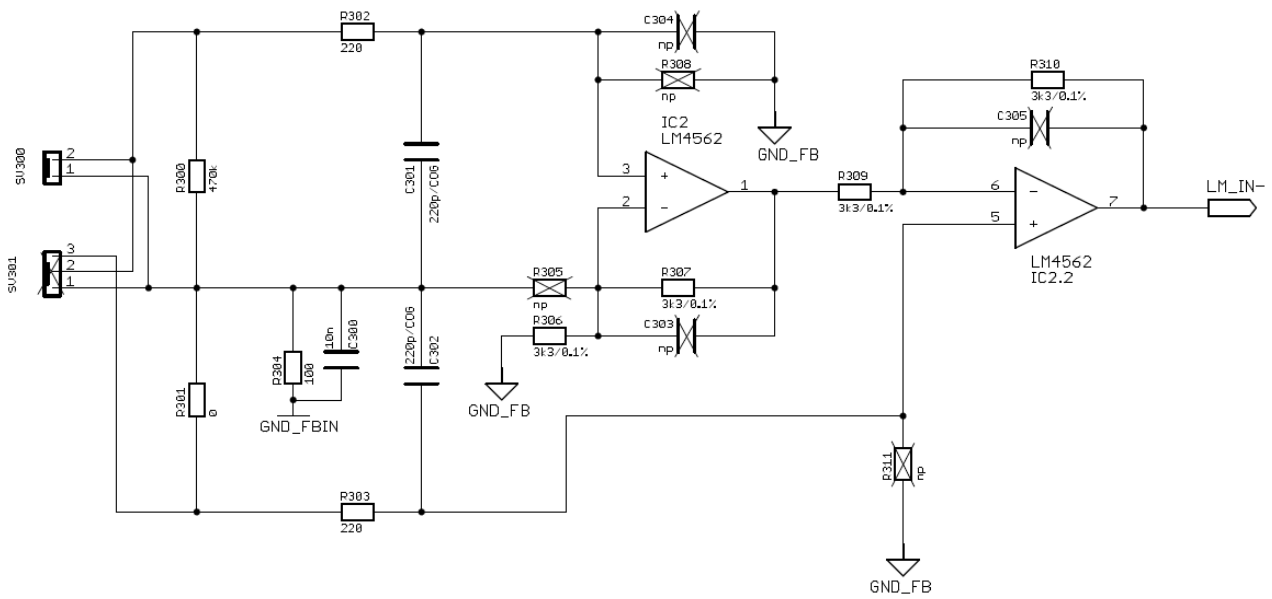


fig 5: Inputstage, unbalanced, High impedance

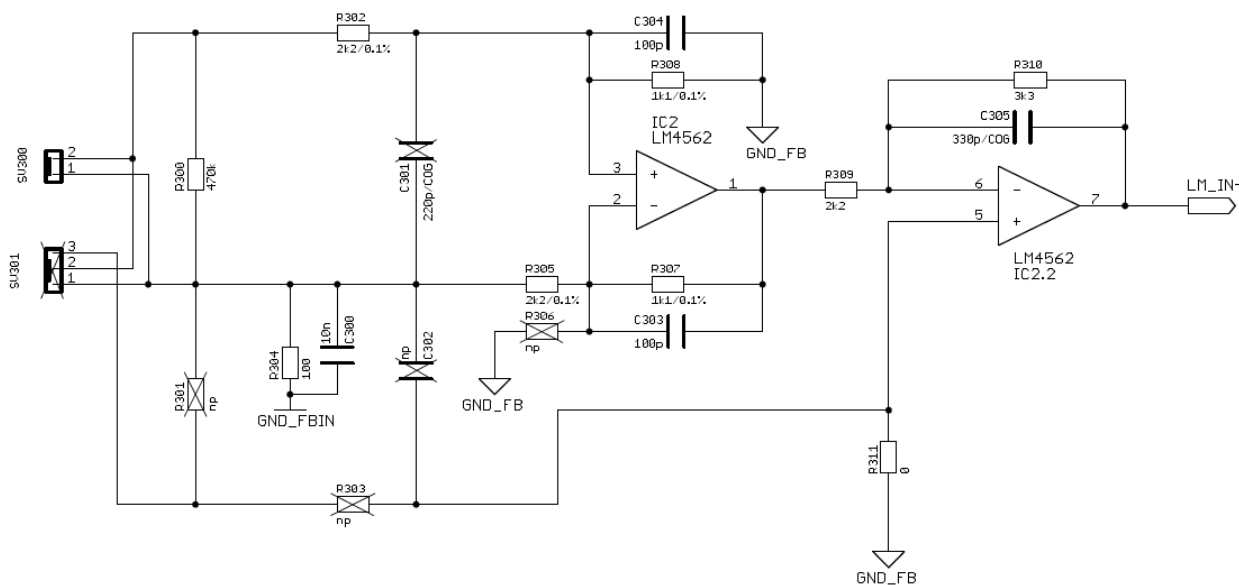


fig 6: Inputstage, unbalanced, Low impedance

The circuit's function is only described for the low impedance, unbalanced Inputstage. IC 2.1 and the resistors R302, R305, R307, R308 form a differential amplifier to remove the noise and distortion from the connection to the previous stage. The negative input will be fed by 100R to the central GND on this PCB and the signal will be measured between the positive and negative input. The output of the differential amplifier is referred to the pseudodifferential SGND line.

Next is the inverting stage around IC2.1 with no gain and it is referred also to the same SGND line. For cleanliness and to remove the unused components from the PCB, the simplest form was realized. The low input impedance is good against the noise and can be driven easily by the preceding input stage (on an external PCB).

