

HTM AUDIO UK

HTM50S

HTM50 Evaluation demo board

PRELIMINARY DATASHEET

Last Updated

20/08/12

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Introduction

Linear amplifiers are still mainstream using energy inefficient designs. HTM products give the highest efficiency possible using high end digital technology . Our designs reduce current consumption and heat lost. This reduces the need for bulky heat sinks and noisy fans. This reduces cost, increases power output, and increase long term reliability. HTM audio utilise the latest energy efficient technology using high spec branded components.

The HTM50 module was developed after 2 years of extensive audio testing. HTM audio tested numerous digital devices. The maxim device is a no fuss solution which offers fantastic features and value. The device is small and includes internal output mosfets. Many similar designs require external mosfets, which increases cost and board space. The Maxim device is one of the few which offers 50watts RMS output and is the best sounding device while remaining the most compact solution.

While HTM push the latest digital audio technology. We avoid using devices which are short term manufactured or difficult to find by using standard easily available components. This helps make our products long term repairable if required. We do not build in obsolescence.

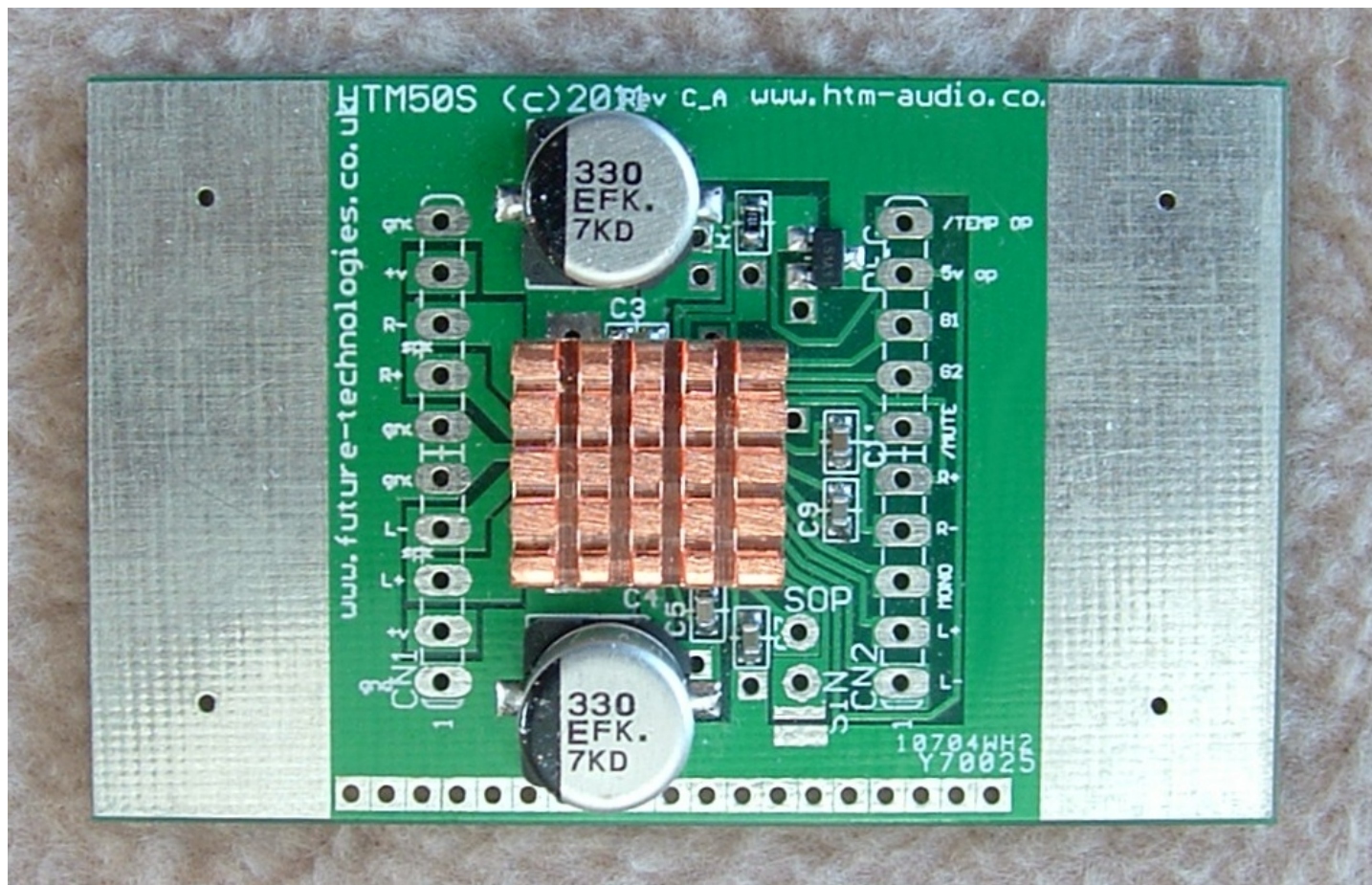
HTM noted that many designs are over-specified and manufactures can be rather clever at hiding that their products are not as powerful as one first thinks. For example, a claimed 400watt 4 channel amplifier at first sounds impressive. Though later we find the limit is 100watts per channel which is quoted into 4ohms. For normal 8ohm operation it is in reality limited to 50watts per channel. Then we notice the figure quoted is at 10% distortion, meaning , it may well only have 30watts actual RMS power. This is often quoted into a 1khz frequency which is not the same as being driven into lower bass frequencies such as 50hz. HTM found that very few designs could deliver rated output power into 50hz and our 30watts example could be substantially lower at 50hz. HTM audio will never aim to trick the user into thinking the product is many times better than it first seems. The HTM50 module is 50watts RMS into 8ohms at 50hz, with no tricks involved!

