

I2V, details on sub-elements

Preface

In previous articles on this site about I/V-converters, the use of an op amp as I2V-converter has been argued. Also the type of op amp has been chosen. Some details have been treated implicitly which will be magnified and enlightened here. Some remarks on the low pass filter aback the I2VC will be made too.

The starting-point is figure 1, an I/V-converter for a PCM63 of those days.

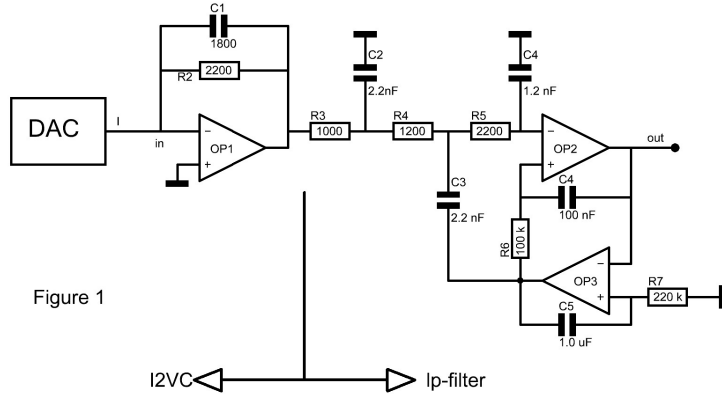


Figure 1

Here we see OP1 as I2VC and the combination of OP2 and OP3 as a low pass filter (lp-filter).

The I2VC

The circuit with OP1, C1 and R2 yield a so called 'virtual ground input'. The output voltage of OP1 is equal to the input current (from the DAC) times the impedance of the parallel circuit of C1 and R2:

$$U_{out} = I \cdot (R2 / (1 + j\omega C1 \cdot R2))$$
 This means for the audio frequencies:

$$U_{out} = I \cdot R2$$
 so that in Figure 1 each

milli ampère from the DAC yields 2.2 volt at the output of OP1.

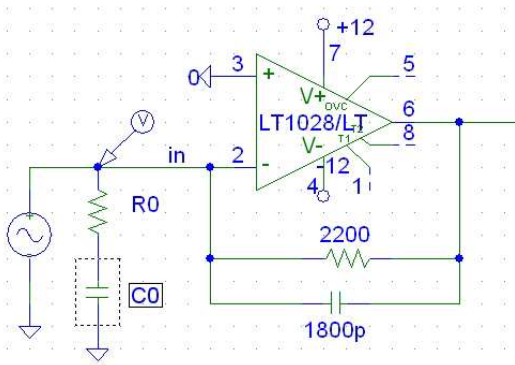
The DAC should be terminated with a low impedance. Well, the “-“ input of OP1 offers a so called virtual ground. The impedance at this point (in the audio range) is $R2/G$, with $G = \text{“open loop gain”}$. With the choice of an LT1028 as OP1, $G = 150\text{dB}@1\text{Hz}$ and $0\text{dB}@50\text{MHz}$, so that the virtual ground input impedance is $1\text{ n}\Omega$ (nano-ohm!) at DC and $2200\Omega@50\text{MHz}$ without C1 ($G \cdot B = 1@50\text{MHz}$ indeed). This are theoretical values but it shows that the input impedance rises with the frequency. With C1 this should be solved but our op amps don't behave any more as an op amp above about 10 MHz.

Does this matter?

Yes, this matters, because the DAC will squirt currents with very high frequencies resulting from the DAC-process. These HF-currents should be left out of the op amp.

The only way is to design an input low pass filter at the input of OP1 but how?