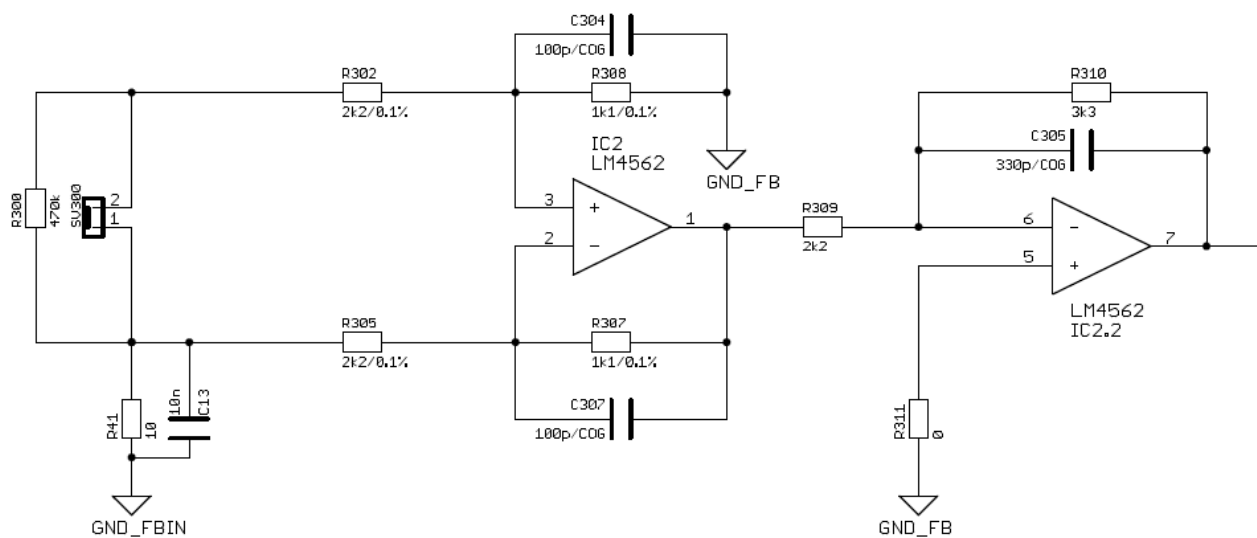


fig 7: Inputstage, unbalanced, pseudo differential, Low impedance

Output Stage

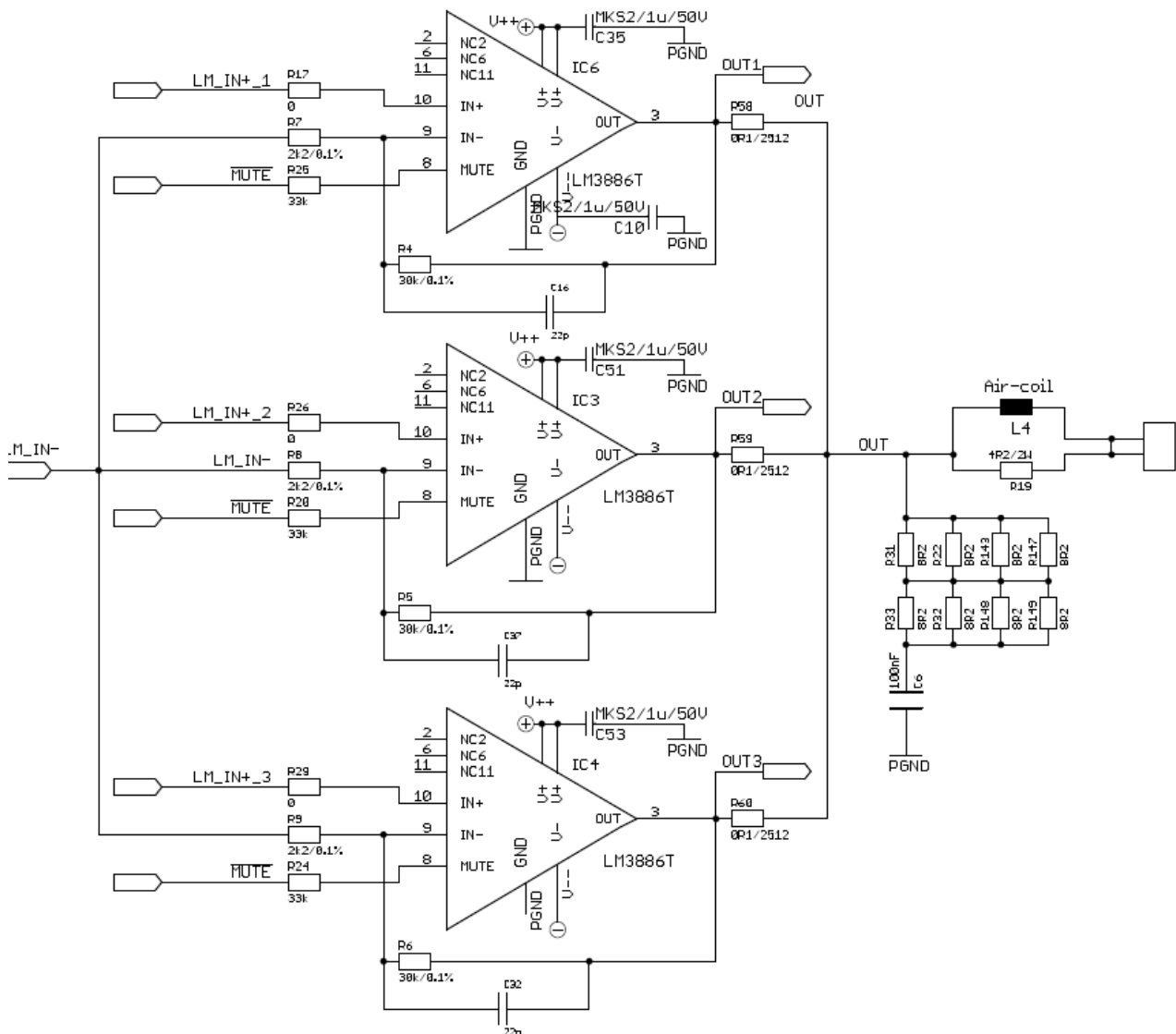


fig 8: paralleled Output Stage

This inverted stage is driving the, also inverted, 3 Power stages in parallel. At the output, all 3 single outputs will be summed over the 100mR output resistors and fed to the positive speaker terminal. Also the three output amplifiers are connected as pseudo differential amplifier and are referenced to the DC-Servo's outputs, removing further distortions in the GND connections.

