DOKUZ EYLÜL UNIVERSITY
GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES

ELECTRONIC CIRCUIT DESIGN FOR AUDIO NOISE ELIMINATION

by
Şekibe Şebnem TAS

July, 2008
İZMİR

ELECTRONIC CIRCUIT DESIGN FOR AUDIO NOISE ELIMINATION

A Thesis Submitted to the
Graduate School of Natural and Applied Sciences of Dokuz Eylül University
In Partial Fulfillment of the Requirements for the Degree of Master of Science
in Electrical and Electronics Engineering

by
Şekibe Şebnem TAS

July, 2008
İZMİR

M.Sc THESIS EXAMINATION RESULT FORM

We have read the thesis entitled **"ELECTRONIC CIRCUIT DESIGN FOR AUDIO NOISE ELIMINATION"** completed by **ŞEKİBE ŞEBNEM TAS** under supervision of **ASSOC. PROF. DR. UĞUR ÇAM** and we certify that in our opinion it is fully adequate, in scope and in quality, as a thesis for the degree of Master of Science.

....

Assoc. Prof. Dr. Uğur ÇAM

....

(Jury Member) (Jury Member)

Prof.Dr. Cahit HELVACI
Director
Graduate School of Natural and Applied Sciences

ACKNOWLEDGEMENTS

I would like to thank my advisor Assoc. Prof. Dr. Uğur Çam for his support, encouragement and valuable guidance. I am grateful to him for letting me share his professional knowledge and experience.

I would like to thank TÜBİTAK for supporting me financially during my master study.

I also would like to thank my colleagues at Vestel R&D department who helped me a lot and gave me the oppurtunity to use R&D laboratories during my research. I am also grateful to Kürşat Sarıarslan for his help and for sharing his experience with me during my study.

Finally, I would like to thank my parents for always helping and supporting me throughout my life. I love them all.

Şekibe Şebnem TAS

ELECTRONIC CIRCUIT DESIGN FOR AUDIO NOISE ELIMINATION

ABSTRACT

As we all know, today's converged portable multimedia devices have more and more features integrated into their smaller systems. Audio is a basic feature of any system marketed with multimedia capability. Engineers often put more focus on portable multimedia devices that have features such as wireless connectivity, video processing, image capture and display. Hence, audio circuits end up wedged in the system wherever space can be found among other components, resulting in mediocre or downrigth poor audio quality.

There are many sources of poor audio quality in portable audio systems, but this thesis focuses on sources of audible noise on analog audio signals. Non-harmonically related noise, whether white(flat) or tonal, can be very annoying to the end user. White noise is perceived as a background hiss, which is very noticeable during quiet audio passages. Tonal noise can be perceived as a buzz, hum or whine, depending on the frequency content.

Audio noise can be encountered in all audio visual equipment such as televisions, dvd players, home theatre systems etc. Although there are different techniques used to reduce the audible noise and to increase the audio quality in such systems, designing a system that is totally silent is almost impossible. In this thesis, a new technique is proposed to prevent the audio noise in portable audio equipments. This new technique is based on a circuit mounted on the motherboard of the equipment and it is intended to prevent extraneous noises emanating from speakers. The designed circuit is implemented on an LCD(Liquid Crystal Display) television and has reduced the audio noise significantly.

Keywords: mains hum, ground loop, audio noise, electromagnetic interference

SESTEKİ GÜRÜLTÜYÜ GİDEREN ELEKTRONİK DEVRE TASARIMI

ÖZ

Hepimizin bildiği gibi, kullandığımız çokluortam cihazları gün geçtikçe daha fazla özelliği daha küçük sistemler içinde entegre edecek şekilde tasarlanmaktadır. Ses bu cihazların sahip olduğu en temel özelliklerden birisidir. Kablosuz bağlantı, görüntü işleme gibi birçok özelliğin çokluortam cihazları tarafından desteklenmesi ses devrelerinin fiziksel olarak daha küçük alanlara yerleştirilmesine ve ses devrelerinin çalışmasının etkilemensine neden olur. Bunun sonucu olarak da cihazlarda ses kalitesinde bozulmalar gözlenmektedir.

Elektronik sistemlerde düşük ses kalitesinin birçok nedeni olabilir, ancak bu tezde analog ses sinyallerindeki bozulmanın nedenleri ve duyulabilir gürültü incelenmiştir. Duyulabilir gürültü, ses sinyalinden bağımsız beyaz gürültü şeklinde ya da ses tonuna bağlı gürültü şeklinde ortaya çıkabilir. Bu gürültü özellikle düşük ses seviyelerinde dinleyici tarafından hışırtı ya da uğultu şeklinde duyulmaktadır.

Televizyon, bilgisayar, ev sinema sistemi gibi birçok işitsel görsel elektronik sistemde ses gürültüsü sistemin kalitesini düşüren önemli faktörlerden birisidir. Bu sistemlerde birçok özelliğin entegre olarak bir arada desteklenmesinden dolayı duyulabilir gürültüyü tamamiyle yok etmek mümkün olmasa da farklı teknikler kullanarak gürültüyü azaltmak mümkündür. Bu tezde, tasarlanan elektronik bir devre yardımıyla cihazdan yayılan duyulabilir gürültünün azaltılması amaçlanmaktadır. Uygulamada tasarlanan elektronik devrenin çalıştırılması amacıyla bir LCD(Liquid Cyrstal Display) televizyon kullanılmıştır. Devre televizyonun anakartına monte edilerek çalıştırılmış ve gürültünün önemli ölçüde azalması sağlanmıştır.

Anahtar kelimeler: şebeke gürültüsü, toprak döngüsü, ses gürültüsü, elektromanyetik girişim

CONTENTS

Page

CHAPTER ONE

INTRODUCTION

The term noise is defined as anything that obscures a signal. It has a physical origin and is random in nature. The noise can be characterized by frequency spectrum, amplitude distribution and the physical mechanism that generates it. Understanding the mechanism and to keep it under control is essential in the design and performance of any device, because it sets limits as minimum signal voltage or maximum speed at which the device can work. In this thesis, different factors which cause the audible noise in electronic systems are investigated and a new technique is proposed to eliminate this noise. Noise measurements are implemented on an LCD television motherboard and the performance of the proposed circuitry is investigated also on the television motherboard.

In the second chapter, the characteristic properties of noise is explained and the physical sources that generate noise in electric circuits are given. Also, a classification is made according to the frequency spectrum of noise which is generally used to define different noise types.

In the third chapter, main focus is the audible noise in electronic systems. Audible noise is called hum or hiss noise depending on its frequency content. This noise is generated due to interference with unwanted signals and poor grounding and is usually heard as hum, buzz, clicks or pops in audio signals or seen as hum bars or specks in video signals. This noise is generally much more noticeable and irritating than the other noises in the designed audio and video systems.

In the fourth chapter, audio noise is measured on an LCD television. The details of the measurement technique and the components of the TV that generate the audio noise are also explained.

In the fifth chapter, the proposed circuitry which is designed to overcome the audio noise problem is explained. The circuitry is connected to the LCD television motherboard and the obtained results are compared with the problematic case. Also, different techniques are investigated to overcome the audio noise problem in the television.

And in the last chapter, a conclusion is given which summarizes the whole study.

CHAPTER TWO
NOISE FUNDAMENTALS

Noise is a term that is defined as anything that degrades or impairs the signal. In electronics, noise can refer to the electronic signal corresponding to acoustic noise (in an audio system) or the electronic signal corresponding to the visual noise commonly seen as snow on a degraded television or video image. In signal processing or computing it can be considered data without meaning; that is, data that is not being used to transmit a signal, but is simply produced as an unwanted by product of other activities. In information theory, however, noise is still considered to be information. In many of these areas, the special case of thermal noise arises, which sets a fundamental lower limit to what can be measured or signalled and is related to basic physical processes at the molecular level described by well known simple formulas. (Noise, 2008)

As noise is random, it can only be predicted by statistical means, usually with a Gaussian probability density function as shown below:

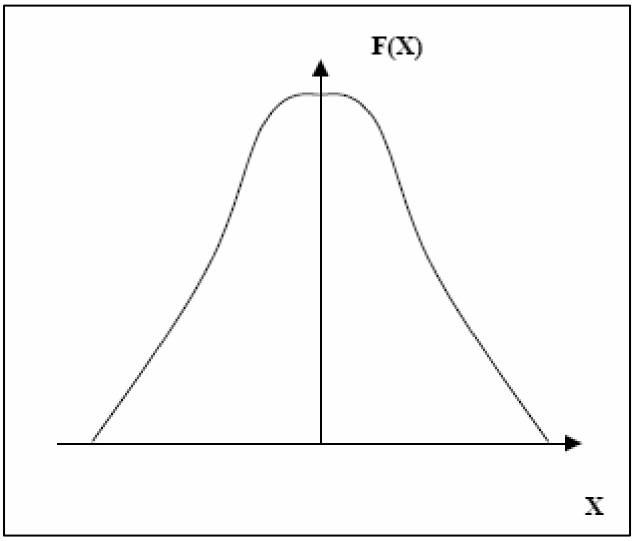


Figure 2.1 Gaussian probability density function
(RFIC Theory Tutorials, 2003)

2.1 Noise Power Spectral Density

Noise power spectral density describes the noise content in a 1Hz bandwidth. It has the units watts/hertz and is denoted by S(f). The noise power can be calculated by integrating the power spectral density function over the specified bandwidth. The figure below shows the power calculation of a noise signal which has a flat power spectral density equal to No.