HTM audio provides easy to follow information which will assist our customers in operating our products operating quickly and efficiently. Many datasheets span 100's of technical pages which may take the user a great deal of time to digest. We aim to reduce the amount of time taken to get our products up and working. The start of the datasheet will assist the user in operating the module as fast as possible. While the later sections aim towards refinements and operating enhancements.

HTM are constantly reviewing their audio products and constantly trying to find ways to improve on them. HTM are also working on high power amplifiers up to and over 2,000watts RMS. HTM aim to bring out a universal design which is scalable from a few watts to many thousands of watts. HTM will aim to never let one of its products become obsolete and will always aim to produce compatible replacements.

Hear The Music
Now, and forever.

QUICK START WIRING DIAGRAM

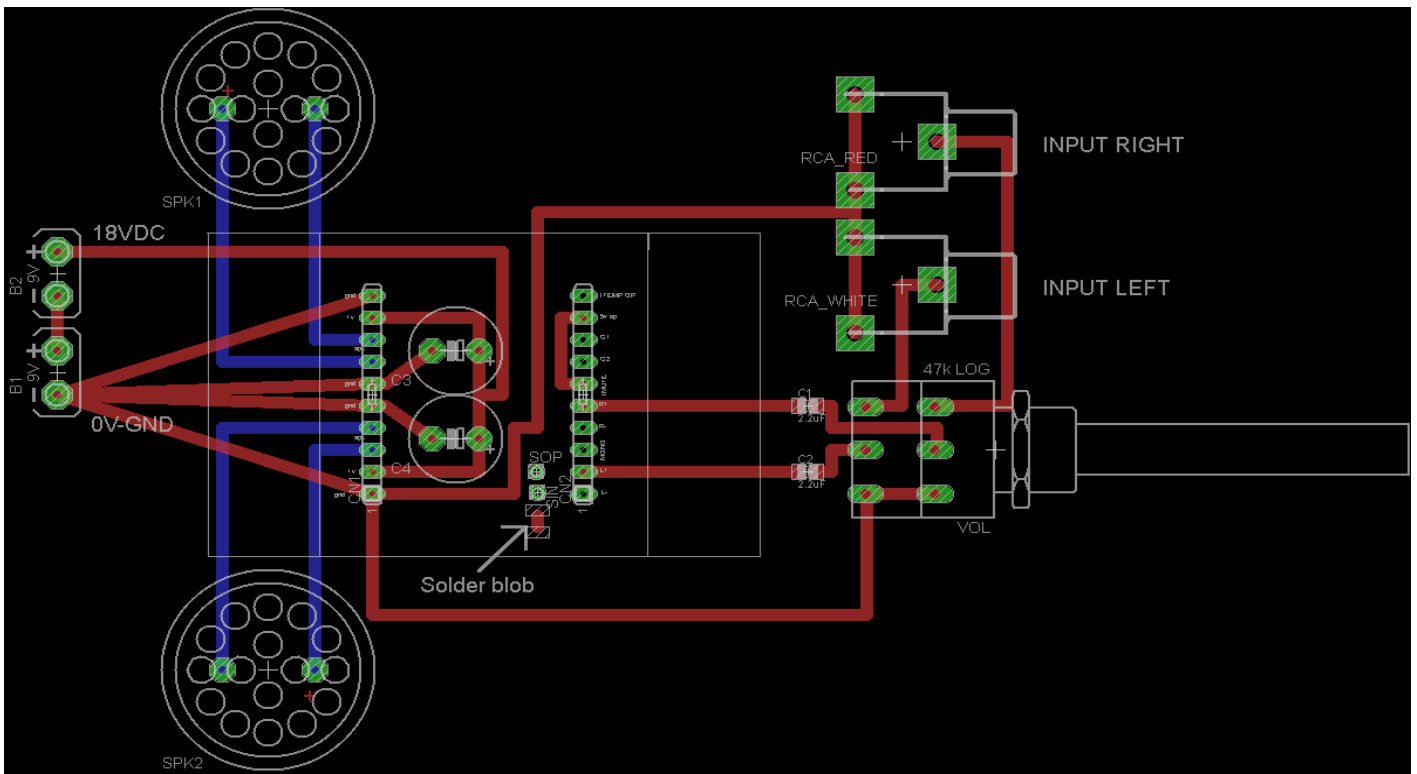


LEFT SIDE

GND
+VDC
OUTPUT RIGHT -
OUTPUT RIGHT +
GND
GND
OUTPUT LEFT -
OUTPUT LEFT +
+VDC
GND

RIGHT SIDE

/TEMP OUTPUT
+5VDC 100mA output
GAIN SETTING 1
GAIN SETTING 2
/MUTE
INPUT RIGHT +
INPUT RIGHT -
MONO/BRIDGE SELECT
INPUT LEFT +
INPUT LEFT -
S_IN sync input (rev C_A only) connect to gnd via a solder pad if not used.
S_OP sync output (rev C_A only)



Above is a typical connection layout. While the HTM50 module has 4 GND connections and 2 +V connections, these are all wired internally on the board so any single combination of GND and +V can be used. However we recommend they all be connected as illustrated. Where each power rail wire is connected direct from the power supply to the PCB. This is a better method than connecting all the GND pins together onboard and then running a single wire. Power supply connections should run direct to the smoothing capacitors C3,C4 then to the module as illustrated. More on connections for power supply is explained in later sections. Input voltages should be typically between 12V and 24V. The module will run down to 10V below this voltage maybe possible, but not recommended. 24V is the maximum voltage level and should not be any higher.

