Modular Analog Audio Mixer

Design in Practice

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***Abstract** - Our goal is to introduce a design procedure for building a high-quality modular audio mixing console. Modular refers to a design that can be split into smaller portions (modules) so when complete, they can be joined together to form one system. A modular audio mixer is formed by assembling some main modules that can be varied in number and/or disposition to suit individual needs. Being determined to motivate electronic engineering students to be involved on analog circuits design from not just a theoretical and mathematical perspective, we have implemented an educational platform in order to encourage learning in practice.*

Keywords - *Operational Amplifiers Topologies, Audio Frequency response, Analog Electronics, Tone Control Circuit, Audio EQ Circuit*

I. INTRODUCTION

An audio mixer, also called a mixing console, is an electronic device for combining, and modifying audio signals. The modified audio signals are summed to produce some combined output signals. Audio mixers can be analog or digital. Digital mixing consoles use Digital Signal Processing concepts and analog mixers are usually based on op-amps (operational amplifiers) electronic circuits. Although digital signal processing is the current trend, analog circuits are still in use due to their simplicity, and low cost.

To illustrate the approach, we will describe our recent efforts to develop a modular design. Modular design is a form of splitting an object into smaller portions such that when they are done, they are joined together to form one complete system. A modular audio mixer is formed assembling some main modules that can be varied in number and/or disposition to suit everyone needs. In a modular mixing console, the constructor can decide how many inputs should be provided. The input audio signals could be anything from microphones, CD players, PC sound cards or any other type of analog audio sources.

Our basic mixer will be able to combine these signals from different signal sources, change the volume of each input channel as well as the overall volume of the output. Later, we will proceed to a more complex structure by adding some circuits for audio equalization. We will discuss about adding circuits for attenuating or boosting a range of frequencies e.g., bass, midrange, and treble on each audio channel and also about a general graphic equalizer to perform general equalization control at the output, a VU meter device and a headphone monitor.

II. AUDIO MIXER BASICS

A. Summing Amplifier

The heart of the mixing console is the summing circuit shown in figure 1. This circuit is also known as the summing amplifier. It consists of an operational amplifier, n input resistors (R1, R2 … Rn) and a feedback resistor (Rf). A summing amplifier sums several (weighted) voltages. Its output voltage Vo is given by the formula: Vo=-(A1*V1+A2*V2+A3*V3+…+An*Vn). Ax is the voltage gain for the xth input and it is equal to Rf/Rx.

When all input resistors (R1, R2,…. Rn), and also the feedback resistor Rf, have the same value then the voltage gain for each input channel becomes equal to the unity and the formula becomes: Vo= - (V1+V2+V3+…+Vn). The minus sign indicates that the summing output is inverted or otherwise phase shifted by 180 degrees. However, phase shift has no audible effect.

If the value of the feedback resistor, Rf, becomes greater than the value of any input resistor, Rx, then the gain for channel x, will be greater than unity (and equal to Rf/Rx).

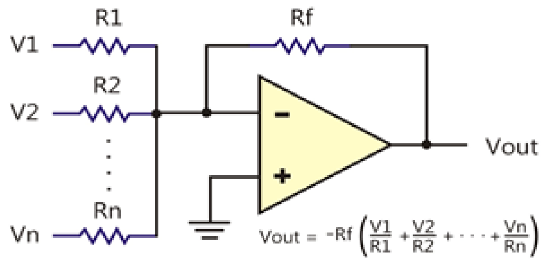

Figure 1. *The summing amplifier is itself a mixer.*

The summing circuit is actually an adder, and a basic mixer is really nothing more than an adder too, so the summing circuit is itself a mixer. This holds true but we must also notice that the summing circuit has not any adjusting elements for adjusting sound volume (voltage levels). An audio mixer usually has, and adding them should be the next challenge.

In our design, we will add some more circuits according to the block diagram of figure 2.We will add a matching circuit at every input to ensure that any signal source is not unduly overloaded. Every matching circuit will serve as a preamplifier and a volume level adjuster for each input channel, and from now on we will call it as the “input module”. The outputs of all input modules will all be combined in a summing amplifier. We will also provide a potentiometer to our summing amplifier which will be the “master” volume adjuster (for adjusting the volume of the output).

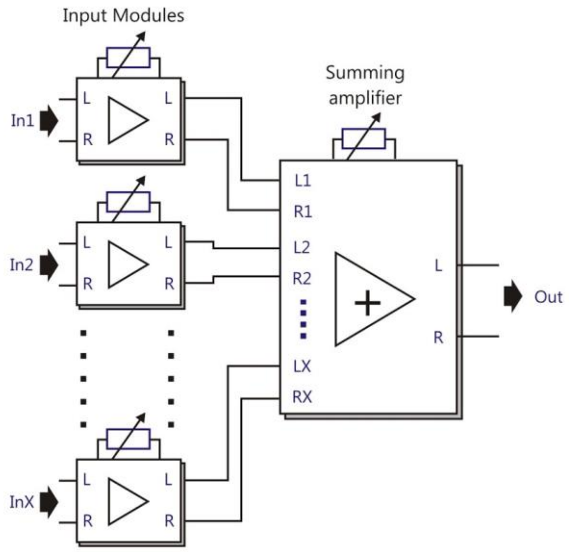

Figure 2. *A simple stereo-mixer.*

B. Line Input Modules

Most audio signals used to transmit analog sound between audio components such as CD and DVD players, TVs, audio amplifiers, and mixing consoles are Line-level signals. As opposed to line level, there are weaker audio signals, such as those from microphones and instrument pickups, and stronger signals, such as those used to drive headphones and loudspeakers. Below, we will present a line-level input module. Later, we will design additional input modules for weaker audio signals.

For designing a line-type input module, we must consider two facts:

- Cables between line output and line input are generally extremely short compared to the audio signal wavelength in the cable. So, there are no transmission lines effects and no impedance matching required. However, line level circuits use the impedance bridging principle, in which a low impedance output drives a high impedance input. A typical line out connection has an output impedance from 100 to 600 Ω. Line inputs present a much higher impedance, typically 10 kΩ or more. The two impedances form a voltage divider (see figure 3) with a shunt element (Rin) having a large resistor value relative to the resistor value of the series element (Rout), which ensures that little of the signal is shunted to ground and that current requirements are minimized. Most of the voltage asserted by the output appears across the input impedance (Rin) and almost none of the voltage is dropped across the output impedance (Rout).

- The approximate nominal voltage level of a line-type signal is about 320 mV root mean square (Vrms) or 1.2 Vrms, for consumer and professional audio equipment, respectively.