DC Servos

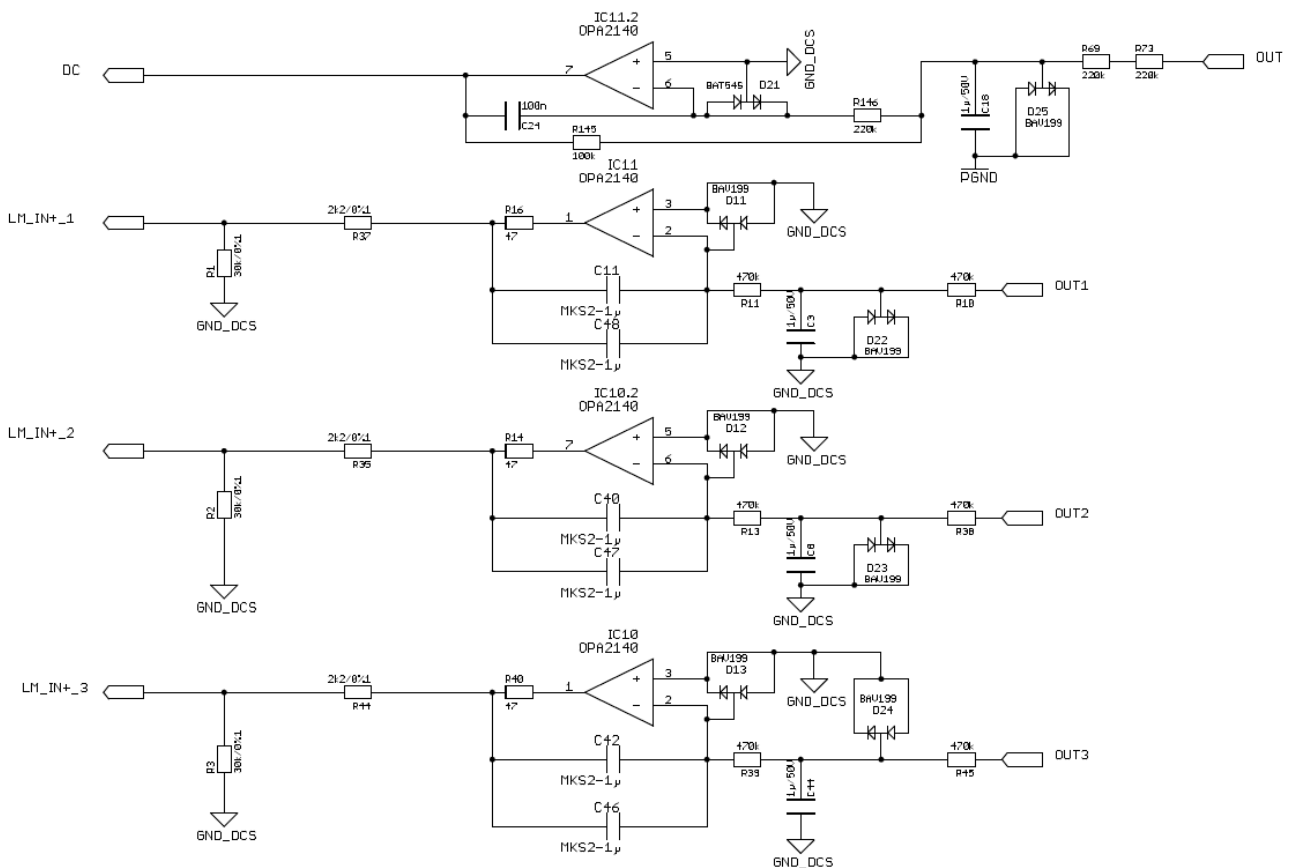


Fig 9: pseudo differential DC-Servos

The DC signals are tapped directly at the LM3886s outputs, before the summing resistors. Every device has its own DC-servo putting its servo signal back into the positive input, resulting in a DC voltage < 1mV at every device output.

