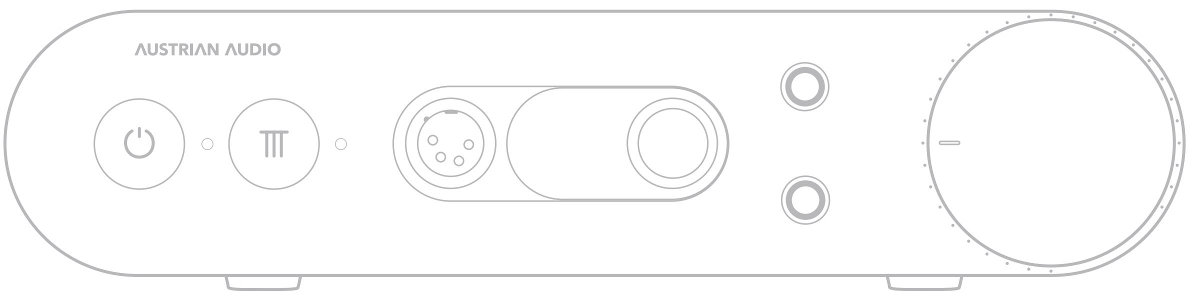
Austrian Audio | Paul Weinreich

Behind the *Full Score one*

a technical essay



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1 Introduction

Over the following pages we will take the time for a small peek behind the curtains of the Austrian Audio *Full Score one*. We will have an in-depth discussion of the circuit technology we use, of measurements and of the intentions we had when designing the product.

When it comes to headphone amplifiers or audio amplifiers in general, many people assume that the last word has long been spoken. Furthermore, there are seemingly as many headphone amplifiers already on the market as stars in the sky, all with very different stunts and selling points, starting with the vanishing low Total Harmonic Distortion (THD) figures of the devices deploying the *THX AAA*[1] circuit, up to the high impedance, comparatively high THD outputs of some tube powered amplifiers. Also, there is a significant amount of people declaring that they are not able to distinguish the sound of different transistor amplifiers at all, or that all of these amplifiers more or less sound the same.

We will try to explain why that is and how many designs used today in audio amplifiers may still be flawed if a truly transparent signal reproduction is pursued. In order for the design goals of the *Full Score one* to be understandable, we first have to define the basics and what we mean by the term *'truly transparent'*. Everything boils down to the question of how objectively measurable parameters can be translated into perceivable differences. Let us start with the easy parameters:

1.1 Standard Measurements and Limits of Hearing

- The smallest perceivable step change in signal amplitude can be detected at about 0.3 dB for a pure tone and at 0.5-1.0 dB for more complex signals*
- The smallest perceivable difference in pitch is 0.2% between 500 Hz and 2 kHz, the in this regards most sensitive region of hearing*
- Lower-order THD is definitely perceived above 1 %, higher order THD above the fifth order already at 0.3%. There is no evidence that differences between 0.001% and 0.005% have any meaningful impact or that a THD of 0.00001% is necessary for a transparent system.*

All three requirements above can be easily met by many amplifier designs. Real perceivable differences may still be found in the time domain. We sometimes call this a 'blind spot' in the characterisation of audio amplifiers, since the complex nonlinearities in the time domain that an electronic circuit can impose on a signal are usually not included in the standard audio measurement suite, let alone been thoroughly connected to possible subjectively perceived differences in the sound. Leaving out the impact on the listener, for now: What are the most important measurements in the time domain?

[1] THX AAA Circuit Website: https://www.thx.com/aaa/, visited on March 12th, 2024

1.2 Measurement Parameters in the Time Domain

- Slew rate: The value of the slew rate is measured in V/µs and describes the amplifier's maximum rate of output excursion. For example, an amplifier with a slew rate of 1 V/µs is able to output a step of 1 V within 1 µs. The slew rate of an amplifier may be signal dependent in a nonlinear manner. There are systems that have a higher slew rate with smaller signals or the highest slew rate in the middle of the output range. Also, many amplifiers, especially audio ICs, have a mismatch between unity gain bandwidth (usually far too high) and the corresponding slew rate (usually not sufficient in combination with the high unity gain frequency).

- Rise time: The rise time (µs) of an amplifier is not necessarily dependent on the slew rate and the bandwidth. Amplifiers can be designed so that they produce a constant rise time, which leads to a variable slew rate. The rise time is normally measured between 10% an 90% of the peak signal level. Let us briefly omit the correct measurement for a short simplified demonstration: An imaginative amplifier has a rise time of 1 µs, which means that the amplifier has a slew rate of 1 V/µs when outputting 1 Vpk and a 5 V/µs slew rate for an output of 5 Vpk. Arguably, a constant rise time would be preferred in most use cases, especially in audio, since an amplifier with a constant rise time treats all output signal levels alike. Consequently, the large signal bandwidth is the same as the small signal bandwidth of an amplifier with a constant rise time. A clean time domain behaviour would be guaranteed by design.

- Unity gain bandwidth frequency: This is the frequency, at which the amplifier drops out of active amplification and - for most systems - it is the frequency with the least phase margin. Looking at the UGBW can be helpful in assessing the stability of a circuit using any form of negative feedback.

- Phase margin: For amplifiers comprising a global negative feedback loop as a form of error correction, the phase margin describes how much phase shift the amplifier can tolerate before the feedback loop becomes unstable. For almost all designs with a low resistance path into the load, the phase margin is especially important, since the load has a lot of authority to influence the transfer function of the amplifier and therefore influence the overall phase margin. Power amplifiers intended to drive passive speakers usually have to provide ample phase margin, as passive speakers with complex passive crossovers can present lots of frequency-dependent phase shift. Even for headphone amplifiers it is very important to have an eye on the phase margin as there is a wide variety of load impedances, ranging from very inductive loads to quite large capacitances of some generously built high-end cables. The phase margin can also be signal level variant for amplifier designs that suffer from a level-dependent bandwidth.

@ 1 Vrms, 10 kΩ load	TL072	NE5532	NJM4580	LME49720	AD797	OPA1612
[k]=%	0.003	0.002	0.0005	0.00003	0.000001	0.000015
OLG	125 dB	100 dB	110 dB	140 dB	120 dB	130 dB
UGBW	5.25 MHz	10 MHz	12 MHz	55 MHz	110 MHz	40 MHz
needed slew rate	47 V/µs	88 V/µs	107 V/µs	489 V/µs	977 V/µs	355 V/µs
actual max. slew rate	20 V/µs	9 V/µs	5 V/µs	20 V/µs	20 V/µs	27 V/µs
min. phase margin	56 °	50 °	58 °	55 °	65 °	67 °

Table 1: Comparison between some of the most popular audio ICs to date

The calculation for the needed slew rate was done using the steepest gradient of a sine wave at the rated unity gain bandwidth frequency with 1 Vrms. We can observe the mismatch between the available and the needed slew rate for many audio ICs at moderate output signal levels already. This is still a simplification since we leave out the fact that all mentioned ICs have a nonlinear rise time and a slew rate over their output range and that all circuits in addition are faced with their own harmonic and non-harmonic distortions, let alone that we arbitrarily decided on the level of 1 Vrms. Also, all amplifiers have their own harmonic and inharmonic distortions (like crossover artefacts) that - when fed back - can exceed the steepness of a pure sine wave.