Speaker connections are straight forward. They are labelled onboard as L and R with + and – denominating how they are to be connected. MUTE pin should be connected to 5V OP to enable the amplifier.

The "solder blob" is normally there by default as it is necessary for testing. This defaults the module to MASTER mode and is generally used for single amplifier modules. SIN has a solder pad, which is next to another solder pad (GND). A simple solder blob will enable the amplifier into master mode ready for normal operation.

Inputs L & R should be via a 2.2uF polyester capacitor as show by C1,C2. Note the HTM50S uses a single supply rail so to interface to AC type inputs capacitors must be used to remove the input DC offset (typically 2.5VDC). Failing to use the capacitors will result in a huge DC offset on the speaker outputs. This will push the speaker cone fully inwards or outwards. Running the module with this fault may damage the module and the speaker. Input capacitors should always be used. There is no need to connect the L- & R- pins as these already have a capacitor connected ready for single ended input operation.

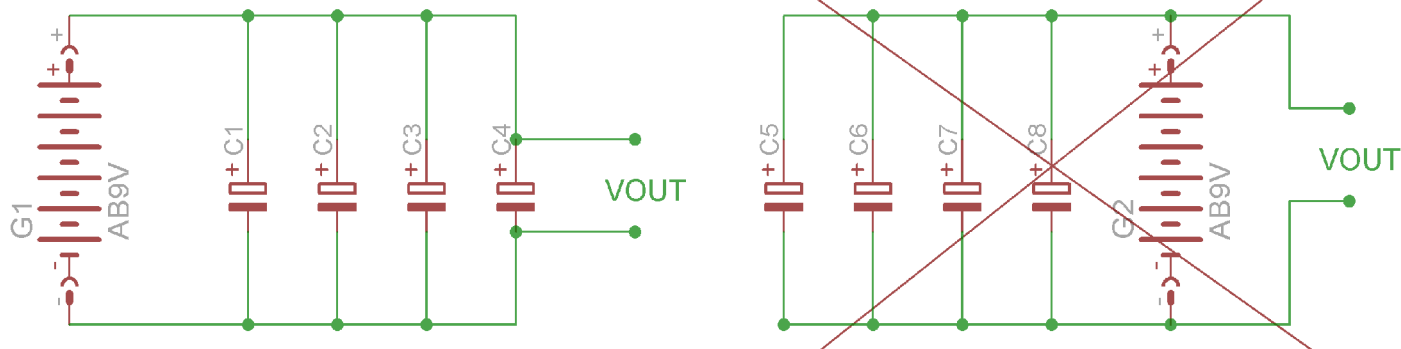
Further connections so a typical volume control connection along with RCA connections. The volume pot should be a LOG type, typically 47K will work in most general designs. All the ground connections so go via separate ground wires to a central point on the HTM50 module. For example we used the bottom ground pin to take all the connections back too. This may well not be the optimal ground point in all designs. Some experimentation may be required to obtain best overall wiring results.

All audio wiring should be used with screened cables. 2 inner cores to carry signal ground and signal itself. Outer sheilding to be connected to the power supply ground point. It is recommended the ground point be connected to mains earth to help prevent mains hum from being introduced between floating grounds between audio equipment.

While the above outlines a typical connection layout, it does not necessarily mean it is the best. What is outlined will allow the user to be able to get the module running as quick as possible with minimal time spent in reading and study. Further information for more advanced wiring can be found in later sections of this datasheet.

Power supply considerations

Below are 2 typical ways to wire a PSU source to the amplifier module. Image to the left shows a typical (ideal) connection layout, while the image on the right maybe mistakenly wired incorrectly as shown. Both images show 4 smoothing capacitors and a power source which is illustrated as a simple battery. VOUT is assumed to be connected to the audio amplifier's power supply rails.



The left image is a typical good connection layout. Where the power supply is physically connected to the capacitors and then to the load, which will be the amplifier module. This has good layout symmetry as its power supply then capacitors then amplifier.

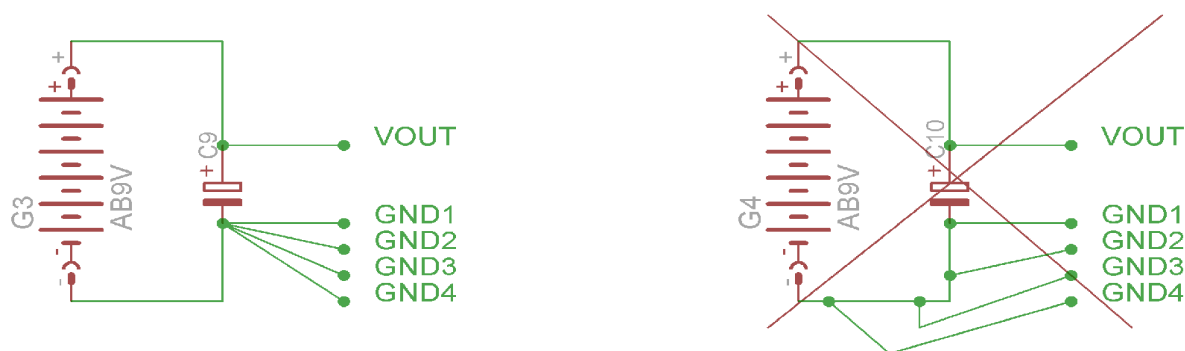
The image on the right is a common mistake which can be easily be wired when building the power supply section. Where the power supply is connected direct to the amplifier module and the capacitors are connected before the power source. The symmetry is on this layout is bad, as in effect, the supply voltage is flowing in 2 directions, to the left then to the right to power the module. This will cause greater power supply ripple.

