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All amplifiers are analogue, but some amplifiers are more analogue than others

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ABSTRACT

This paper intends to clarify the terms “digital” and “analogue” as applied to class-D audio power amplifiers. Since loudspeaker terminals require an analogue voltage, an audio power amplifier must have an analogue output. If its input is digital, digital-to-analogue conversion is necessarily executed at some point. Once a designer acknowledges the analogue output properties of a class-D power stage, amplifier quality can improve. The incorrect assumption that some amplifiers are supposedly digital causes many designers to come up with complicated patches to ordinary analogue phenomena such as timing distortion or supply rejection. This irrational approach blocks the way to a rich world of well-established analogue techniques to avoid and correct many of these problems and realize otherwise unattainable characteristics such as excellent THD+N and extremely low output impedance throughout the audio band.

1. INTRODUCTION

Digital is any form of information that can be represented as a string of discrete symbols. Since any such message can be coded using only two symbols it is common to do so.

Analogue means a proportional representation of one physical quantity using another physical quantity, for example air pressure represented as voltage. A transducer performs the actual conversion. The

meaning of the word analogue is generally extended to apply to the original physical quantities as well, so the air pressure itself is also called analogue. In other words, physical quantities such as voltage, current, position, pressure and time are all analogue.

Digital information can be transmitted or stored completely error-free using an analogue signal or analogue state respectively, by agreeing on a coding method. In fact, digital information is always

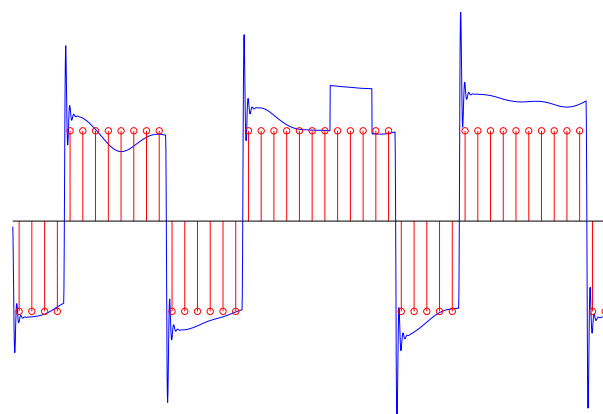
represented using some physical quantity. Information itself is abstract, time-less, space-less, ethereal and fairly useless if not physically available somehow. Degradation of the analogue transmission will not affect the transmitted information as long as the individual symbols can still be distinguished. An analogue signal may contain a digital message, but at the same time this signal has analogue features such as amplitude, spectral density, a noise floor and offset, to mention just a few. This reality is the cause of much confusion.

2. SHAPE VERSUS MEANING

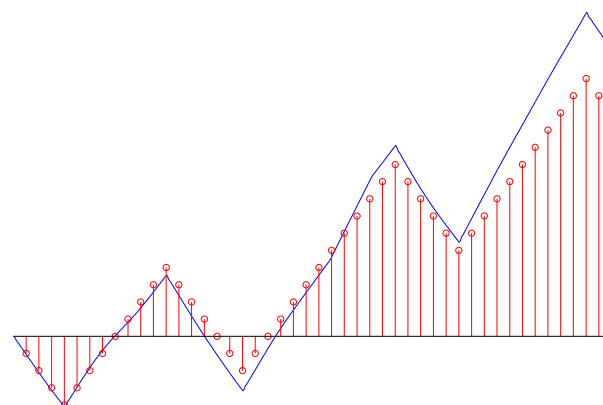
What makes a signal digital is whether the recipient interprets it as such i.e. if there has been an agreed method of coding symbols into it, used by both ends. In a “digital signal”, analogue is the form, digital the content. A square wave can be digital or analogue, a continuous-time signal can be digital or analogue too. It all depends on what happens downstream. Numbers can’t move a speaker cone any more than mere thoughts can. Because of this, there is nothing one can do to make an amplifier “more digital”. Or less. All amplifiers are analogue..... Think of a fax-machine receiving a coded message while you listen in on the phone. You get a nice picture and an ugly sound based on the exact same signal.

The distinction may seem like splitting hairs, but in the context of class-D amplifiers the consequences are rather more than philosophical. Since power stages and the necessary *LC*-output-filter deal with currents, voltages and time, they are profoundly analogue. Figure 1(a) shows a series of data bits and one valid way of representing them electrically. Figure 1(b) shows the same signal integrated by two different integrators. One is digital. It reads the data embedded in the electrical signal and makes a running sum. The other is analogue, and integrates the analogue waveform over time. The outcomes are dramatically different, because the digital integrator takes the signal for what it means, whereas the analogue integrator takes it for what it looks like, regardless of any meaning.

Like the analogue integrator, loudspeakers are sensitive to all these analogue properties. It does not have a clue about the fact that its input signal might be generated by some digital PCM-to-PWM converter and then enlarged in amplitude by a power-stage.



(a) Analog PWM output $u(t)$, re-sampled $x(k)$.



(b) Analogue: $\int u(t)dt$. Digital: $\sum x(k)$.

Fig. 1: An analogue signal (blue) and its one-bit digital interpretation (red), differences are exaggerated for clarity.

