



## 1. CONCEPT

“UcD” stands for “Universal class D amplifier”. This is a reflection of the requirements put forward when it was developed, and of the extent in which it embodies them.

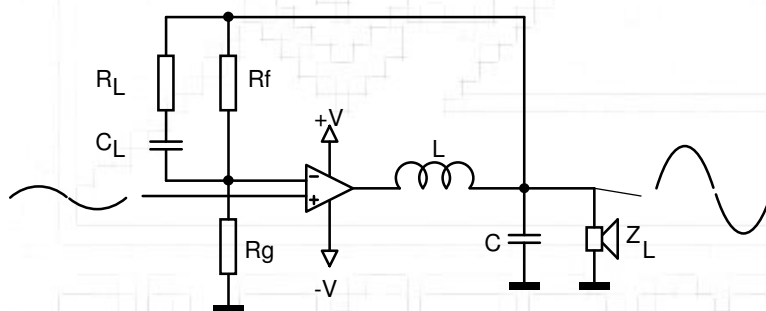
### 1.1. Requirements

Universality meant that the new amplifier be able to replace linear audio amplifiers in all fields of use. Following are the chief considerations made:

- The amplifier should be as easy to use, if not easier, as a linear amplifier. As a module, its application should not require special EMC knowledge.
- Its audio performance should not depend on special thought from the user either. This dictated good PSRR and the use of differential signal inputs.
- It should have excellent EMC performance. Many products use it in one box with a radio tuner. The reception quality should not be appreciably affected even if the antenna was a piece of wire dangling next to the speaker wires.
- It should be simple circuit-wise. This would reflect itself in cost (the company’s main desire) and in its usefulness in audiophile applications (my main desire). Self-oscillation becomes the automatic choice. Less common was the decision to construct the active electronics with discrete parts only.
- It should be completely load-invariant. The loudspeaker-dependent frequency response deviations that other class D amplifiers exhibited, while usually being of a euphonic nature, were to me an impediment to their use in true high-end audio. Euphonic colouration is still colouration and therefore not acceptable.
- THD should be low enough such that it would not produce any sound colouration. My experience with tube amps and low-feedback amps of various sorts was that 0.05% THD in itself does not manifest itself as colouration, as long as it is independent of frequency and as long as 2nd and 3rd harmonics dominate. The spec was pinned at a maximum of 0.03% up to half rated power and loop gain had to be constant across the audio range. Reducing THD at lower frequencies is not hard to do (class D amplifiers delivering 0.005% at 1kHz were already on the market) but sound quality would actually be worse off.