1.3 Listening Test concerning the Time Domain

What is the influence of these parameters on the sound of an amplifier? Listening to the first prototype of the *Full Score one*, the product manager of the project was pleased by the sound details the amplifier left untouched, but at the same time he was surprised by the harsh honesty presented by the system in combination with some recordings. Born was the idea of a toggle switch between two different modes, one slightly dialling back the speed of the amplifier and one providing a very short rise time. To answer the question of how far the amplifier should be dialed back, we at first had to answer the question: What is the minimum rise time requirement for an audio amplifier to be imperceptible? Austrian Audio conducted the following three double-blind listening tests. The first two tests were conducted with the same 40 participants, the third with merely 22 participants (due to Covid restrictions). The listening setup was composed of a DAC routed through or bypassing a meticulously parameterized test preamplifier. The main amplifier always stayed the same and was not unlike the main amplifier circuit used in the *Full Score one* today.

1. In the first run the listening panel faced a double-blind test to decide whether a preamplifier circuit's presence in the signal chain could be perceived. The goal of the first round was to determine the critical rise time at which the preamplifier could be noticed. Therefore, the rise time of the test amplifier was increased from 250 ns to 1 µs in 200 ns steps. 65% of listeners could make out the preamplifier above 650 ns. All other electrical properties of the preamplifier remained unchanged during the test.

2. The next test was to determine, whether a constant rise time amplifier was to be preferred over a constant slew rate circuit. The constant slew rate circuit could be identified below a slew rate of 100 V/µs by 80% of listeners against the constant rise time circuit adjusted to the same slew rate at the maximum peak signal level.

3. The third test was done by another, smaller group during the Covid pandemic. The panel had to vote for their favourite out of three different kinds of preamplifiers, slowing down the whole signal chain, when the TTT function is disabled. The results were inconclusive, as there was no clear favourite.

All test results have to be taken with a large grain of salt. Subjective (listening) tests are prone to be influenced by manyfold (outside) factors even when they are done doubly blind. Small deviances in the test setup can scramble the conclusion. Plus, our two different test panels were too small, even though the first panel delivered seemingly significant results. Nevertheless, we used some of the outcomes as guidelines to formulate the design goals in accordance with our own listening experience.

After these tests, we are now very curios to hear about the reaction on the market as the listener base widens, and to learn from many more people about how the sound of the *Full Score one* is perceived, with and without enabled *TTT*.

2 Design Goals

2.1 Total Harmonic Distortion Target

As stated in the introduction, of the essay, the total harmonic distortion target is less important than a good behaviour in the time domain for the design of the *Full Score one*. The THD value shall be low enough for there to be no evidence that the amount can be perceived, and so that the value is (in practice) guaranteed to still be dominated by any headphones. This demands a value somewhere below 0.01%, as long as there are no excess higher-order distortion artefacts present.

2.2 Noise Target and Gain

There is a wide variety of sensitivities in today's headphone market. The sensitivity of a headphone defines the usable output voltage range of the headphone amplifier in practical use. Less sensitive headphones need more voltage to be fully driven. The headphone amplifier also has to provide more voltage gain. Sensitive headphones need less voltage to reach their maximum SPL-level, leaving some of the voltage headroom of the headphone amplifier unused. In this case, the noise floor of the amplifier can become a problem if the noise voltage is moved into the hearing range. Many devices solve the noise problem by providing different gain settings, as the equivalent input noise of many designs is fixed and amplified by the gain. For the *Full Score one* we chose not to implement a gain switch in favour of the user's convenience. Therefore, the voltage noise of the main amplifier has to be extremely low while still providing enough voltage gain for the most insensitive headphones. With just 0.9 $\mathrm{nV}/\sqrt{Hz}$ the main amplifier is almost at the physical limit of voltage noise at room temperature. In order to still be able to drive the most insensitive headphones, we had to use 10 dB voltage gain, also amplifying the input noise of the main amplifier by 10 dB. Still, the noise stays below the practical hearing threshold, even with high-sensitivity headphones. The Austrian Audio *The Composer* can be considered rather high-sensitivity. In this example the noise stays below 9 dB SPL(A), which is almost always masked by the environmental noise. However, the decision against a gain switch also has another implication. For the volume control to still be accurate in the low regions with high-sensitivity headphones, we had to custom-order a special taper curve of the ALPS RK271 series potentiometer, which starts with a much shallower slope and gets very steep in the end.

2.3 Frequency Response Target

To eliminate subsonic frequencies that only lead to intermodulation of the drivers, it is wise to implement a roll-off below at least 5 Hz. A reasonable high frequency roll-off is a matter of stability of the circuit. If there is a mismatch between the slew rate and the rise time of an amplifier (as is the case with many audio ICs), one must fear HF interference and other odd behaviour whenever the bandwidth is not limited in a meaningful way. The Austrian Audio *Full Score one* instead has a much higher slew rate than is needed for the available bandwidth. Therefore the high frequency roll-off can fall together with the main amplifier's edge frequency, which is far outside of the audio band.

2.4 Channel Separation Target

The industry standard value of at least 60 dB channel separation should be a good starting point. 6.35 mm jacks are commonly unable to provide this value with low impedance headphones (< 32 Ω), and can even fall below 40 dB. This is caused by the shared ground connection of the left and right channel within the jack. For headphone impedances < 32 Ω we recommend using the 4-pole XLR instead, since it has separate ground returns for a channel separation above 90 dB.

2.5 Rise Time Target

The rise time targets were defined in accordance with our listening tests. The *Full Score one* shall have a rise time below 650 ns, TTT enabled. Since the last listening test was inconclusive, we went with our personal preamplifier choice and the artificially prolonged rise time of 6 µs when TTT is off.

2.6 Phase Margin Target

The phase margin of a headphone amplifier cannot be high enough and should be at least 90° in the worst-case scenario. With more than 90° phase margin, no first order inductive or capacitive load can lead to the instability of the global negative feedback loop.

2.7 DC Balance Target

The DC balance of a headphone amplifier is measured at its outputs with reference to the headphone ground. The *Full Score one* uses a DC servo circuit, which constantly monitors and nulls the DC offset of the main amplifier. The offset value shall be kept below 1 mV, in order to avoid a constant shift of the drivers from their mechanical centre position. In-ear headphones are particularly sensitive to DC offset voltages. This is the only time we use ICs in the *Full Score one*. The DC balance of the usually laser-trimmed integrated circuits cannot easily be achieved by amplifier designs made from discrete components. The relevant audio signal above the edge frequency of the DC servo is not affected by the ICs.

2.8 Input Impedance Target

The input impedance shall be high enough so that the *Full Score one* can easily be driven by a high impedance transformer-less tube amplifier without causing any additional distortion. In practice we chose the value of 100 kΩ. Please note that this rather high input impedance causes audible noise when the inputs are left unconnected and the volume is raised. This is normal. When the inputs are terminated by a signal source, the *Full Score one* performs as low-noise as can be expected.

2.9 Output Power Target

The industry has seen an increase in the average needed power to drive headphones, some even requiring more than 1 Vrms to get above 94 dB SPL at an impedance below 20 Ω. As the Austrian Audio *Full Score one* is a large desktop headphone amplifier, we can make use of the large cooling area and provide a future-proof power rating of > 1 W per channel.