Nevertheless, most class-D practitioners actively avoid certain analogue techniques believing that these compromise the “digital essence” of their circuit. The drawbacks of this approach lie in the analogue errors of the switching power stage. The switch timing is heavily current-dependent, causing outright distortion. The power supply voltage fluctuates, amplitude-modulating the audio signal. The output filter is non-linear and has an undefined frequency response in loads other than one specific resistive impedance. Typically, error control (feedback) is considered “very analogue” whereas the use of a heavily regulated power supply is not. Hav-

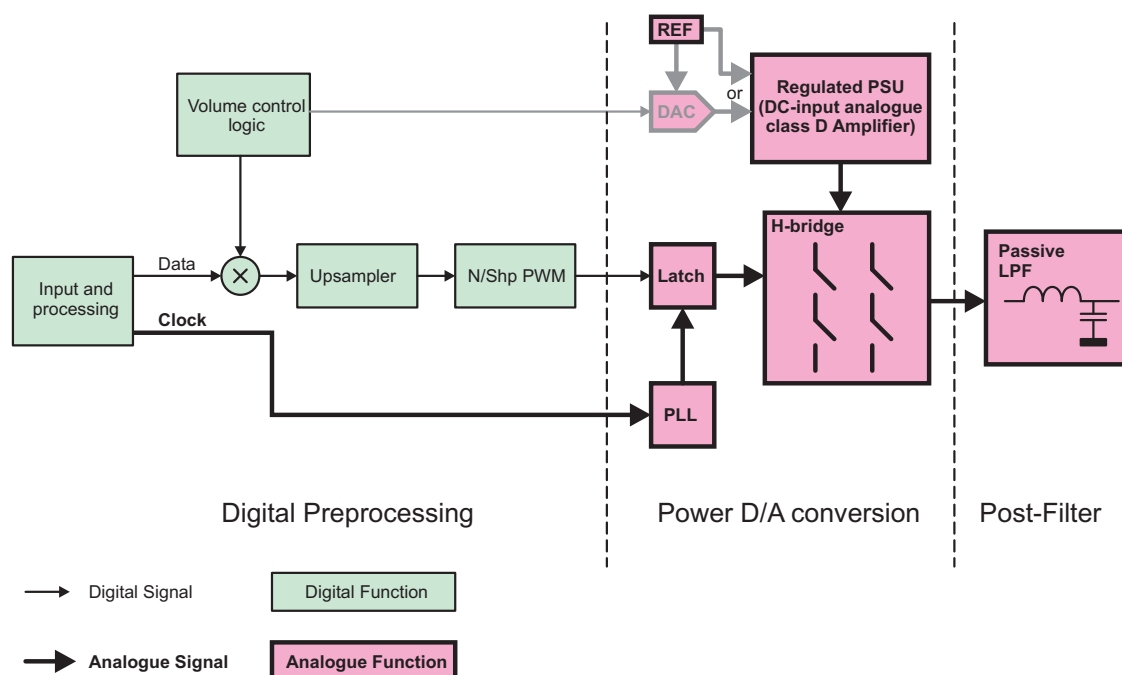


Fig. 2: Open-loop PWM amplifier

ing ruled out feedback as too obviously analogue, many designers go on to contrive solutions for some of these problems individually, while simply accepting others as “a fact of life” and talking down their importance. Indeed, output impedance simply cannot be made reasonably low without using feedback.

2.1. Real life

The archetypical “digital amplifier” of figure 2 derives a pulse width modulated signal from an incoming pulse-code modulated signal. The meaning of the PCM signal is a series of numbers ranging from -1.00 to +1.00. The PCM signal is first interpolated, typically eight-fold for CD audio data. This data is subsequently word-length reduced to typically 8 bits through noise shaping. This 8 bit word is then translated into a string of ones followed by a string of zeros, for a total of 256 positions. This conversion process is entirely digital. Numbers go in, numbers come out. For every 16-bit (or 24-bit) number that goes in, 2048 single-bit numbers come out.

It was quickly understood [2] that this conversion introduced a significant non-linearity because the signal-variant position of the 1/0 transition falsified

the z transform on which the PCM data and the noise shaping process relied. Usually a forward error correction is applied during the noise shaping stage to preempt this. Only much later was it realized that this distortion mechanism is academical compared to what the intended power D/A conversion stage does to the signal next.

The output of this PCM-PWM process is a string of 1-bit numbers. Numbers don’t drive speakers (voltages and currents do) so they now have to be converted to a time-varying voltage. A first crucial step is to clock out the data as a square wave voltage synchronized to a low-jitter clock. The quality of the clock is important, because the correspondence between z and s transform (as defined by frequency) is only as accurate as the clock used to convert numbers into an analogue representation. The clock is thus the bridge between timeless numbers and a timed physical representation. The second step brings a voltage reference into the equation. Like the PCM input data, the 1-bit output data has a quantity between -1 and +1 as its meaning. This dimensionless number is converted into a voltage by multiplying it with a reference voltage.

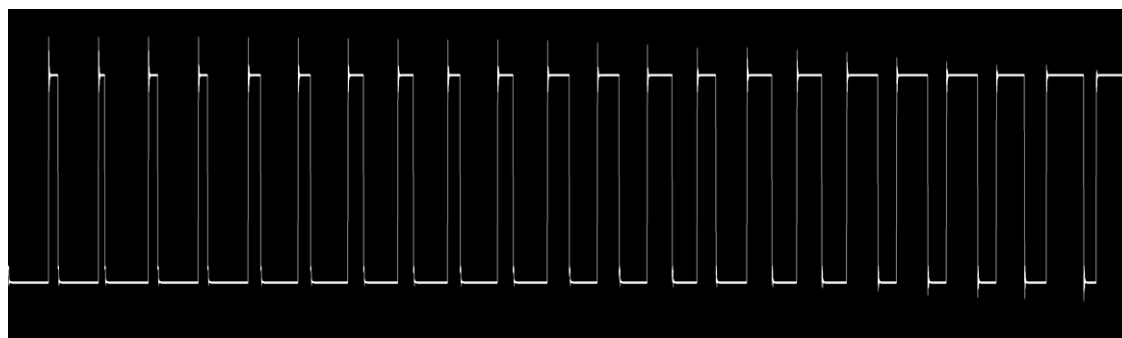


Fig. 3: Analogue bridge output voltage.

