USB powered Guitar Amplifier

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1 Introduction

One may wonder why someone would want to design a guitar amplifier, a piece of electronics more appropriate for wizards than engineers. Well, not being a guitar player myself I got little interest for this subject until recently. I changed my mind when some guy came looking for help for its own guitar amplifiers and the problem was always the same: Too much humming. Last time the troublesome amplifier was a brand new unit that, apart from the LED in the pilot light, could have been manufactured seventy years ago. Yes, it uses valves (This little hummer vox was British, so lets name them "valves" instead of "tubes"). The problem with the hum was related to its poor power supply design. It was not more than a pair of silicon diodes and a capacitor, so it is no wonder if it hums like crazy. But its disgusting sound was considered acceptable by the manufacturer and left for the final user to fix it, probably because it is understood that to mess around in a chassis with hundreds of volts awaiting the opportunity to electrocute you is a lot of fun.

Yet, this gave me the opportunity to study the amplifier using a simulator, which is much safer than touching the real thing, and to find that it can deliver 4 watts of power to the speaker at the cost of some 25 watts at the power cord (not including the valve heaters). At this point I started thinking that you can sell anything to a guitar player as long as it has at least one valve in it (a search on Internet about this particular amp finds many reviews with lots of praise, nobody complains about the factory-built hum), and also I began to consider the possibility of designing an equivalent amplifier myself. Of course, using modern electronics.

The first design goal was to be able to power the amplifier with an ubiquitous cell phone charger. With 5 volts and 1 amp, it would be enough to achieve more or less the same output power if an efficient output stage is used.

The second goal was to get more or less the same gain range than the valve amp, and to include some sort of soft clipping in the case of overdrives because I think this is the thing the players talk about when they mention this or that "brand-name sound". BTW, the 75dB of maximum gain of the valve amplifier is nonsense when the guitar can deliver peak voltages in the hundreds of millivolts, unless you are intentionally overdriving the amplifier which if often the case. And those guys get no satisfaction with an square wave at the speaker.

2 Amplifier design

The basic diagram of the amplifier is shown in Figure 1. At the end I settled with a design with a switched, class D, power stage capable of converting almost all the power it consumes into speaker output. With 5V supply and a bridge configuration this can achieve a theoretical maximum power of 3.125Watts into a 4Ω speaker, which is near the output power of the valve amplifier and loud enough to disturb neighbors. Notice that the output power almost doubles when the output stage is driven with an square wave instead of a sine. A good news for shredders, I think.

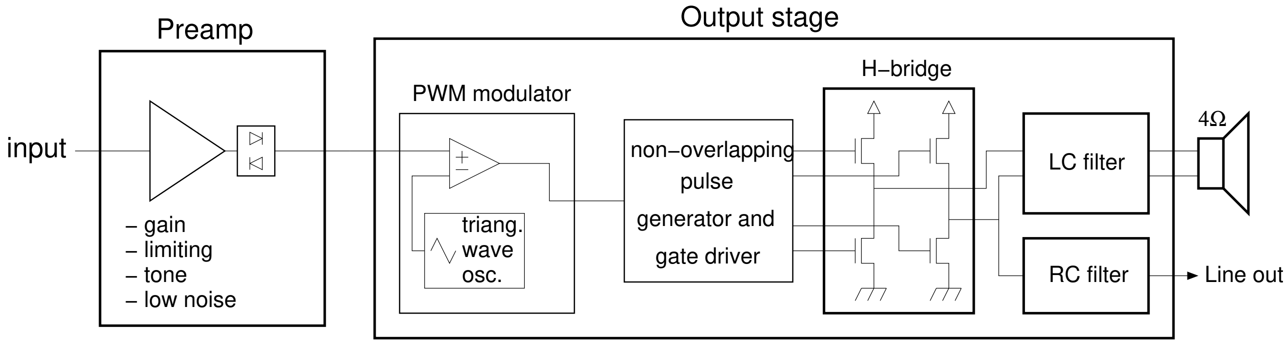

Figure 1: Block diagram of the propossed amplifier

The main problem with this kind of amplifier is the non ideal behavior of components that usually ends translated into distortion. But wait a minute, this is an amplifier for a guitar and players do love distortion! So it isn't a problem after all. In fact the first prototype was more linear than expected and, risking being tagged as a "HiFi" instead of "guitar" amp, something had to be done to include more distortion. But not in the power stage, distortion was generously added to the preceding preamp that also includes the gain, in this case about 40dB. This maximum gain looks lower than that of the valve amplifier, but you must add another 15.6dB of gain from the power stage, and to take into account the lower speaker impedance (the valve amplifier used a 16Ω speaker). Even with this reduced gain the amplifier can go into clipping for only 16mV of input amplitude when the corresponding pot is set to maximum.