2.10 Output Voltage Target

We regularly observe confusion between the power and voltage requirements of headphones. In reality it is a matter of efficiency. The impedance of a headphone is not necessarily connected to its efficiency. There are very efficient headphones with a high impedance, which therefore still need a higher voltage output from the headphone amplifier, but not a lot of power. There are also headphones which are very inefficient but still require a higher drive voltage in spite of a low impedance. Since the *Full Score one* has a gain of 10 dB (like many headphone amplifiers in the market at their high gain setting), we arrive at 3.16 Vrms output with our specified input voltage of 0 dBV (1 Vrms). If even more output voltage is needed, the *Full Score one* can be 'overdriven' by a signal source delivering more than 1 Vrms. The maximum non-clipped output voltage (about 9 Vrms) at full volume is only limited by the rather high supply voltage of the main amplifier. The preamplifier is not easily driven into clipping. It only acts as a 0 dB buffer in front of the volume attenuator and can handle input voltages up to 15 Vrms.

2.11 Output Impedance Target

In theory, the output impedance of a headphone amplifier should be as low as possible to keep the frequency response deviation of the connected headphones as low as possible. Also, a low output impedance in theory helps dampen down headphone resonances. In practice, the output impedance requirement for the amplifier is not as severe as one might think, since the impedance of the headphone cable is usually much higher. To rule out any implications before they even arise the output impedance target of the *Full Score one* is treated like it would be for power amplifiers. An output impedance below 0.1 Ω should guarantee the unaffected operation, even if a headphone has extreme impedance variations. The *Full Score one* output impedance measures below 0.02 Ω, directly acquired at the output terminals.

2.12 Power Supply Design Target

The *Full Score one* shall use a simple, conventional linear power supply. Many customers still prefer linear power supplies over switch mode units because of their lower repercussion noise artefacts back into the mains. Furthermore, linear power supplies usually have much superior durability. The power supply of the *Full Score one* shall have an automatic mains voltage selector between the two mains voltage regions, 100-120 V~/220-240V~. That way we can rule out any user error. Regulations for the off mode of consumer and household electronics require a maximum power usage of < 0.3 W. In order to meet this requirement, we were forced to implement a small switch mode power supply, which is only in use when the unit is powered off. The switch mode power supply gets disconnected when the *Full Score one* is powered on.

2.13 Summary

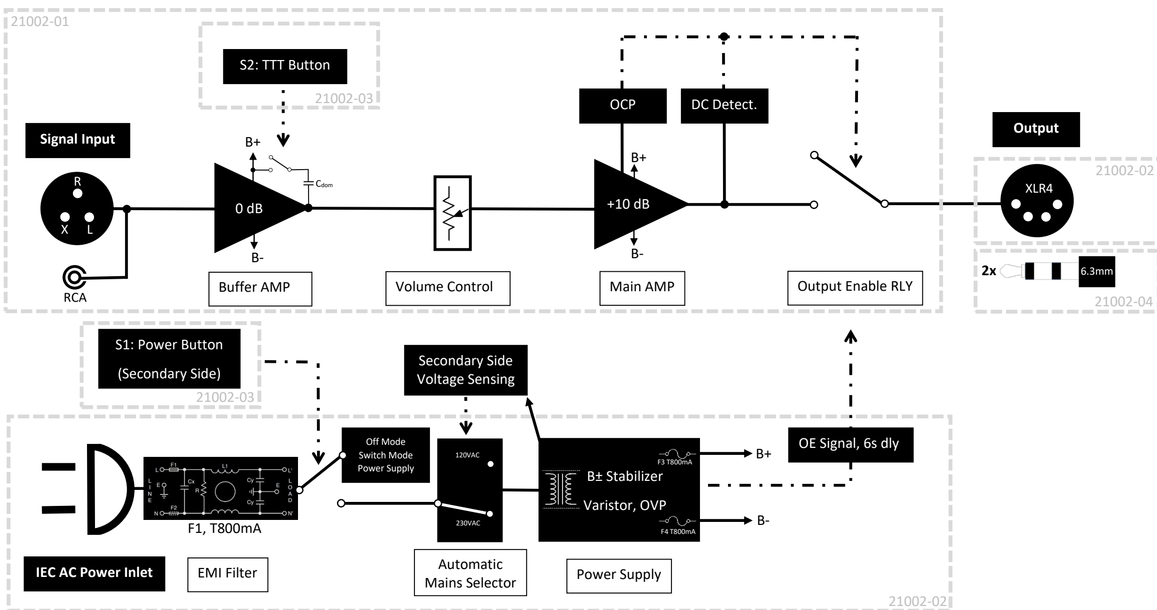
To summarize, the requirements for the amplifier circuits cannot be met in their entirety by off-the-shelf solutions using common amplifier ICs. We need to be able to precisely control every electrical parameter in order to achieve the combination of a reasonable phase margin and speed while still maintaining good immunity against any kind of electrical interferences. For this reason. we turned to discretely built wide-band amplifier circuits, carefully tailored by lead/lag compensation. The derived design targets are as follows:

	design target	actual design
[k]=%	$<$ 0.01, 20 Hz - 20 kHz	$<$ 0.002, 20 Hz-20 kHz, $<$ 0.0004 @ 1kHz
noise	lower than acoustic noise	0.9 nV/$\sqrt{Hz}$, 1.5 µVrms (A) with 10 dB gain
gain	the usual high gain setting	10 dB
channel separation	$>$ 60 dB	90 dB @ 1 kHz
rise time	$<$ 650 ns	200 ns, TTT active
phase margin	$>$ 90°	97°
DC balance	$<$ ±1 mV	$<$ ±0.6 mV
input impedance	$>$ 47 kΩ	100 kΩ
output power	$>$ 0.6 W @ 10 Ω	1 W @ 10 Ω , 1 Vrms input
output voltage	$>$ 1 Vrms	3.16 Vrms, 1 Vrms input
output impedance	$<$ 0.1 Ω	$<$ 0.02 Ω

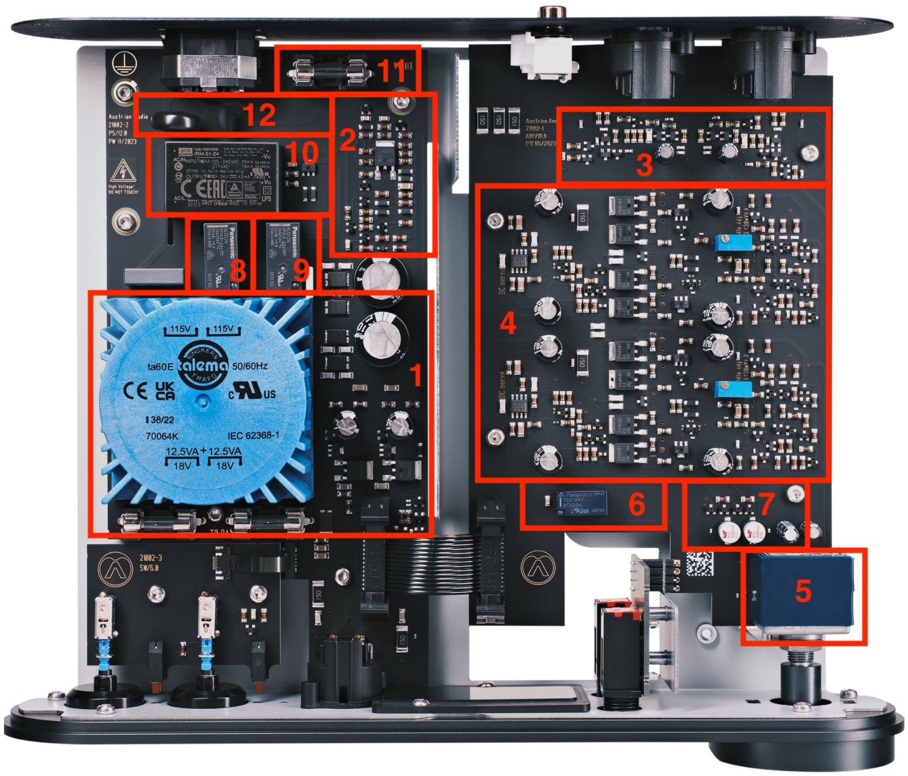
Table 2: Design targets at full load in comparison with the actual design