The solution is more simple as it appears to be. The simulation of the input impedance of an LT1028 as an I2VC (OP1) yields:



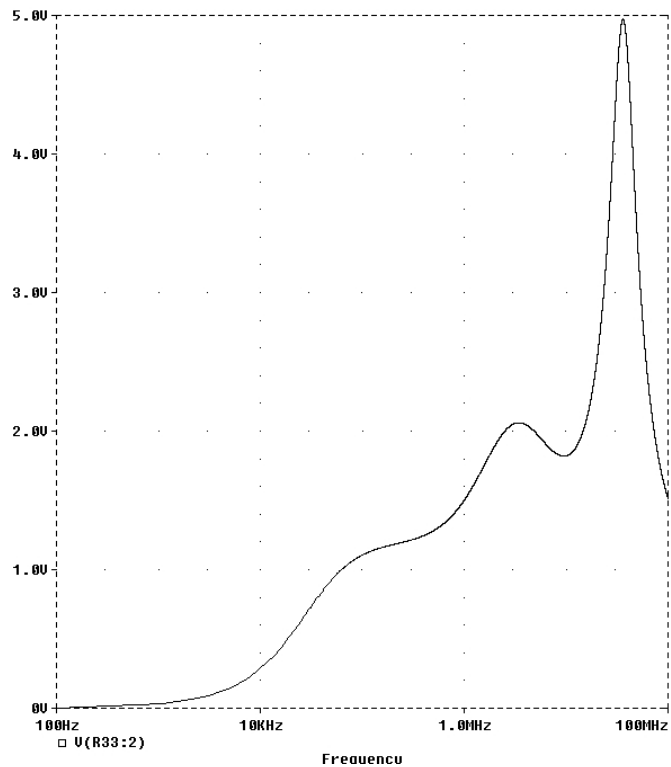
as we saw before in earlier articles:

- 1 nΩ at DC,
- 50mΩ@2kHz,
- 543mΩ@20kHz, etc.

with $R0 = \text{infinite}$ and $C0 = 0$.

This counts globally also for other values of the RC-combination in the feedback because of the large open loop gain of the LT1028.

(The simulation is done with a current of 1 mA so that every volt corresponds with 1 Ω input impedance.)



I doubt if the results of the simulation with the old MICROSIM8 are reliable for high frequencies. However one thing is clear: the input impedance increases with frequency.

If we take $R_0 = 2.2 \Omega$ and C_0 short circuited, the input impedance will become as the graph at the right. The input impedance can't get over 2.2Ω of course, but at 20 kHz it has decreased to 476 m Ω which means that too much current flows into the 2.2Ω shunt so that the gain of the I2VC will decrease about 2 dB! Of course we could make the 1800 pF capacitor smaller but there is still another problem: the gain of the I2VC has been raised from unity gain (1) to $2200/2.2 = 1000$ so that each μV input offset will result in 1 mV output offset!

The solution is to introduce C_0 , but what should be its value? After trial and error the best value appear to be 1 μF . Its reactance is $8 \Omega @ 20\text{kHz}$. Moreover this value 'helps' to enlarge the input impedance to its original value (thanks to the inductive tendency of a virtual ground circuit) as shown in the graph at the right.

Demands on R_0 and C_0

The components must work at high frequencies so their parasitic resistors, capacitances and inductions should be minimal.

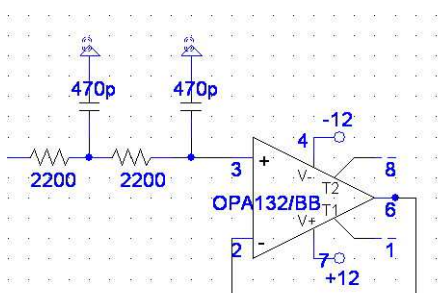
For R_0 we take an SMD resistor and for C_0 a ceramic chip-C (MLCC) which are very sensitive to the (DC-) voltage applied to them, but in our application the voltage is and stays null. No worries.

R_0 and C_0 must be connected directly to OP1 or to the DAC-output pins. The connection from the DAC to the I2VC should be made with a solid screened thin cable or coax if the connection is larger than a few centimeters.