In addition to the speaker, the amplifier also includes a low amplitude output, "line out", because I much dislike the common practice of putting a microphone in front of the amp. A line-out socket will provide a better quality signal without ambient noise to the sound engineer and will save an expensive microphone. This output is derived from the same terminals that carry the speaker voltage in order to capture any non-ideal effect that would also be present as sound.

The two stages of the amplifier are described next.

3 Output stage

The output stage is the most complex part of the amplifier and its schematic is shown in figure 2. A pulse-width modulator is built around an astable multivibrator, U2, and a comparator, U3. In the classic 555 multivibrator the timing resistors have been replaced by two current sources. One current source, Q2, is fixed at about 500μA, while the other, Q1, can be switched on and off using the discharge output of the 555 and sinks twice the current. This results in linear slopes during the charge and discharge of the capacitor C10, and a triangular wave of 1.66V peak to peak and 68kHz, at the negative input of the comparator. The audio signal is compared to this triangular wave giving a sequence of variable width pulses at the modulator output.

Then follows the block labeled as "non-overlapping pulse generator and gate driver" in figure 1. This block is built with the ICs U4 and U5, and its purpose is to guarantee around 100ns of "all-off" time for the four MOSFET transistors in the H-bridge. The required delay is obtained by means of the resistors R23 and R24, along with the input capacitance of the NAND gates U4:C and U4:D. Without this circuit the MOSFETs Q3 and Q5 (and also Q4 and Q6) can be both in conduction for a short time during the PWM edges resulting in high current spikes and power dissipation. This circuit is therefore one of the two key components for achieving low power consumption, the other is the inductor L1, of the speaker filter.

The H-bridge consist of four MOSFET transistors, all rated around 3A of maximum current. But these power devices come in tiny SOT-23 packages unable to dissipate much more than a quarter of watt, so efficiency is mandatory in order to not turn these transistors into smoke. The bridge applies plus or minus 5V pulses to