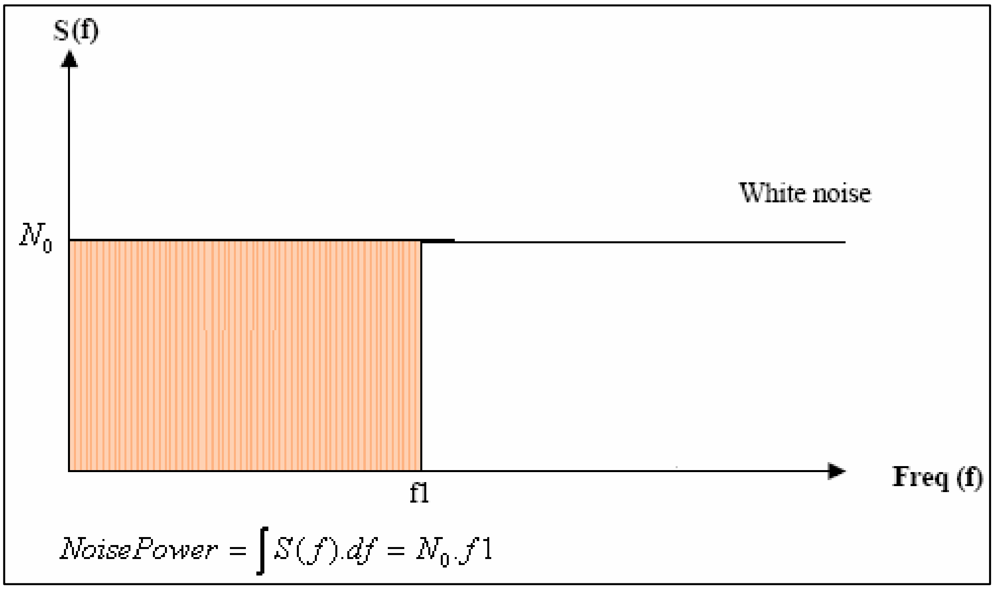


Figure 2.2 White noise power spectral density function and the power calculation (RFIC Theory Tutorials, 2003)

2.2 Equivalent Noise Bandwidth

Equivalent noise bandwidth is defined as the frequency span of a noise power curve with an amplitude equal to the actual peak value, and with the same integrated area. In other words, the equivalent noise bandwidth describes the bandwidth of a 'brick wall' system with the same noise power as the actual system. The figure below shows two examples.

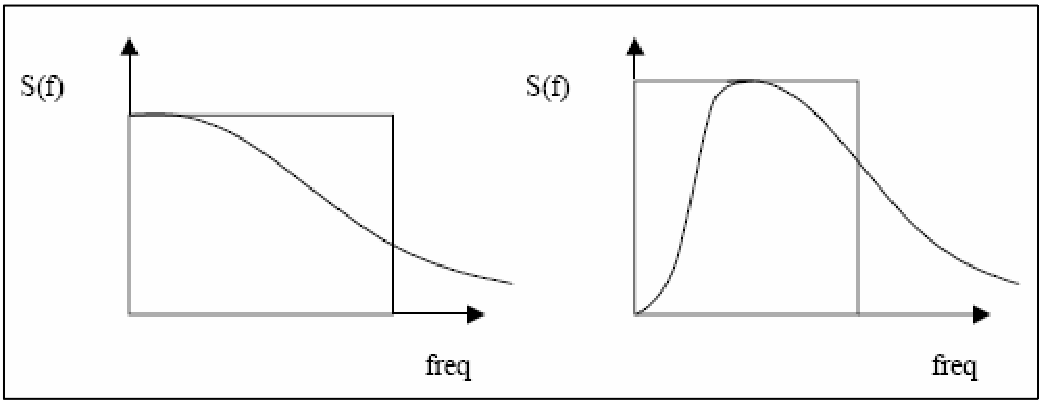


Figure 2.3 Equivalent noise bandwidth of different random functions (RFIC Theory Tutorials, 2003)

2.3 Signal to Noise Ratio

Signal to noise ratio(SNR) is defined as the ratio of the signal power to the noise power corrupting the signal. Signal to noise ratio compares the level of a desired signal to the level of background noise. The higher the ratio, the less obtrusive the background noise is. SNR can be formulated as below:

$$SNR = \frac{P_s}{P_N} \tag{2.1}$$

(P_s : signal power, P_N : noise power)

SNR is frequently expressed in decibels as;

$$SNR(dB) = 10\log_{10}\frac{P_s}{P_N} \tag{2.2}$$

SNR parameterizes the performance of optimal signal processing systems.

These definitions implicitly assume that the signal and the noise are statistically unrelated and arise from different sources. In many applications, corrupting signal arises from man-made sources and can be statistically related to the signal. For example, a cellular telephone's signal can be corrupted by other telephone signals as well as noise. Such non-signals are termed as interference and other parameters should be used to define the performance of the system like signal-to-interference ratio, abbreviated as SIR.

2.4 Noise Figure/ Noise Factor

The noise figure, NF, of a device specifies how much additional noise the device will contribute to the noise already from the source. (RFIC Theory Tutorials, 2003). Noise figure is given by,

$$NF = SNR_{in} - SNR_{out} \tag{2.3}$$

where every variable is a dB figure.

Noise factor, F, is specified as the numerical ratio form of noise figure. Noise factor is a straight ratio of SNR ratios. Noise figure is the decibel equivalent of noise factor.

$$F = \frac{SNR_{out}}{SNR_{in}} \tag{2.4}$$

Ideally F=1.

2.5 Colours of Noise

Even though noise is a random signal, it can have characteristic statistical properties. Spectral density (power distribution in the frequency spectrum) is such a property, which can be used to distinguish different types of noise. This classification by spectral density is given "color" terminology, with different types named after different colors, and is common in different disciplines where noise is an important factor (like acoustics, electrical engineering, and physics).

The color names for these different types of sounds are derived from a loose analogy between the spectrum of frequencies of sound wave present in the sound and the equivalent spectrum of light wave frequencies. That is, if the sound wave pattern of blue noise were translated into light waves, the resulting light would be blue, and so on. (Federal Standard 1037C, 2002)

2.5.1 White Noise

White noise is a signal or process with equal energy per cycle (hertz), producing a flat frequency spectrum in linear space, named by analogy to white light.

In other words, the signal has equal power in any linear band, at any center frequency, having a given bandwidth. For example, the range of frequencies between 40 Hz and 60 Hz contains the same amount of power as the range between 4000 Hz and 4020 Hz has.

An infinite-bandwidth white noise signal is purely a theoretical construct. By having power at all frequencies, the total power of such a signal would be infinite. In practice, a signal is white if it has a flat spectrum over a defined frequency band.

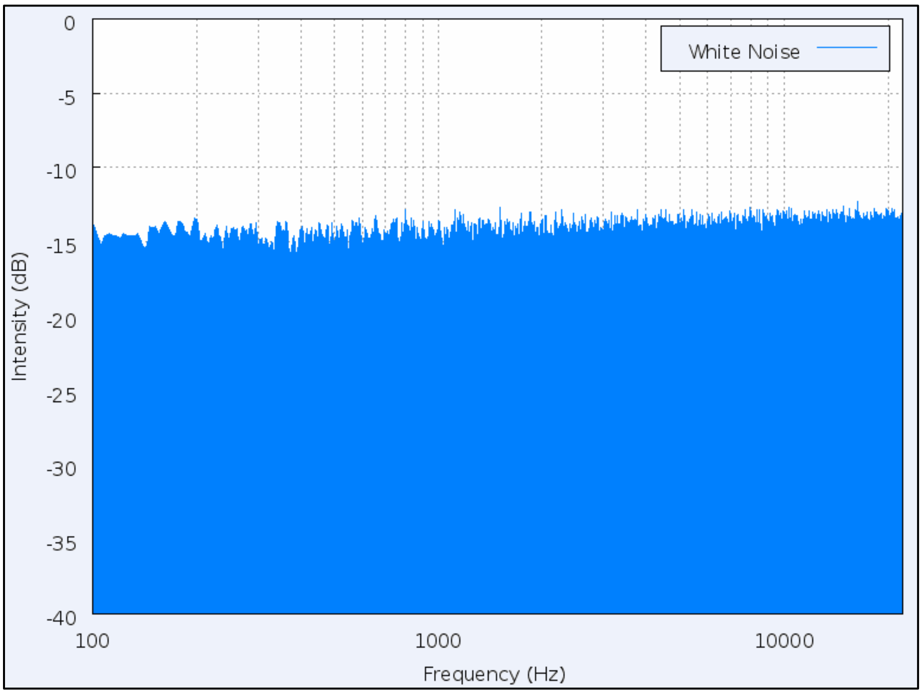


Figure 2.4 White noise frequency spectrum (Colors of Noise, 2008)

2.5.2 Blue (Azure) Noise

Blue noise's power density increases 3 dB per octave with increasing frequency (density proportional to f) over a finite frequency range. In computer graphics, the term blue noise is sometimes used more loosely as any noise with minimal low frequency components and no concentrated spikes in energy. This can be good noise for dithering (Mitchell, 1987); retinal cells are arranged in a blue-noise-like pattern for this reason (Yellot, 1983).

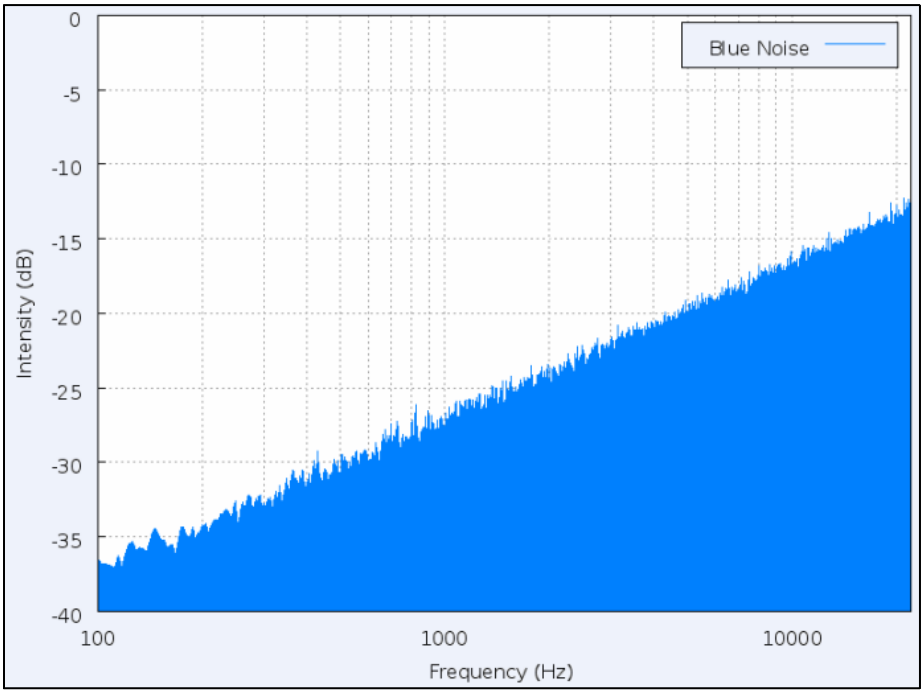


Figure 2.5 Blue noise frequency spectrum (Colors of Noise, 2008)

2.5.3 Purple (Violet) Noise

Purple noise's power density increases 6 dB per octave with increasing frequency (density proportional to f^2) over a finite frequency range. It is also known as differentiated white noise or violet noise.