While both setups will work, it is preferable that the smoothing capacitors are located AFTER then power supply source, and the capacitors supply the energy to the amplifier module. It is preferable to supply the amplifier module from as smooth as source as possible, which will always be from the smoothing capacitors.

Power supply connections should always be as short and as direct as possible. The wire size should be higher rated than the highest current draw. For example, if we expect 2 amps full current draw, we should look towards using at least 5amps rated wire. Resistance in the power supply wiring will decrease amplifier performance and in some extreme cases can contribute to oscillations and stability issues. Smoothing capacitors should be located as close to the amplifier as possible. It is also advised that a toroidal transformer be used with a quality bridge rectifier and high ripply smoothing capacitors.

Switchmode powersupplies are not recommended for class d audio designs. Class D amplifiers typically switch at 100khz or greater, Switchmodes can typically switch from 10khz to 100khz. As such switchmodes can introduce switching noise into the audio range. While this may not seem to be a problem as it is typically out of audio range, the harmonics introduced can cause audio distortions down to 1khz. Further compounding the problem, beat frequencies can happen between the switching of the amplifier module and the switchmode power supply. For example, if the amplifier switches at 100khz and the switchmode switches at 101khz, there will be a 1khz beat frequency. This 1khz frequency will then have harmonics spanning up and down the audio spectrum. These frequencies will then produce multiple other beat frequencies and harmonics to the point the amplifier will develop severe distortions and audible noise.

Below is a typical example as good and bad ground connections. To the left, the ideal "star" earth connections. To the right a poor method of wiring along the ground rail.



The left image shows ideal star earth connections. It is important to note all the ground connections are connected to the 0volt connection of the smoothing capacitor and not the 0volt rail of the supply.

In some cases audio engineers introduce a ground bus-bar and connect all ground connections to it. While this is a good idea for low resistance, it is extremely bad for inductance. All wiring has inductance and the longer a wire is, the higher the inductance will become. This is extremely problematic to overcome as we cannot do away with wiring, but we can help to minimise the effect by keeping wires as short and direct as physically possible.

Inductance will introduce noise into any wire, be it a audio signal wire or power supply. It is not uncommon for a 12inch screened connection signal wire to obtain 100mV of audio noise, simply down to the length and inductance of the wire. A contributing factor is also resistance of the wire. Where voltage drops can happen between various ground connections and also introduce noise into the wiring.

We put together a simple test to show the amount of noise per mm of wire!

1mA = 1.6mV noise, 1A = 160mV noise @ 200khz

1A = 8.20mV, 1mA = 0.072mV @ 10khz

So if we assumed a 10cm cable length carrying 1mA signal current, this would be $0.072\text{mV} \times 100 = 7.2\text{mV}$. Which isn't a great deal. However once our wiring inductance increases due to longer lengths of connections, it is probable to obtain 50mV or more in the audio signals. This is the main results of "noise" when turning the volume control up or down. Its not down the the volume control being bad, as often this is not the case. It is more probable that there is a voltage introduced onto the pot which then causes noise in the audio. The worse the noise sound is, the higher the voltage is on the ground connections.

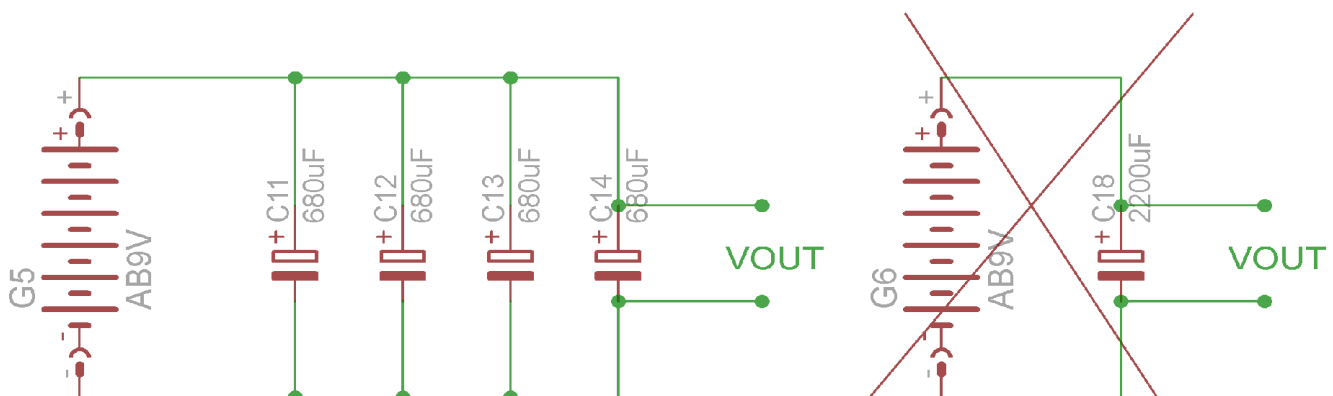
As for the actual supply rail noise, We can obtain as much as 160mV in just 1 mm of wire! At 10cm cable length we have $160\text{mV} \times 100 = 16\text{volts!}$ Of course this is only a hypothetical calculation. In the real world it would be difficult to obtain more than 100mV of noise, though it can be possible to hit 800mV as we have seen in some designs!