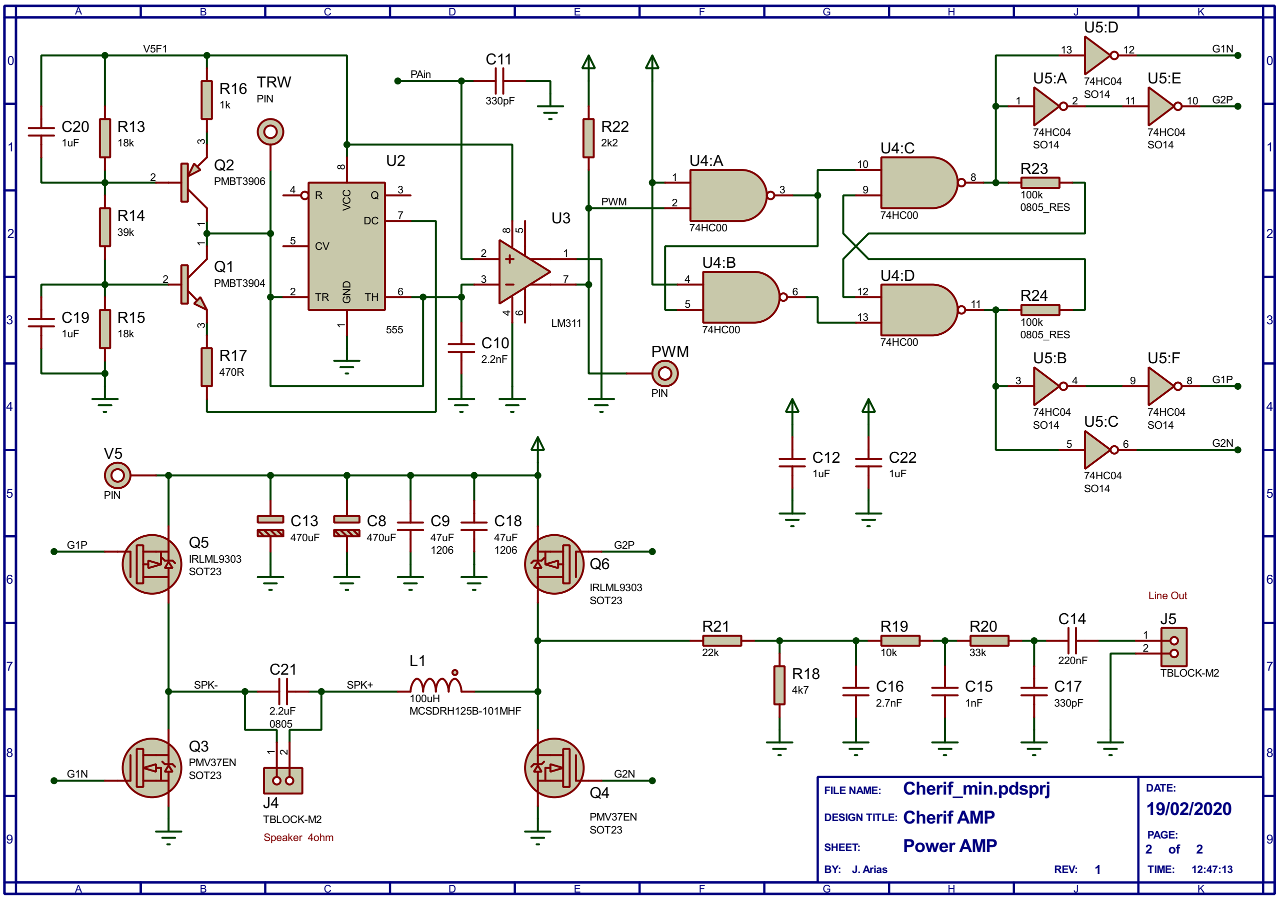

Figure 2: Output stage schematic.

the speaker filter that, after removing the PWM carrier, will result in up to 10V peak to peak analog output.

Finally, the speaker filter, built with inductor L1 and capacitor C21, filters the PWM carrier out of the speaker voltage. But this isn't the main purpose of the filter. With the carrier at about 68kHz the filter is already included in the listener ear and there is no need for electrical filtering. In fact, the inductor L1 was included only for efficiency purposes as it returns to the supply the excess current consumed during the previous PWM semicycle. In order to illustrate this lets consider a PWM signal with a 50% duty cycle, which is what we get when there is silence at the audio input, and lets assume the capacitor C21 is big enough to be modeled as a short. In figure 3 the four different states of this bridge during one PWM cycle are shown. First, during the state 1, the current builds up in the inductor at the cost of consuming it from the power supply, until the voltage levels change. Then, in state 2, the conducting transistors are others, but the current in the inductor keeps flowing in the same direction, against the voltage potential, and returning charge and energy to the power supply. During this state the current in the inductor decreases until it reaches zero, and then, in state 3, it starts to flow in the opposite direction, consuming charge from the power supply. Finally, in state 4, the bridge switches again and the inductor current is also returned to the supply. In overall, the average inductor current and the supply current are both zero and no power is consumed at all, at least while there are no resistive losses in the inductor and switches.

Even during the short time intervals when all transistors are off the inductor current keeps flowing to the power supply via the parasitic drain-bulk diodes of the transistors (see figure 2, Q3-Q6), resulting in small voltage overshoots on the two sides of the bridge.

If the duty cycle of the PWM signal is not 50% a net current builds up in the inductor, but in this case the capacitor C21 gets charged until its voltage is precisely the one needed to make the average current zero again, and this voltage depends linearly on the duty cycle. Only by connecting a resistive load, the speaker, in parallel