3 Circuit Description

3.1 Block Diagram



Picture 1: Austrian Audio *Full Score one* block diagram



Picture 2: Austrian Audio *Full Score one* inside view

1. Main, linear power supply
2. Automatic mains voltage selector circuit
3. Preamplifier, left and right channel
4. Main (power) amplifier, left and right channel
5. Volume attenuator
6. Output enable relay
7. Part of the DC offset protection circuit
8. Power-on relay
9. Mains voltage selector relay
10. Off-mode switch mode power supply
11. Primary fuse
12. Mains over-voltage protector (left side), inrush current limiter (on the right)

The Austrian Audio *Full Score one* is comprised of three main parts, a linear power supply (1) with an automatic mains voltage selector (2), a preamplifier (3) and main power amplifier (4), separated by a volume attenuator (5). The preamplifier is set to an amplification factor of 0 dB and responsible for the implementation of the *True Transient Technology* switch. The main amplifier provides a gain of 10 dB. All three output terminals, the two 6.35 mm jacks and the 4-pole XLR are connected in parallel to the main power amplifier.

3.2 Power Supply

The power supply of the *Full Score one* can be kept very simple. Both preamplifier and main amplifier use circuits with self-balancing working currents and voltage references so that there is no need for a stiff, regulated power supply voltage. The power supply rejection of the preamplifier is about 70 dB, as it is for many simple current feedback amplifiers, and the main amplifier has a power supply rejection of 100 dB in the audio frequency spectrum. This is why we chose a simple pass transistor arrangement with a capacitor multiplier in order to smooth out the last bit of mains-injected noise. This construct applies an additional 90 dB of ripple rejection above 100 Hz which leads to 170 dB of ripple rejection for the preamplifier output and 190 dB for the main amplifier. As this is not a voltage regulator with a fixed target but just a smoothing circuit which tracks the average rectified secondary voltage of the transformer, the ripple rejection always stays intact, even with very low mains voltage or during mains voltage dropouts.

3.3 Preamplifier and TTT-Function

For the preamplifier, we use a very simple and fast, two-stage current feedback amplifier. The first stage is the input stage with a cascode-connected JFET. The second stage is both output stage and gain stage at the same time. Using a JFET, the input impedance can be high, steady and low-capacitance. As this amplifier is only loaded by the 10 kΩ volume attenuator, there is no need for a dedicated output stage. We have a push-pull relation between the input transistor and the gain stage transistor, which is necessary for a short, symmetrical and constant rise time, and a high slew rate. Current feedback amplifiers are known to be excellent performers in the time domain but display rather average THD figures. Our *True Transient Technology* is realised already at the preamplifier stage in order for its effect to be independent from the volume attenuation - in other words, independent from the user's listening level or choice of headphones. When the TTT function is off, additional capacitors get patched into the circuit, limiting the rise time of the preamplifier. We can partially 'simulate' the effect of a slower headphone amplifier. Still, we have a constant rise time, not a constant or even non-linear slew rate. Higher THD can be measured, especially above 1 kHz, but not in the region we would consider perceivable. Second order effects (like the demodulation of out-band noise of DACs and additional intermodulation with self-inflicted out-of-band distortion artefacts) are also left out of the 'simulation'. In this way, we do not go all the way back to the performance in the time domain compared to many audio amplifier circuits. We get the following technical data for the preamplifier:

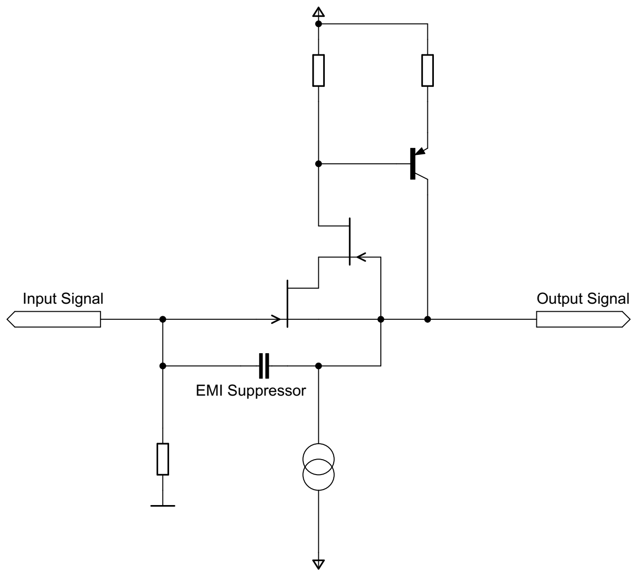


Fig. 1: Austrian Audio *Full Score one* preamplifier: Simplified schematic

@ 1 Vrms, 10 kΩ load	TTT = ON	TTT = OFF
[k]=% @ 1 KHz	0.0004	0.003
OLG	134 V/mA	134 V/mA
UGBW	0.3 MHz	0.02 MHz
bandwidth (-3 dB)	3.3 MHz	0.075 MHz
max. slew rate	2000 V/µs	5 V/µs
rise/fall time	20 ns	6 µs
phase margin	175 °	120 °
noise density	3.5 nV/$\sqrt{Hz}$	3.5 nV/$\sqrt{Hz}$

Table 3: Technical data of the *Full Score one* preamplifier

The preamplifier is lead/lag-compensated in order to retain a good balance between speed, phase margin and distortion, and a more than reasonable roll-off at 3.3 MHz.