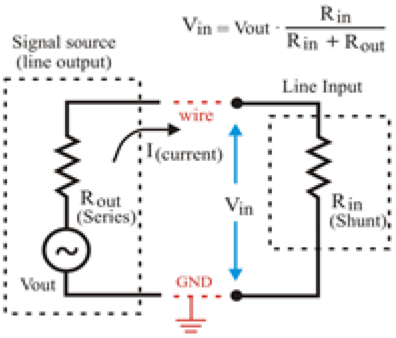

Figure 3. *Line-output and line-input impedances form a voltage divider.*

An appropriate input module circuit which conforms with the above criteria is shown in figure 4.

The circuit of figure 4 is actually an Inverting Amplifier. An inverting amplifier is a very common circuit, based on an op-amp. Its basic function is to scale (or amplify) and invert the input signal. The inversion is equivalent to a phase shift and has no audible effect.

Referring to the left-hand channel (the right-hand channel is, of course, identical) and as long as the op-amp gain is very large, the amplifier gain is determined by the external resistors (the feedback resistors R2A and R3 and the input resistor R1). The voltage gain is equal to the ratio of (R2A+R3)/R1. Moreover, the input impedance (Rin) of the circuit is approximately equal to R1 because the operational amplifier's inverting (i.e., −) input is a virtual ground.

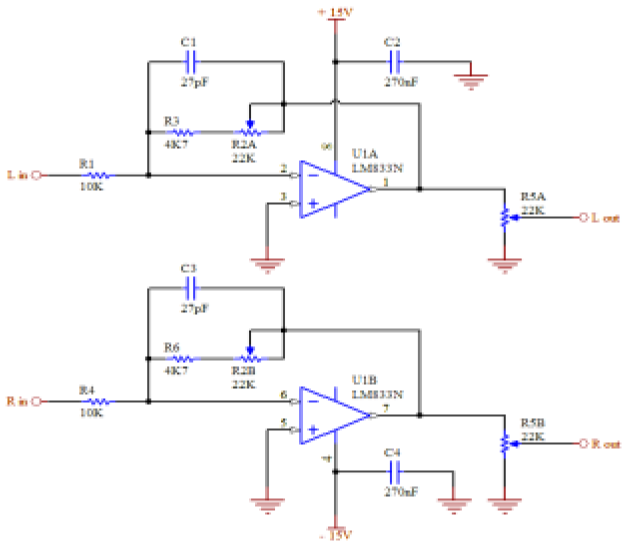

Figure 4. *A line-type input module. Left and right channels are identical. R2 and R5 are logarithmic-type stereo sliding-potentiometers.*

We have chose R1 to be equal to 10K in order to comply with the criterion of the minimum input impedance required on a line-type input. Intentionally, we use a minimum value for R4 in order to avoid thermal noise (we will discuss about this later). R3 is equal to 4.7K, and R2A can be varied from 0 to 22K, so that the voltage amplification (R2A+R3)/R1 can be varied from about 0.5 to 2.6 (from -6.5 to +8.5db). This way, R2 acts as a gain adjuster, and the nominal output level can be adjusted from 160 to 830mV rms or from 0.6 to 3.12 V rms, for consumer and professional audio equipment, respectively.C1 is used for high-frequency noise filtering and for preventing oscillations. Together with R2 and R3, it forms a low pass filter. Filter's cut-off frequency is above the upper limit of the audible range (20Hz-20KHz), and varies inversely proportional to the change in value of R2 (see figure 5). The R5 potentiometer implements an adjustable voltage divider and acts as the volume level adjuster.

The circuit off figure 4 is a good example of what can be used as a line-input module in a mixer. Of course, many other circuits can also used for the same purpose. Any voltage preamplifier which has adequate input impedance (equal or higher than 10K) and has also a volume level adjuster can be used at the input stages of a mixer. Actually, there are some additional requirements for the right candidate. The input stages must not produce any noise or distortion, they must have flat frequency response from 20Hz to 20KHz (audible range) and must also be stable. Almost everything depends on the right choice of the operational amplifier, the use of appropriate filtering and also the use of as small as possible resistor values.

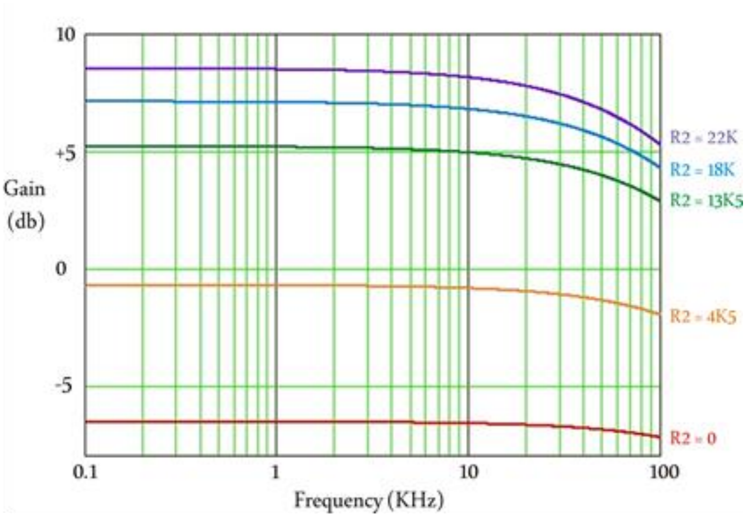

Figure 5. *Line-input module frequency response for different wiper positions of R2*

In the prototype (see photo 3), we use the classic LM833 dual operational amplifier which has been designed with particular emphasis on performance in audio systems. We also use as small as possible resistor values in order to avoid thermal noise. It is well known that any resistor produces some thermal noise voltage which is proportional to the resistor's value.

So, minimizing resistor values is essential for minimizing thermal noise level. Unfortunately, for a single stage circuit as this of figure 4, choosing the input impedance to be at least 10K, somehow limits our choices.

Thermal noise power is also proportional to the total bandwidth. Keeping the total bandwidth as low as possible is essential for reducing noise level. For this purpose, low pass-filtering is used in every audio mixer for rejecting spectral content above 20KHz (the upper limit of the audible range). Although filtering improves the overall SNR (signal to noise ratio), it also affects the frequency response of the preamplifier, and it is somehow difficult to ensure both good filtering and flat frequency response (at the entire audio range 20Hz-20KHz) at the same time. In our circuit we use rather simple filtering and we mostly focus on achieving flat response (see figure 5).

