EEL 4924C
Preliminary Design Report

Amprotector

Designed by Team: Audiophiles

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Project Abstract

Amprotector is a limiting audio device to automatically calculate how much limiting of the small signal is needed to protect a specific speaker cabinet. We want to build a “smart” limiting device that knows the voltage signal after the power amplifier stage to determine the limiting needed on the compressor. The user of the device would set-up speaker impedance and speaker power rating. The audio amplifier's output could then be fed back to the limiter where the limiter can detect when the power amplifier’s output wattage surpasses the power the speaker can take. This is shown in figure 1. The limiter could then lower the sound level of the audio signal while keeping a low level of distortion.
Project Features/Objectives

The most characteristic feature of our device compared to similar devices on the market currently is our capability to accommodate a wide variety of speakers, covering almost literally any typical speaker sold commercially today. We accomplish this by inserting a couple of user interface knobs and switches whereby the user can specify the impedance and power specifications of his particular speaker. This is shown in figure 2 on the bottom of page 5.

The current industry standards for speaker impedance values are set at discrete values of 2, 4, or 8 Ohms. To accommodate this setting, there will be a three-position switch on our user interface panel allowing the user to set their speaker’s impedance accordingly. The power consumption levels of a typical speaker ranges between 50 and 1000 Watts. The user would be able to specify their speaker’s power rating by adjusting a knob on the user interface panel that would correspond to that typical range of wattages.

Another distinguishing feature of our device is use of compression and not complete obliteration of the signal as it approaches dangerous levels. Our device must first calculate the maximum level of output from the power amplifier that the speaker
with the user-identified specifications will be able to tolerate. Once we have that ceiling established, we will then set up an algorithm through our microprocessor that will feature different levels of compression depending on how close to that maximum ceiling value the signal is. For example, when the output from the amplifier gets within 10 percent of its maximum value, we could set up a compression ratio of 15:1. This means that once the audio signal approached damaging levels (the top 10 percent of its range of safe sound levels) any signal that would have otherwise become 15 times louder would now only be allowed to grow by 100 percent. Then we would set a second level so that if we ever detected the output of the amplifier at or above the predetermined ceiling value, we would limit the signal back to be slightly under the ceiling value. This allows the sound to be less noticeably affected by our device as it approaches the top range of allowable sound levels and therefore produce a more aesthetically pleasing, and less distracting, effect.
Figure 2
The main technological implementation issue we faced in coming up with our design was deciding how we were going to implement our signal compression. We knew that the more complex the algorithms we were to use in our compression the more gradually we could allow our signal to transition between being unaffected by our device (no attenuation) and being completely cut off from getting any louder once it hit the maximum allowable amplifier output. Therefore, our first thought was to use a digital signal processor that would allow for very advanced manipulation of the signal once it was digitized. With this option we would even have the option of going above and beyond the scope of our project’s objective (of protecting speakers) to also incorporate other possible effects such as upwards and downwards compression (even compression in addition to that required to protect the speakers), companding, equalizing, etc. Unfortunately, upon investigation of this option, we first found that we would need to use a development board to use our DSP chip. Even if this was approved for use in our project we came across even more important barriers such as a very high cost of DSP chips, stories from other students of unreliable DSP chip programmers, and advice from DSP-specializing professors that using a DSP chip would likely prove to be time-consuming.
and complicated, especially given that neither of us have any solid background in digital signal processing (in fact, we are both currently enrolled in that course).

As an alternative, we decided to use an Atmel microprocessor controlling a digital potentiometer to limit the amplifier’s gain. The microprocessor would take in the signal from the amplifier, analyze how much the signal needed to be attenuated based on the user-specified impedance and power requirements of the speaker, and then adjust digital potentiometer’s resistance value to attenuate the amplifier signal based on the concept of voltage division. Using this approach will still allow us to program in some simpler algorithms into the microprocessor in order to get a gradual sense of attenuation as the signal approaches dangerous levels, yet it will be much simpler, cost-effective, and overall a more reasonable approach in achieving the objectives of our project.