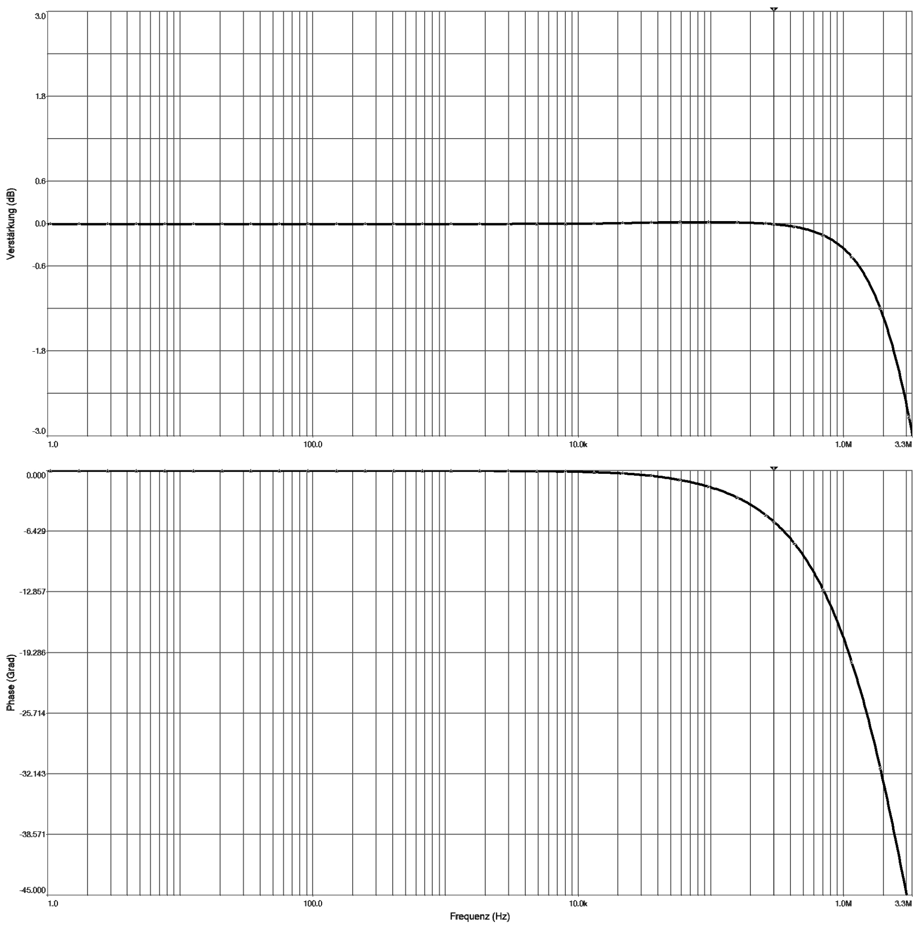


Fig. 2: Austrian Audio *Full Score one*: Preamplifier gain vs phase shift, - 3dB @ 3.3 MHz

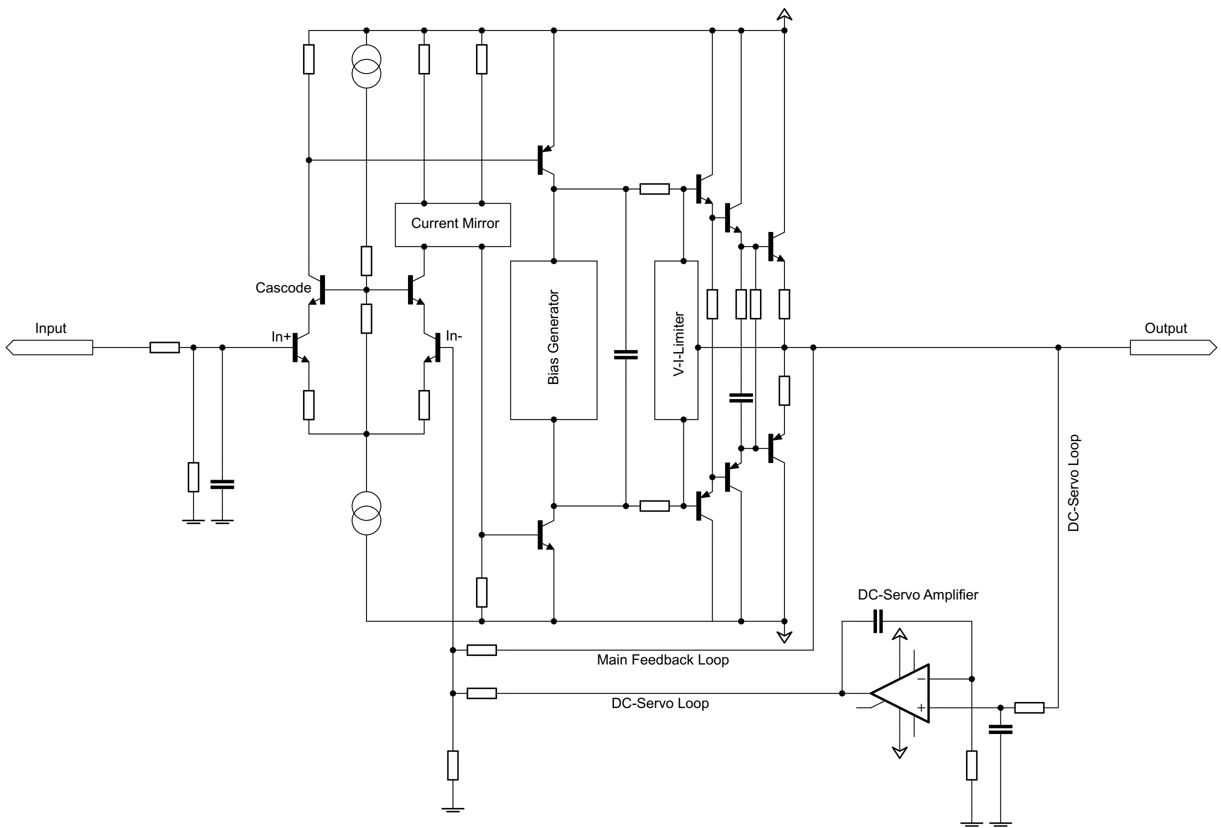


Fig. 3: Austrian Audio *Full Score one* main amplifier: Simplified schematic

3.4 Main Amplifier

The main amplifier of the Austrian Audio *Full Score one* is a wide-band voltage feedback amplifier. It is configured like a small power amplifier in order to handle all common dynamic headphones known to date, like high impedance headphones above 300 Ω which require more voltage, low impedance and high sensitivity headphones below 32 Ω like in-ear headphones, or even planar magnet drivers combining both low sensitivity and low impedance.

@ 3 Vrms, 10 Ω load	
[k]=% @ 1 KHz	0.00017
OLG	65 dB
UGBW	8 MHz
bandwidth (-3 dB)	2 MHz
max. slew rate	300 V/µs
rise/fall Time	200 ns
phase margin	97 °
noise density	0.9 nV/$\sqrt{Hz}$

Table 4: Technical data of the *Full Score one* main amplifier

3.4.1 Input Stage

The input stage is formed by a parallel differential amplifier, which has the following advantages:

- Very good linearity
- Very good DC balance
- All semiconductors can be well-matched since the two 'arms' of a parallel differential amplifier use components of the same polarity (as opposed to serial differential amplifiers).
- Low noise
- A powerful lead network can easily be implemented to increase the phase margin of the feedback loop.

There is one major disadvantage:

- The fixed current source of the common emitter limits the maximum current flowing through each tail of the differential amplifier, leading to a limited slew rate as opposed to a fixed rise time. To retain a constant rise time for the main amplifier as a whole, we had to maximise the standing current of the input stage and to decrease the gain by local feedback. When the gain of the differential input amplifier and the common mode voltage are low enough in comparison to the gain stage, the rise time is dominated by the gain stage instead. In this way, we can still use a parallel differential amplifier in a constant rise time design.