Because of using dual opamps for the servos, one left free and handles a lowpass filter which filtered output is connected with the ADC of the micro to detect DC at the output.

Miscellaneous

There also exist the "MUTE" signal, connected in parallel for all 3 output devices. This signal can be controlled by the micro by pulling the line low. Also the driver of the SSR can be turned ON/OFF by the control line xxx. The rail voltages are divided down so that they can be measured by the micro, similarly the current sense signals are handled. The low power +/-12V are only tested for a minimum voltage (+/-7.5V) and the result is present as digital signal. The DC-lowpass output is prepared to be 2.5V when idle and can then go up/down by 2.5V to be totally between 0V and 5V.

On the microcontroller board exist 3 optocouplers, 2 going out and one going in. They are used to communicate to the Main amp controller and a single channel can be turned ON and OFF by a signal through the incoming coupler. The two outgoing signals will report errors. With no error, both couplers are ON (LED inside light), so that in case of supply fail, an error will be generated.

Output Protection

Regarding the output protections and precautions, the standard way hasn't been left.

There's an output inductor with some windings of isolated copper, isolating the output from the load if the load is very capacitive and the frequency is high. This inductor is paralleled to a power resistor, giving some resistive load if, at high frequencies, a capacitive load looks like a short circuit. In parallel to the load is the Zobel network, giving a resistive load if the output is very inductive. You can also find the standard flyback diodes to absorb output voltages above the supply rails. There seems to be a mechanism in the LM3886 which will do that, but for safety, this two diodes are added which wouldn't cost the world.

In case of DC, the SSR was added, removing the load from the amplifier. Also this SSR prevents plops at power up/power down and can also be used in case of overloading the amplifier.

Short circuit protection is done internal to the LM3886 devices and a longer time of overstress will be detected by the current flowing in the supply rails.

See the next paragraph of how all errors will be handled.

Controlling

Nowadays, microcontroller are that cheap and easily available that you didn't find a reason to not use them.

The micro will measure continuously the input voltages (+/- 40V), the rails voltages (+/-35V), the low power supply (+/- 12V), the rail currents and the DC at the output with its ADCs.

At startup, the voltages must reach a minimum level and the current must be in a valid range for idle mode. If this is OK, the micro will un-mute the LM3886 devices and after the DC-servos are stable also set the SSR to ON.

Now, the DC at the output, the rail currents and the voltages are permanently checked to be valid. If one parameter fails, the SSR will be set to OFF and the micro will wait some seconds, dependent of the error type. Explaining the whole software would go far outside the range of this article.....

Conclusion

If you have read the article, and I hope you have, then you see that designing audio amplifiers is not black magic, it's only using well known engineering technics and using your brain in the right way.

Beside the use of the integrated power stages (LM3886), all things can also be used when designing discrete amplifiers and even in small signal stages.

Literature:

- [1] D.Self, Audio Amplifier Design, Audio Small Signal Design
- [2] B.Cordell
- [3] AppNote NatSemi/TI, Bridge/Parallel Mode of LM3886
- [4] Common Mode THD, see [1] and [2]
- [5] TI, Opamps for everyone, esp. noise
- [6] OPA27 datasheet from BB, now TI

Appendices:

Appendix A	SSR for Audio, ACAV
Appendix B	Real world measurement results of the Amplifier, ACAV
Appendix C	Differential, single Ended and pseudo differential Audio signals, ACAV
Appendix D	Voltage regulators for Power Amplifier, ACAV