### 1.2. Operating Principle

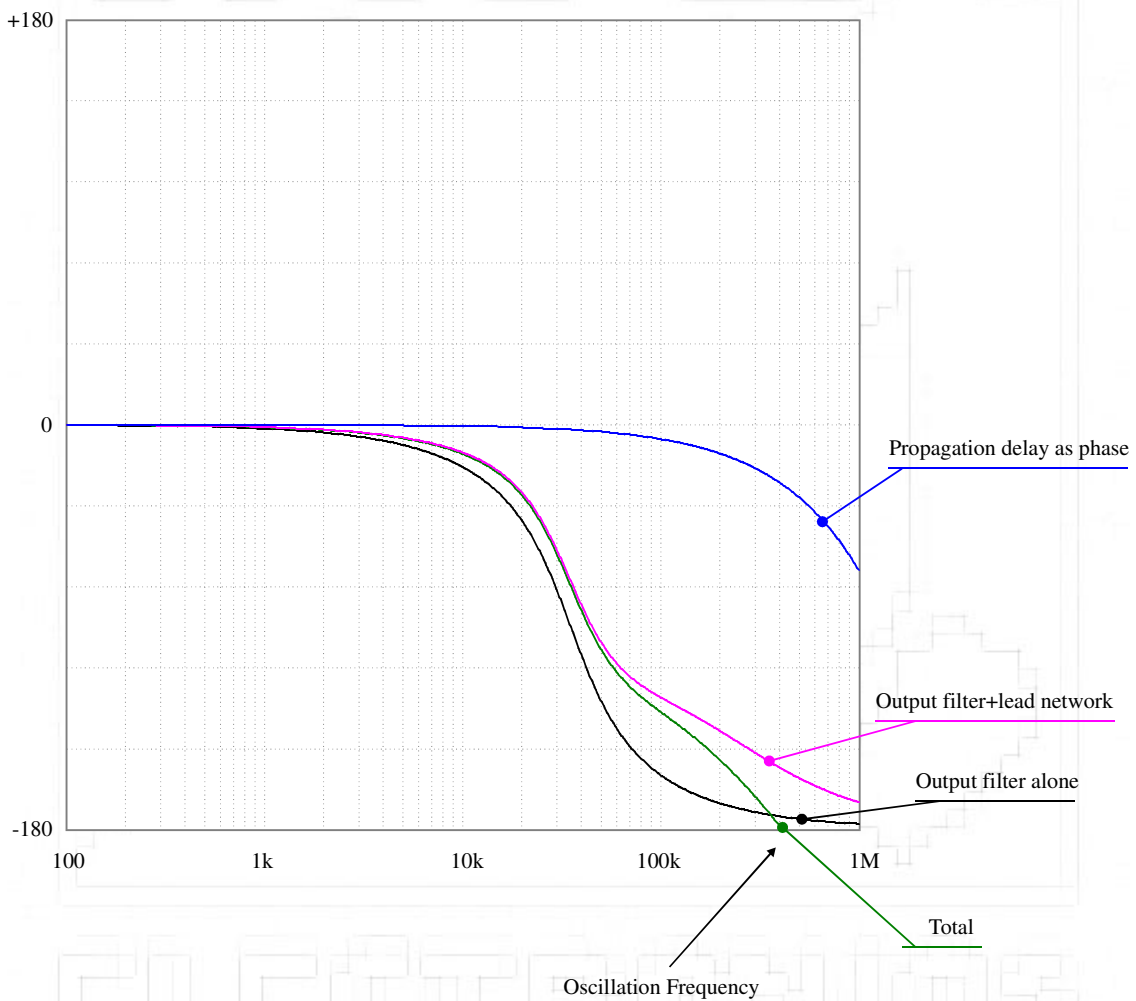
It is unclear what exactly happened next, but the first concept sketch looked like this:





The output is connected to the inverting input of the comparator through a voltage divider equipped with a phase lead network. The switching frequency is set at 10 or more times the corner frequency of the filter. At this frequency the phase lag of the output filter is very nearly 180 degrees, pretty much irrespective of the attached load. Oscillation occurs at the frequency where phase shift is exactly 180 degrees. The phase lead network is set to combine with the propagation delay of the power stage to create a phase response that transitions through 180 degrees sufficiently steeply, producing a clearly defined switching frequency. In doing so, realistic deviations from 180 degrees produced by the output filter are made largely irrelevant to the oscillation condition.

The following graph demonstrates the idea.



In this example, the filter's corner frequency is 35kHz. In a phase plot this is the point where phase shift is 90 degrees. The switching frequency is put at 400kHz. With respect to the oscillation condition, the function of the output filter is only to provide roughly 180 degrees of phase shift, with the feedback network and propagation delay standing in for the precise