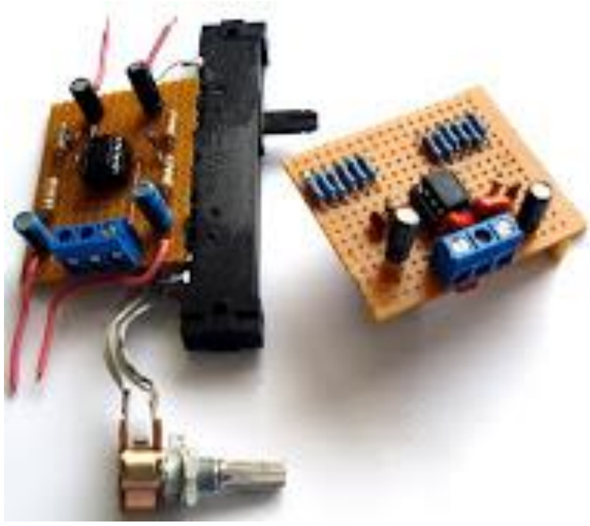

Photo 3. *The Line-input module (at the left) and the summing amplifier module (at the right), built on prototyping matrix-boards.*

There is also another important notice regarding the circuit of figure 4. We use DC-coupling for flat response even at very low frequencies. This is an advantage as far as the input source does not have any DC-leakage (offset). If there is any DC at the input, it will be amplified and will pass at the output, and may saturate the next stages. Adding a DC-blocking capacitor at the input (in series with R1) will solve the problem. The capacitor will limit somehow the flatness at very low frequencies and some low frequency content may be attenuated, unless a large enough value is used (the recommended value is about 10uF or more).

C. The Summing Amplifier Circuit

The summing amplifier circuit of our mixing console is shown in figure 6. Referring to the left-hand channel (the right-hand channel is identical), R7A is the feedback resistor and R1L, R2L,....RXL, are the input resistors. The feedback resistor is a 4.7K stereo potentiometer (R7) which is used to enable the output level to be matched to the sensitivity of the unit to which the mixer is connected. In other words, R7 acts as the master-level adjuster.

The input resistors (R1L-RXL or R1R-RXR) have the same value and they are all equal to 4.7K, so that the gain, which is equal to R7/R1, can be varied between 0 and 1 (negative infinity to 0db). C8 performs basic filtering, and is used to attenuate high-frequency signals in order to reduce noise level. There is also a series RC network at the output. The purpose of this network is to prevent any DC-offset appearing at the output of the mixer and also obviate any tendency to oscillation caused by a capacitive load (such as long screened cables). Using the modular-design concept (see photo 4), the summing amplifier module can be easily assembled on a prototyping matrix-board (see photo 3).

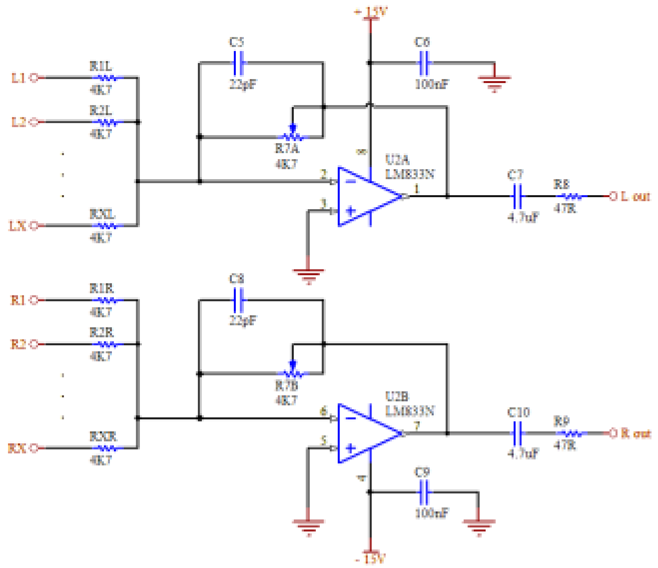

Figure 6. *The summing amplifier circuit. Left and right channels are identical. R7 is a logarithmic-type stereo sliding-potentiometer.*

Photo 4. *The mixer uses the modular-design concept. However, building some modules on the same board minimizes the number of cable connections required.*

D. Additional Outputs

Besides the main output, most mixers usually have additional outputs. The main output is usually connected on a final amplifier and other outputs are usually intended to be used as signal sources for monitoring or recording equipment.

Adding outputs is rather a simple task and it is based on the concept of using one or more voltage followers. A voltage follower is a basic op-amp circuit. It is called a "follower" due to its ability to "follow" its input without any loss. It is actually a unity-gain voltage amplifier which has very high input impedance, and very low output impedance at the same time, thus providing maximum isolation between its input and output. The voltage follower circuit is shown on figure 7.

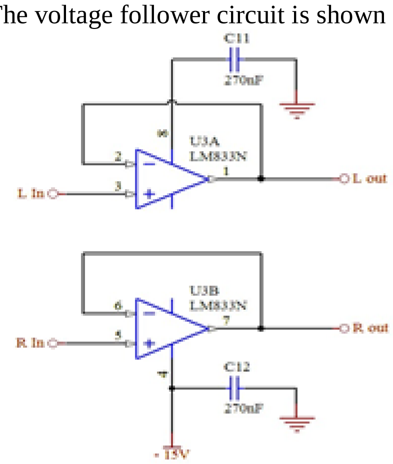

Figure 7. *The voltage-follower circuit. It is actually a unity-gain voltage amplifier which provides maximum isolation between its input and output.*

By adding one or more voltage followers, having their inputs connected on the summing amplifier's output, we will be able to build a mixer (see photo 5) with as many outputs as we wish. Due to the high input impedance of the voltage-follower circuit, we can connect as many followers as we wish on the summing amplifier's output, without overloading it. (see figure 8). All outputs from the followers will provide the same audio signal but they will be totally isolated one from another

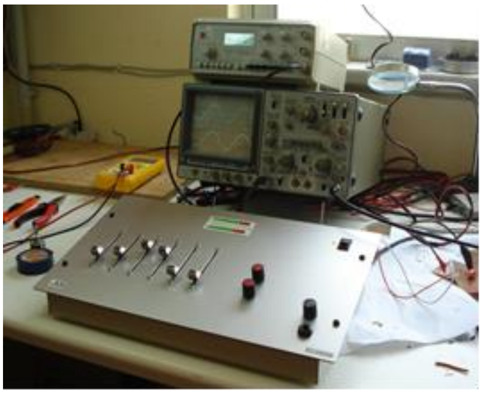

Photo 5*. Testing the mixer at the lab.*

If we wish to be able to independently adjust the signal level in each output, we may use a potentiometer at the input of every voltage follower. The wiper of each potentiometer should then be connected to the input of each voltage follower (and the other two pins of each potentiometer should be connected to the summing amplifier's output and the ground, respectively). The specific arrangement is shown on figure 9.These potentiometers will then act as volume-adjusters for each output, and one of them may serve as the “master” volume adjuster. In this specific arrangement, R7 (see figure 6) is no more needed to be used as a volume adjuster, and it may be replaced with a fixed value resistor (or a trimmer resistor).