For other DAC's

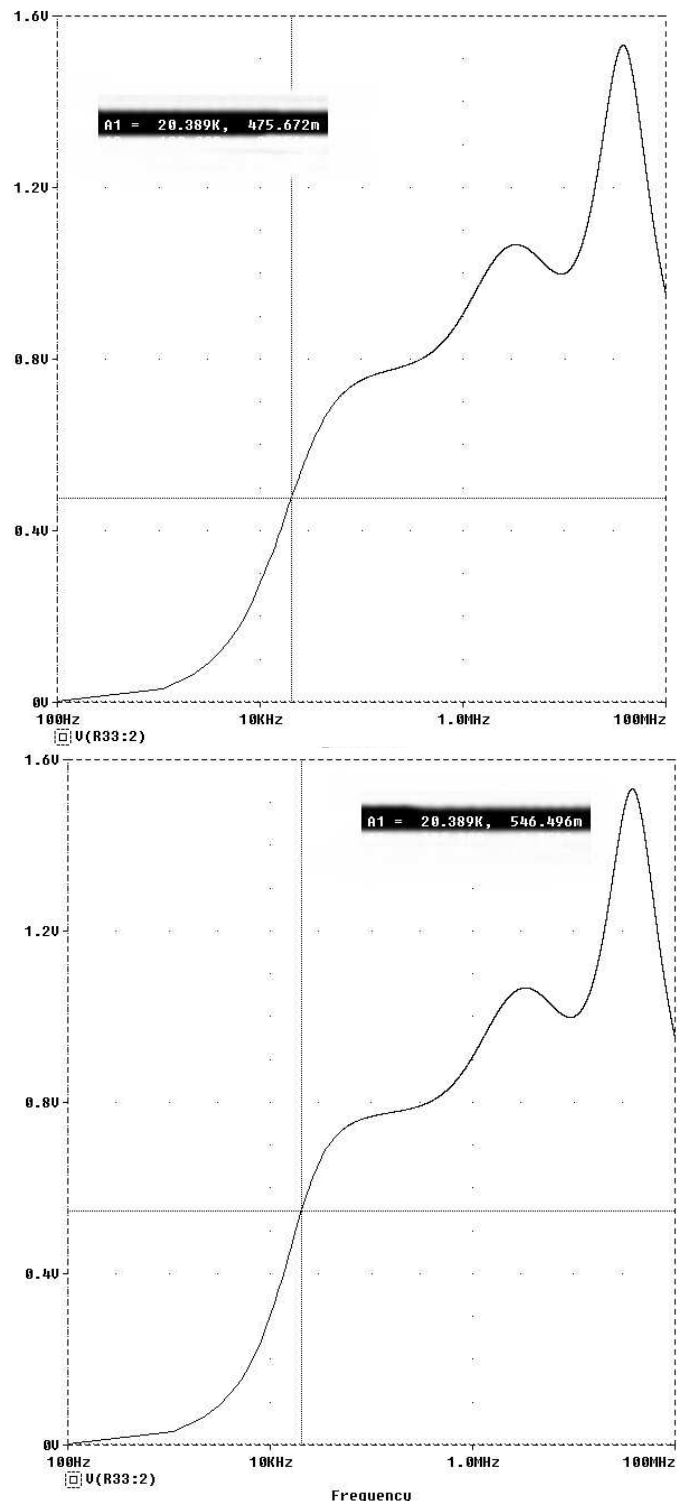
The current from DAC's as the PCM1792 is larger than 2 mA. In case the feed back circuit should be adapted accordingly. With the 7.5 mA from the PCM1792 the resistor in question could very well be 820 Ω with a capacitor up to 2200 pF or even 560 Ω with 3200 pF. We come to the right values later.

The Ip-filter



The Ip-filter in figure 1 is a servo-circuit: OP3 'measures' the offset of OP2 arisen from the DAC, the I2VC and/or OP2. It drives the "+" input of OP2 until the output offset is zero. Moreover the unknown impedance from the filter (via C_3) in the feed back is kept away from OP2. To get the loop stable $C_5.R_7 \geq 20.C_4.R_6$.

I do not trust this circuit any more. The audio from I2V-converters as in figure 1 moves in the stereo image if the music becomes louder: the instruments 'endorse larger' and move in the direction of the loudspeakers. It is hardly to explain why this circuit should be responsible for this effect. Such dynamic behavior can't be simulated. It has cost me much time and effort discovering this and I'm still not sure.