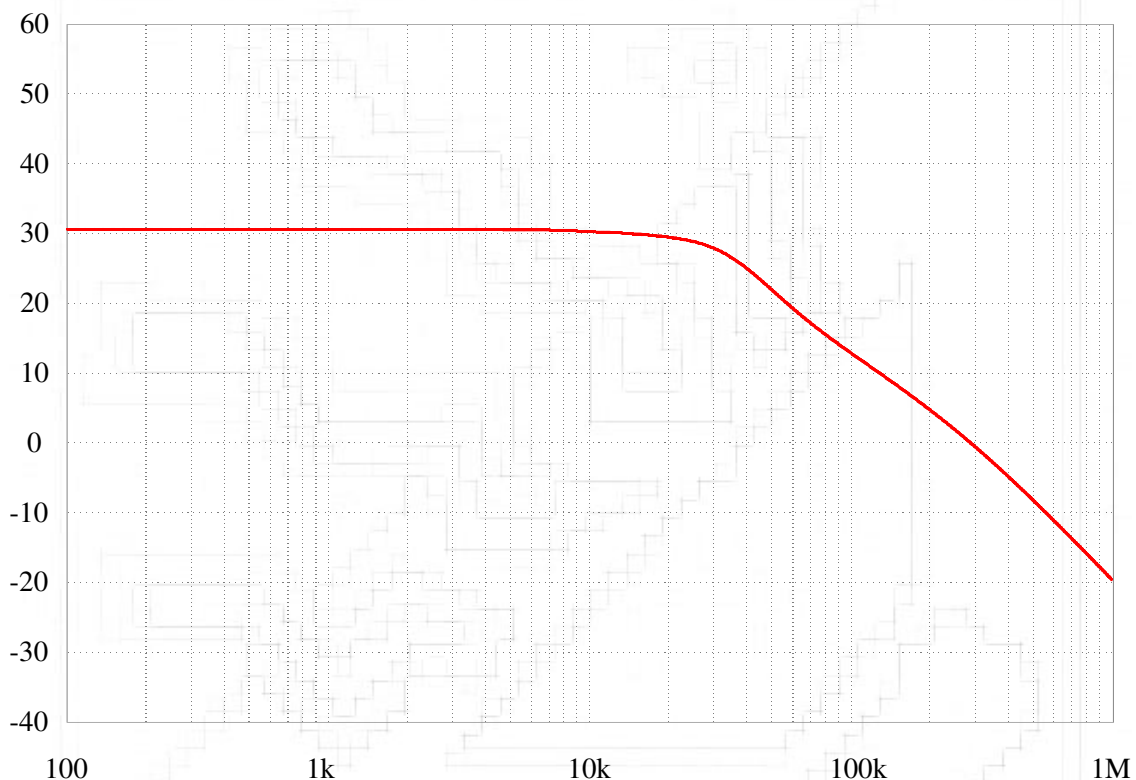


oscillation frequency. The impact of load or tolerances on the output filter components on switching frequency thus becomes very small.

### 1.2.1. Loop Gain

DC (open loop) gain of a class D power stage is a function of the power supply voltage and the amplitude of the carrier component at the comparator input. Since the carrier derives from the power stage itself, gain depends only on the total attenuation of the output filter and feedback network. Loop gain vs frequency further proceeds as dictated by the output filter and the feedback network.

Shown below is the loop gain of the 2<sup>nd</sup> order modulator used in the Hypex UcD modules.



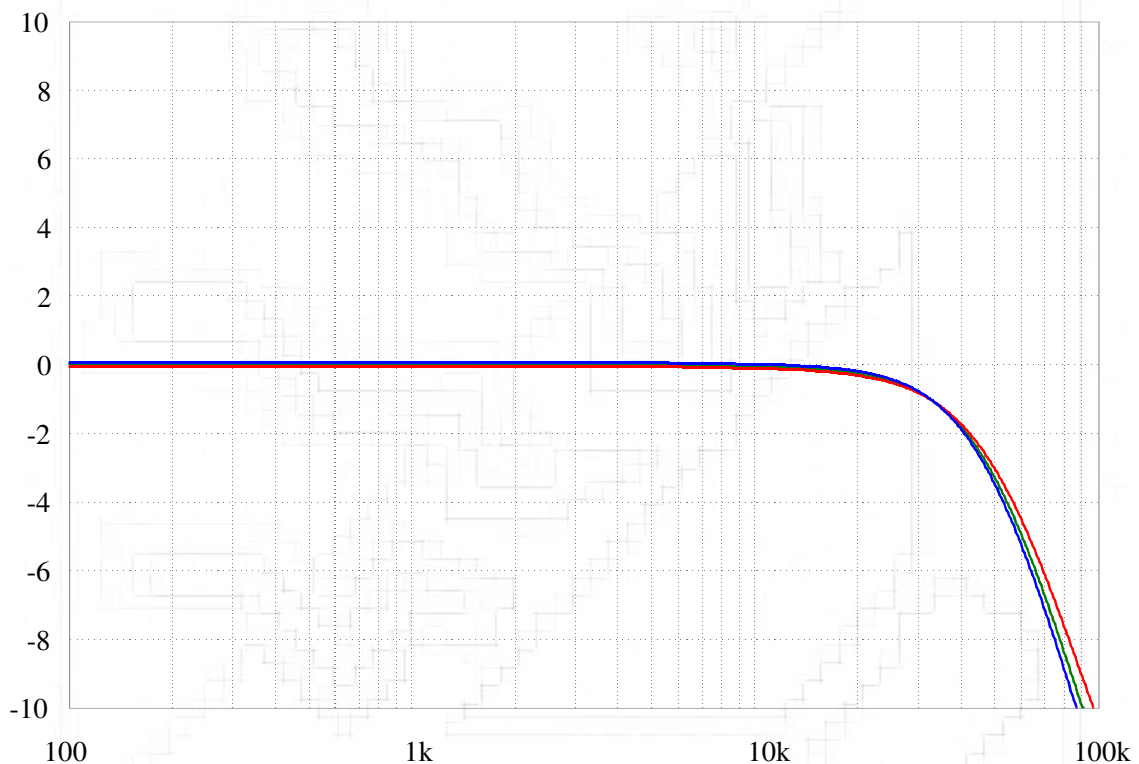
Loop gain 34 times (30.5dB), its  $-3\text{dB}$  point is 30kHz. This is a conscious choice. A loop gain that increases at lower frequencies is simply a matter of substituting an active pole for the passive pole and very spectacular THD figures at low frequencies would be the result. It is not only my own experience, but also that of nearly the entire *collective audiophilia* that such designs actually sound worse than one with modest THD figures that are constant throughout the audio band. The whole “zero-feedback” phenomenon is a reflection of this, as is the use of vacuum tube circuitry.