The multiplication is performed using an H bridge (4 power MOSFETs) running between ground and a regulated power rail. The latter is the voltage reference, and is usually a switching power converter itself. Interestingly, this power converter effectively constitutes an analogue controlled class D power amplifier in its own right, driven by a DC input, as described in [1].

Figure 3 is taken from an amplifier constructed along these lines. The pulse repetition rate is 705.6 kHz ($16f_s$). Rising edges occur at fixed 16 bit intervals, the falling edge is variable in steps of 88.6 ns ($1/256f_s$). The waveform is good enough to extract the original digital data, and store it as a string of 1-bit numbers. This string can be analyzed using an FFT program. The result is shown in figure 4(a).

Note that the frequency scale of an FFT done on a string of numbers is a dimensionless number ranging from 0 to 0.5. Recalibration of the scale to “real” frequencies requires knowledge that is not part of the numbers itself (the sampling rate). Numbers don’t have frequency, in the same way that they don’t have time.

The digital spectrum in figure 4(a) reveals two test tones sitting in the middle of a narrow range that is otherwise astoundingly devoid of any other components. Surely an analogue representation of this signal, suitably low-pass filtered, must be the cleanest two-tone test signal ever seen?

Close inspection of the analogue waveform in figure 3 reveals several non-idealities. For one, the edges are not located exactly at 88.6 ns intervals. Depending on the pulse widths, deviations of up

to 10 ns are seen. The waveform also shows overshoot and ringing, sometimes concentrated at the rising edges, sometimes at the falling edges. Both effects are caused by the current out of the output stage. Although superficially the voltage waveform looks like a “digital signal,” the current waveform does not. Figure 5 shows the output current of the power stage. The current waveform is in fact a pair of sinusoids with a high-frequency triangle superimposed on it. It stands to reason that the myth of the “digital amplifier” would have never taken off if it had been as common to show current waveforms as it is to view voltage waveforms. Depending on the orientation and amplitude of the current, the actual transition timing will hog the turn-off of one FET or the turn-on of the other. In the latter case, dV/dt will be steeper, resulting in more pronounced overshoot and ringing on this edge.

Although hardly visible in the voltage plot, the amplitude of the pulses isn’t perfectly stable either. The load current is effectively modulating the voltage which is supposed to act as the D/A conversion voltage reference.

So far, we’ve seen that the signal found on the power stage of this amplifier is a useful representation of a string of 1-bit numbers. In fact, timing as tight as 10 ns is not necessary to keep the eye pattern open, and neither is a stabilized power supply. We’ve shown that interpreting the data as representing digital information does produce a valid outcome.

Unfortunately, an LC lowpass filter doesn’t take numbers as its input. It takes a time-varying voltage and integrates it, warts and all. We must therefore take an analogue spectrum of the signal coming from

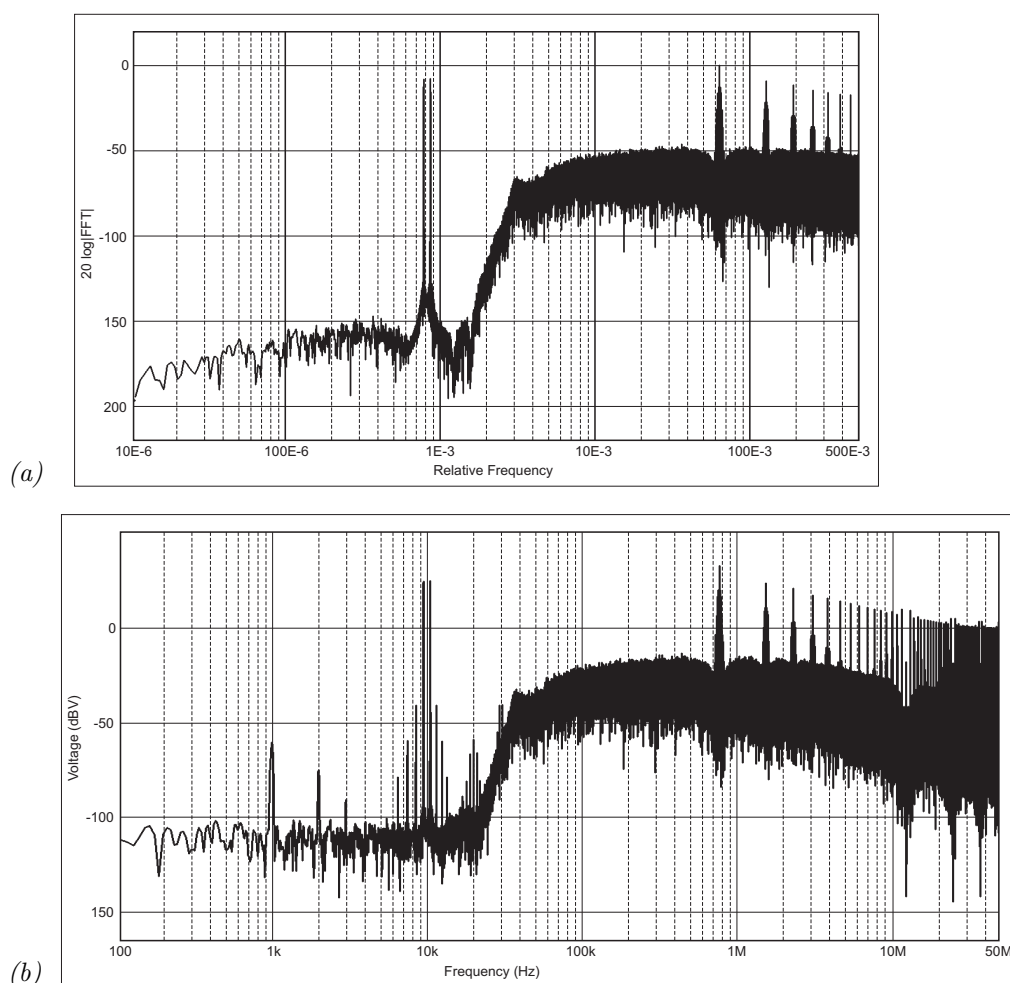


Fig. 4: (a) Spectrum of digital numbers (frequency is relative to f_s). (b) Spectrum of analogue output signal (frequency is absolute in Hz).