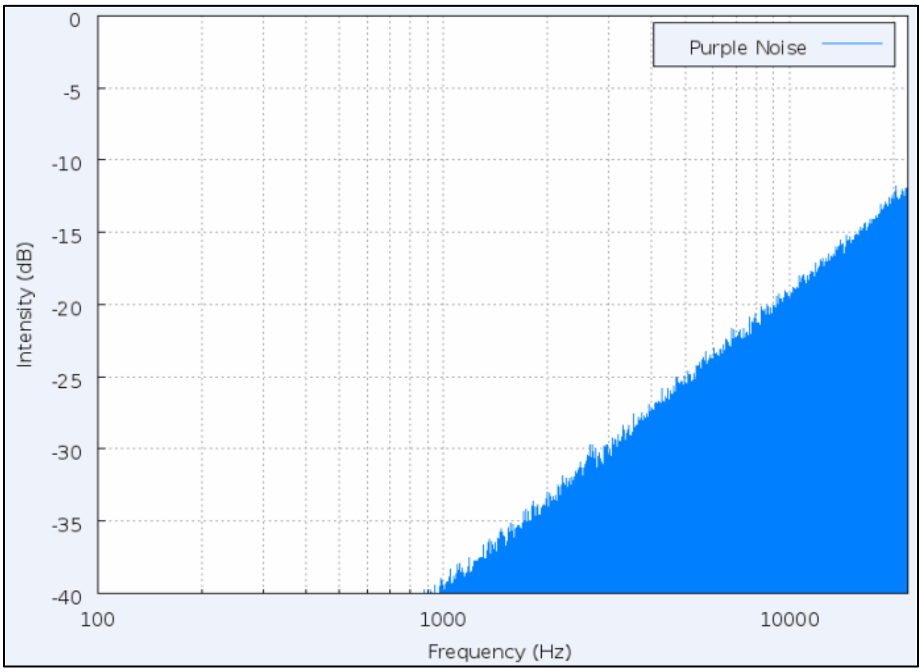


Figure 2.6 Purple noise frequency spectrum (Colors of Noise, 2008)

2.5.4 Grey Noise

Grey noise is random noise subjected to a psychoacoustic equal loudness curve such as an inverted A-weighting curve over a given range of frequencies, giving the listener the perception that it is equally loud at all frequencies. This is in contrast to white noise, which is in fact equally loud at all frequencies but not perceived as due to the natural bias of the human ear.

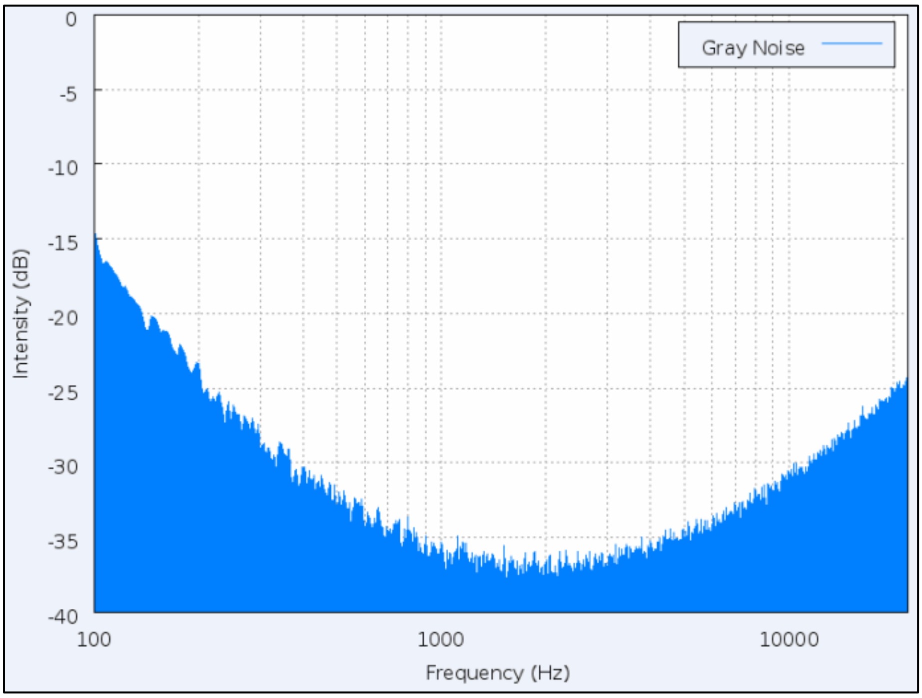


Figure 2.7 Grey noise frequency spectrum (Colors of Noise, 2008)

2.6 Electronic Noise

Electronic noise is the unwanted signal characteristic of all electronic circuits. Depending on the circuit, the noise put out by electronic devices can vary greatly. This noise comes from many different electronic effects. The main constituents of noise in a system is due to; (Cortina, 2007)

- Shot noise
- 1/f noise
- Johnson noise
- Burst noise

2.6.1 Shot Noise

Shot noise consists of random fluctuations of the electric current in an electrical conductor, which are caused by the fact that the current is carried by discrete charges(electrons). The strength of this noise increases for growing magnitude of the average current flowing through the conductor. Shot noise is to be distinguished from current fluctuations in equilibrium, which happen without any applied voltage and without any average current flowing. These equilibrium current fluctuations are known as Johnson-Nyquist noise. Shot noise is important in electronics, telecommunication and fundamental physics. Shot noise is a Poisson process and the charge carriers which make up the current will follow a Poisson distribution. Low noise active electronic devices are designed such that shot noise is suppressed by the electrostatic repulsion of the charge carriers. Shot noise in optical devices are called quantum noise. (Noise, 2007)

2.6.2 1/f Noise

1/f noise, also known as pink noise, is a signal or process with a frequency spectrum such that the spectral density is proportional to the reciprocal of the frequency. Sometimes pronounced as one over f noise, it is also called flicker noise. There is equal energy in all octaves. In terms of power at a constant bandwidth, 1/f noise falls off at 3 dB per octave. The human auditory system, which uses a roughly logarithmic concept of frequency approximated by the Bark scale, does not perceive with equal sensitivity all audible frequencies. However, humans may still differentiate between white noise and pink noise with ease. From a practical point of view, producing true pink noise is impossible, since the energy of such a signal would be infinite. That is, the energy of pink noise in any frequency interval from $f1$ to $f2$ is proportional to $\log(f2 / f1)$ and if $f2$ is infinity, so is the energy. Practically, pink noise is only pink over a certain frequency interval. (Noise, 2007)

2.6.3 Johnson Noise

Johnson noise (thermal noise, Johnson-Nyquist noise) is the noise generated by the equilibrium fluctuation of the electric current inside an electrical conductor, which happens regardless of any applied voltage, due to the random thermal motion of the charge carriers (the electrons). Thermal noise is to be distinguished from shot noise, which consists of additional current fluctuations that occur when a voltage is applied and a macroscopic current starts to flow. For the general case, the above definition applies to charge carriers in any type of conducting medium (e.g. ions in an electrolyte). (Noise, 2007)

2.6.4 Burst Noise

Burst noise is a type of electronic noise that occurs in semiconductors. It is also called impulse noise, bi-stable noise, or random telegraph signals (RTS noise). It consists of sudden step-like transitions between two or more levels as high as several hundred microvolts, at random and unpredictable times. Each shift in offset voltage or current lasts for several milliseconds, and the intervals between pulses tend to be in the audio range (less than 100 Hz), leading to the term popcorn noise for the popping or crackling sounds it produces in audio circuits.

It was discovered during the development of one of the first semiconductor op-amps; the 709. Its sources are not currently known, but appear to be related to trapping of charge carriers, imperfections in semiconductor devices such as surface contamination and heavy ion implants. The worst case noise seems to be at low temperatures and with high thermal noise from external resistors.

Individual op-amps can be screened for popcorn noise with peak detector circuits, to minimize the amount of noise in a specific application. (Noise, 2007)

CHAPTER THREE
NOISE IN AUDIO SYSTEMS

In general audio noise at different audio equipments arises from two sources:

- Poor grounding and/or ground loops
- Electromagnetic Interference(EMI) between components and circuits

3.1 Grounding in Electronic Circuits and between Instruments

The purpose of grounding is to electrically interconnect conductive objects in order to minimize voltage differences between them. The ground in any system must serve two purposes. First, it is the return path for all currents flowing to a device. Second, it is the reference voltage for both digital and analog circuits. Grounding would be a simple exercise if the voltage at all points of the ground could be the same. In reality, this is not possible. All wires and traces on the PCB(Printed Circuit Board) have a finite resistance. This means that whenever there is current flowing through the ground, there will be a corresponding voltage drop. Any loop of wire also forms an inductor. This means that whenever current flows from the supply to a load, and back to the supply, the current path has some inductance. The inductance increases the ground impedance at high frequencies. Ground is also used to prevent the propagation of electromagnetic noise in the circuit. Because of these diverse uses, care must be taken in laying out the PCB ground system to avoid interference between different areas of the board.

A ground loop occurs when there is more than one ground connection path between two pieces of equipment. The duplicate ground paths form the equivalent of a loop antenna which very efficiently picks up interference currents. Lead resistance transforms these currents into voltage fluctuations. As a consequence of ground loop induced voltages, the ground reference in the system is no longer a stable potential, so signals ride on the noise. The noise becomes part of the main signal.

Ground loops are the most common cause of AC line frequency hum in sound systems. Ground loops can be generally identified by a low hum (60Hz in the US, 50Hz in Europe) through the sound system. A ground loop in the power or video signal occurs when some components in the same system are receiving its power from a different ground than other components, or the ground potential between two pieces of equipment is not identical. Ground loop makes the system sensitive to pick up interference from mains wiring which can lead to erratic operation of the equipments or even damages to the equipments. Wiring and grounding problems account for up to 80 percent of all power quality related problems related with sensitive electronic equipments like audio/video systems.

Always when operating with grounding issues it should be known that there is no absolute ground. There is a certain amount of resistance to electrical current between all grounding points. This resistance can change with humidity, temperature, connected equipment and many other variables. No matter how small, the resistance can always allow an electrical voltage to exist across it. The ground wires between wall sockets and power company transformers are not perfect conductors and neither is the shield of the cables that are used with equipments. If all the wires were perfect conductors, ground loops would not be a problem. Effects of ground loop in video pictures are in the form of a black shadow bar across the screen or as tearing in the top corner of a picture. This is caused by different earth potentials in a system.

The finite resistance of the grounding path may couple the noise generated by the power elements into the signal path of the precision circuit elements, as shown in Figure 3.1.