### 1.2.2. Frequency response

Frequency response is determined entirely by the feedback network. In the “elementary” 1st order case described under 1.2, the frequency response will be a first-order lowpass function. The control circuit used in most UcD implementations such as the Hypex modules is second-order, so the frequency response will be a second-order function as well. The frequency response is chosen as a compromise between Thomson (Bessel) and Butterworth filters. Thomson filters have the best phase and impulse responses but start drooping early in the frequency domain. Butterworth filters are exquisitely flat, but exhibit greater overshoot and ringing. This tradeoff is a constant theme in discussions concerning audio (A/D and D/A) converters and is equally important here.

Shown below is a family of frequency response plots, taken with loads of 3, 6 and infinity (open circuit) ohms.

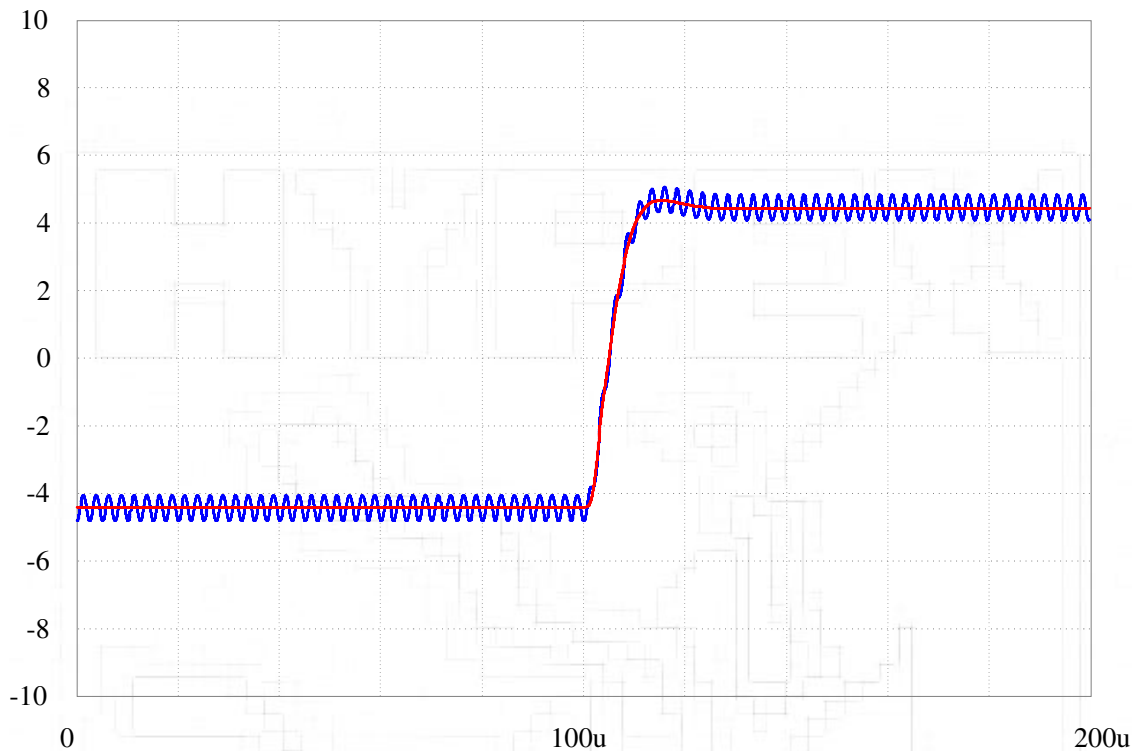


This is a breath of fresh air compared to other class D amplifiers that have a very wobbly frequency response that can vary over several (if not tens of) decibels, depending on what happens to be attached.