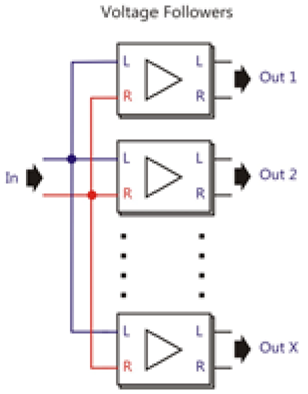

Figure 8. *Adding outputs using one or more voltage followers.*

Using potentiometers at the inputs of the voltage followers, reduces their input impedances. The total impedance “seen” by the summing amplifier will be reduced also, and will be about Rpot/N (where Rpot is the resistor value of each potentiometer, and N is the total number of potentiometers). This will limit somehow the maximum number of voltage followers can be connected at the main output. However, by using potentiometers of about 47K each, adding up to 4 or 5 outputs will not be a problem.

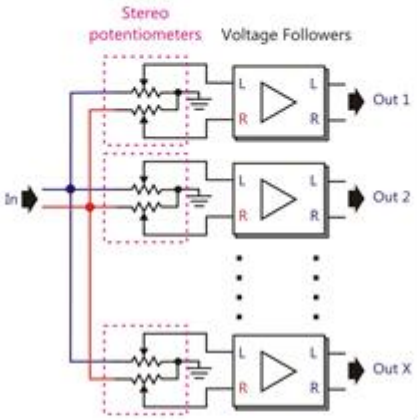

Figure 9. *Using potentiometers at the inputs of the voltage followers enables independent volume adjustment for each output.*

D. Adding Tone Control

Tone control allows listeners to adjust sound to their liking. It also enables them to compensate for recording deficiencies, hearing impairments, room acoustics or shortcomings with playback equipment.

Our mixing console is of modular design. Not only we can build as many input channels as we wish but we can also upgrade the design to use some additional modules for equalization (tone control). Such an upgraded design (see photo 6) which uses an additional module for tone control at each input is presented in figure 10. Obviously, this is not the only possible arrangement. Different arrangements may use tone control modules only at specific inputs, instead of all. You may also use one tone control module at the output (at the output of the summing amplifier).

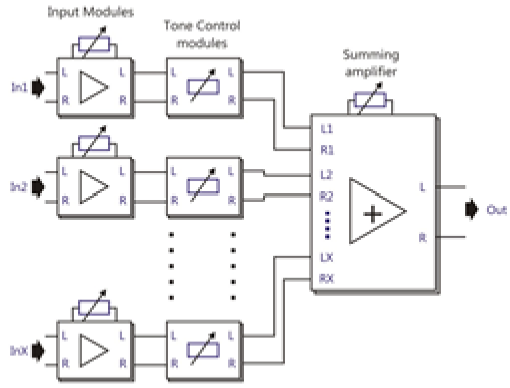

Figure 10. *An audio mixer which uses tone-control modules.*

Usually, the additional module required for tone control is a 2-bands or a 3-bands tone control circuit. Example designs for tone controls circuits, are presented in figures 11 and 13.

Photo 6. *This prototype has 12 inputs and uses 3-bands tone control modules at all of them.*

The circuit of figure 11 is a 2-bands tone control, and it is able to attenuate or boost bass and treble (2-frequency bands). The circuit is based on design formulas, found on page 14 of LM833-N's datasheet. R1 and R2 potentiometers provide independent control of bass and treble frequencies, respectively; both bass and treble can be boosted (when wipers is on left) or cut (wipers on right) and with both controls at their mid positions, the circuit provides a relatively flat frequency response.

The 2-bands tone control circuit (see photo 7) uses one low pass and one high pass filter at each audio channel. Both filters are almost flat-top at their pass-band, and the flat-top gain of any filter can be varied between -20 and +20db, independently from the other one. The cut-off frequencies (-3db points) for the low and the high pass filters are preset at 34Hz and at 11 KHz, respectively (see figure 12). If you wish to alter them, refer to the design formulas.

The electronic schematic is quite simple and the circuit can be easily built on a universal board (prototyping matrix-board). However, we must take care to use low tolerance components to ensure the same response for both (R and L) audio channels. The circuit operates normally when connected to a low-impedance voltage source (like the line-input module of our mixer). Due to DC-coupling, any DC present at the input may cause some problems. So, use DC-blocking capacitors, if necessary.