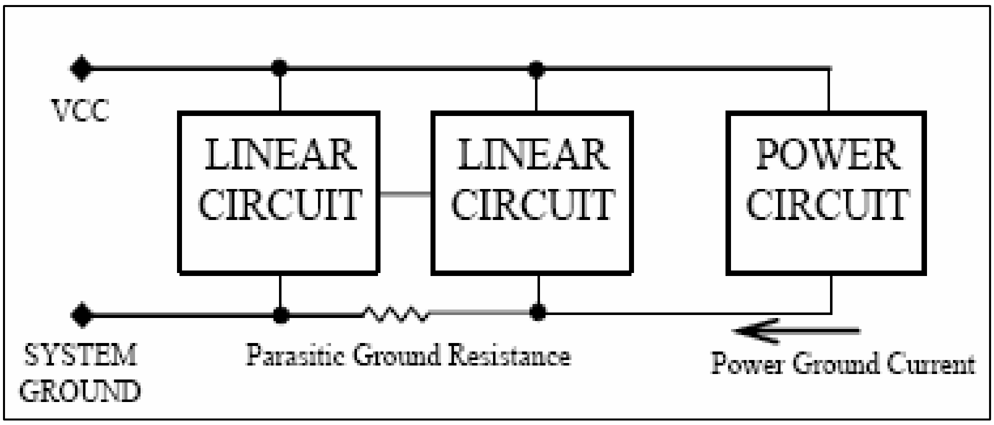


Figure 3.1 Coupling of ground current noise (Fairchild Semiconductor, 2005)

If a ground connection carries substantial current, a voltage drop will develop in the ground because of the basic resistance of the ground material. This voltage drop makes local ground different than zero volts, depending on circuit topology and element location. If the current is high and if there are ground sensitive circuits connected to this current path, the voltage drop across the parasitic ground resistance will produce offsets in the voltage measurements of the sensitive linear circuit. For example, in the case of a switching regulator, where the power current is time varying, noise is induced in the system. Linear circuits are referenced to ground and some parameters of the converter such as load regulation and overall output voltage accuracy are affected by this noise. Another related issue of the ground system is its tendency to carry inductively induced currents. Since the ground system can be spatially extended, it is possible to generate circulating currents in the ground caused by changing external magnetic fields. Because the ground system has usually low resistance, the ground loop currents can become quite large. This often happens in larger systems that magnetically couple to the 50 or 60 Hz power supply lines, which are common in many areas (Fairchild Semiconductor, 2005). These circulating currents can then induce noise in the sensitive linear circuits through parasitic ground resistance, as shown in Figure 3.2.

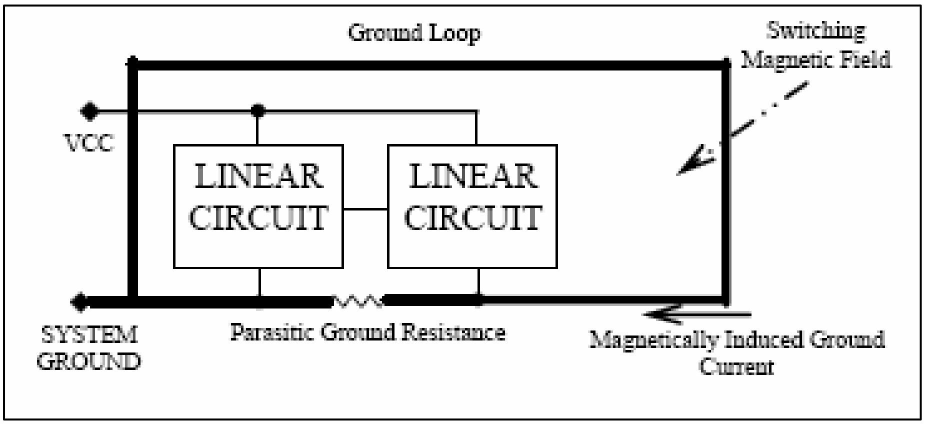


Figure 3.2 Coupling of magnetically induced ground current noise
(Fairchild Semiconductor, 2005)

Another most common source of loop areas are those formed by power and ground traces as shown in Figure 3.3. Whenever possible, multilayer PCBs with power and ground planes should be utilized. Multilayer boards not only minimize the loop area between power and ground, but also mitigate the effects of electromagnetic interference(EMI) that are produced by the high frequency signals.

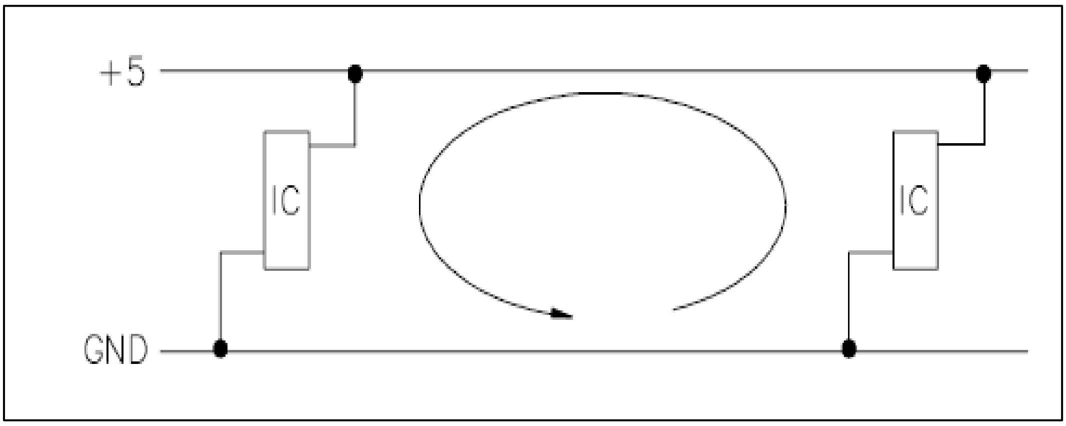


Figure 3.3 Power and ground PCB loop (Semtech Corporation, 2002)

The most common technique used to decrease the ground loop problems in electronics systems is the star grounding technique. In star grounding, a common point is chosen to which all ground lines are connected. A physical point is chosen on the circuit board to serve as the star ground point and all grounds radiate from that point to the separate parts of the system. Any one of the ground lines radiating from the star ground may star again, forming a local star ground for a sub-circuit. Special care must be taken to separate the signal ground from the power ground at a star, such that no power current will flow through any signal line ground. In Figure 3.4, a cell phone system with a star ground connection is shown. Cell phone system design is complex as it compounds various sub-circuits such as analog baseband, digital baseband, audio amplifiers, LCD controller, power and battery management, and RF transceiver. Ideally, all grounds would converge on the main star and a single point ground reference would be generated at that location. It is physically difficult to bring all the traces on a board to a physical location and make contact at a mathematical point to avoid current generated interference. Therefore, the main star might be broken into sub-stars. This shifts the burden to coupling signals from one star to another, which have slightly different, time variable ground reference potentials. Signal coupling between different blocks would have to be differential to avoid all ground problems.

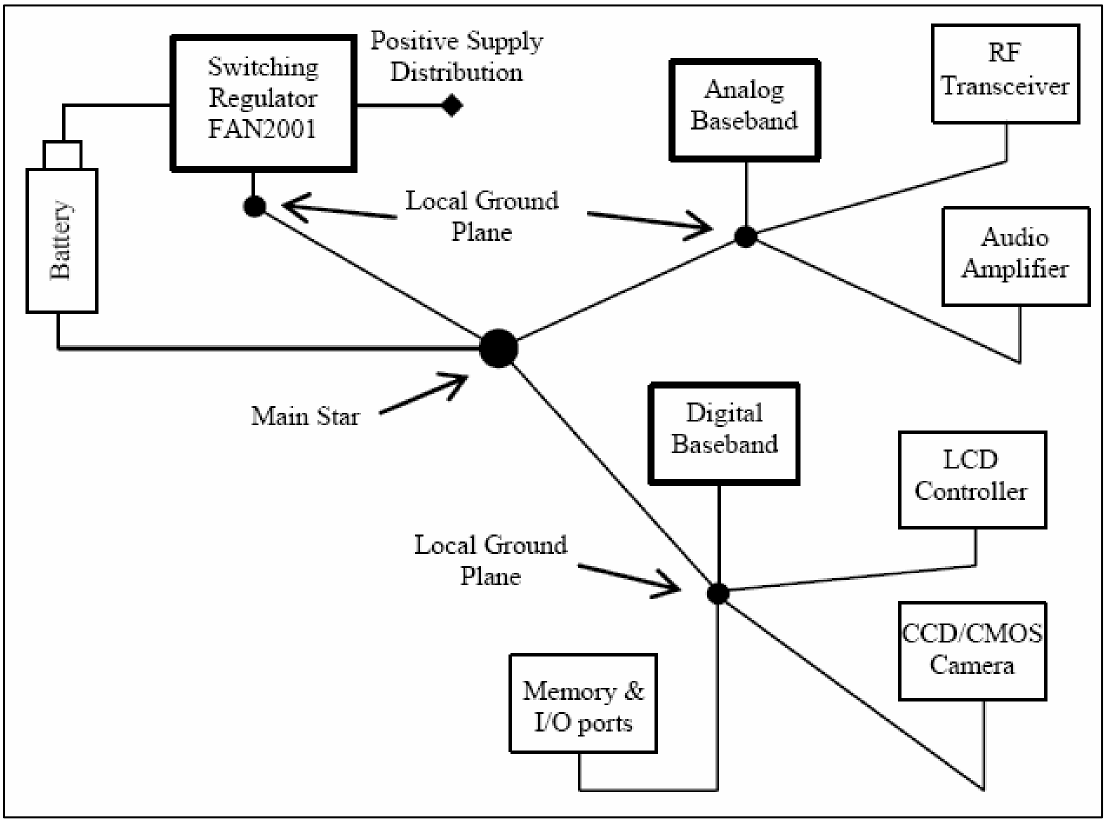


Figure 3.4 Star ground connection for a cell phone (Fairchild Semiconductor, 2005)

Noise can also be added to the signals due to using the same ground for analog and digital circuits on the PCB. The digital circuit noise can get to the analogue signal path if separate grounding systems are not used for digital and analogue parts. Digital grounds are invariably noisier than analog grounds because of the switching noise generated in digital chips when they change state. For large current transients, PCB trace inductances cause voltage drops between various ground points and signal lines on the board. For digital lines this is not a problem unless this voltage level cross a logic threshold. For analog lines it is plain noise to be added to the signals. If all the analog and digital circuits are placed on separate parts of the PCB, the ground currents will naturally be isolated. For this to work well, the analog section must contain only analog circuits on all layers of the PCB.