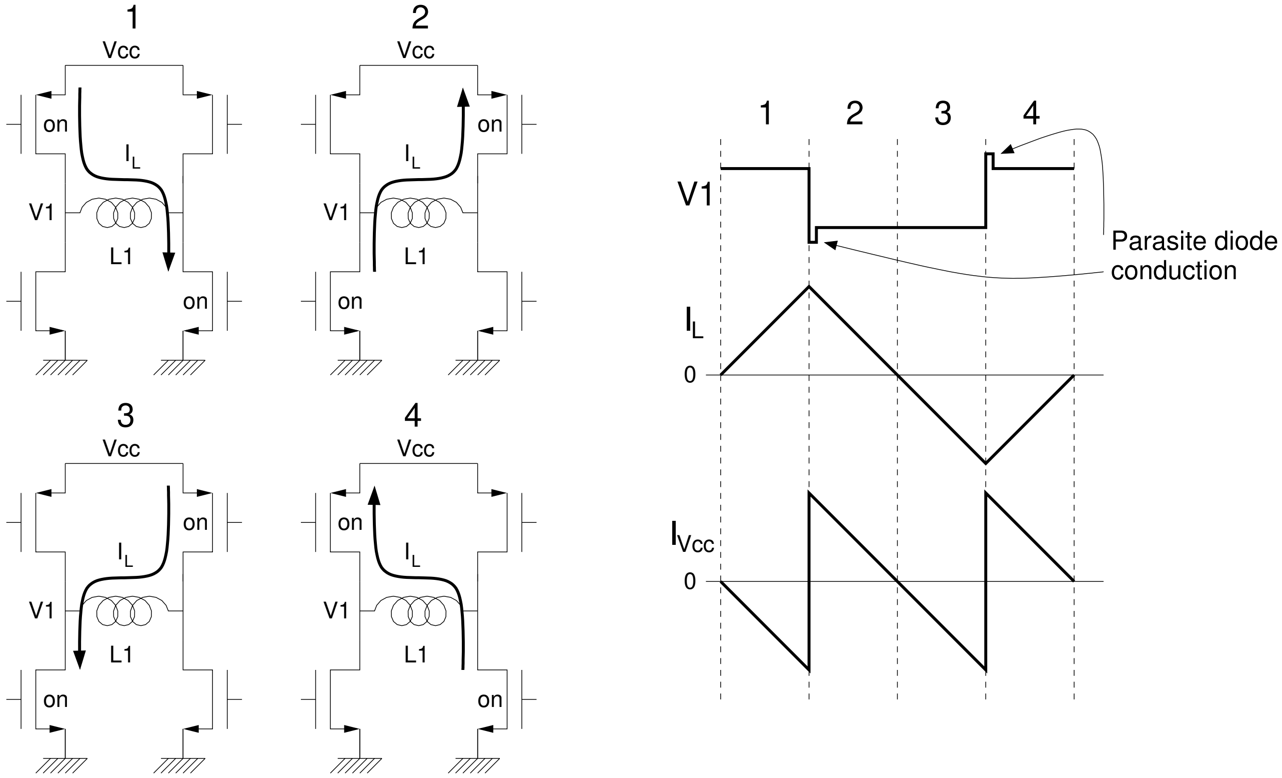

Figure 3: Currents in the bridge and inductor during one PWM cycle.

with C21 we can get some power consumption.

This is the way the output stage gets its great power efficiency. Without the resistance of the MOSFET switches and inductor the efficiency would be near 99% (the parasitic diodes decreases this figure a little from the ideal 100% because they disipate power). In more realistic SPICE simulations a near 90% efficiency seems to be achievable.

The line out signal is also obtained through a filter, but in this case it uses resistors and capacitors because very little power is involved. Also, its input signal is obtained from one side of the bridge and it includes the same voltage overshoots that could affect the speaker sound.

4 Preamp

The schematic of the input stage of the amplifier is shown in figure 4 along with other ancillary functions of the circuit, such as power filtering and expansion connector. All amplification is provided by a single, rail to rail, operational amplifier in a non-inverting configuration. The gain of this stage is controlled mainly by RV1, and its value is $gain \approx [1 + RV1/(R4 + R1/2)]$. R12 and C3 are included to avoid the amplification of the offset voltage of U1, but otherwise they don't affect too much the characteristics of the stage. With this configuration the input impedance is about ten times R8, or $220k\Omega$.

The nonlinear feedback network of the operational amplifier deserves a more detailed discussion. Its goal is to provide a transfer characteristic like the one shown in figure 5 and, therefore, a soft clipping for overdriving voltages at the input. It works as follows (assuming JP1 is shorted):

For low input amplitudes the output voltage is small and the four diodes, D1-D4, are off. The slope of the transfer curve is the gain, and it is almost proportional to the value of RV1 (the minimum gain with RV1=0 is one, or 0dB).

When the opamp output is about ±650mV D2 or D3 enters conduction, effectively connecting VR2 in parallel with RV1 and reducing the gain. The amount of gain reduction is controlled by RV2. With increasing input voltages the output keeps also increasing, at a lower rate, until it reaches about 1.3 volts.