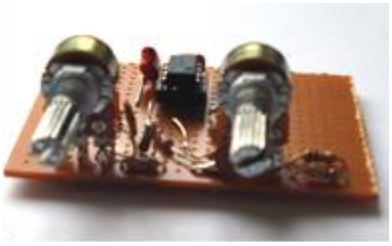

Photo 7. *A 2-bands control module, built on a prototyping matrix-board.*

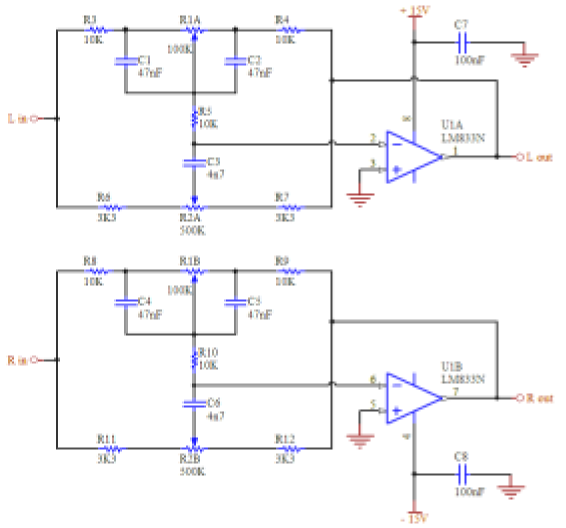

Figure 11. *A 2-bands tone control circuit*

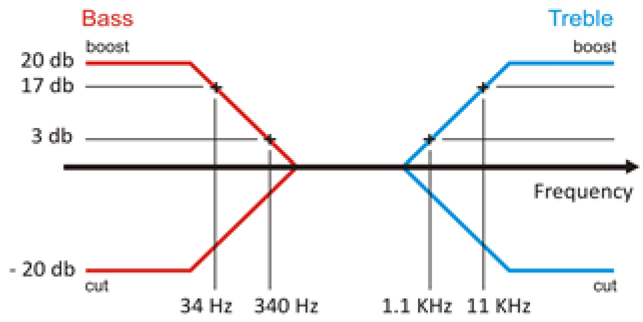

Figure 12. *2-bands tone control circuit frequency response (Bode plot)*

The circuit of figure 13 is based on the popular low–noise NE5532 operational amplifier and has three separate adjustable filters. The first one is an adjustable-gain low pass filter; the second one is a band –pass adjustable filter and the third one is an adjustable high-pass filter. Each filter enables the adjustment for bass, mid and high frequencies, respectively (see figure 14). The cut- off frequencies (-3db points) for the bass and high frequency filters are about 20 Hz and 8 KHz, respectively. The central frequency of the band-pass filter is about 1 KHZ. The normal gain, G, (when pots are at midrange) is given by G=20*LOG (R2/R1). By using equal resistor values for R2 and R1, the normal gain becomes equal to 0db.

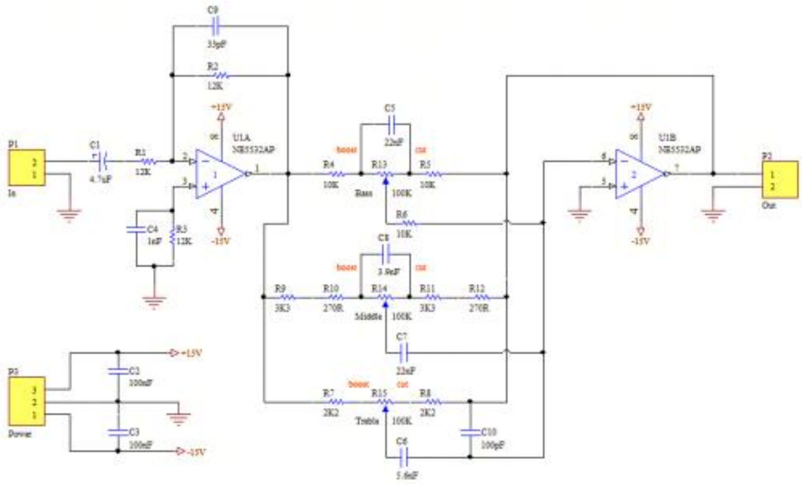

Figure 13. *A 3-bands tone control circuit*

In figure 13, and due to limited space available, only the right-hand audio channel circuit block is presented. The same circuit block, should be also used for the left-hand audio channel

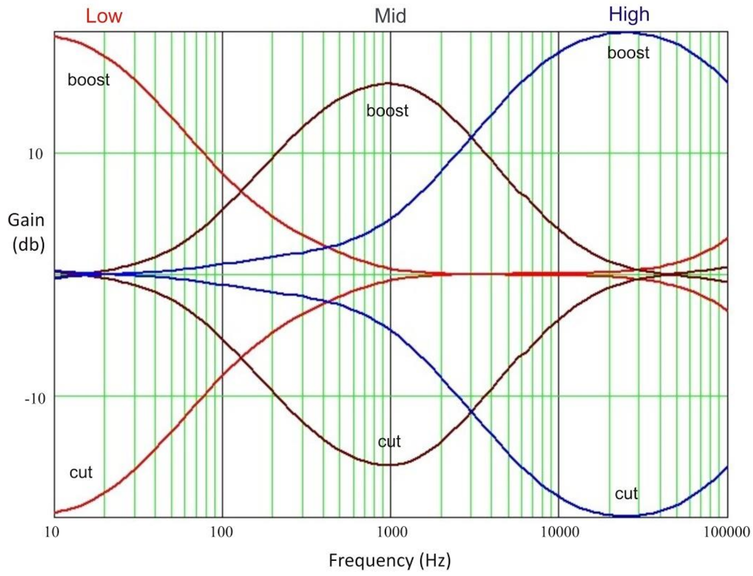

Figure 14. 3-bands tone control circuit frequency response (simulation on Electronics Workbench software)

D. Adding an Equalizer

At this point, our mixer uses relatively simple filters for limited adjustments. Graphic and parametric equalizers have much more flexibility in tailoring the frequency content of an audio signal than a simple tone-control module. An audio equalizer is actually a bank of many adjustable filters. Using the modular concept, we can use one equalizer at every input of our mixer. However, since an equalizer is a quite complex and expensive circuit, it is more practical to use it only once, at our mixer's output as shown in figure 15.

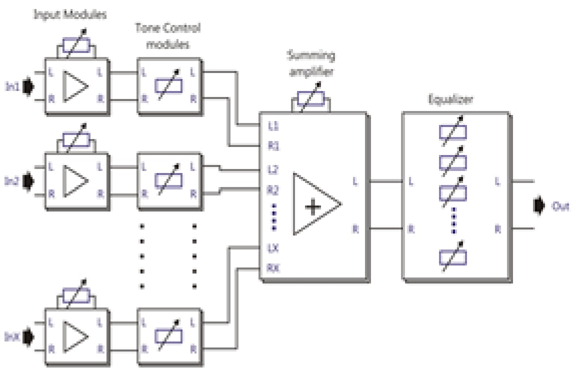

Figure 15. *Adding an equalizer at mixer's output*

A reference design for a graphic audio equalizer is presented in figure 16. The specific circuit is based on Philips Semiconductors Application Note 142 (first published on October 1984). The circuit itself has great performance and uses a top performance operational amplifier; the NE5532. The graphic-equalizer consists of an input buffer (IC1 -a), several variable-boost/ cut active filters (lC2-a), and an output summing amplifier (IC1-b). The IC1-a circuit is designed for unity gain and is used mainly for impedance-matching between the input source and the equalizer filters. Each filter is a variable-bandpass or notching device, depending on the setting of the control potentiometer R2 (see frequency response on figure 17).