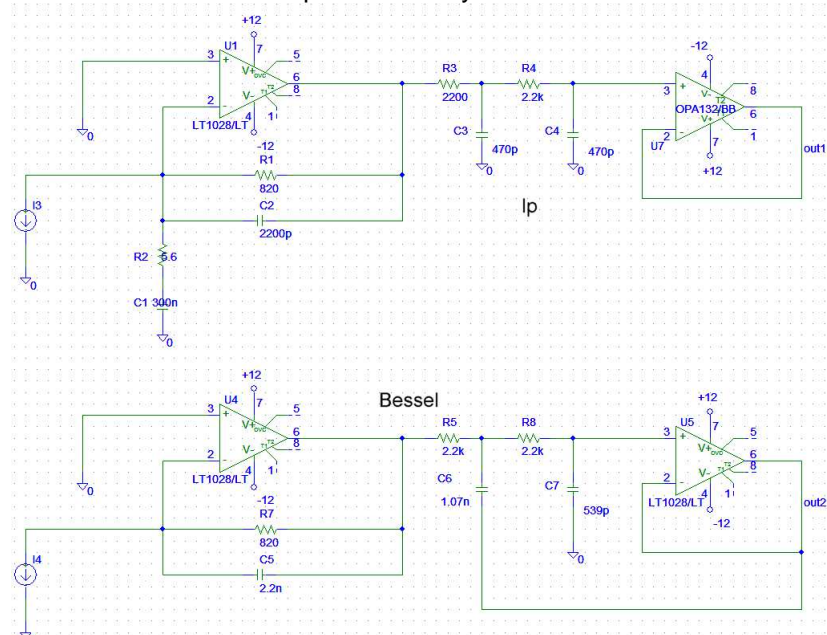
Therefore the lp-filter has been replaced with the most simple circuit: a passive 12dB/octave filter with an OPA134 as follower. The filter is even no part of the feed back! Moreover, the non-linear input junctions of the OPA134 are small and shunted by a 470 pF capacity so that the non linear junction capacitance distortion is reduced to a minimum.

The RC-times seem to be small. The cut off frequency starts at 154 kHz, but otherwise the roll off of the complete I2V-converter starts too early. At 20 kHz it is already 0,6 dB down! The phase linearity however is within 0,5° in the audio band if the RC-product in the I2VC equals 1.804 ...($\Omega \mu\text{F}$) together with the low pass filter as shown above. The components must be selected within 1%.

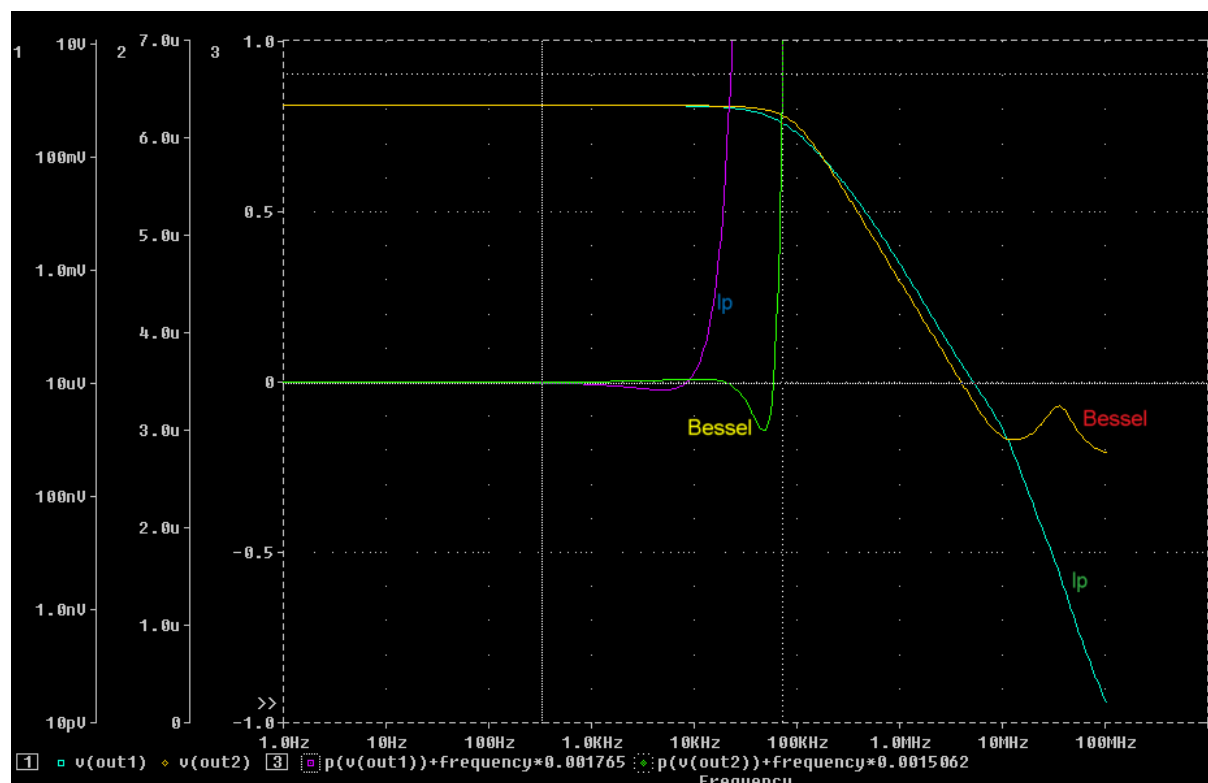
With 8 or 16x oversampling this filter will satisfy.