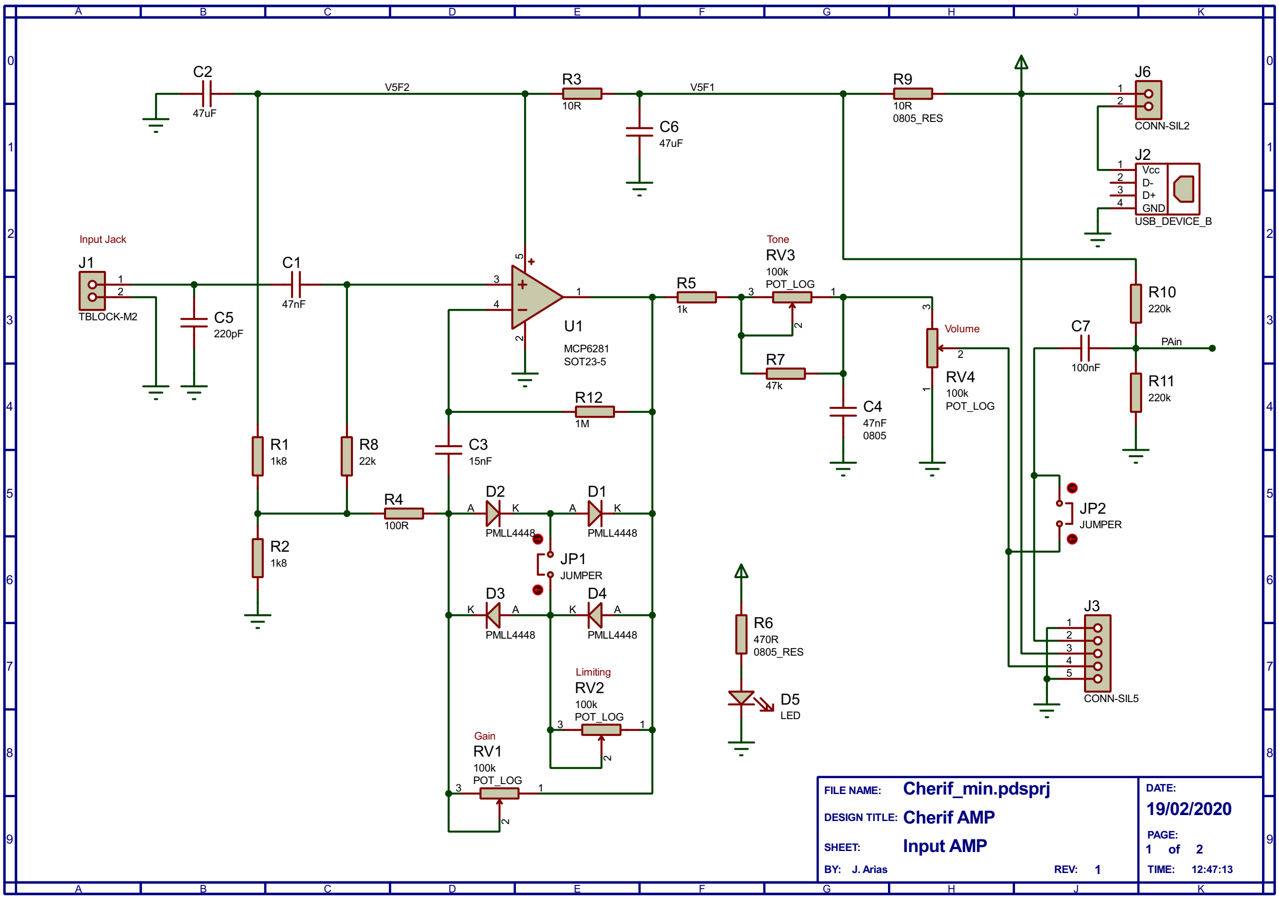

Figure 4: Schematic of the input stage.

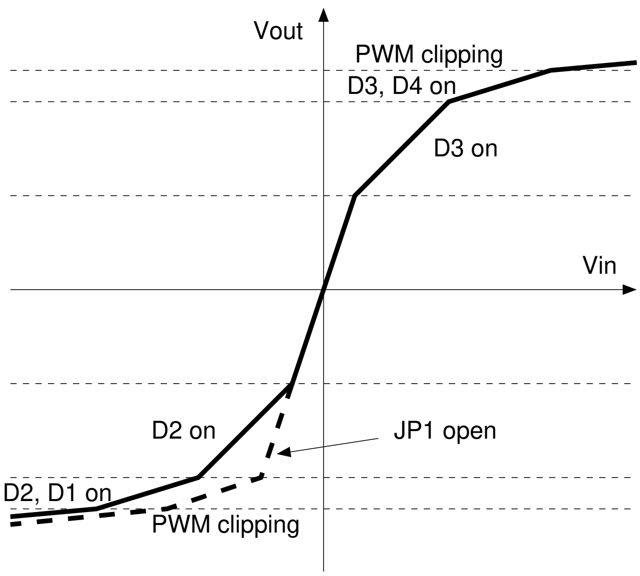

Figure 5: Transfer characteristics of the input stage.

Figure 6: Photograph of the prototype board

Now, the two series diodes D3 and D4, or D2 and D1, are in conduction and their effect is to short RV1 completely, resulting in a unity gain.

With still higher input voltages the output keeps slowly increasing until the audio amplitude reaches that of the triangle wave oscillator in the PWM modulator and hard clipping is finally reached. This can happen if the master volume control, RV4, is set to maximum. With lower volume settings hard clipping is reached when the operational amplifier output is near the supply rails. This would require a quite big input amplitude because now the gain is only one.

This behavior can be altered by opening the jumper JP1. In this case the first clipping is applied only to the positive side of the output wave. This asymmetric clipping can have the interesting result of generating more amplitude for the second and fourth harmonic of the input tone, while mainly only odd harmonics will be generated otherwise.

The input stage also includes a tone circuit that is just a low-pass filter with a cutoff frequency controlled by RV3.

The connector J3 is included to allow for the inclusion of other effect boards to the amplifier, with a digital reverb/echo board in mind. If no expansion boards are connected to J3, JP2 has to be shorted.

5 Experimental results

A prototype board for the amplifier was built and tested. While the photograph of figure 6 shows a board with almost all the components installed in fact we followed an incremental mounting. First, only the ICs and their related passives were installed on the board and their basic functionality tested. Then, the power MOSFETs and through-hole components, like the pots, were added.