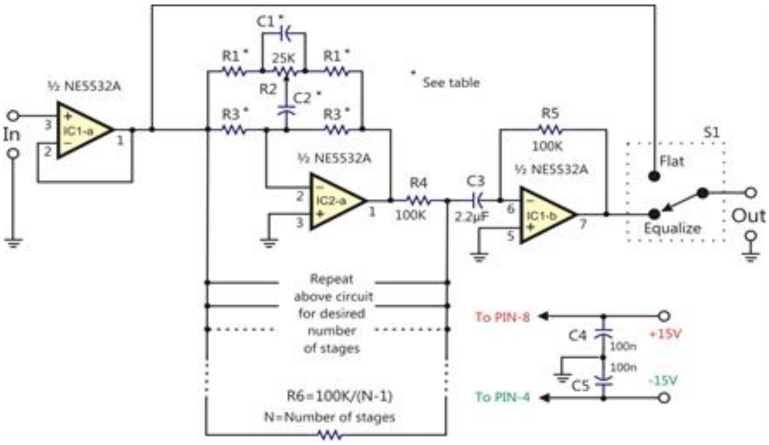

Figure 16. *A reference design for a graphic audio equalizer*

Any number of equalizer filter-stages can be used within the range of about 20 Hz to 20 KHz. The more stages you have, the easier is to boost or cut a particular frequency without affecting the response at adjacent frequencies. All the filter stages use the same R-C feedback-network configuration, to provide a maximum of about 13-dB of boost or cut at Fo, their center frequency. Different values for C1 and C2 are used in each stage, for setting the value of Fo.

A list for the values of R1, R3, C1 and C2 for a 7-bands graphic equalizer is presented on table 1. Values for different arrangements can be found on page 3 of the Philips Semiconductors Application Note 142 (which now is available for download from NXP's web site). C1 is about ten times as large as C2 and the values for R1 and R3 are both related to the value of R2, approximately by a factor of 10. The center frequencies of the list have been selected so that C1 and C2 are standard, off-the-shelf, values. The equalizer uses linear slide potentiometers for R2.

The value of R6 depends on the number of filter stages used. It insures that the gain across the equalizer is unity when all controls (R2's) are in the FLAT or 0 dB position. The nominal value of R6 is 100K (the value of R4-R5) divided by N-1, where N is the number of stages used. For a 7-bands equalizer, the nominal value of R6 is 100K/6, which equals about 16.7K. Note that only one audio channel is shown in the circuit schematic. In order to build a stereo version of the Audio Graphic Equalizer you'll need two of those circuits.

R1=2.4K, R3=240K, R2=25K, R6=16.7K		
Fo	**C1**	**C2**
60 Hz	0.47uF	0.047uf

158 Hz	0.15uF	0.015uF
425 Hz	0.056uF	0.0056uF
1082 Hz	0.022uF	0.0022uF
2382 Hz	0.01uF	0.001uF
6000 Hz	0.0039uF	390pF
15880 Hz	0.0015uF	150pF

Table 1. *7-bands Audio Equalizer component values*

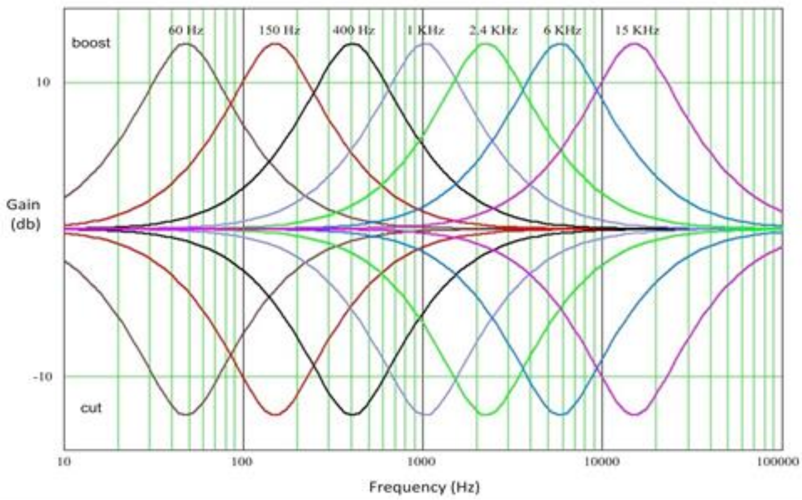

Figure 17. *7-bands audio equalizer frequency response (simulation on Electronics Workbench software)*

The circuit described above is based on the concept of adding signals from several filters outputs. There are also other interesting reference designs for audio equalizers, based on different concepts. One of them is the one described on Texas Instruments AN-*435 (Designing with the LMC835 Digital-Controlled Graphic Equalizer)*. The topology described on AN-435 is based on the gyrator concept rather than summing filters outputs.

E. Adding a VU meter

A Vu meter (Volume Unit Meter) is a device which is used to display a representation of the signal level. Think the Vu meter as a special kind of voltmeter which can be connected directly to any audio signal line of interest similarly to a high impedance voltmeter or oscilloscope input, measuring the voltage at the specific line of interest while drawing minimal current (and hence minimal power) from the source. As long as the vu-meter has enough input impedance it will not affect the performance of the mixer circuits.

Most mixers have only one vu meter at their output lines, but some quite expensive mixing consoles offer independent signal level indication for every input channel. Since we have a modular design we can use as many vu-meters as we like by simply connecting them directly to the signal lines of interest (for example, at mixer's output or at any input-module's output).

An accurate LED-type stereo VU meter (see photo 8), based on the well-known Texas Instruments LM3915 IC is presented on Figure 18. The specific VU meter operates from a single supply voltage in the range of 3V to 6V. The maximum supply current at full scale is about 400mA and it is not dependent on the power supply voltage.

Photo 8. *A bar-graph LED volume unit (VU) meter module, based on LM3915 IC*

The LM3915 senses analog voltage levels and drives ten LEDs on a bar-graph, providing a logarithmic 3 dB/step analog display. We use 2 x LM3915 ICs; one for the Right (R) and another one for the Left (L) audio channel. We also use the MCP6022– dual operational amplifier IC as a Precision Half-Wave Rectifier and Preamplifier.

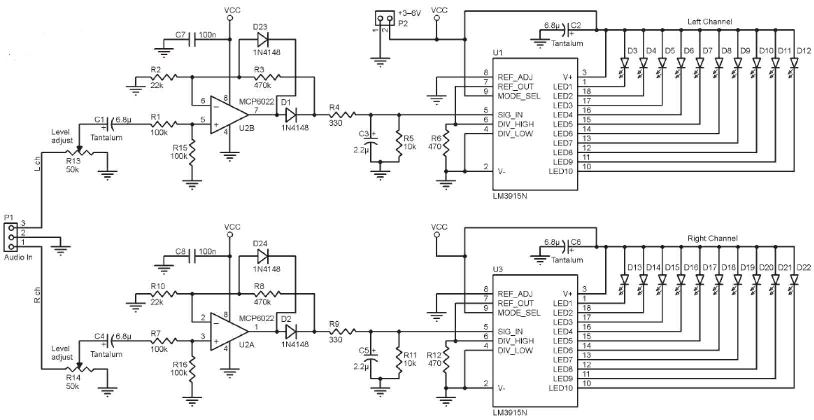

Figure 18. *The stereo volume unit (VU) meter circuit*

The input impedance in each channel (R and L) is more than 33K. R13 and R14 are used for full-scale voltage level adjustment. The minimum full-scale voltage level can be as low as 150mV. Vu-meters used at mixers outputs, are usually adjusted for full-scale reading at about 1.4V. C3, R4, R5 and C5, R9, R11 networks are simple integrators and they adjust the vu-meter's rise and fall times for the L and R audio channel, respectively. The VU-meter intentionally "slows" measurement, averaging out peaks and troughs of short duration. The "speed" of measurement is preset according to author's preferences but can be easily set to another value.