The ideal configuration of grounding between instruments, as far as grounding is concerned, would involve having each instrument grounded internally to a single point, and then having each instrument joined together at a second single point.

Signals would then be coupled from one instrument to another through single-wire cables, since all instruments would use one universal ground as their reference.

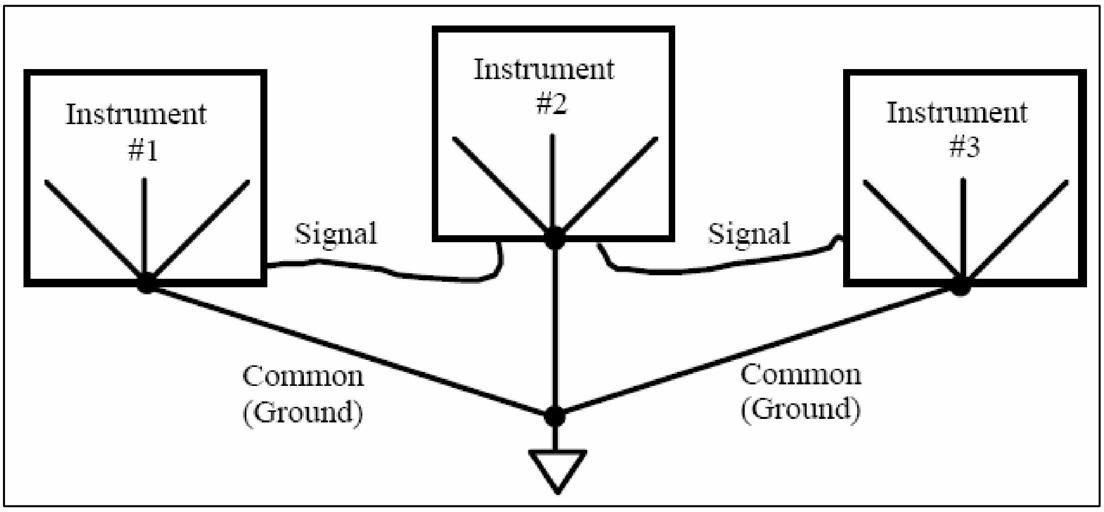


Figure 3.5 Grounding between instruments (Noise in Analog Circuits, 2005)

3.2 Electromagnetic Interference

Electromagnetic Interference(EMI) is defined as the influence of unwanted signals on devices and systems, making the operation of the device difficult or impossible. EMI is generated by a varying electric or magnetic field and transmitting them by means of conductive, inductive or capacitive coupling, through free space or a combination of these means. An electromagnetic signal must have a source or origin. The source then needs a coupling path to facilitate the transmission of the disturbance signal to the victim. The coupling between the source and the victim lines may be due to one or more of the following mechanisms:

- conductive coupling
- capacitive coupling
- inductive coupling

3.2.1 Conductive Coupling

Conductive coupling is the transfer of energy achieved by means of physical contact. Such coupling may be achieved by wire, resistor, or common terminal such as a binding post or metallic bonding. Conductive coupling passes the full spectrum of frequencies including dc.

Conductive coupling occurs when two circuits or channels have a common branch as depicted in Figure 3.6. If Circuit 1 is considered as being a power line and Circuit 2 as a telecommunications channel, the parasitic current introduced into Circuit 2 from Circuit 1 is detrimental because of its intensity. In commercial cabling installations, the occurrence of conductive coupling is fairly common when the bonding and grounding systems used for power and telecommunications are not sufficiently isolated.

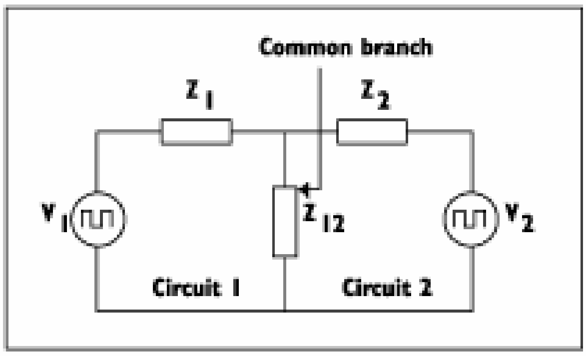


Figure 3.6 Conductive coupling between two circuits (Electromagnetic Interference – Considerations in Structured Cabling Systems, 2002)

3.2.2 Capacitive Coupling

Capacitive coupling is the transfer of energy within an electrical network by means of the capacitance between circuit nodes. Capacitive coupling is typically achieved by placing a capacitor in series with the signal to be coupled. Capacitive coupling is often unintended, such as the capacitance between two wires or PCB traces that are next to each other. Voltages changing in one wire will induce a current flow in adjacent wires according to the normal capacitance relationship:

$$I = C * \frac{dV}{dt} \tag{3.1}$$

Because the currents are proportional to the rate of change of voltage, capacitive coupling is generally most serious in high-frequency circuits, but low-frequency circuits can also be affected by capacitive coupling. For example, if it is assumed that a capacitor is inserted between a signal source with finite output impedance and an amplifier assumed to have infinite input impedance as shown below, it is easy to understand capacitive coupling between adjacent PCB traces. Changing the voltage on one plate of the capacitor, which is actually an adjacent wire in the circuit, induces a current such that,

$$I_{fluctuating} = C\frac{dV_{fluctuating}}{dt}. \tag{3.2}$$

This current in turn induces a voltage drop equal to

$$\Delta V_{in} = Z_{source} I_{fluctuating} = Z_{source} C \frac{dV_{fluctuating}}{dt} \tag{3.3}$$

where Z_{source} is the output impedance of the signal source. Here, it is seen that the capacitively coupled current scales with the impedance of the signal source in voltage-amplification circuits. (Noise in Analog Circuits, 2005)

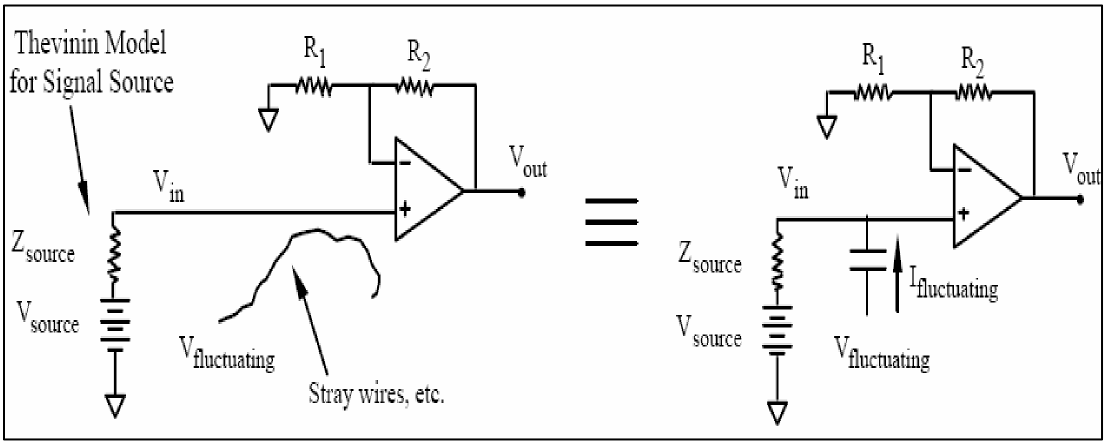


Figure 3.7 Capacitive coupling mechanism (Noise in Analog Circuits, 2005)

Thus, high impedance voltage sources, such as pH electrodes and some kinds of optical detectors, are particularly susceptible to interference due to capacitively coupled currents. In contrast, capacitive coupling between op-amp stages, for example, is small because the output impedance of op-amps is small. In current amplification circuits, such as an I-V converter, the capacitively-coupled current looks identical to a real signal current, and will be amplified; again, this is the same as in voltage amplification circuits because a current source is an ideal battery in series with a very large resistance. Capacitive coupling can be minimized in several ways. First, a shield can be put around signal wires, with the shield connected to a good ground. In that case, the capacitive currents can flow through the shield without inducing voltage drops; the signal wire sees that it is surrounded by a ground potential and does not see any fluctuating electrostatic potentials. This is basic idea behind coaxial cable and shielded twisted pair. In both cases, by surrounding the signal wires by a good conductor at ground potential, the effects of capacitive coupling can be reduced.

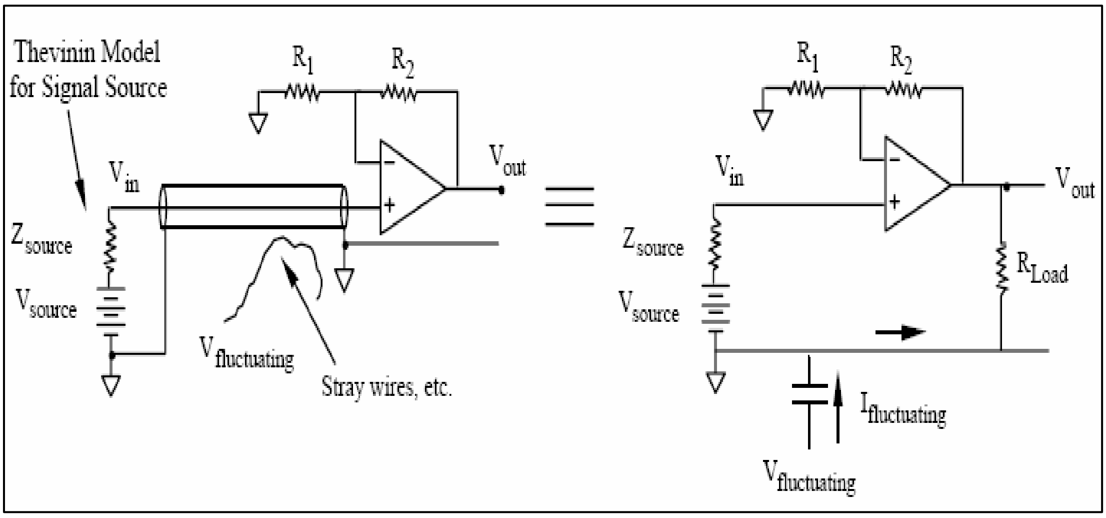


Figure 3.8 Reducing capacitive coupling by shielding (Noise in Analog Circuits, 2005)

External fluctuating voltages now induce currents in the ground line, rather than the signal line. Since the external load usually connected to an op-amp is on the order of kilohms or tens of kilohms, the effects are reduced by many orders of magnitude. If Rload<<Zsource, then all this induced current will flow through Rload. The op-amp will keep Vout constant, so that current induced on the ground line will not produce any change in the ouput voltage measured with respect to ground assuming that the ground line has zero resistance. One way of further reducing capacitive coupling is to use triax and/or shielded twisted pair wire, with a differential input on the voltage amplifier. The idea here is that there are essentially two grounds: one serves as a reference point for the voltage measurement and should have only small amounts of

current flowing through it, and the other shield acts as a capacitor plate to establish a constant potential and carry off the capacitively-induced currents to a good ground. Any remaining capacitive coupling which propagates into the two central conductors will usually affect both wires similarly, so that a differential amplifier which responds to the difference in voltage between the two inputs can often get rid of the remaining common-mode pickup.