The load-insensitivity of the frequency response (ie. the output impedance) of UcD is even significantly lower than that of most class A amplifiers! This insures that irrespective of the kind of loudspeaker used with UcD, it will produce exactly the impulse response it was designed for.

### 1.2.3. Impulse Response and Slew Rate

As hinted earlier, impulse response and frequency response are inextricably linked. Shown below in blue is the small-signal impulse response of UcD and in red the impulse response of a 2nd order lowpass filter designed to mimic the UcD’s frequency response.



They are identical, with only the HF residual to show the difference!

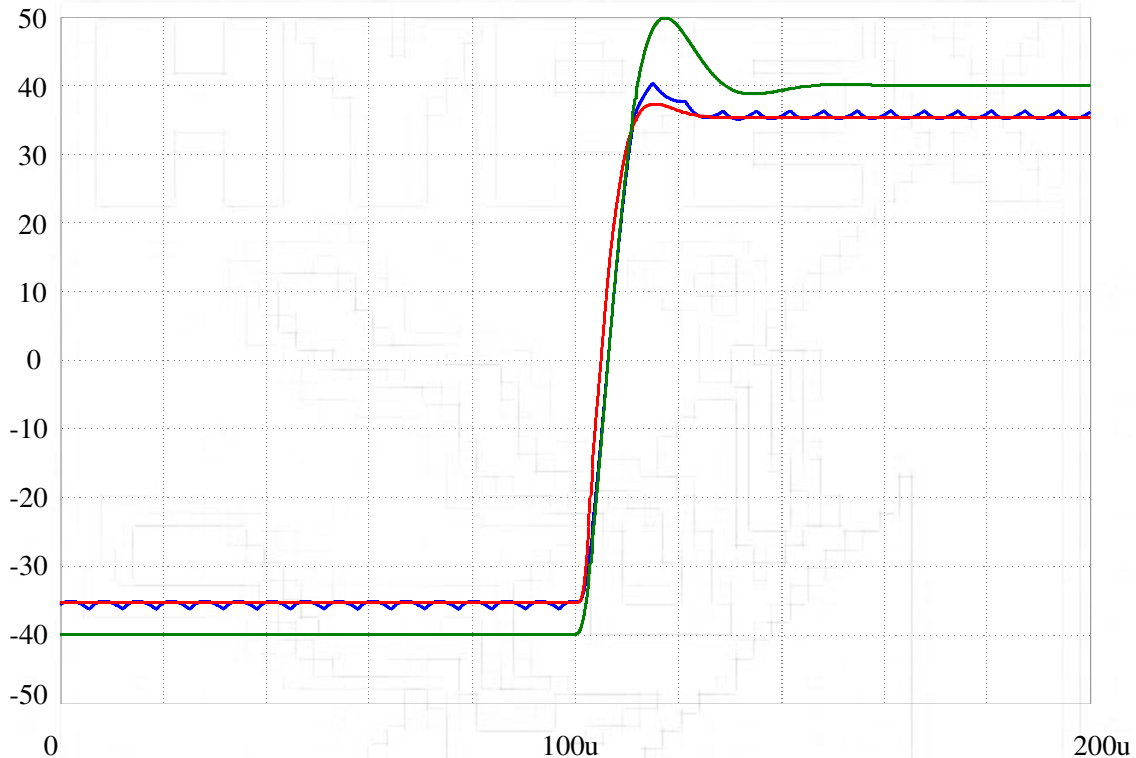
Slew rate does not have the same significance in class D amplifiers as in linear amplifiers. In linear amplifiers, slew rate is dictated by the bias current of the driver stage. A very high slew rate is required to insure that the open-loop THD does not increase with frequency inside the audio band. Typically, a good linear amplifier is designed to have a slew rate at least an order of magnitude above what is required for full power at 20kHz.