3.4.2 Gain Stage

For the gain stage we implemented a fully complementary push-pull circuit. The current of the parallel differential input stage is mirrored in order to create a level-shifted complementary signal for the negative (or pull) side of the gain stage. Push-pull gain stages have the advantage of a constant rise time, since the available current is proportional to the signal level (the higher the voltage swing, the higher the current). But the designer of the amplifier has to take special care because the amplification factor of a push-pull gain stage also varies with the non-fixed current which could lead to a runaway situation and to the instability of the feedback loop with larger signals. Furthermore these types of gain stages are more susceptible to the non-linear load by the output stage. Thankfully, all above-mentioned challenges can be met. If some tricks are applied to

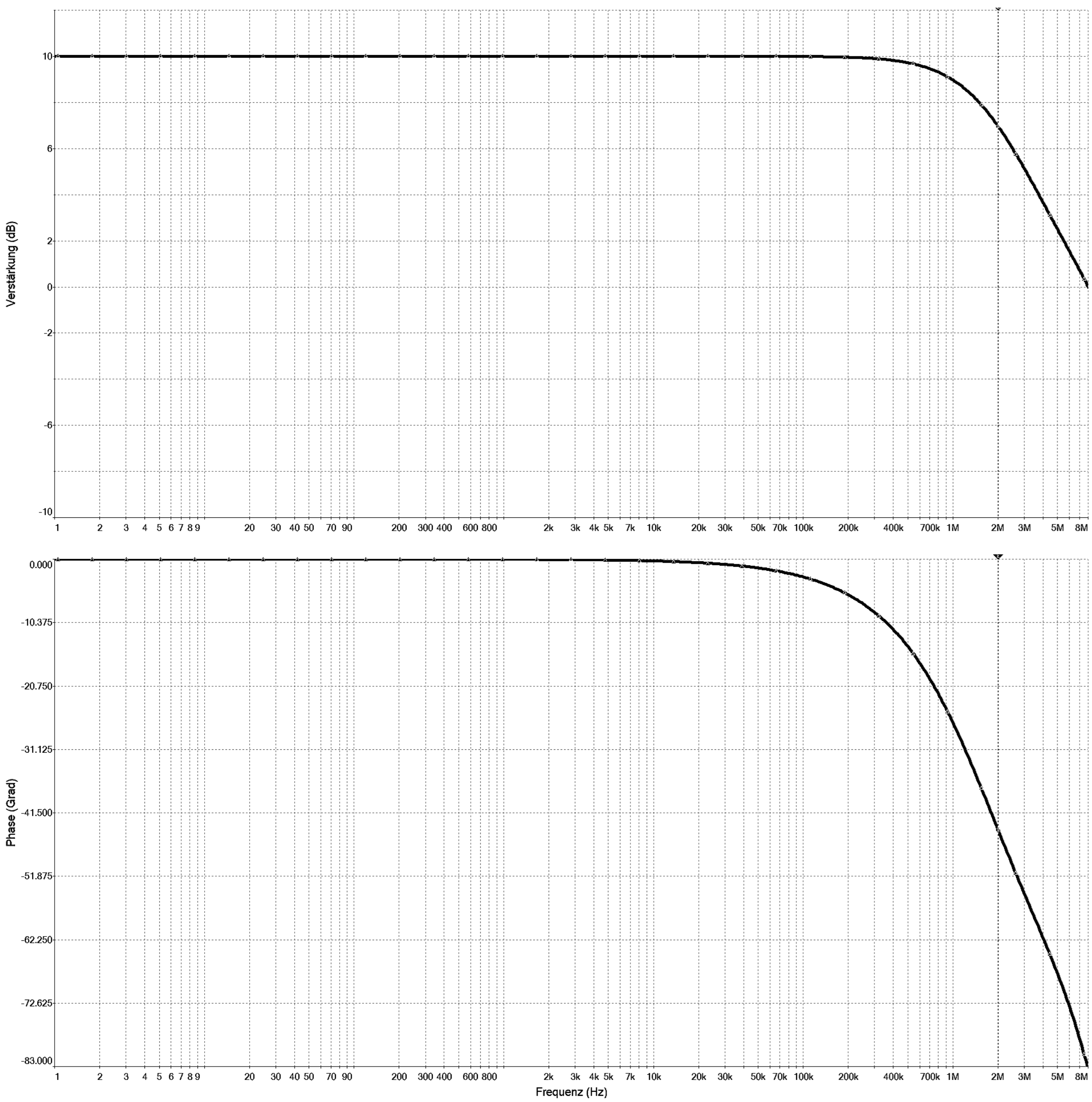


Fig. 4: Austrian Audio *Full Score one*: Main amplifier gain vs phase shift, - 3dB @ 2 MHz

keep the voltage gain under control and the currents flowing into the output stage low a push-pull gain stage can be very stable and offer very low distortion.

3.4.3 Class-B Output Stage and Biasing

The Austrian Audio *Full Score one* uses a complementary, triple emitter follower Class-B output stage, composed of bipolar transistors. The advantage of a triple configuration instead of a Darlington pair lies in the much higher current gain. This is important in combination with push-pull gain stages, which need to be relieved from the load-dependent currents flowing in the output stage. Concerning output stage biasing, there seems to be some confusion (even in the literature on the subject) about the definition of Class-B and Class-AB. Class-AB is broadly referred to an output stage that is biased into Class-A for smaller signals, drifting into Class-B whenever the Class-A idle current is exceeded. When it comes to pure Class-B, there are some definitions stating that there is no idle current in the output stage at all, which would indeed be a disastrous crossover distortion generator. An output stage lacking any idle current should (in our opinion) not be called any class. A real Class-B output stage is biased just enough to have the least distortion in the crossover region without already touching Class-AB currents. One could also use the term 'optimally biased'. For further reading we suggest the paper of Dr. Bernard Oliver, *Distortion in Complementary-Pair Class-B Amplifiers* or the chapter 17 of Douglas Self's *Audio Power Amplifier Design*, p. 446. The consensus amongst amplifier designers is to either go full Class-A with the drawback of massive power consumption at idle or to have optimal Class-B biasing with reasonable idle power-draw and slightly more distortion. Class-AB biasing usually leads to more distortion than Class-B, as two more crossover regions from Class-A to Class-B are introduced into the output signal whenever the Class-A current is exceeded. This is why we chose a Class-B output stage for the Austrian Audio *Full Score one*.

3.4.4 DC Servo

We implemented a DC servo circuit that performs a constant DC offset correction of the main amplifier. The circuit forms a second control loop around the global negative feedback loop. Only the 4 Hz low-pass filtered signal of the main amplifier is processed by a very low offset amplifier IC, which compares its input to signal ground, thereby setting the low frequency roll-off.

3.4.5 Protection

Most headphone amplifiers face daily short circuits at their outputs, simply by plugging and unplugging the headphones while music is playing. The reason for this are the 6.35 mm jacks which form a connection between the left and the right output channel in the process of connecting or disconnecting the headphones. In the past, there were many headphone amplifiers simply using power resistors in series to protect their outputs. As with many other more modern designs we tried instead to keep the output impedance as low as possible which in turn makes protecting the amplifier much more difficult. A fast, low impedance amplifier also needs a fast protection. The Austrian Audio *Full Score one* has a two-stage approach. The first stage is a fast-acting V-I-limiter which has to be even faster than the main amplifier's regulation to keep the output transistors within their safe operating area. The second stage is a circuit that monitors the average power dissipation of the output transistors and finally cuts the output relay entirely if the overload situation persists. Our headphone amplifier also has over-temperature and over-voltage protection. Last but not least, there is the DC fault protection, which disables the output relay of the *Full Score one*, protecting the headphones if a DC voltage is present at the output of the main amplifier.