3.2.3 Inductive Coupling

Inductive coupling refers to the transfer of energy from one circuit component to another through a shared magnetic field. A change in current flow through one device induces current flow in the other device. The two devices may be physically contained in a single unit, as in the primary and secondary sides of a transformer, or may be separated as in the antennas on a transmitter and a receiver. For low-noise circuits operating at audio frequencies, it is usually more problematic than the capacitive coupling. Inductive coupling usually occurs at a few well-defined frequencies: 50 Hz and multiples thereof (100 Hz, 150 Hz, etc.) from 220-Volt power lines, 30 kHz -50 kHz from computer monitors, and higher frequencies (hundreds of MHz) from instruments like NMR's(Nuclear Magnetic Resonance).

According to the Maxwell’s equations from electromagnetic theory, a loop of wire in a time-varying magnetic field will have an induced voltage. If it is assumed that the magnetic field B is constant in space but varies in time, the voltage induced around the loop will be:

$$V = -A.\frac{dB}{dt} \tag{3.4}$$

where V is the voltage induced in the loop, B is the magnetic field strength (in Tesla, where 1 Tesla = 10^4 Gauss) and A is the area of the loop in square meters.

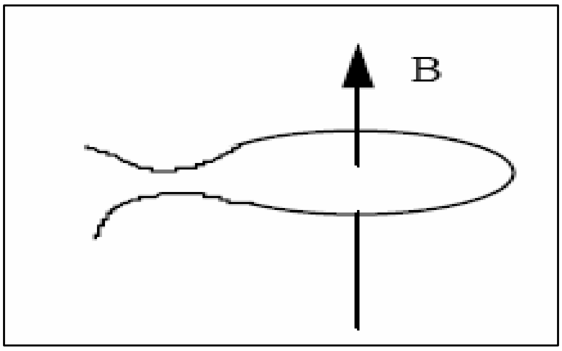


Figure 3.9 Loop wire in the magnetic field
(Noise in Analog Circuits, 2005)

Magnetic induction is also quite serious from some other souces; for example, computer monitors use electrostatic coils to deflect an electron beam which strikes the phosphor screen; the raster-scan of the electron beam is typically at 30 -50kHz for a super-VGA monitor, which means that there's a time varying magnetic field at 30 -50 kHz. Although B might be small, at this higher frequency dB/dt can be quite large, again producing voltages at 30 kHz. Magnetic pickup of this sort is quite problematic in circuits when it is needed to measure time-varying signals in the range between 60 Hz and about 240 Hz. While a lot of attention is typically given to grounding and shielding for high-frequency circuits, the magnetically-induced voltages can be a far more serious problem. The stray magnetic fields from transformers and other coils basically look like a magnetic dipole, and decays rather slowly with distance: approximately r_0/r , where r_0 is the size of the coil generating the magnetic field. Because $1/r$ is not a particularly strong function of distance, it means that it's generally impractical to eliminate magnetically-induced voltage by simply moving the power supplies farther away, although sometimes it solves the problem. In general, the primary solution to magnetically-induced voltages is to make the circuit immune to pickup by making all circuit paths small, effectively decreasing the area A. One way of minimizing the area A is to use coaxial cable, in which the signal is applied to the central element and the return current flows through the shield. The insulator between the shield and the wire is about 1 mm thick (=0.001 meter), making the effective area of the loop about 0.001*cable length (in meters). Because the shield completely surrounds the central conductor, it is considered as two loops, one constituting the central conductor and the top part of the shield, and the second loop comprised of the central conductor and the bottom part of the shield. The current induced in each of these loops will be either clockwise or counter-clockwise depending on whether the magnetic field is increasing or decreasing, which means that the currents induced in the top loop and the bottom loop will cancel one another. If the magnetic field is inhomogeneous such as a coaxial wire running right next to a transformer, then the fields will not cancel and the cable still will be affected by magnetic induction.

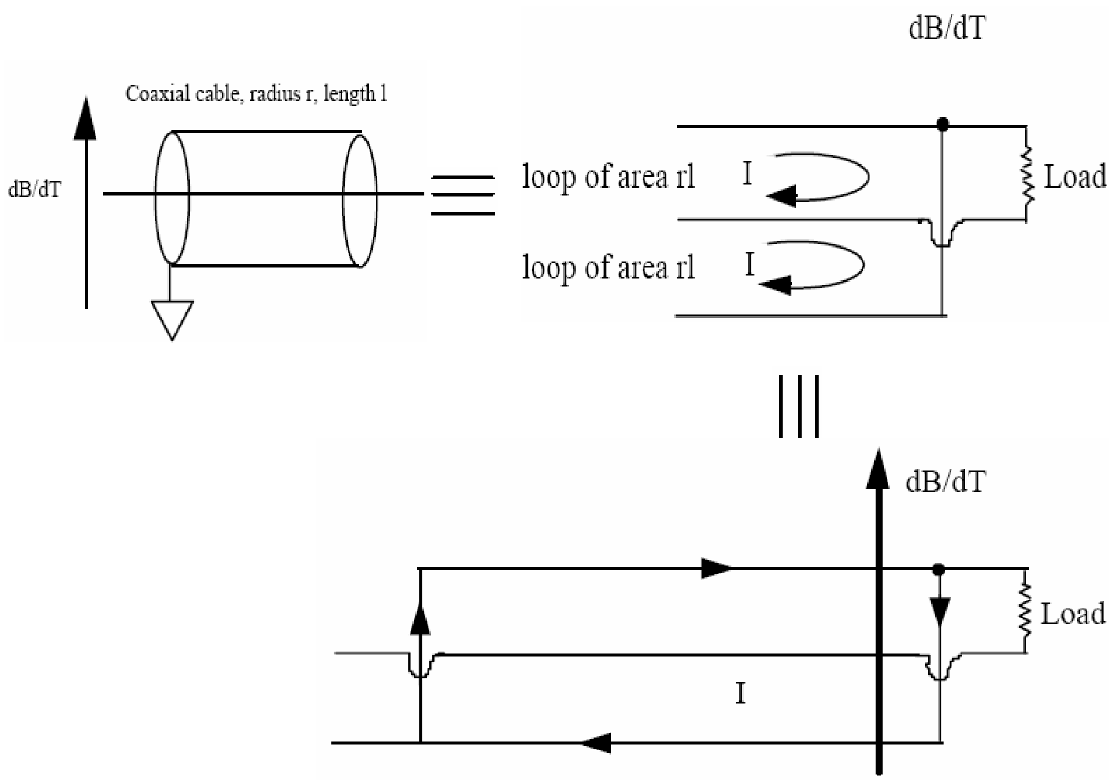


Figure 3.10 Coaxial cabling (Noise in Analog Circuits, 2005)

Another way of reducing the loop area A is to use twisted-pair conductors, in which the signal and ground wires are twisted around one another. Each "twist" can be thought as being two half-twists of 180 degrees each.

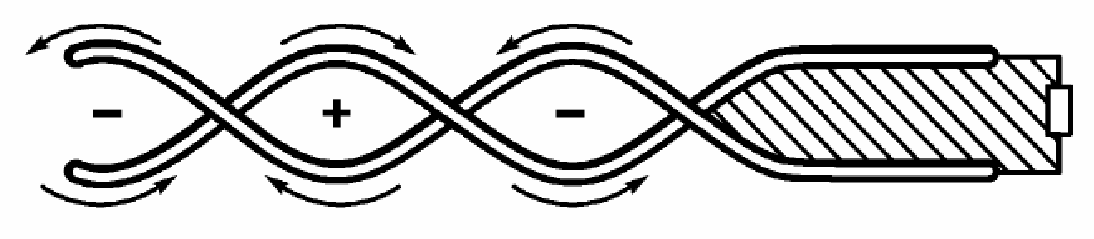


Figure 3.11 Twisted pair cabling (Broeders, Meywes & Baker, 2008)

Any current induced in the area of the first half-twist will be cancelled by the current induced in the next half-twist because the signal and ground wires are exchanged. As long as the magnetic field B doesn't change significantly on the distance scale of one twist, the magnetic inducion will be zero. Twisted-pair usually gives a better solution than coaxial cable because the insulators are usually thinner resulting in smaller loop areas.

CHAPTER FOUR
AUDIO NOISE TEST AND MEASUREMENT IN LIQUID CRYSTAL DISPLAY(LCD) TELEVISIONS

4.1 LCD Television

Liquid Crystal Display Televisions (LCD TV) are display devices that use LCD technology for visual output (producing an image). The technology used is generally thin film transistor (TFT), because this allows for size, especially depth reduction, lower weight and reduced energy consumption. These LCD screens are also used as computer monitors. LCD technology is based on the properties of polarized light. Two thin, polarized panels sandwich a thin liquid-crystal gel that is divided into individual pixels. An X/Y grid of wires allows each pixel in the array to be activated individually. When an LCD pixel darkens, it polarizes at 90 degrees to the polarizing screens. The pixel darkens in proportion to the voltage applied to it: for a bright detail, a low voltage is applied to the pixel; for a dark shadow area, a higher voltage is applied. LCDs are not completely opaque to light, however; some light will always go through even the blackest LCD pixels. LCDs do not produce their own light, so a light source behind the display is also necessary; this is accomplished by using several florescent light bulbs and a diffuser plate to help distribute the light evenly. This back light must be powerful, as polarized glass and liquid crystal materials absorb more than half of the light that passes through their layers. LCD televisions can support both the analog and digital video broadcasting.