By replacing C3, R4, R5 and C5, R9 and R11 and after some trial and error you will be able to find the ideal values according to your preferences. Once a time, there were some standards for VU meters response but I think they mostly concern the case of old passive electromechanical devices.

F. Adding Microphone Inputs

Since now, the input stages are unable to provide significant gain for a microphone source. Our mixer has only “line-type” audio inputs. We can overcome this limitation by adding a microphone preamplifier at any channel we wish to convert it from a “line-type” audio input to a “microphone-type” one. The specific method is illustrated in figure 19.

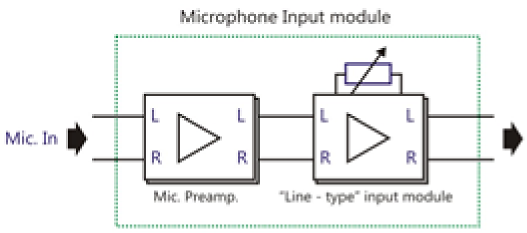

Figure 19. Converting a line-type audio input to a microphone-type one.

Two reference designs for microphone preamplifiers are presented in Figures 20 and 21. The circuit of figure 20 is based on the MAX4468 IC from Maxim, and it is optimized to be used with an electret microphone-capsule. The second circuit (figure 21) is a balanced microphone preamplifier based on the NE5534 operational amplifier, and it is an excellent choice for low impedance dynamic microphones.

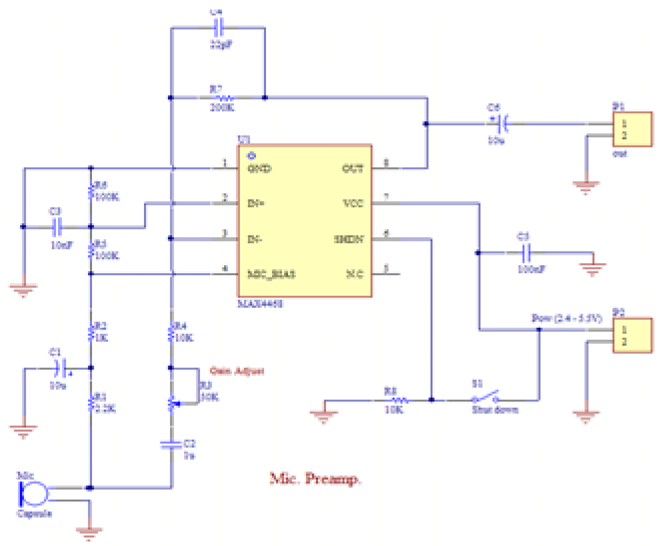

Figure 20. A microphone preamplifier, optimized optimized to be used with an electret microphone-capsule

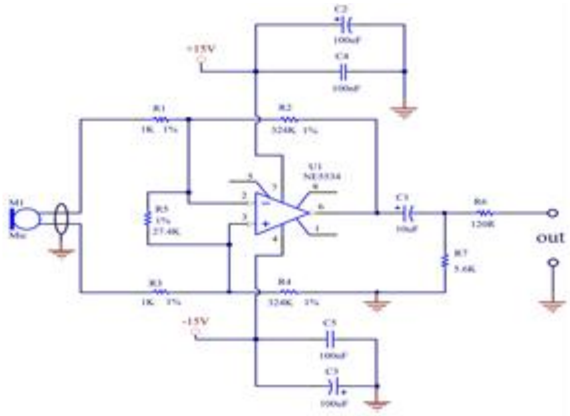

Figure 21. *A balanced microphone preamplifier for low impedance dynamic microphones.*

G. Adding “Phono” Inputs

A “phono” input refers to a Phonograph input. It is a special type input which can accept signals from an analog turntable or a magnetic guitar-pickup (or some specific other types of equipment). A phono input always uses a special circuit to boost the incoming signal and also provides the RIAA equalization necessary to restore the original sound. If you are still enjoying vinyl sound or you have a guitar which uses a RIAA magnetic pickup, you definitely need a phono input in order to be able to connect your classic turntable or your beloved music instrument to the mixer. Again, all you need is to convert one “line-type” audio input to a “phono-type” input by simply adding a phono preamplifier as illustrated in figure 22.

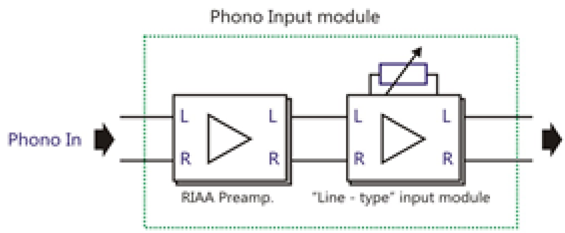

Figure 22. *Converting a line-type audio input to a phono-type one.*

Reference designs for RIAA preamplifiers can be found on page 14 of LM833-N's datasheet, on Texas Instrument's AN-346 application report on High-Performance Audio Applications of the LM833, and on Philips Semiconductors Application Note 142 (which is available for download from NXP's web site).

H. Adding Headphones Monitor

If you wish to add headphone monitoring capability in your audio mixing console, you need to use a headphone amplifier. With it you will be able to monitor the output of the mixer or any input. A headphone amplifier is just a small stereo power amplifier which provides a sufficient output to operate a pair of standard headphones.

A stereo headphone monitor circuit which is especially designed to be embedded in a homemade audio mixer is shown in figure 23. The circuit is equipped with a stereo volume-control potentiometer (R2) which allows the sound to be adjusted to a comfortable listening level. Due to its small size (see photo 9) it may be mounted inside an audio mixer by using the volume-potentiometer bush fixing alone.