In a class D amplifier, slew rate is determined solely by the output filter. The actual power stage will happily swing from minus to plus full scale in 20ns (something it does all the time), corresponding to 4kV/us on a 100W amplifier.

The output slew rate is ultimately limited by what comes out of the output filter in response to a full-scale step. Since the limiting factor is a linear circuit (the passive output filter), the distortion mechanism that linear amplifiers need to avoid by targeting high slew rates is not present in class D amplifiers. As long as slew rate suffices to reproduce 20kHz at full power, there is nothing to worry about.



Shown below in blue is the large-signal step response of a UcD running off 40V rails. The red curve is the step response of the lowpass filter (the ideal response) and green is the response of the output filter only, to a 80V step.



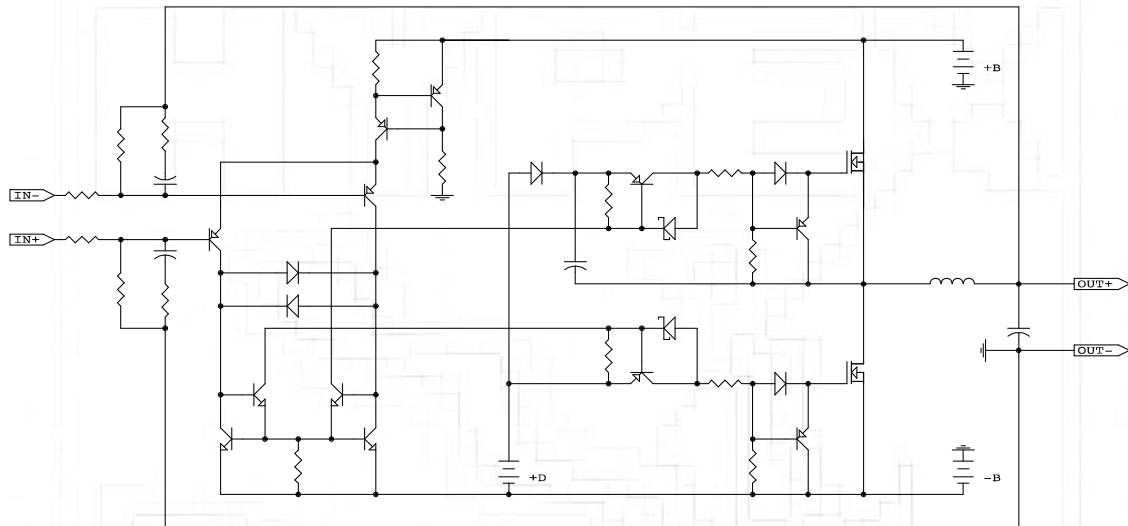
The difference between red and blue curves show that indeed the amplifier cannot entirely follow a full-scale step. The green curve shows why: the output filter determines the slew rate limit. Slew rate at 0V in this graph is 9V/us, sufficient to reproduce a 35kHz full-power signal. An alternative way of viewing the situation is that the power bandwidth of the amplifier will correspond to the bandwidth of the output filter -which just happens to be 35kHz in our case.

The fact that the difference between the red and blue graphs is still quite small reflects another deliberate design choice, namely to restrict small-signal bandwidth to a value not much higher than the power bandwidth. This insures that the input signal will have to be something fairly nasty before slew-rate induced distortion sets in. During the course of the development of UcD, some versions were made with >100kHz small-signal bandwidth, but these could not cope with the HF output signal of an SACD player.



## 2. IMPLEMENTATION

The basic discrete UcD circuit is drawn below. Not shown are an additional passive pole and a few extra parts for startup and thermal tracking.