the power stage, and see the signal the way the output filter sees it. Figure 4(b) shows the outcome. The two test tones are still there (now physically at 9.5kHz and 10.5kHz) but the differences are more marked than the commonalities. For one, the spectrum extends up to infinity. In terms of spectral content in the band of interest, the test tones are joined by a sprinkling of intermodulation spurious and the relative noise level has increased significantly.

One is reminded that this is the exact same signal (figure 3) that the spectrum of figure 4(a) is taken from. The only difference is that this time it isn't interpreted. It's shown in all its analogue glory. This

is how the output filter sees it, and this is how it comes out of the loudspeakers.

By now it should be clear that the signal at the output filter is not digital. The signal ceases being digital once it is referred to a time base. Even at the time of writing, amplifiers according to this design are still touted as “more pure, because the D/A conversion stage is avoided”. This is not quite the case, unfortunately. A simple, high-quality small-signal D/A converter is substituted for by a mediocre discrete design using power FETs.

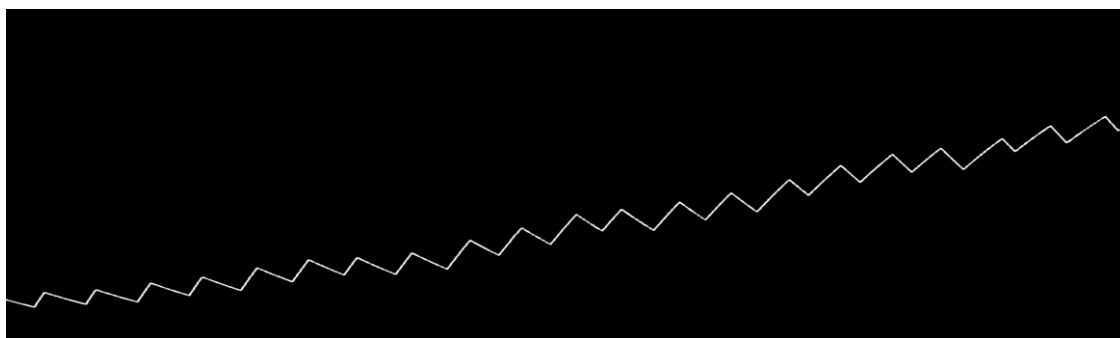


Fig. 5: Analogue bridge output current.

3. THE HANDICAP GAME

In sports, a common occurrence is playing with a handicap. It allows less experienced players to play against more experienced players by making the game more difficult to play for the more experienced player, thus equalising the chances of winning. Adding ballast to previous winners' cars in motor-car races is one example. One gets the impression that designers of open-loop digital amplifiers are playing with a voluntary handicap. The linearity problems confronting power stage designers are all very analogue in nature. Yet, many equipment designers expressly specify that no error control be used in addressing these problems. The motivations for such a specification are varied, but mainly boil down to the following four:

- Belief 1: that error control contravenes certain unstated psychoacoustic principles ("negative feedback sounds bad").
- Belief 2: that a switching power stage is digital and that adding error control pollutes the digital essence of the design.
- Belief 3: that adding error control makes the design more complicated.
- Belief 4: that direct digital drives down cost.

Discussing unstated psychoacoustic principles is outside the scope of this (or any) paper while essential concepts are firmly in the domain of neurology and anthropology. The latter two beliefs on the other hand are subject to this discussion.

If system design is simplified by an all-digital design tactic, this is not borne out by the work published on this subject. In keeping with the sports analogy, it has taken the best hands in the industry to make power stages delivering THD performance comparable to the simplest closed-loop designs based on much more rudimentary power circuits, while still remaining unable to control the voltage appearing at the speaker terminals.

However, efficiency analysis gives a much more compelling argument in terms of system complexity.

3.1. Degrees of Freedom

The audio system is best viewed as two chains. The signal chain goes from disc to speaker. The power chain goes from wall plug to speaker. More and more, the power chain consists entirely of switching power converters. Each power converter adds one degree of freedom to the system. In total, at least as many degrees of freedom are needed as there are variables to be controlled. An audio amplifier with a power-factor corrected SMPS has two controlled variables, namely the mains current and the loud-speaker voltage.

