ISCE

The Institute of Sound and Communications Engineers

Engineering Note 9.3

Optimising the overall efficiency of a system

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Introduction

It is fairly obvious that there is no point in feeding a loudspeaker with signals at frequencies at which it produces no useful sound output, which we may define (as in IEC 60268-5) as frequencies at which the response is more than 10 dB down on the average mid-range response (see the definition of 'effective frequency range' in clause 21.2 of the standard). The loudspeaker is less stressed, and the voice-coil runs cooler, if we restrict the frequency range applied to it to match its effective frequency range. Alternatively, we can drive it harder for the same rise in voice-coil temperature, and thus get more sound pressure from the same combination of amplifier and loudspeaker.

The energy content of high-frequency audio signals is quite low, so not much is to be gained (and intelligibility may be lost) if we restrict the high-frequency end of the signal spectrum. But many of our loudspeakers do not have a very extended low-frequency response, and by cutting out the ineffective low-frequency signals at the input of the power amplifier, we free-up amplifier headroom and power to deal with signals that the loudspeaker *can* reproduce. However, for loudspeakers in open-back or 'not noticeably designed' enclosures, we must be careful not to exceed the cone excursion ('x_{max}') limit.

How to do it

Ideally, as an add-on to a simple or existing system with a minimum of effort, we would use a passive LC high-pass filter, needing no power supply and giving a cut-off rate of 12 dB/octave. Unfortunately, the inductor would need to be about 1 H, and inexpensive inductors of this value have far too much d.c. resistance to work . A 2-stage RC filter would give a 12 dB/octave cut-off rate, but because the second section shunts the first, it is only achieved at very low frequencies indeed in this application. However, most 100 V line amplifiers and some designed for low-impedance loads, already have a low-frequency roll-off, being about 3 dB down at 40 Hz. We can take advantage of this to simplify our efficiency-boosting filter to *just a series capacitor.*

The input impedance of a sensibly-designed sound system amplifier should be close to 10 kΩ, not the 47 kΩ or more used for household equipment (a hangover from the days of valves!). For smallish wall-mounted or ceiling loudspeakers, a filter corner frequency (-3dB point) of 100 Hz is often suitable, and we can achieve this by passing the input signal through a 150 nF capacitor to the 10 kΩ input resistance:

$$C = 1/(2\pi fR) = 1/\ 2\pi.100.10^4 = 159 \times 10^{-9}\ \text{F}$$

This capacitor can often be built into the free connector bringing the signal to the amplifier. For really small loudspeakers and for horns, a higher corner frequency may be desirable, in which case, it is just a matter of scaling, noting that a higher frequency requires a smaller capacitor:

Corner frequency Hz	Capacitance nF
100	150
150	100
200	68
250	56
300	47
400	33

We really need to check the input impedance *at the corner frequency*, even if it's specified by the manufacturer because that is likely to be at 1 kHz and it might be different at other frequencies.

Measuring the input impedance

Set up the amplifier with a suitable dummy load resistor and a sine-wave generator connected to the input through a variable resistor, preferably a 100 kΩ log potentiometer, using just the 'earthy' end of the track and the slider, not all three terminals, of course. Connect an audio voltmeter across the dummy load. With the pot set to zero resistance, adjust the input signal voltage, at the wanted corner frequency, to get a convenient number of volts at the output. For a 100 V line amplifier, for example, 20 V would be OK, while for a 200 W 8 Ω amplifier (maximum output voltage 40 V) 10 V might be suitable. It isn't critical at all. The reason you use a log pot is that you can measure a wider range of input impedance without having to adjust the pot very carefully, nearly at one end of the track.

Now increase the pot resistance until the voltage drops to half what it was. Disconnect the pot and measure the resistance (between the same two terminals as you used before!). That resistance is equal to the input impedance of the amplifier.