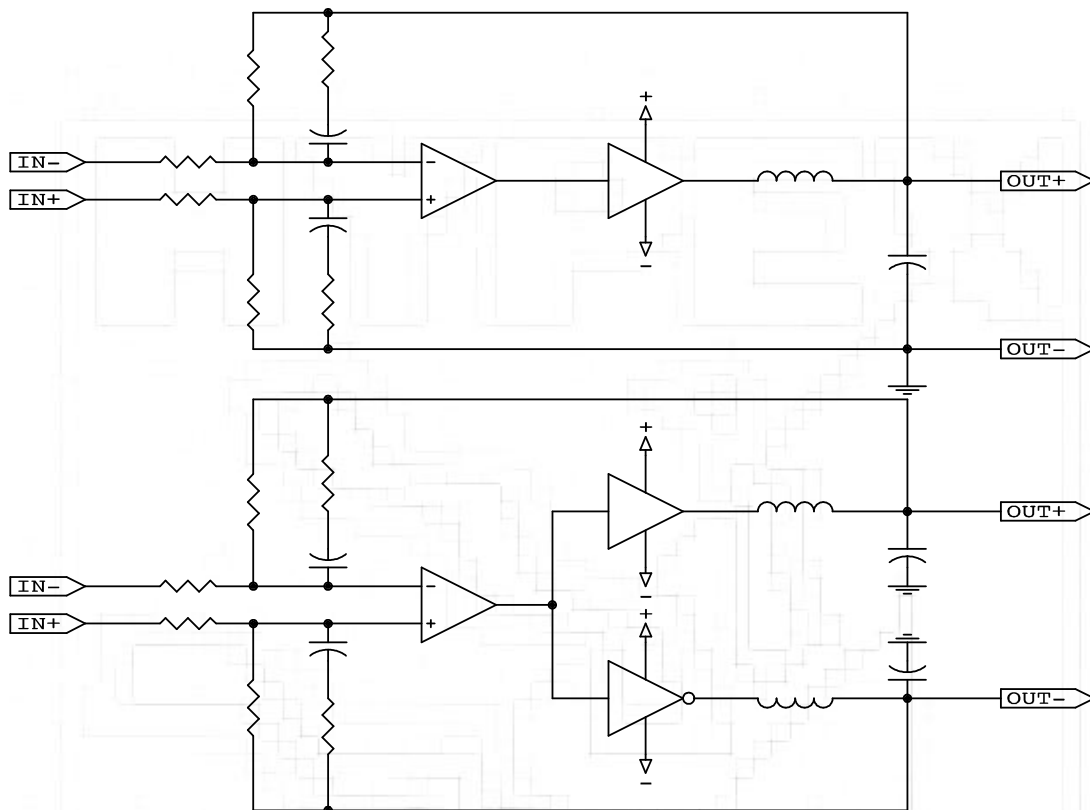
One could be forgiven for confusing this circuit with that of a linear amplifier. This is how practical discrete class D can get!

The comparator is discrete and is current-coupled with the MOSFET drivers. Usually, current consumption on the +D supply is low enough to regulate it directly from the negative supply, so the whole circuit operates on +/-50V rails only. The amplifier is turned on/off by controlling the tail current source.

Note the arrangement of the feedback network. It is duplicated and the noninverting side is set up to sense the ground at the speaker terminal. The inputs effectively form a differential amplifier, which can be used to sense the signal ground remotely.

### 2.1. Half-Bridge vs Full Bridge Amplifiers

A minor myth surrounding UcD is that it be only possible to make half-bridge UcD amplifiers. Probably it arose because there is no "Full Bridge UcD Patent". The real reason why no such patent exists is because conversion to full bridge is too trivial to mention:



The full bridge version can be made to run off a single ended or dual supplies. Good matching of the DC portions of the feedback networks is required for good PSRR.

### Reasons for choosing full bridge

- Voltage swing. At the current state of MOSFET technology, diode recovery problems become prohibitive for devices of  $V_{BRDSS} > 150V$ . If more than, say 50Vrms of output swing is required, either current-steering diodes must be added, or a full-bridge configuration selected. We are aware of people building class D amplifiers using 200V MOSFETs without using current-steering diodes, but these designs have serious EMI and efficiency problems because of this.
- DC operation. In non-audio applications where DC output voltages and currents are required, bus runaway (pumping) pretty much rules out half-bridge designs.

### Not reasons for choosing full bridge

- Pumping (in an audio application). The storage capacitance needed to produce a suitable DC supply is already sufficient to render pumping effects pretty much a non-issue. Much is made of the pumping problem by people who have full bridge amplifiers to sell.