4 Measurements

4.1 Measurement Setup

The following measurements were done with the Audio Precision APx525 measurement system. We used the asymmetrical BNC outputs of the signal generator and the symmetrical input of the analyser to break the ground loop. All those familiar with the APx525 and 555 systems know that it is not always possible to avoid mains-leakage induced hum into the APx analyser when the device under test is mains-powered by a toroidal transformer. The grounding scheme of the Austrian Audio *Full Score one* is optimised for a floating load like a headphone, in favor of better immunity against ground loops at the input. One can lower the impact of the mains induced leakage a little bit by connecting the chassis ground jack of the APx directly to the ground of the RCA input connectors of the *Full Score one*. The measured hum is not present in real life with headphones.

4.2 Measurement Results

4.2.1 Dashboards

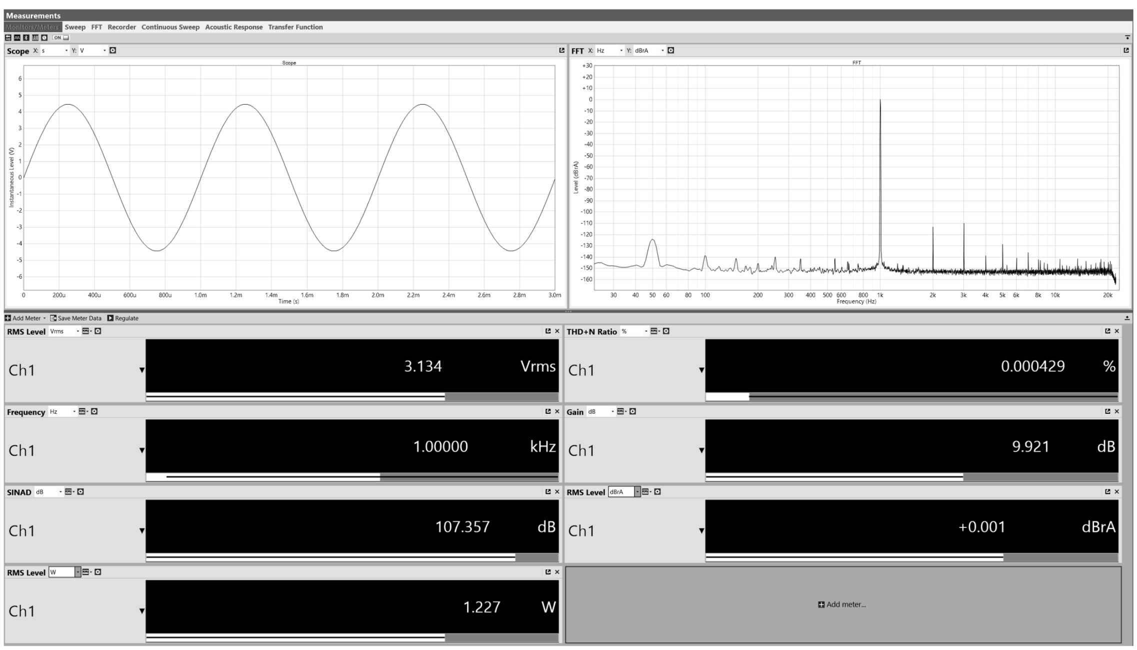


Figure 5: Dashboard inspired by the *Audio Science Review* forum
20 kHz bandwidth | 1 Vrms input | both channels loaded with 10 Ω | full specified power

We can observe the mains leakage spikes, beginning with the largest peak at the nominal mains frequency of 50 Hz and the expected overtones. The *Full Score one* exhibits dominant second and third order distortion in this full-load scenario, although it is still far below the limits of audibility.

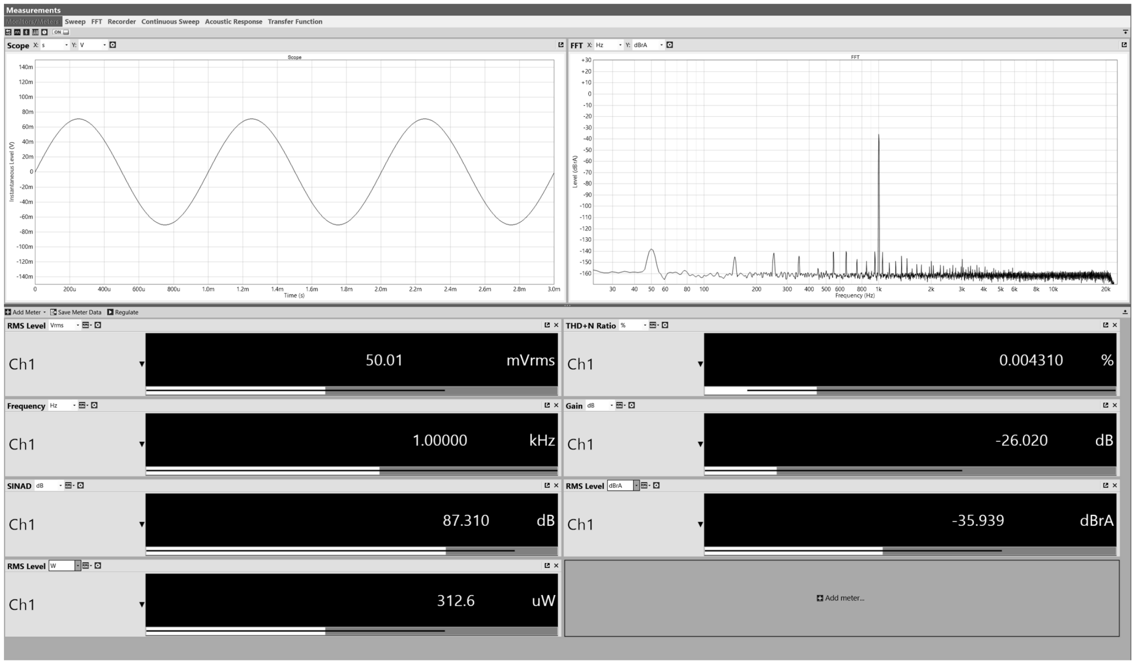


Figure 6: Dashboard inspired by the *Audio Science Review* forum
20 kHz bandwidth | 1 Vrms input | both channels loaded with 10 Ω | 50 mVrms output

The second dashboard, measured at 50 mVrms, shows the same mains leakage artefacts, which can be displayed in more detail, as both the noise floor of the APx signal generator and the preamplifier noise of the *Full Score one* get attenuated by the volume potentiometer. Both dashboards show the absence of output stage crossover artefacts.