How did I come to these values?

Well, Paul Vancluysen did some investigations on the subject. He simulated two circuits to compare them with each other for phase linearity:



On top is my practical suggestion of the circuit (Ip) and the second one (Bessel) is the circuit as applied in many applications. The results are shown below. Mind that the scale for the phase linearity runs from -1° to +1° !! Both curves stay within $\pm 0,5^\circ$ in the audio range. However over 10 MHz is the amplitude curve worse for the Bessel filter as predicted.



If we remain to 2200 Ω for R3 and R4 in combination with 470 pF for C3 and C4 in my Ip-filter, the R1.C2-product must be 1.804 μs to keep the phase linearity within 1°. In case, the value of R1 must be in the range of 500 – 1000 Ω . The desired gain, depending on the current from the DAC at 0 dB, could be adjusted with the gain of the OPA134.

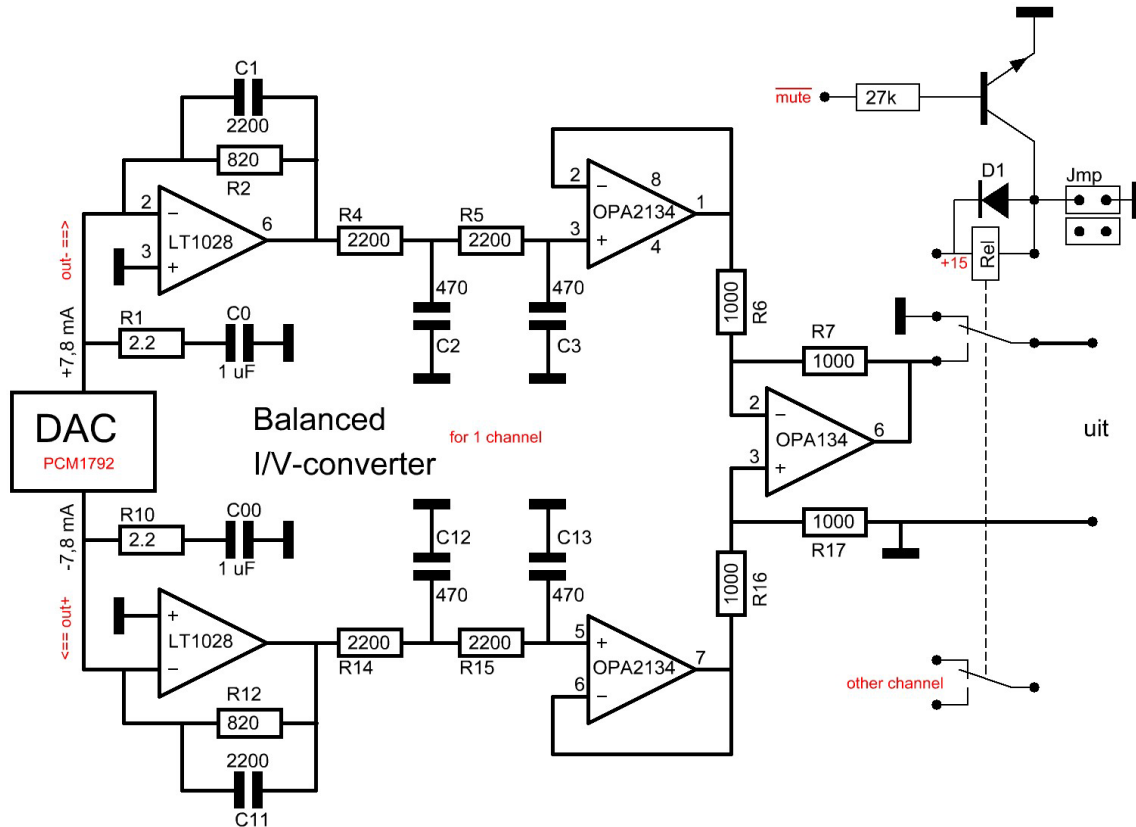
Offset

This I2V-converter is DC-coupled. If the offset is larger than, say, 10 mV, you could compensate it. The best way to do this is to supply a small current into the virtual ground with a rather large resistor from the “-” or “+” (depending on the polarity of the offset) power supply. Don't be afraid for the noise from it because the input impedance < 550 mΩ in the audio range so that the 'power supply rejection' stays large.

The I2V-converter should be supplied from noise-free stabilizers like TentLabs shunt regulators (see: <http://www.tentlabsshop.com/DetailServlet?detailID=1673>)