When the power amplifier has full control over its duty cycle, it will be capable of converting any power supply voltage to any output voltage, provided that the former is greater than the latter. This allows the power supply to use its degree of freedom (also duty cycle) to control the shape of the input current while maintaining control over the output voltage only over the long run. Single-stage power factor corrected (PFC) input isolated supplies are a proven technology, showing that indeed a two-stage

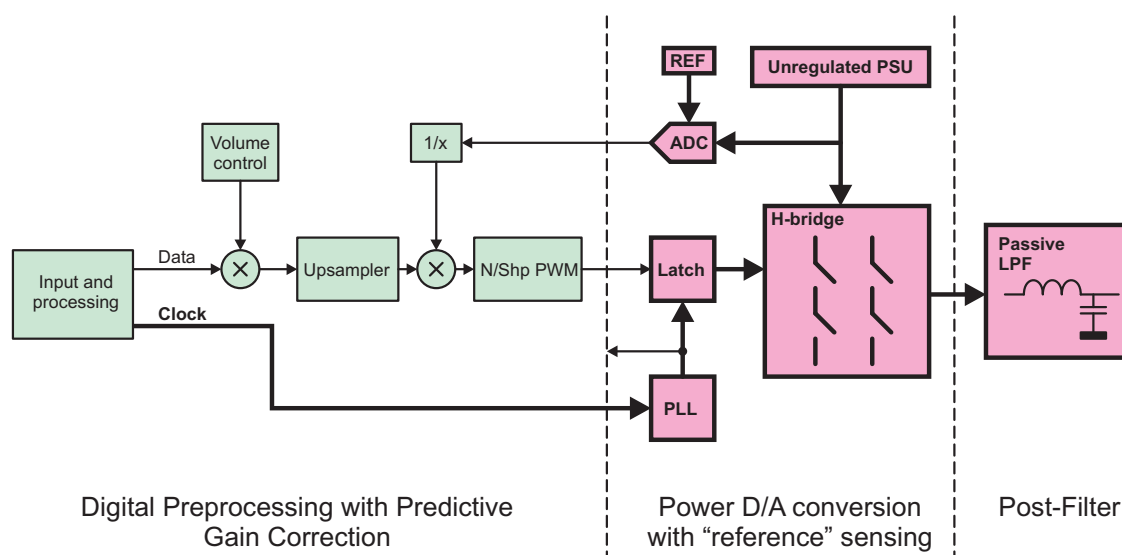


Fig. 6: Power supply feed forward scheme.

approach is sufficient to control both speaker voltage and mains current shape at once.

If the input current is not an issue, single-stage power amplifiers are feasible, e.g. using phase shift synchronous demodulation circuits. The usefulness of such amplifiers is limited by the fact that adding PFC requires adding a boost input stage, bringing the total number of stages back to two. Given the complexity of the synchronous demodulation stage, this topology is warranted only when no PFC functionality is planned.

In an “open loop digital” amplifier (figure 2), the final stage has zero degrees of freedom. The duty cycle is imposed externally by a circuit that has no knowledge of the state variables of the power circuit. What is controlled is the duty cycle, not the output voltage. This necessarily adds a second power conversion stage to the process in order to insure a stable voltage. If PFC is required, a minimum of three conversion stages is needed.

The unavoidable conclusion of this analysis is that requiring open-loop digital control necessarily adds a power conversion stage and therefore adversely affects system efficiency. All other requirements equal, digital open-loop control increases power losses and drives up cost and complexity.

4. SOME INTERESTING AVENUES

Many designs have been published and commercialised that attempt to address some of the practical (analogue) problems encountered while retaining the immaterial thing called “digital essence”. Many ideas have been developed by several people, with minor adaptations. Some of the more interesting ones will now be discussed.

4.1. Power Supply Feedforward

The most obvious cause of errors in switching power amplifiers is power supply ripple. One way to reduce sensitivity to power supply variations is to measure the power supply voltage using an ADC (see figure 6) and to scale the input to the noise shaper per the inverse of the measured supply voltage [3]. If the ADC is fast enough (it is usually run at sampling rates significantly greater than the audio sampling rates), acceptable reductions in power-supply induced modulation can be achieved. Well up to 30dB at 20kHz is possible. One disadvantage is that the signal-to-noise ratio of the converter directly impacts the system SNR.

Timing-wise the power stage remains in function as DAC. Voltage-wise, the D/A function is the inverted ADC, as indicated by the voltage reference that feeds it. Now that the voltage reference function has shifted from the power supply to the small-

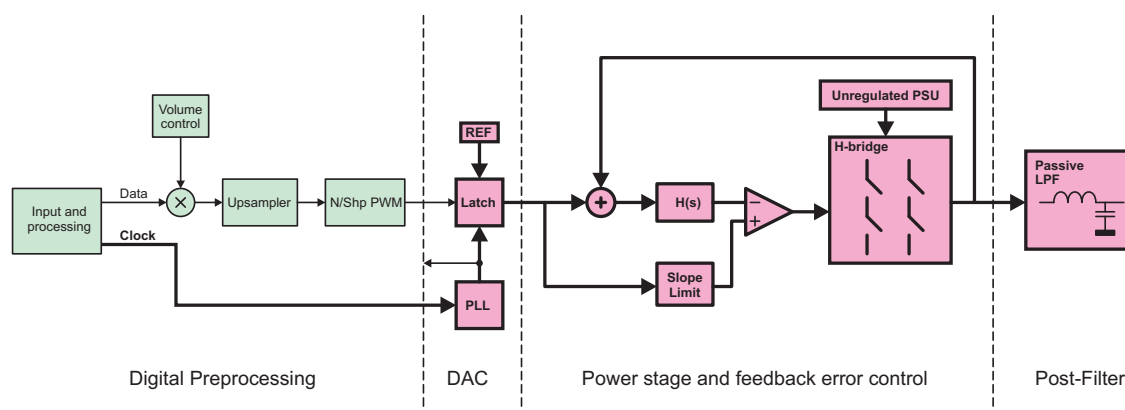


Fig. 7: Digital PWM with Pulse-Edge Postcorrection.

signal domain, the power supply is given back one degree of freedom. This degree of freedom may now be spent on dropping one stage (when a PFC is used) or adding PFC functionality. Since PFC is still a rarity in power amplifiers, no advantage is taken of this potential economy. This accounts for the fact that most manufacturers have dropped previously announced plans to start implementing power supply feed-forward in their products. Most OEMs have standardised on regulated flyback PSU's without PFC, regardless of the supply requirements of the power amplifiers.