4.1.1 LCD Television Block Diagram

Motherboard(chassis), power board, LCD panel and the panel inverters are the main components of an LCD television. The building blocks of the chassis are tuner, IF demodulator, video processor, scaler/deinterlacer, audio processor, audio amplifier, audio and video switches, microcontroller and memory. The block diagram of the chassis can be seen in Figure 4.1.

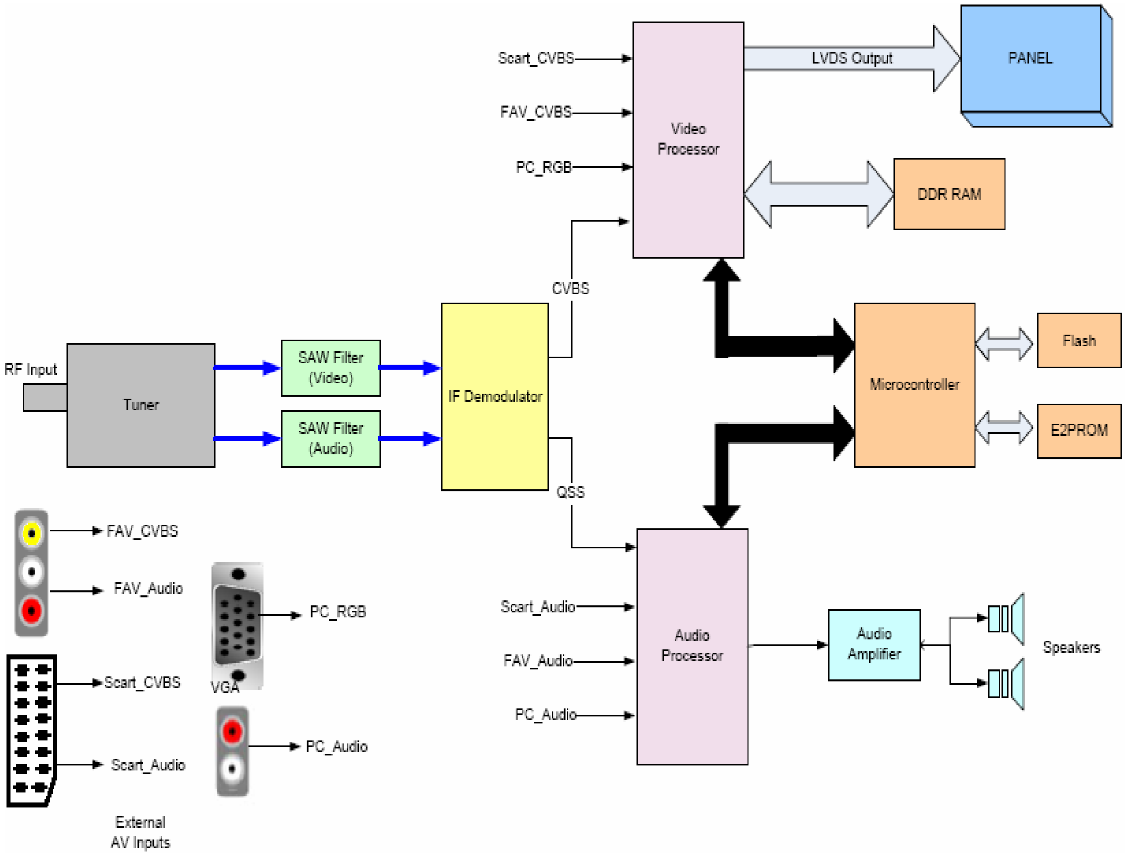


Figure 4.1 Simplified block diagram of an LCD television

4.1.1.1 Tuner

A tuner is a circuit module or free-standing equipment which detects radio-frequency (RF) signals usually of low amplitude, amplifies and converts them to a form suitable for further processing. A television tuner converts an RF television transmission into audio and video signals which can be further processed to produce sound and picture. Different tuners are used for different television standards such as PAL, NTSC, ATSC, SECAM, DVB-C, DVB-T, T-DMB, open cable. A typical TV signal as described above requires 4 MHz of bandwidth. When the sound is added, a TV signal requires 6 MHz of bandwidth. Therefore, the FCC(Federal Communications Commission) allocated three bands of frequencies in the radio spectrum, chopped into 6-MHz slices, to accommodate TV channels:

- 54 to 88 MHz for channels 2 to 6
- 174 to 216 MHz for channels 7 through 13

- 470 to 890 MHz for UHF channels 14 through 83

The composite TV signal can be broadcast on any avaliable channel. The tuner locks to a specific channel and converts the RF signal to intermediate frequency(IF) around 38MHz. For PAL systems, video carrrier is at 38.9MHz and audio carrier is at 33.4MHz.

4.1.1.2 SAW (Surface Acoustic Wave) Filter

The intermediate frequency signal at the output of the tuner enters the SAW filter. SAW filters are electromechanical devices commonly used in radio frequency applications. Electrical signals are converted to a mechanical wave in a piezoelectric crystal; this wave is delayed as it propagates across the crystal, before being converted back to an electrical signal by further electrodes. The delayed outputs are recombined to produce a direct analog implementation of a finite impulse response filter. The SAW filter is used for pre-filtering on the TV signal. It limits the signal bandwidth to 8MHz and suppresses major parts of the adjacent channels. Two SAW filters are used at the output of the tuner, one for audio and another one for video signal. Because sound carrier is split at the output of the tuner and another SAW filter is used for sound, it is called split carrier sound process. Figure 4.2 shows the frequency response of audio SAW filter retrieved from the datasheet of K9656M. It has a pass band for sound carriers between 32.35MHz and 33.40MHz. α_{rel} indicates the relative attenuation with respect to the insertion attenuation at 33.40MHz.

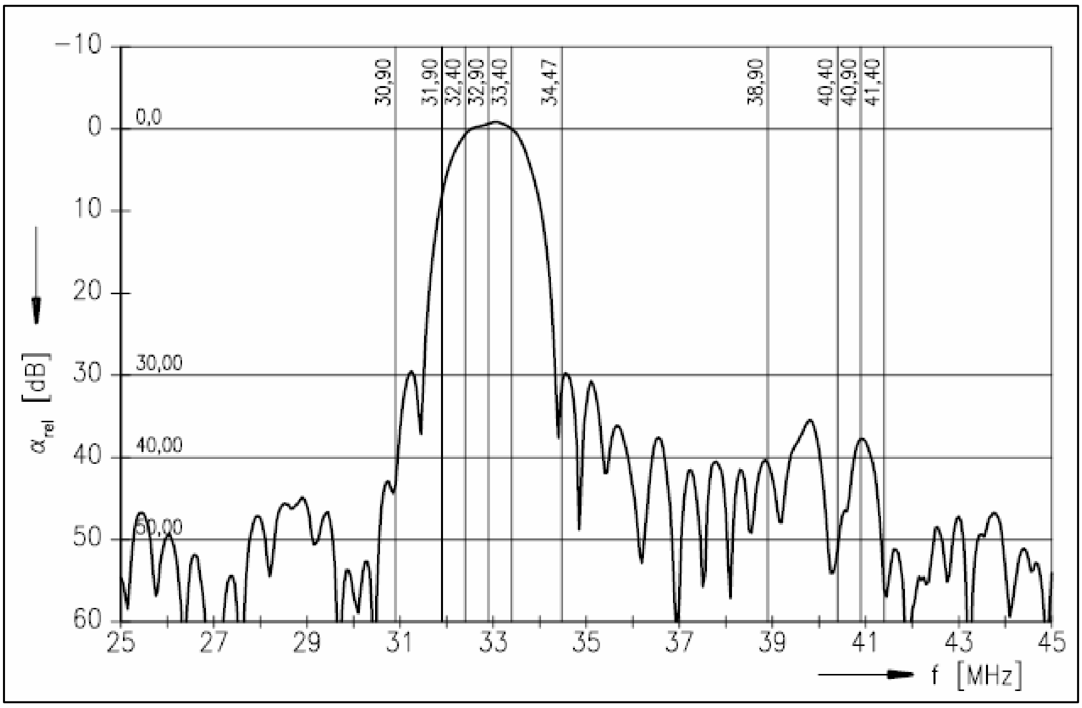


Figure 4.2 Frequency response of K9656M audio saw filter (EPCOS Electronics, 2001)

Figure 4.3 shows the frequency response of video SAW filter for TV applications retrieved from the datasheet of K3953M. It has a pass band between 33.90MHz and 38.90MHz. α_{rel} indicates the relative attenuation with respect to the insertion attenuation at 37.40MHz.

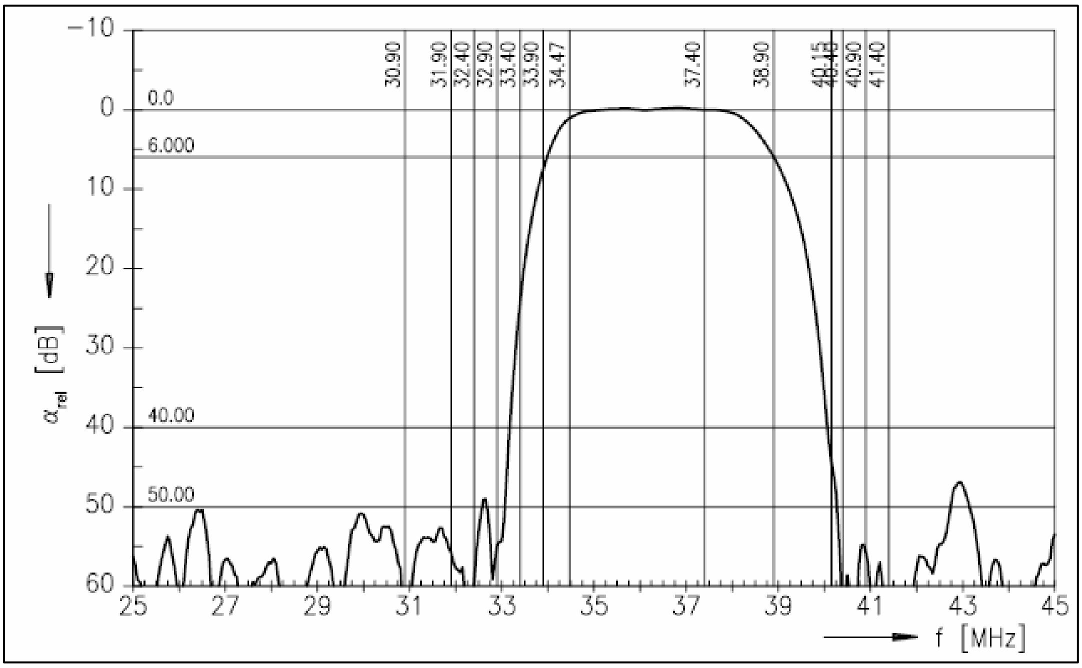


Figure 4.3 Frequency response of K3953M video saw filter (EPCOS Electronics, 2001)

SAW filters provide better SNR and channel separation in TV applications.