The first signal checked, apart from supply voltages and current consumption, was the output of the triangle-wave generator. In figure 7 we can see this signal along with some measurements. The frequency was measured at about 53kHz, and the peak-to-peak voltage was almost 2V, higher than the theoretical 1.66V expected. The higher than expected amplitude and the corresponding lower frequency were probably due to the internal delays

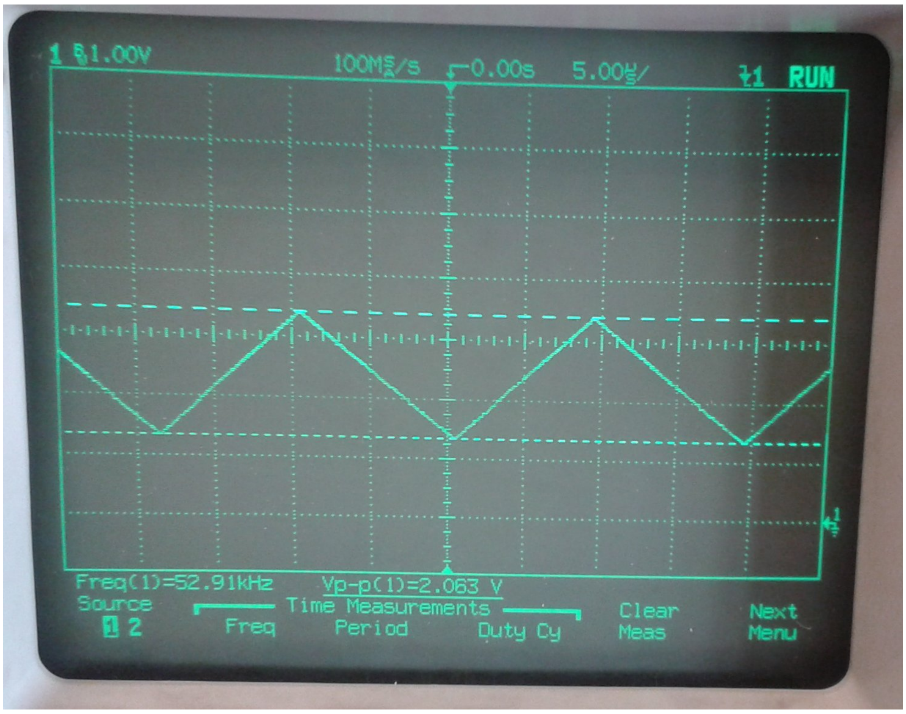

Figure 7: Measured triangle-wave of the PWM modulator

of the 555 IC. But in any case, the 2V amplitude defines the linear input range of the power stage.

With the power MOSFET now installed, their gate voltages were checked. The resulting curves are shown in figure 8(a), where the delay between turning off one MOSFET and turning on another is clearly visible. This delay is around 100ns, but isn't easy to measure accurately because the large gate capacitance of the MOSFETs slows down the driving signals. Also, in figure 8(b) we can see the voltage at one output of the MOSFET bridge, and the 700mV overshot we get when all the MOSFETs are off and the inductor current flows through the parasitic drain-bulk diodes of the MOSFETs. Here we can conclude that the actual time when all MOSFETs are actually off is around 50ns, or slightly less.

Next, the performance of the output stage was measured. In order to bypass the preamp the input signal from the generator was connected directly to the output side of the jumper JP2. The generator was a 12-bit DAC driven with a synthesized 1kHz sine wave with variable amplitude. A dummy 4Ω load was connected at the speaker terminals and the resulting differential voltage was low-pass filtered, attenuated, and converted to a single-ended signal using an operational amplifier before being applied to a 12-bit ADC. Both the DAC and the ADC were running at 100K samples per second under the control of a dsPIC33 DSP. The digitized samples for each amplitude were uploaded to a computer and further processed in order to obtain the harmonic content of the signal via Fourier transform. The final result is shown in figure 9.

Here we can see the lineal increase on the first harmonic amplitude until the onset of clipping. Also notice that all the other curves have amplitudes relatives to that of the first harmonic. The total harmonic distortion, THD, is dominated by the second harmonic amplitude up to the clipping region, and its value is mostly under -40dB. Just before clipping the THD is around -45dB, or conversely, about 0.6% that is a quite good value for a power amplifier.

The power at the supply and the dummy load were also measured. With a constant voltage at the input the amplifier draws only 23.7mA of current from the 5V supply. I want to mention that the LED in the pilot light was not mounted in order to avoid adding a noticeable current to this quiescent power. So low it is that the amplifier could be powered by AA batteries for days if there is no sound at the speaker.