It is worth noting that noise on the power supply rail will be radiated as RF noise which can be easily introduced into signal wiring. Care must be taken to screen all audio connections, where a 2 core screen cable is used. The outer screen is to be connected direct to the power supply star earthing point, and inner cores to be used for audio signal and grounds. We must not use the outer screen for ground connections. This connection will have a long route back to the power supply and will be prone to pick-up RF noise, which will then be introduced as audio noise, which as stated previously will be noticeable when altering the volume control.

It is also worth noting that separate wires in parallel offer lower inductance and resistance than a single wire. So it can be a great advantage to run multiple cables from point to point rather than a larger single wire.

The HTM PCB layout uses a ground plane where the entire bottom of the PCB is ground. The top layer has as much ground area as possible around all parts to help prevent radiated noise. Also a foil shield is used to limit radiated noise above the board. All parts such as resistors and solder joints can all act as micro antennas, so the foil helps to minimise the radiated noise throughout the design.

The following images is for guidance and can improve supply current while reducing parts cost.



The left image shows multiple smoothing capacitors of lower value. The right image shows just one smoothing capacitor at a higher value. While there is nothing wrong with using a single capacitor of a higher value, often the ripple current suffers and the price is often higher than using smaller less value capacitors.

For example, a 2,200uF capacitor may have a 1000mA ripple current and can be £3 each. However, we found smaller value capacitors such as 680uF, while in general may have a lower ripple current, some have 2,500mA or more. So in using several smaller capacitors we can obtain a ripple current of 10amps easily. Which is more than we will ever need. However we must not forget the capacitor values either. If about 2,200uF is our desired value, Using 4 capacitors of 680uF will be 2,700uF in total at 10,000mA ripple. A higher value capacitor of 2,200uF will generally not be able to supply 10,000mA of ripple current. So we have improved our design in using many smaller capacitors. It is generally found smaller values are cheaper than larger ones, typically we can obtain 680uF for as little as 50p of generally good spec. This means we saved £1 over buying a single capacitor, and we have increased its capacitance rating slightly higher, and greatly increased the current capability. It is also worth noting that parallel capacitors will have a lower ESR resistance, which also helps to increase the current capability. As a added bonus, often there are better price breaks at buying more capacitors, so overall its a win-win situation.

However, we must not be tempted to use 100's of smaller capacitors thinking this will improve things further. As this is not the case. We have seen High end amplifiers "upgraded" from using a couple of high spec capacitors, into using a huge array of smaller values while apparently improving the power supply. However this is not the case.

There becomes a design trade off between spec and cost. We must not forget that in wiring multiple capacitors we are actually adding in more inductance and resistance. These losses will make it harder for the amplifier to pull current when it is needed even though we have a high current capable smoothing capacitor array. The wiring losses may well reduce the overall efficiency which is what we was trying to avoid! Capacitors must be located as close to the amplifier module as physically possible. There is no use building a high amp capacitor array only to result in several feet of wiring in between the capacitors themselves and the actual amplifier. Unfortunately we have seen such "upgrades" offered at a rather high price on some high end amplifiers.

The HTM module uses 4 smoothing capacitors. These are located either side of the HTM50S amplifier module. These are high ripple types which can supply more than enough current demand. Capacitor values only need to be small, as typically the module will only pull 100mA of supply current. However we have enough headroom to supply enough current to those drum and bass listeners!

Gain Setting

The HTM50S has 4 gain settings, 22,25,29.5 & 36dB default. The gain setting can be easily altered by connecting the G1 & G2 pins to either GND or 5V OP pins as shown in the table below.

G1	G2	dB
Low (GND)	High(5V OP)	22
High (5V OP)	High(5V OP)	25
High (5V OP)	Low (GND)	29.5
Low (GND)	Low (GND)	36 - Default

Normal amplifier gains are in the order of 24dB. We choose the highest gain as in our tests we found that 36dB was better suited for a varying range of audio sources from CD players to computers. This also offers a ideal situation for some users who wish to use a passive cut or boost tone control. Typically as amplifiers are 24dB and 36dB is selected, this is a difference of 12dB which is the normal boost & cut level for tone controls. While HTM audio never recommend using tone controls to boost treble or bass as it will greatly add distortion to the audio, it is however useful for a passive cut filter. For example, suppose a PA system was used for vocals only, any audio under 100Hz can be cut to limit the current drawn from the amplifier. This is ideal where the amplifier maybe run from batteries or other low current sources. It is also worth nothing that the amplifier can be driven in effect to a higher volume output level as most thermal loading is related to sub 100Hz frequencies. See output power section for more information.

If the volume control is half way and the amplifier is driven into clipping, then a lower dB setting can be used to compensate for the high audio input level. Over all 36dB has proven to be the most useful default setting, though as described can be very easily altered.

OUTPUT WATTAGE

the HTM50 module is rated at 50watts total genuine RMS power into 8ohms from a single supply rail of 24VDC. The module can be configured for 2 channels of 25watts RMS each into 8ohms, or a single mono channel of 50watts each into 4ohm. It is important to note than the speaker impedances in relation to the output power.