4.2.2 THD vs Frequency

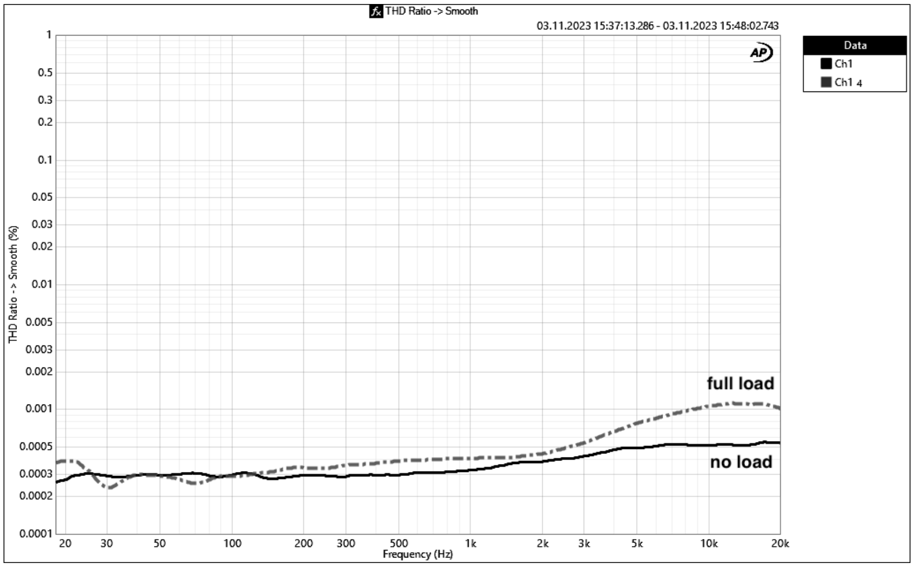


Figure 7: Austrian Audio *Full Score one* | THD vs frequency | 90 kHz bandwidth | solid line: no load | dashed line: full load (10 Ω, 1.2 W)

The THD measurement was done at both extremes of the load spectrum to illustrate the load dependency of the *Full Score one*. We leave it up to the reader's imagination to draw the intermediate graphs either when the load impedance is raised or when the output voltage is lowered, thus reducing the power output.

4.2.3 Frequency Response

Please note that the following extended frequency response measurements were done using the stepped sweep function of an oscilloscope. The phase value not only includes the cascade of both amplifiers, the preamplifier, and the main amplifier, but also the EMI filtering circuits in the signal path, which introduce additional phase lag compared to fig. 4 and fig. 2.

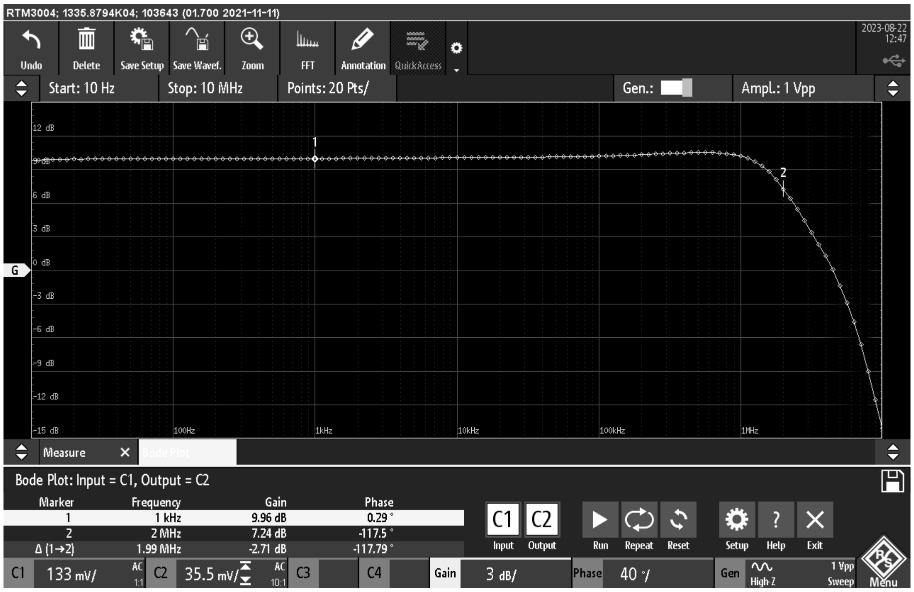


Figure 8: Austrian Audio *Full Score one* | frequency response | *TTT* active

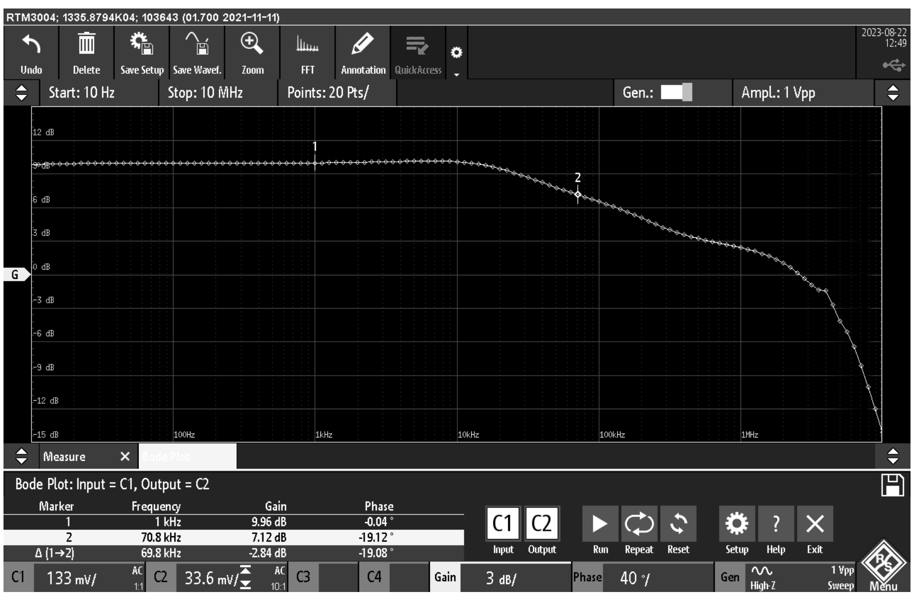


Figure 9: Austrian Audio *Full Score one* | frequency response | *TTT* disabled

5 Technical Data

The following data sheet was recorded at 230 V~/50Hz mains voltage, 25 °C room temperature.

Parameter	Value
min. input impedance, 4 Hz - 20 kHz	100 kΩ, effective 9 pF
max. output voltage, 4 Hz - 20 kHz	19 dBV, 9 Vrms, < 0.1 % THD
recommended max. load, 4 Hz - 20 kHz	10 Ω - 600 Ω
THD + N, 4 Hz - 20 kHz	< 0.001 % @ 10 Ω, measuring bandwidth = 90 kHz
THD + N, 1 kHz	< 0.0004 % @ 10 Ω, 1.0 W, measuring bandwidth = 22 kHz***
SINAD, 22 kHz bandwidth	87 dB @ 50 mVrms /107 dB @ 3.12 Vrms ***
amplification, no load	9.96 dB
volume attenuation	-100 dB***
power bandwidth	5 Hz - 2 MHz, *TTT* active
max. slew rate	300 V/µs***
rise time	0.2 µs***
damping factor, 4 Hz-20 kHz, XLR 4	200 @ 8 Ω
self noise, no input, volume at min.	EIN = 136 dB(A) / 1.5 µVrms (A)

Table 5: Technical data of the *Full Score one*

***This is the average value. Other publications of Austrian Audio show different values compensating for possible dispersion of the series production.

6 Further Reading

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- Self, D.: Audio Power Amplifier Design (Sixth Edition), *Focal Press*, 2013