With a sine wave and near the clipping amplitude the amplifier draws 0.56A from the 5V supply and delivers 2.9V rms to the dummy load that was measured to be 3.77Ω. This results in an output power of 2.3W and an

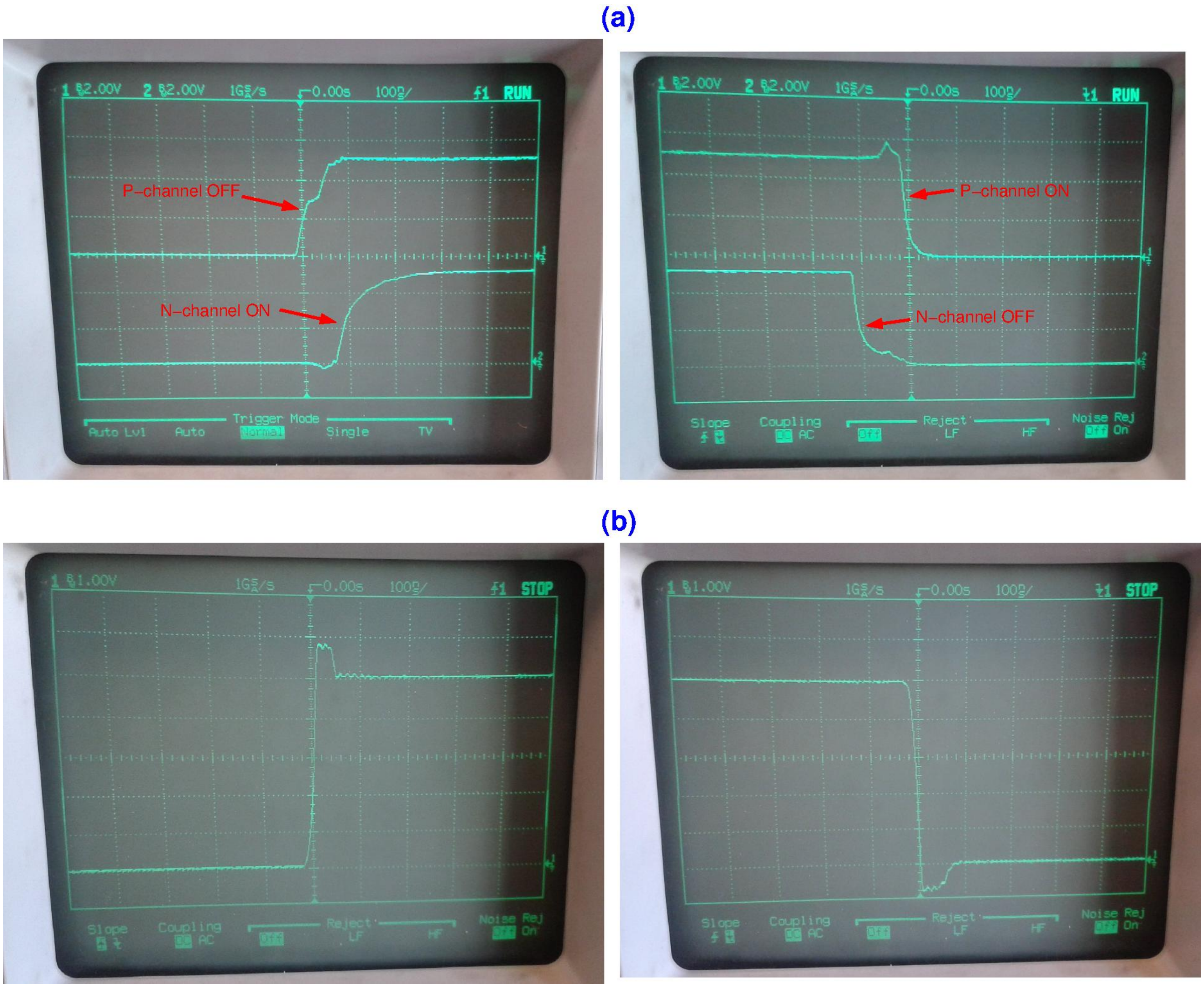

Figure 8: (a) Gate voltages of the MOSFETs Q3 and Q5 showing the all-off time. (b) Voltage overshoots in the bridge output during the all-off time of the MOSFETs.