**Reasons for choosing half bridge**

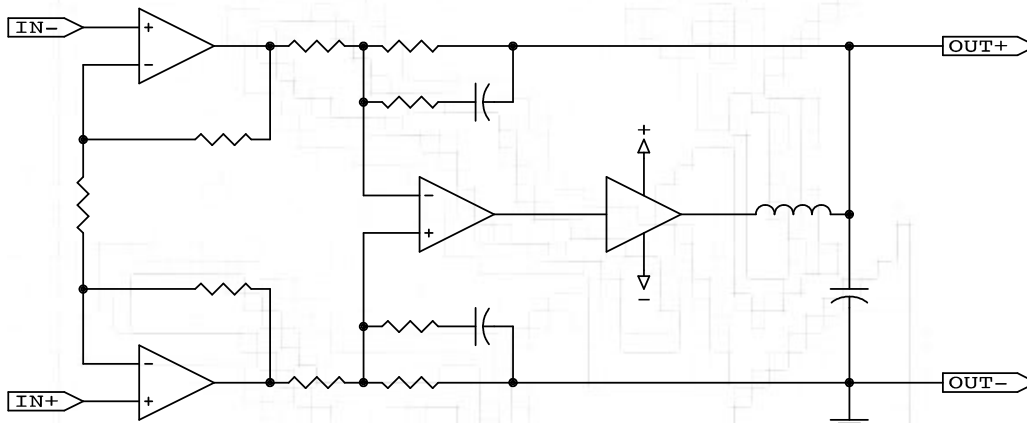
- Economy. For up to a few hundred watts, half bridge power stages are the lowest cost solution, and are also the most compact. Otherwise put: for the vast majority of applications, half-bridge amplifiers are the default choice.

**Not reasons for choosing half bridge**

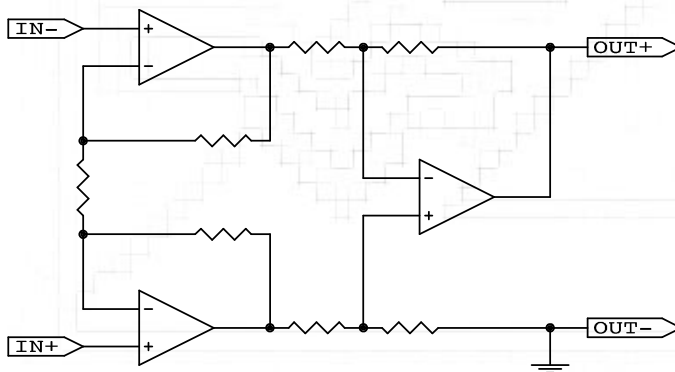
- “Better performance or sound quality”. Given equal care in design, there is no difference in attainable performance in full-bridge or half-bridge amplifiers. No sonic differences have been found caused by the power stage arrangement.

**2.2. Input stage**

It is customary, though by no means required, to equip the UcD circuit with a two-op amp difference amplifier at the input. Such a buffer is present on all Hypex UcD modules.



One is reminded of the traditional three-op-amp instrumentation amplifier:



The similarity is more than passing. The UcD con op-amp buffer is a functional instrumentation amplifier with high-impedance inputs and with a CMRR which is improved by a factor equal to the gain of the first stage. The gain structure in a typical implementation is 5 for the input stage and 4.5 for the subsequent UcD. CMRR-wise it would have been better to have all the gain realised in the first stage, but a standard IC op amp wouldn't cut it because of the voltage swing it would have to deliver.



There is a misconception that an amplifier having inverting and noninverting inputs must necessarily be driven by a balanced signal source. This is quite incorrect. The only thing it does is measure the voltage difference between one signal and another. If one input is connected to the output of an unbalanced signal source and the other input to the ground at that source, this will work fine.

### 2.3. Protections

A complete, “boxed-up” amplifier typically contains two main protections

- Overcurrent: basically the short-circuit protection to prevent blowing power FETs. Two varieties exist. One shuts down the amplifier for a short period, the other “clips” the output current. Both have been made to work on UcD.
- DC: Security regulations (and good engineering practice!) require a design to take account of an eventual failure, regardless of how well protected it may be. A shorted power FET is a fire hazard, because it will cause a large DC current to flow through the loudspeaker.

Hypex UcD modules have an overcurrent protection but no DC protection for the simple reason that a failed power stage does not readily respond to pleas for shut-down. It is up to the module user to add shutdown capability to the power supply.

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