Most users will assume that MONO means "bridged" output and will aim to configure the speaker connections assuming such. However this is incorrect. The outputs of the HTM50 are already bridged into the 2 channels. What the MONO configuration does is internally link the left and right channels in parallel. This means you can now drive a single 4ohm speaker from the outputs. Note that in stereo mode the output of 25watts per channel is into 8ohms! 4Ohms will of course pull twice the current, so in order to satisfy this, the 2 output channels are linked in parallel giving twice the current output capability. This is of course set by driving the setting the MONO pin. By default MONO is linked to GND for stereo operation, and is connected to 5V OP for MONO mode.

Below is a overlay of the output power vs input voltage and speaker load ~1% THD.

Voltage	4R (MONO – SINGLE CHANNEL)	8R (STEREO PER CHANNEL)
12V	15 WATTS	7.5 WATTS
14V	20 WATTS	10 WATTS
16V	25 WATTS	12.5 WATTS
18V	33 WATTS	16.5 WATTS
20V	38 WATTS	19 WATTS
22V	46 WATTS	23 WATTS
24V	54 WATTS	27 WATTS

It is worth noting that it can be possible to drive , for example, 24V stereo into 4ohms. This will in effect give 54watts per channel for 108watts total RMS. However, this will hit the thermal and current limits of the module so it is not recommended. However, if intended for vocal/PA types where sub 100Hz are not required, then it can be possible to run stereo into 4ohms and not overload the module. A High pass filter can be used to eliminate sub 100hz frequencies.

Another method is using stereo into 4R is to reduce the supply rail voltage as to not hit the thermal overload or overcurrent protection. For example, 12V into 4R will be the same power output as 24V into 8R. In both cases the output power will be 50watts per channel. Though as stated, 24V into 4R will hit 100watts per channel (200watts total) which is pushing the module beyond is recommended operating limits. HTM recommend only following the specified ratings as shown in the above table.

It is possible to link multiple modules in a number of configurations to gain more output power or channels. For example, using the SIN/SO (Sync input & sync output) pins, multiple HTM50 modules can be linked to allow 4 channels or more. It is possible to run 2 modules in MONO mode to obtain 2 channels of 50watts into 4R. More modules can be linked for surround sound systems where 4 channels of either 25watts or 50watts can be configured.

MASTER SLAVE OPERATION

The Sync pins (SIN & SO) provide the necessary interface to operate multiple modules. Users new to class D design will often attempt to use 2 or more modules as a stand alone setup, then spend endless hours trying to work out why there is mass distortion on the amplifier. As already mentioned in the power supply considerations section, beat frequencies can develop between multiple class d amplifiers. For example, if module 1 has 100khz modulation frequency and module 2 has 101khz modulation frequency, this generates a 1khz beat frequency. This will result in break up of audio at 1khz and upper and lower harmonics. This often will sound like whistles of interferences with a great deal of distortion.

To overcome this, most class d designs will have sync pins. Where one module is chosen to provide the master modulation frequency. Then, every other amplifier must be used as a slave where it will be feed with the master modulation frequency. The HTM modules have master and slave pins, where SO (sync output) connects to SIN (sync input) on all the slave amplifiers. It is also important to note SO should be pulled high with a 10k resistor.

PROTECTION SYSTEM

The HTM50 series is programmed to automatically mute when the die temperature of the module goes above about 160degC. A indication there is something wrong will be that the heatsink will be hot to the touch. During regular music at even moderate volume levels the heatsink and board will remain cool. However, driving at full volume with heavy bass music will result in the module becoming warm to the touch which is normal. Extremal heatsinks and fans are not required. Though if the user seems to think they are, then he must consider the module is being driven too hard. Either by to low a speaker impedance on 24V or there is some other system fault. See the output wattage section for more information on safe operating conditions.

Once a thermal overload is detected, the amplifier will mute for approximately quarter of a second minimum. The amplifier will automatically resume normal operation once the die temperature reaches a safe operating temperature The advanced PCB layout will typically cool the module very affectively, and the mute period will typically never be longer than 2 seconds. HTM audio have over driven the module on purpose to test such conditions and typically 1 second is the general overload mute period.

During such a thermal overload the /TEMP pin will be pulled LOW for the duration of the overload. This should not be connected directly to a LED , though can be connected direct to a micro controller or other low input current device. If a LED is required, the /TEMP pin should be buffered with a generic transistor. The user may wish to add a small time delay to show the overload more clearly, as often overloads can last as little as quarter of a second and this may not be a long enough time to visually see the LED light.

It is also generally assumed that overloading the amplifier will always cause a higher current to be drawn and a high temperature to be reached. Thus assuming the /TEMP pin will always be activated during faults, however this is not always the case. During short circuits where e heavy current is drawn, it can often take several seconds to heat up the module before the thermal trip is reached. However, The module does have internal current limiting. Unfortunately Maxim have not documented this in any great detail. However HTM audio have found that the output current may be limited to approximately 2 amps. Assuming a total short-circuit at 24VDC into 2amps is 48watts. This would seem logical as the amplifier is designed to operate at 25watts per channel. During tests HTM audio have been unable to draw more than 2 amps maximum from the power supply rail. So in general a fast blow fuse of 1 to 2 amps is recommended in the supply rail. Fuses can also be used in the speaker leads, though some experimentation maybe required to find the optimal fuse rating for the design under construction.