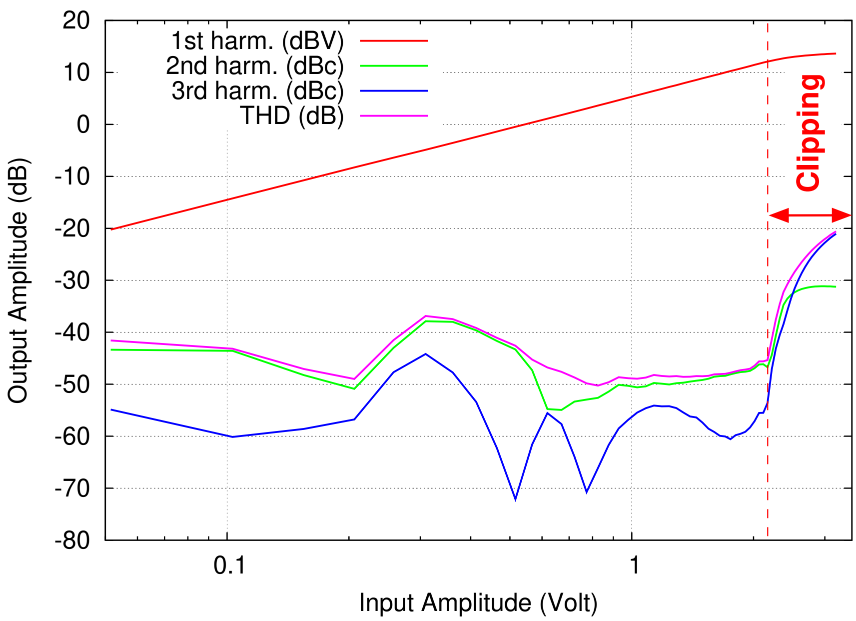

Figure 9: Measured distortion of the output stage. (1kHz input tone)

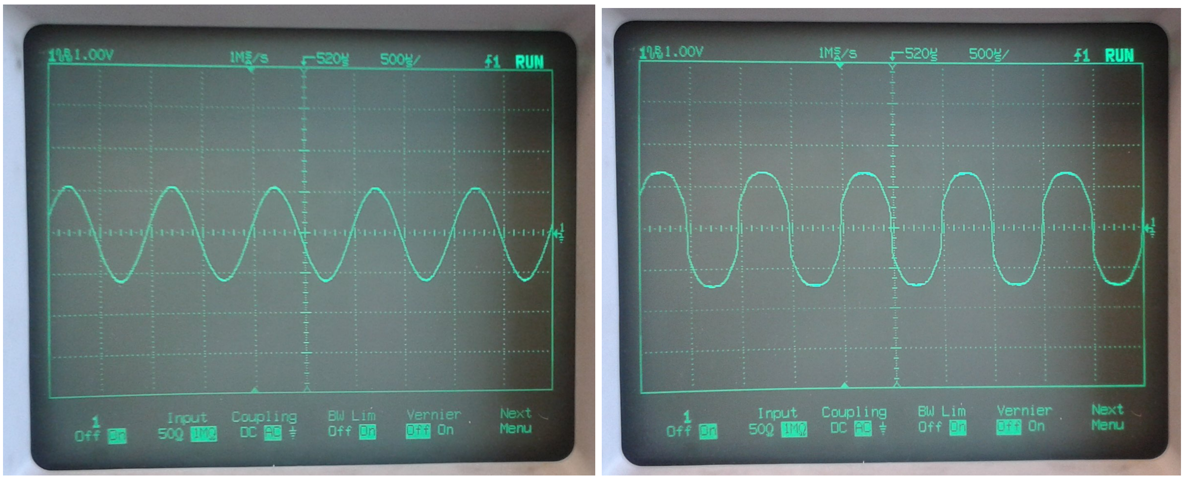

Figure 10: Waveforms at U1 output for two different pot settings. The second trace shows the effects of smooth limiting.

efficiency around 82%. Going further into clipping we measured 3.25W at the speaker with an efficiency of 85%.

These values are lower than the theoretical maximum, and the main reason for this is the on-resistance of the p-channel MOSFETs that is a bit too high (0.22Ω) But don't try to put another MOSFETs with lower resistance because they all have a very high gate capacitance and the driver IC wouldn't be able to turn them on and off in a reasonable time. The series resistance of the filter inductor is also responsible for a noticeable fraction of the power loss, and it is surely more than expected due to the skin effect.

But in overall there isn't a big difference in sound loudness with the valve amp (remember that the ear follows a logarithmic curve for volume), and the power efficiency is so good that a battery powered version seems to be quite practical. Of course, a continuous sine wave with half an amp of current will drain the battery quickly, but that signal isn't very representative of the actual sound of a guitar where maximum amplitude bursts are short and the average volume is low.

With the pots already in place, the input stage can also be tested. In figure 10 two scope snapshots are shown. The first has a low gain setting and almost no limiting (pot RV2 at maximum), resulting in a near perfect sine wave, while the second has a higher gain and a stronger limiting, showing a severe distortion. These waves were recorded at the U1 output, before the low-pass filtering of the tone control and the attenuation of the volume control. They look pretty on the scope screen, but their sound is another question (I personally don't like the sound of distortion very much, but I'm not a musician)

6 Conclusions and improvements

An efficient, class-D, guitar amplifier is designed and tested. This design can be powered with a cell-phone charger with an USB cable, and it shows a performance comparable to some much more expensive and dangerous valve amps. In the circuit the components with a lower maximum rating for their supply are the digital gates, U4, and U5, and the rail-to-rail operational amplifier, all of them with 7V maximun. Therefore it is possible to power this amplifier using four AA alkaline cells instead of a phone charger. This would increase the supply voltage to 6V and the maximum linear output power to 3.3W while making the amplifier portable.