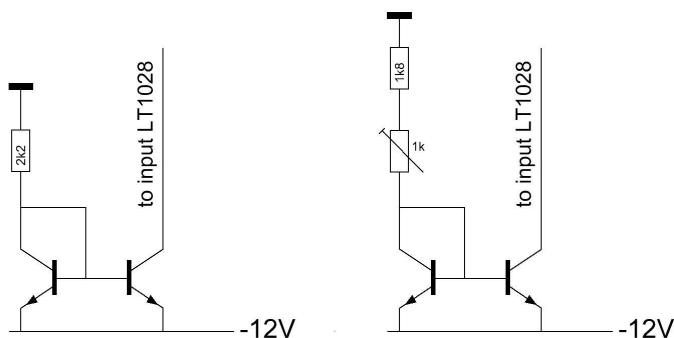
Balanced I2VC

What about the compensation of the current from a DAC with single analog power supply?



This diagram has been shown before in the article in Dutch: 'Ombouw van een CD624 met FPGA en PCM1792'. The 1792 is a DAC with balanced outputs but connected to a single analog supply of 5 V. The four currents from this DAC without signal are 5 mA DC with a differential audio of 7,8 mA_{pp} superimposed (at 0 dB). With R2 and R12 = 820 Ω the output of the LT1028's is about 4.3 volt DC with an amplitude of 3.2 V audio superimposed on it.

This means that the power supply voltages on the OPA2134 should be rather high, in practice >±12 V. The output swing of these op amps is from nearly 1 to 7.5 volt. Wouldn't be more convenient to move this to about -3.2 to +3.2 volt? In case the inputs of the LT1028's should be 'compensated' with a DC current of > -5 mA.



If one of these currents (per channel) is adjustable, the offset at the output (of the OPA134) becomes tunable. This could be implemented with two current mirrors to the negative power supply (per channel) as shown below.

A more obvious method is to connect the non-inverting inputs of the two LT1028's to half the power voltage of the DAC, 2.5 volt with a PCM1792, but be careful! Any

noise on this supply will be amplified and degrade the signal to noise ratio!

Some DAC's, as the PCM1738, do have a reference voltage output Vcom for this purpose.