In terms of performance, this operation may have freed up one degree of freedom, but it is not used optimally. The power stage distortion is not addressed and the loudspeaker voltage is not controlled.

4.2. Pulse Edge Postcorrection

The other most obvious cause of distortion in switching power stages is timing error distortion. As discussed earlier, the output current influences the transition timing of the power stage. The premise of the implementation of figure 7 is that the timing difference between the switching output and the “digital input” is measured and the timing of the next switch transition corrected accordingly. A practical description is given in [4]. Unfortunately, one cannot compare an analogue signal with a digital signal. To solve the problem, a clocked latch output is nominated “digital” while in fact delivering an analogue signal that is a more precise representation of the audio content than that produced by the power stage. The first path taken by the signal is through a

circuit that slows down the edges and then through a comparator that sharpens them up again. The other comparator input allows some modification of the duty cycle by slicing the slope-limited PWM signal at varying offsets. This comparator input is controlled by an integrating loop function that measures the difference between the idealised PWM from the latch and the distorted PWM from the power stage. If the attenuation in the feedback circuit matches the voltage ratio between the power supply voltage and the supply voltage of the latch, the circuit indeed ends up correcting only timing errors. If there is a mismatch between the supply voltages, the error control will also correct the gain error by pulling harder at the duty cycle. Clearly this circuit limits the amount of tolerable power supply variation before it runs out of modulation space.

The most important observation to make is that if the latch output were set to a fixed 50% duty cycle and an audio signal were superimposed on it, the audio signal would still end up amplified. So for all intents and purposes, the output from the latch is an analogue signal and the latch is the DAC. The supply that powers the latch is the voltage reference. This circuit is fundamentally equivalent to a DAC followed by an analogue controlled class D amplifier. In general, it can be said that the main objective of the pulse edge postcorrection is to reclaim the lost degree of freedom while avoiding triggering belief 2 in customers by having a too obvious DAC and feedback loop in the circuit. A technical solution to a psychological problem, so to speak.

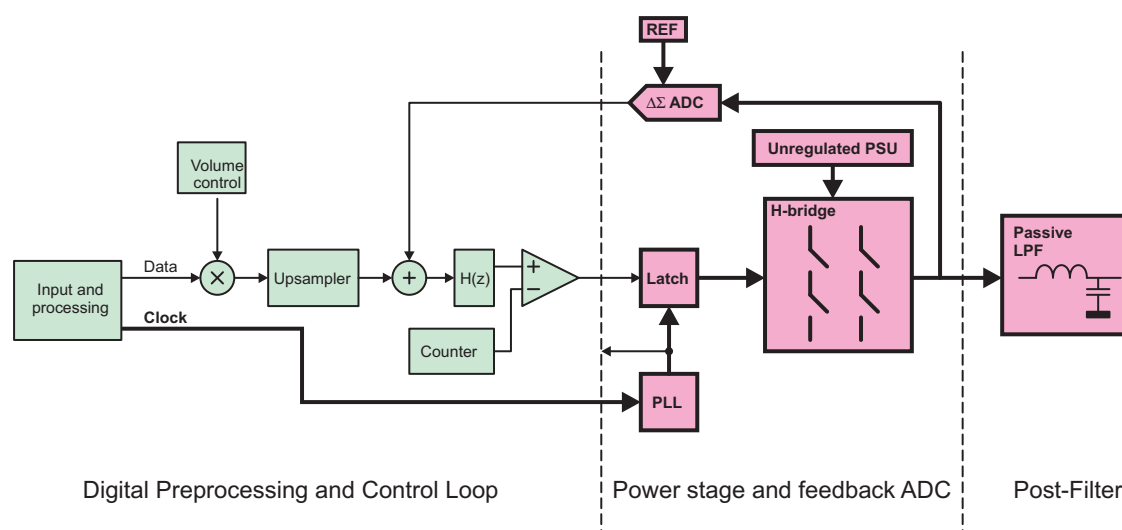


Fig. 8: ADC feedback scheme.

Nevertheless, careful implementation of the scheme can have objective technical advantages. The carrier component entering the actual modulator through the error control loop is largely canceled. This removes carrier related nonlinearities from the modulation process, allowing lower distortion at large modulation indexes. On the other hand, this method implies using feedback only before the output filter. The duty cycle is used to control a variable, but it is the wrong variable. The output voltage is the only variable of interest. Other state variables should be derived from the output voltage, not from other nodes inside the amplifier.

4.3. ADC feedback

It is unclear if any practical implementations of this idea have been made yet [5], but in principle it is possible to move the summing junction of a control loop into the digital domain by using an ADC in the feedback path. The drawing of fig 8 shows the control function performed in the z domain. In fact, the structure of many PWM noise shapers is very similar to this control loop, except that the loop is closed inside the digital domain. Such noise shapers are readily converted to digital loop controllers using a fast ADC. This ADC needs to have only a few bits, but must provide at least as good a SNR and THD performance inside the audio band as the total system is supposed to. Indeed, since the ADC is in

the feedback path, it is out of reach for the control system. Such an ADC would normally be a low-bit $\Delta\Sigma$ type with a continuous-time-domain loop. A switched-capacitor converter would alias HF components from the power stage back into the audio band. This circuit, too, has an explicit DAC. It is the DAC used inside the ADC. Again, the main objective of this circuit is to achieve full error control without triggering belief 2. To the person who is more interested in the method than in the result, this solution is even more digital than an open loop amplifier. In addition, belief 1 may be defused too, by recruiting the notion that digital signal processing is magically capable of doing things that are impossible in the analogue domain. This notion is correct in forward systems, but not in feedback systems. Some cost advantage is to be had from this method, though. If the order of the loop function is greater than the order of the $\Delta\Sigma$ loop in the ADC, the total amount of analogue circuitry is reduced. Also the size of the integration capacitors will be smaller, thus allowing a very complex low-frequency loop function to be realised at the cost of a simpler, high-frequency one.