Another challenge in technological implementation we face is converting the signal coming from the power amplifier into a low voltage signal that can easily be interpreted by our microprocessor in order to accordingly adjust the attenuation level. This is shown in figure 6, the software flow chart. First, we will use analog step-down circuits involving an array of op amps in order to convert the high level signal from the
amplifier to fit within the 5-volt limit into our microprocessor. Next, we need a continuous digital representation of the signal level in order for the microprocessor to be able to correctly interpret the signal amplitude and adjust it quickly if needed. In order to accomplish this, we plan to use a wave rectifier circuit with a capacitor to obtain constant positive signal amplitude and then to pass this rectified signal through a low pass filter to steady the signal level further. This is shown in figure 3. Once this is all accomplished, we will have a nearly steady signal level between 0 and 5 volts that we will be able to send through the A/D converter of the microprocessor and thus obtain an accurate measurement of the amplifier’s signal level that can then be interpreted and acted on further by the microprocessor.
User-defined inputs (impedance, power of speakers)

Calculate maximum signal level speakers could handle safely

Signal from amplifier output (after stepped down)

D/A conversion

Is signal from amplifier within 25% of its maximum level?

No

Do nothing

Yes

Is signal within 10% of max level? Beyond the max?

Within 25% only

Tune digital pot for light proportional compression

Beyond the max

Within 10%

Tune digital pot for heavy proportional compression

Implement emergency algorithm to quickly attenuate signal to safe region
Components & Cost

Materials would include a housing box to house all the circuitry and a faceplate for mounting all the user interface and input/output components. Included in our circuitry would be a microprocessor, digital potentiometer, and many analogue circuit components, such as a wide array of op-amps and resistors.

The prototype will not only have more features than current models, but cost much less as well. Most mosfet transistor op-amps will not surpass $20. Many chips can be order as samples. Much of the cost will go into the mechanical casing of the device. Along with input/output jacks, leds, potentiometers and a custom casing design the prototype could total $150. However, similar devices such as the dbx DriveRack PX Powered Speaker Optimizer cost $400.
Competition

Currently, limiters are used to protect speakers from sudden changes in dynamics. However, it is common practice to apply the limiter stage on the end of the effects chain or through the effects loop. The end of the effects chain then goes into the power amplifier. This can still cause a blowout in the loudspeakers because the final power amplifier stage can simply amplify the limited signal too much.

A regular audio limiter serves to reduce (or compress) the dynamic range of the input signal. A picture is shown in figure 4. This can be done for a wide range of reasons including aesthetic effect, to conform to sound level limitations for television or radio programs, to eliminate background noise, or to conform to the technical limitations of your audio equipment.

Amplifiers vary greatly in the range of power they use to drive speakers. The power can range anywhere from five watts to thousands of watts. In addition, speakers vary in the amount of power they can handle from a power amplifier. In many cases, the power driving the loudspeakers can make the speaker's cone move farther than it was designed to do safely. As a result the loudspeaker cone rips and the speaker is destroyed. Since an
audio limiter prevents the signal level from exceeding a preset limit, it could protect the loudspeakers.

United States Patent #4173740\(^1\) is a similar invention but instead simply “cuts off the supply voltage to a power amplifier circuit in disconnecting a loudspeaker from the output terminal of the power amplifier circuit when an overvoltage is developed at the output of the power amplifier circuit.” Our device would lower the gain of the audio signal through sophisticated hard knee or soft knee compression algorithms rather than to cut the signal entirely.

A low frequency loudspeaker processor called the LSP-1 is built with a high signal input line to monitor exactly what the gain of the power amplifier is. However for every different model of loudspeakers this device requires a PCB card that the Australian company must custom program to the specification of the loudspeakers. (http://www.arx.com.au/pdf/old/LSP1manual.pdf)

Our speaker processor would monitor the high signal being fed back and also allow the user full customization for this device to work with any general loudspeaker model.

\(^1\) [http://www.freepatentsonline.com/4173740.html](http://www.freepatentsonline.com/4173740.html)
## Division of Labor

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Table 1
# Gantt Chart

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Table 2