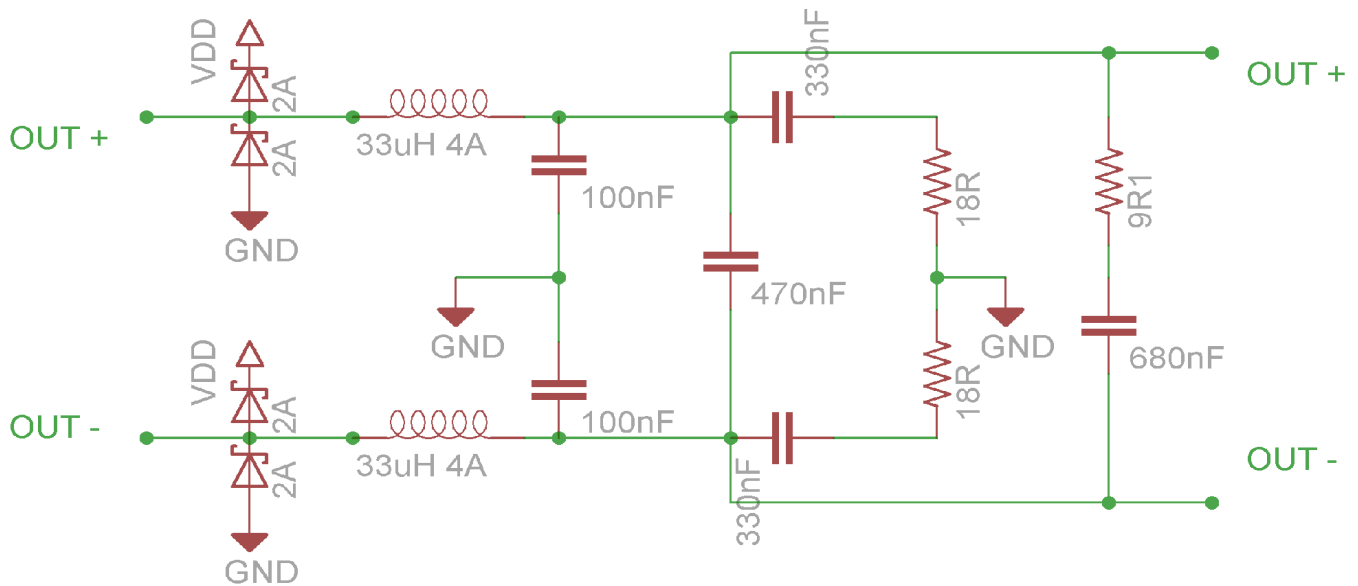
OUTPUT FILTER

The HTM50S module does not include a output filter. The HTM50S is a "spread spectrum filterless" amplifier. It is important to know what "filterless" actually means. For example, like any other digital amplifier a modulation frequency is used. This basically switches the output stage on and off, which causes a high frequency wave to be driven across the speaker. The speaker will then pull current (at 100khz or higher) and can often pull several amps or more even though no audio input signal has yet been applied. In this respect a output filter is used to remove the modulation frequency from the output of the amplifier and practically eliminate the current draw. This is where "spread spectrum" comes into action. Where the modulation frequency is changed randomly, the speaker will not pull any significant current.

For example, if we had a set 100khz frequency and pulled 1 amp of current, we can then progress to 2 frequencies for example 100khz and 99khz, or 101khz. As such the 1 amp of current drawn is now split between 2 frequencies resulting in only 0.5 amp current drawn. The HTM50S module will typically change the frequency enough times to result in a almost nil current drawn. This allows the user to operate the amplifier without a bulky output filter. However, this does not remove the RF energy totally from the speaker cables and can cause interference in the AM RF bands. It is therefore only recommended to use the filterless mode where the entire amplifier and speaker is enclosed in a metal enclosure to shield the radiated RF energy.

The HTM50EVM however, includes the output filter and is aimed to provide the end user with a complete setup to plug in and go. The filter section also includes more overload protection in relation to voltage transients and reverse currents. A typical output filter with additional protection can be seen in the diagram below.

It is important to understand that digital filters should not be driven into audio clipping. The output filters in digital audio systems are designed to operate in a AC type operation. When audio clipping occurs, This places DC into the output filter and considerable current then flows though filter network. In conjunction to the output protection system, such faults will trip the over current protection. It is also worth noting that clipping can occur when the apparent audio volume seems very low. For example, a 15khz audio wave at full volume level can be almost inaudible, and yet in itself draw a very low current level. However if driven into clipping, considerable current will flow though the output filter causing the over current protection to trip. Such operations should not be confused with bass frequency overloads which will trip the over current protection also.



Inductors are generally 33uH 4amps rated current. These should be EMI shielded. While lower resistances are preferable for losses. HTM have found high peak currents can develop in the inductor and cause a false overcurrent cut out condition. Where the amplifier will shut-down for approximately 1 second. Though typically should not be higher than 80mR as this will decrease efficiency.

It is important to note inductors have a tolerance of 20-30%. We found in general a 20% inductor will be 30% lower than the specified value. Another factor is current rating. Often the saturation current is listed by default, where the actual usable current rating is typically 20%-50% the stated value. As current increases though the inductor the value decreases. A 33uH inductor must be able to maintain that inductance up to 4amps, which is rarely possible with off the shelf parts. We found 68uH inductors rated at 3 or 4 amps to be the overall best solution. Where 68uH @~ 30% is 47uH. Where as current increases the value may fall another 10% down towards 40uH. Typically 4amps rating is good up to 3amps on good inductors and 2amps on poorer ones.

Users may be tempted to use even higher values but this also has its downfalls. The output filter is designed to operate with a particular inductance, typically 33-47uH. Higher inductances will reduce amplifier performance and efficiency. Typically 100mR inductor at 2.5amps will dissipate 0.25watts of heat in the inductor alone. This may not sound much but a little heat can built up over time to the point of melting. It is also important to note as the inductors are in effect in series, you have to add the value to 200mR. This is 200mR at 2.5amps results in 1.25watts of heat lost. If we assumed 25watts total output power, then minus the inductor losses will result in $25 - 1.25 = 23.75$ watts output power. This then equates to a overall efficiency loss of 5% per channel.