4.4. Multiphase power stage

Not a logical progression from the previous three examples, this design serves to illustrate further the effects of self-imposed handicaps. The designer of this circuit effectively played with a double hand-

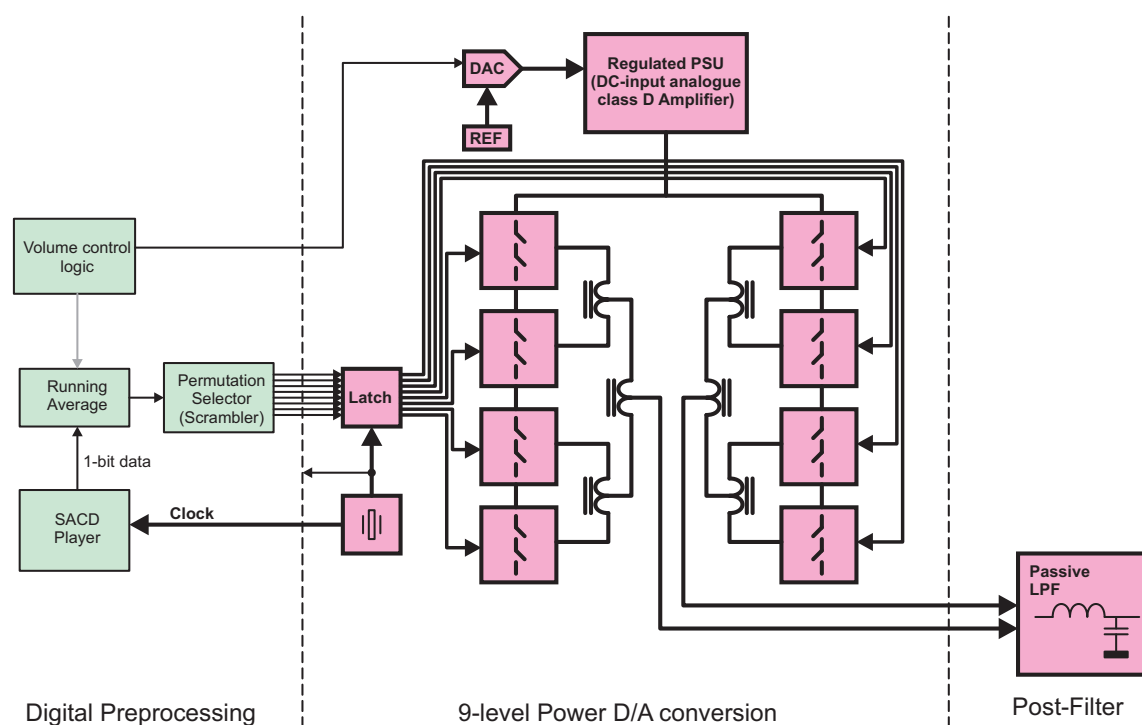


Fig. 9: Multiphase power stage

icap. Not only was the amplifier not allowed to have any analogue error correction, it also had to process 1-bit $\Delta\Sigma$ modulated data (DSD) without any digital processing [6]. The obvious thing to do when confronted with a DSD signal in an open-loop digital setting would be to run it through a PWM noise shaper, but doing so would have apparently destroyed the “one-bit essence” of the signal. Instead, a power stage is built consisting of eight smaller (and thus faster) output stages that are combined into one using center-tapped inductors. The 1-bit signal is distributed over the eight power stages using a running average and a scrambler that estimates the current in the summing inductors and prevents saturation by routing state changes to the power stage that needs it most. Indeed, a running average is a trivial operation and if it is set to 8 taps, each incoming DSD bit can even be said to have been assigned to a particular power stage. The first handicap (the required absence of error control) is translated into the addition of a local switching regulator. The second handicap (the prohibition to convert the DSD signal to another digital format) translates into a very

complicated power stage. The multiphase technique itself does have some advantages. A power efficiency of over 97% is available and the open loop distortion is very low. Again, in its open-loop implementation the efficiency advantage is immediately lost in the added regulating conversion stage, but with added loop control this technique could provide exceedingly low distortion, wide bandwidth and excellent system efficiency.

5. OPTIMIZING PERFORMANCE

A commonly heard claim by manufacturers of open-loop digital amplifiers is that the product performs “as good as the best closed loop amplifiers”. For one, this is a very one-sided claim, for it concerns only THD performance. In terms of noise and especially output impedance, this is simply not correct. In terms of THD, the question should be asked the other way around: why do closed loop amplifiers not deliver vastly better THD performance than the open loop ones? Or more precisely, why are they all in the 0.01% area? The answer lies in the design process itself. At the start is a set of minimum

performance requirements. The designer then determines if they can all be met at the same time, and how this is best done. Useful performance metrics are those that can be evaluated with the amplifier operating as a black box. Audio performance specifications like distortion, noise, frequency response as well as heat output, current draw and EMI are all valid criteria upon which a designer can make design choices concerning what goes into the box. Subjective sound performance may be an impractical specification, but it is still one that can be evaluated without peeking inside the box. Non-performance requirements on the other hand are direct specifications concerning the principle of operation. When the method used to achieve a result is specified, the engineer loses one or more degrees of freedom, directly impacting other performance aspects, cost or both. Nevertheless, designers often do find themselves confronted with such requirements.