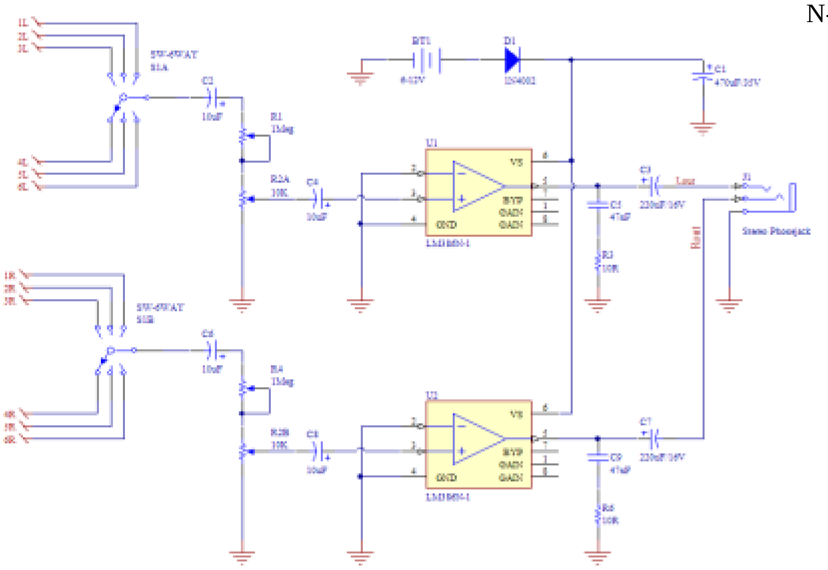

Figure 23. *A stereo headphone monitor circuit especially designed to be embedded in a homemade audio mixer*

The circuit comprises two sections and a small number of components common to both sections. The first section is based on U1 and it is associated with the Left audio channel. The second section is based on U2 which is responsible for the Right audio channel. Integrated circuits U1 and U2 are LM386 amplifiers. LM386 can provide up to 325mW into an 8Ω load. Standard headphones usually have greater impedance than this, so the available output will be reduced. However, this does not matter because only a very small output is sufficient to fully load the headphones. Since the components used on both audio sections are identical, only a description of a single channel is needed:

For any input signal level, the preset potentiometer R1 must be adjusted so that, when volume control (R2) is at maximum, there is minimal distortion combined with sufficient volume. R4, which is the preset potentiometer for the Right audio channel, must also be adjusted so that there is a correct balance (equality) in the volume between the left and the right channels.

The amplified signal of the Left channel appears at the output pin (pin 5) of U1 and AC-coupled to the left-hand headphone threw C3. Capacitor C5 , which is connected in series with R3, is used to stabilize the amplifier and prevent any oscillation that might otherwise occur. The circuit can be powered from a 6-15V power source threw D1. D1 is used to provide reserved-polarity protection.

S1 is a stereo, 6-way rotary pickup selector switch, which is used to select a specific input source (from 6 in total). In order to use the headphone monitor in your homemade audio mixer, you should connect inputs 1L, 2L,...6L and 1R, 2R,...6R, to the L and R outputs, respectively, of the modules you wish to monitor. Think the headphone monitor as a special kind of a signal probe which can be connected directly to any audio signal line of interest similarly to a high impedance voltmeter or oscilloscope input. Using a 6-way selector switch, you will be able to monitor up to 6 modules. For N inputs, you will need a N-way selector switch.

Photo 9. *The headphone monitor module (without the rotary pickup selector switch)*

I. Power Supply Unit

For powering the audio section, we use a simple linear power supply unit which is based on 7815 and 7915 linear regulators. Referring to the power supply electronic schematic (figure 24) , U10 and U11 are used to provide +15V and -15V respectively. A current of less than 1A is enough to power up to 50 modules.

We use a separate 5V power supply unit (see photo 10) for the VU-meter. The PSU used to power the VU-meter is based on Texas Instruments LM7525N-5 switch-type regulator (see figure 25).

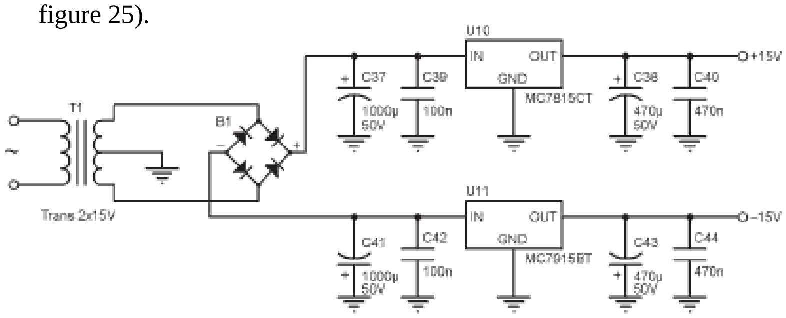

Figure 24. *Simple linear power supply unit which is based on 7815 and 7915 linear regulators*

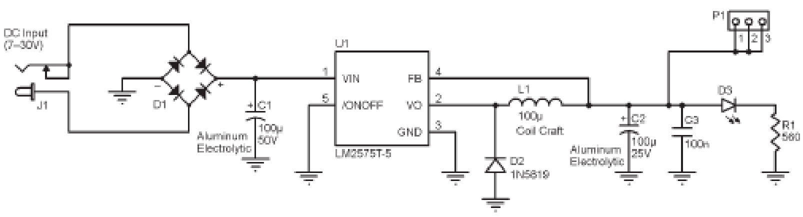

Figure 25. *The PSU used to power the VU-meter is based on Texas Instruments LM7525N-5 regulator*

Photo 10. *The small PSU, used to power the VU-meter*

CONCLUSION

The target of this article was to introduce a design procedure for building a high quality, modular, audio mixing console. The modular design procedure refers to a design which can be split into smaller portions (modules) such that when they are done, they can be joined together to form one complete system. A modular audio mixer is formed assembling some main modules that can be varied in number and/or disposition to suit everyone needs. The examples presented here are only for reference. The constructor is free to use the examples as they are or to use different configurations than those described here. The circuits described here are also for reference. They have been built and tested and exhibit good performance. However, we can not guarantee that they are the best of their kind. There is always room for improvements and you may always refer to bibliography, if you are looking for them.You may notice that building an audio mixing console from modules, requires many wires. However, after taking decisions about the number of modules you wish to use and their specific arrangement, you may design a printed circuit board which will minimize the number of cable connections required.

Photo: *The complete Modular Analog Stereo-Mixer Console with one microphone input,four line-inputs,a voltage unit (VU) meter, bass and treble tone controls, and a headphone monitor.*

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