The HTM50 EVM uses 56uH inductors. These have been tested to give a value of about 42uH. They will maintain a good value up to 3amps which is more than sufficient. If such a inductance of 40uH is used, the 470nF capacitor in the filters should be reduced to 330nF. HTM-audio often use custom made inductors for low resistance and higher power ratings. Users can email for a quotation or visit our website for more information.

Resistors and capacitors can all be 0805 SMT parts. As with the power supply considerations, all wiring and connections between parts should be kept as short as possible. The output filter should be enclosed in a metal can to prevent radiated RF noise which can be introduced into audio leads and back into the amplifier inputs. If this is suspected, a simple low pass filter can be used on the audio inputs to prevent high frequency noise entering the amplifier inputs. Though generally this should not be required.

ONBOARD REGULATOR

The HTM50 module includes a 100mA onboard 5V regulator. On early HTM50 modules a TO92 type was used. On later revisions a SMT parts was used. We found the SMT series offered better regulation and almost indestructible. After some short circuit tests we found the part has a decent thermal overload protection scheme. HTM opted for the SMT regulator due to the fact it can survive short circuits whereas most 5V regulators cannot. The 5V rail acts as a pull up voltage for the module. Though as this only draws a very low current, the user is free to take advantage of the useful 5V supply rail to power external circuitry such as micro controllers or microphone pre-amps.

SOUND QUALITY

We understand a lot of people have been put off digital designs in the past as they sounded awful, and we totally agree. However it is not the fault of digital operation, its the poor design implementation which is at fault. Having worked in the audio industry for many years we are aware of many designs from old to new and tested many.

There has been a lot of debate about digital adding "switching noise" into the audio. While this is true , A lot of linear designs can distort the audio up to 80% with feedback and high gain inputs often adding in multiple harmonic & phase distortions making the sound harsh and "scratchy". This "feature" of linear amplifiers became the "boom and tizz" sound found in most designs today.

People often forget that pure class A designs often were regarded as sounding better than class B designs, even though class A could have up to 10% audio distortion over class B designs. So we should not be put off by a fraction of a percent in class D switching noise. The noise figures are often so low that it would be inaudible even at extreme volume levels.

Digital design may well often look worse on paper than linear, but in audio tests we find this is seldom the case. Linear designs are quoted as having fractions of a percent in distortion in audio tests, though this is often true, real audio is much more complex and very difficult to accurately measure distortion.

In order to obtain less distortion a more complex feedback stage is required which then adds more phasing problems in the audio which are actually increasing distortion by 50% or more plus all the additional harmonic distortions. All this distortion is then fed back into the next feedback stage which just further compounds the distortion problem. However using a single sinewave input the distortion figures quoted on paper look totally fantastic.

There is sometimes so much clutter in the audio path that some of the detail is lost along with in some cases the ability to lose some instruments being played. Where such audio is now lost, distortions are now in play, where the tweeter then over emphasises the distortion into sounding like bright crisp treble. This "sound" then becomes the established "norm" which is actually incorrect.

At first users may think there is no much high frequencies upon initial listening. This is, and isn't correct. The problem with linear designs is distortions and harmonic distortions can distort the sound waves well over 50% even at low volume levels. Tweeters do not like being drive with such distortion and begin to sound very loud giving the impression or "bright clean treble" which is actually incorrect.

Some tweeters will start to resonate around 7khz with such distortions which further gives the impression to bright clean treble. What most users do not understand is the "bright crisp treble" is actually very bad distortions generated by the amplifier and compounded with some types of tweeters. In some reviews this is referred to as "shouty" treble. Unfortunately most amplifier's are riddled with such distortion problems and the listener has probable never heard the music how the original composer intended.

To further compound the problem is the quality of speakers used. In our own speaker designs we find that tweeters are much more efficient than the bass drivers to the point we usually have to add several ohms of resistance in series with the tweeter to limit the volume output. While most bass drivers are generally 85-90db per watt. Tweeters can often be 100db per watt so care must be taken to accurately match both units, something which is seldom done in a mass produced market. As a result of such, when switching to digital designs there may be a lack of treble which is a direct result of bass to tweeter efficiency mismatching. During listening tests the tweeter may sound 50% louder than it should due to distortions, so the manufacture is tempted to use a less efficient tweeter to compensate for this in a attempt to make the audio sound more like it should. Now if we have music with much less distortion then the tweeter will then appear to output 50% less volume. This is not a fault of the amplifier, just a ongoing problem with poor quality designs.

Years ago when we was constructing high power class A designs we soon found that tuning a amplifier to have the least distortion "on paper" always resulted in the worst audio performance. While we could have low distortion figures on paper, the actual audio sound quality would always exceed our expectations with higher distortion figures. The novice will often look for the lowest distortion figures as possible and thats what manufactures aim to supply, amplifier's with ultra low distortion figures. However during actual listening tests a higher distortion amplifier may well easily out perform the lower distortion design. This brings us back to the old debate that Class A always had higher distortion figures but generally sounded better. This has been a ongoing debate between audiophiles over the past decade.

HTM designs do not suffer from such distortions and offer a much smoother sounding audio which will hold more depth and detail. In some cases we have listened to music and discovered there is more going on in the music than we initially thought.