5.1. Optimizing THD performance

In terms of distortion there appears to be a general consensus that THD in the -90dB area at 1kHz or -80dB over the full audio bandwidth is sufficient for high-quality music reproduction, with some disagreement on whether it should be allowed to rise at higher frequencies within the audio band. Thus, amplifiers targeted at quality audio are typically designed for around 0.01% THD at 1kHz, regardless of other specs. Requiring that the distortion performance remain constant over the full audio bandwidth limits available loop gain to the value attained at 20kHz. Typically this will be around 30dB for a simple 2nd order loop. Open loop distortion is then allowed to become 0.3%. Such a figure is currently achievable in power stages up to around 1kW (half bridge) or 2kW (full bridge), while the timing remains relaxed enough for easy EMI control. Beyond this power level, either the distortion requirement will have to be lessened or the “frequency independent” requirement must go. Another factor is idling losses. If very low idling losses are required, the blanking delay will need to be long enough for zero-voltage switching to occur at idle. This automatically pushes open-loop THD to 1% or higher, making low idling losses hard to reconcile with frequency-independent low THD. Noise and linear performance parameters such as frequency response and output impedance are completely independent of the power

stage design. As long as loop control is available, ruler-flat frequency response, low output impedance and low noise are always possible (see figure 10 and figure 11).

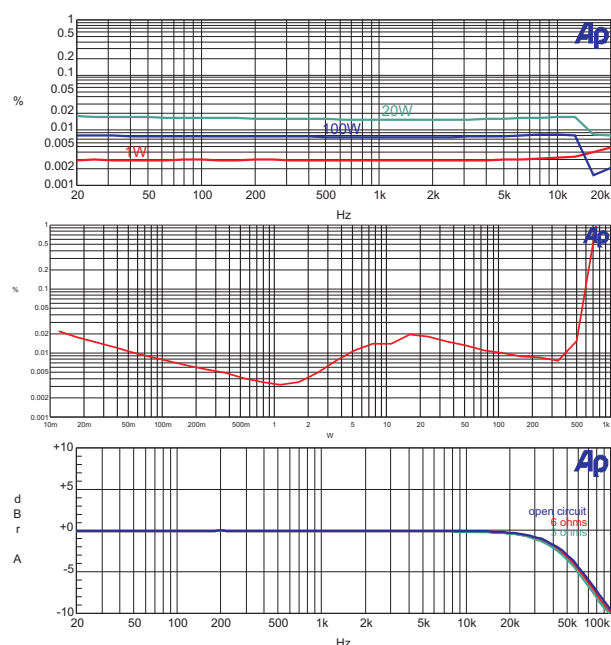


Fig. 10: A 700W class-D module with filter in the loop [7].

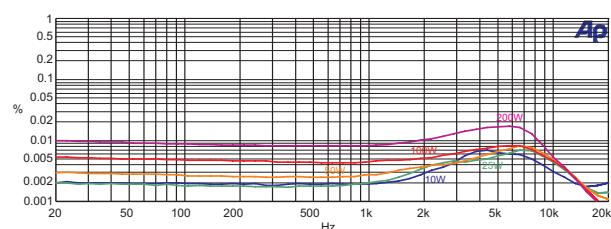


Fig. 11: Distortion of a 200W class-D module with specific analogue output feedback [8].

If an extraneous requirement such as the absence of error control is introduced, the picture shifts dramatically. Open loop THD will need to be 0.01%. Such linearity is nearly impossible to achieve at power levels exceeding 100W for a full bridge circuit. In addition, the fast switching required will increase idling losses and worsen EMI performance considerably. In systems employing noise shaping, noise performance becomes highly dependent on power stage timing as well.

This already answers the question. Most amplifiers produce around 0.01% THD because that was the target spec. Their ability to combine this with other criteria like output impedance, efficiency and EMI is where the true differences lie.

5.2. Output impedance

To eschew error control also renders low output impedance and load-independent frequency response unachievable. None of the “digital amplifiers” is capable of post-filter feedback (i.e. to control the output voltage as it appears at the speaker terminal). Hence, all of these amplifiers suffer from a significant and possibly the most audible error of all: frequency response aberrations due to speaker- LC filter interaction. Loudspeakers are designed to be driven by a nearly perfect voltage source. Only then will they produce their specified frequency response. Inserting an LC filter between it and a voltage source effectively de-tunes the cross-overs, producing response deviations up to several dBs, not to mention the attendant phase error. A few practical figures illustrate this. At 20kHz a typical $30\mu\text{H}$ output coil has an impedance of $3.8j\Omega$, a $1\mu\text{F}$ capacitor’s impedance equals $-8j\Omega$ at 20kHz. It is obvious that different loads have a significant impact on frequency response. The only remedy is to control the output voltage using a correctly designed feedback loop, as is done in [7] and [8], where even at 20kHz an output impedance below 0.05Ω is achieved. Another clear advantage of post-filter feedback is the reduction of distortion products arising from saturation-induced non-linearities in the output inductor.

6. CONCLUSION

A device delivering power to a loudspeaker is by definition analogue. The non-idealities of class D amplifiers are analogue in nature as well, and must be solved using analogue means. Error control (feedback) is an important tool in the reduction of non-linearities, and is often the only available tool to address linear errors such as response deviations. Since the notion of “digital” is not applicable to amplifiers, attempting to make amplifiers more digital by abolishing error control is a futile and irrational endeavour. All amplifiers are analogue, and the appreciation of this fact can only lead to more and better analogue amplifiers.

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