4.1.1.3 IF Demodulator

The filtered video and audio IF signal at the output of the SAW filters are applied to the IF demodulator. Modulated video and audio signals are demodulated by IF demodulator. The demodulated composite video baseband signal(CVBS) and baseband sound signal are available as analog output signals. Quasi split sound (QSS) processing is performed for stereo applications, which improves audio SNR quality.

4.1.1.4 Video Processor

The baseband video signal from demodulator and other analog and digital video signals are the input of the video processor. Video processor provides video decoding by separating the composite video signal to chroma and luma components. It also provides analog to digital conversion of the decoded video signal, scaling and deinterlacing functions, picture enhancement applications like brightness, contrast, hue, saturation, sharpness, gamma adjustments and noise reduction algorithms. Scaling and deinterlacing functions may be in another chip form to be placed between the video processor and the panel interface circuit.

Scaler is the video processing part used for converting video signals between one arbitrary resolution/aspect-ratio and another resolution/aspect-ratio. With upscaling function, a low resolution image is converted to a high resolution image and with downscaling function, a high resolution image is converted to a low resolution image. Scaler makes some signal conditioning to make the image resolution suitable with the display device resolution. This does not mean that placing a video scaler before a limited-capability display device will remove the limitations of that display device. The scaler accepts a video signal and converts it to what the display is expecting.

Deinterlacing is the process of converting interlaced video, like common analog television signals, into a non-interlaced form. Video signal consists of a series of images played in succession, each of these images is known as a frame. For a variety of reasons, analog television standards break these frames into a number of lower-resolution fields, each one containing a portion of the image. The interlaced display draws only half of the lines on the screen for each frame, alternately drawing the odd and even lines for each frame. This reduces flicker by taking advantage of the persistence of vision effect, producing a refresh rate of double the frame rate without either transmitting each frame twice or holding it in a buffer so it can be redrawn. Digital display devices like LCDs and plasma displays are made of pixels and displaying the interlaced video would result that half of the pixels remain black and in halving of brightness. Therefore, all digital display devices require deinterlacing. There are two basic methods of deinterlacing: combination, where the even and odd frames are combined into one image and then displayed, and extension, where each frame (with only half the lines) is extended to the entire screen. Improved deinterlacing algorithms buffer several fields and use techniques like edge detection in an attempt to find the motion between the fields. This is then used to interpolate the missing lines from the original field. The processed video signal is converted to 8-10 bit LVDS (low-voltage differential signaling) format and sent to the panel interface.

4.1.1.5 Audio Processor

QSS signal from IF demodulator, baseband analog audio from peripherals of the TV and digital audio are the inputs of the audio processor. The analog to digital converters built in the audio processor provide digital conversion of the analog audio. The digital signal processing block processes the digital audio by improving the audio quality and implements some special effects used in TV systems. The processed audio signal is converted back to analog and outputted from the audio processor.

To achieve the desired audio output power from the speakers, an audio amplifier is placed between the audio processor and the speakers. Audio output power requirement depends on the panel size of the LCD television. Depending on the requirement of the output power different types of audio amplifiers are used.

4.2 Audio Noise Problem in LCD Televisions

The audio noise heard from the TV speakers is a bass tone at 50Hz, along with its harmonics at higher frequencies which may sound like a buzz. This noise can be easily detected by the user even it has low level. It is generally produced by internal components(power supply, driver circuits, power cable etc.), audio amplifier, poor or broken internal grounds and bad connections of audio cables. These unwanted small signals are amplified by audio amplifier and become noticeable. The figure below shows the block diagram of a standart LCD TV system audio part.

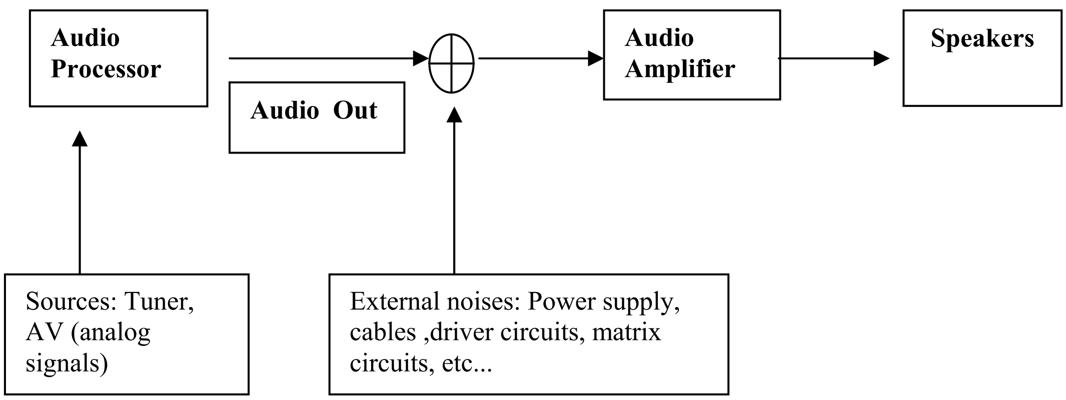


Figure 4.4 Block diagram of a standart LCD TV system audio part

As can be seen from the block diagram, audio noise is generally not related with audio source. Sound processor output is the clear audio source and noise is added to audio signal due to the EMI from external components and circuits and poor grounding in the TV chassis. The major noise sources in an LCD TV are power supply, panel inverter, high frequency signals and bad cable connections.

Power supply is the most important cause of audio noise in LCD televisions. Power supplies are generally described as either linear mode or switched mode, according to the method by which standard household and commercial AC line input voltage is transformed into a DC output voltage suitable for use by electronic equipment. Simplified block diagram of both design topologies are shown in Figure 4.5:

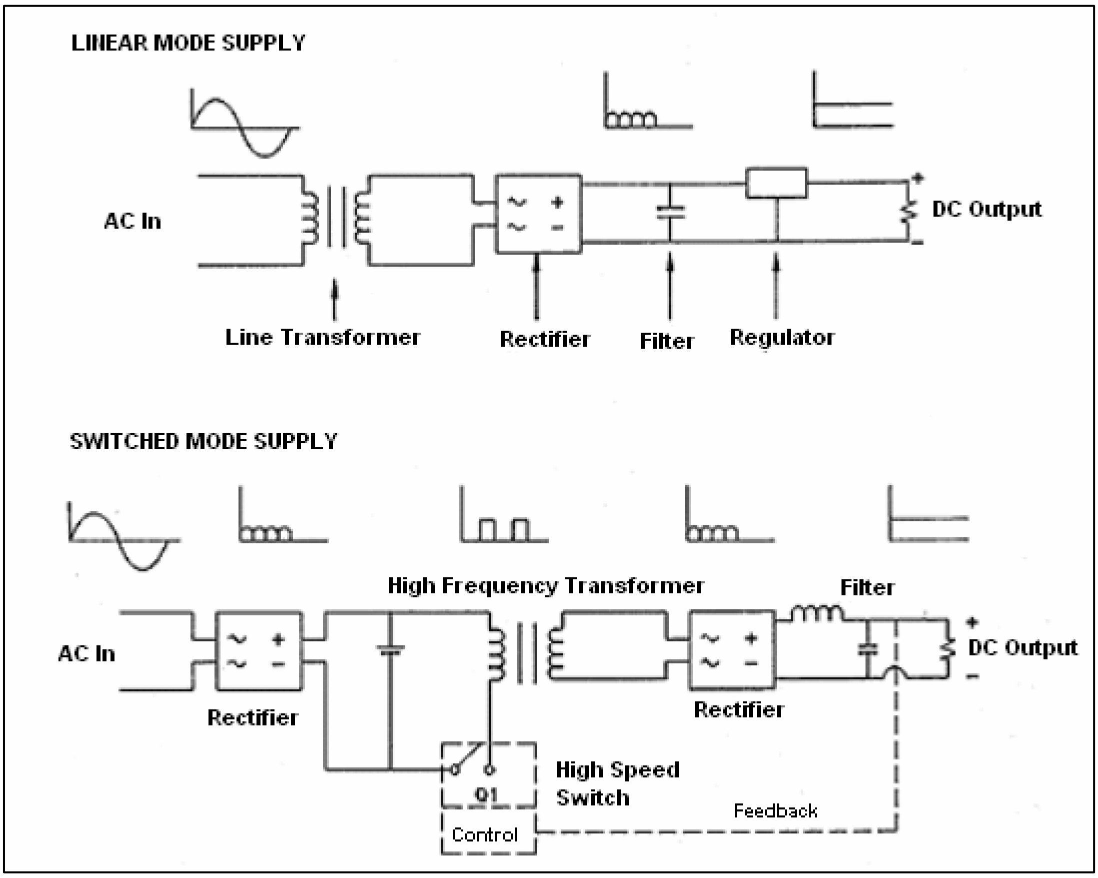


Figure 4.5 Simplified block diagrams of linear and switch mode power supplies (EMI Control Applications Notes For Typical Computer Subsystems, 2008)

A switched mode power supply (SMPS) is used in LCD television to obtain the required dc supply voltages in the TV chassis and to feed the cold cathode fluorescent lamps (CCFL) used for backlighting the LCD panel. SMPS rapidly switches a power transistor between saturation (full on) and cutoff (completely off) with a variable duty cycle whose average is the desired output voltage. The resulting rectangular waveform is low-pass filtered with an inductor and capacitor. The main advantage of this method is greater efficiency because the switching transistor dissipates little power in the saturated state and the off state compared to the semiconducting state (active region). Conventional series-regulated linear power supplies maintain a constant voltage by varying their resistance to cope with input voltage changes or load current demand changes. The linear regulator can, therefore, tend to be very inefficient. Other advantages include smaller size and lighter weight from the elimination of low frequency transformers which have a high weight and lower heat generation from the higher efficiency. Disadvantages include greater complexity, the generation of high amplitude, high frequency energy that the low-pass filter must block to avoid EMI, and a ripple voltage at the switching frequency and the harmonic frequencies thereof. The major source of EMI that is actively generated in switch mode supplies is associated with the switching behavior of the rectifier diodes located in the converter transformer secondary circuit. These rectifier diodes are selected to pass the full output load current (often tens of amperes) and efficiently switch at twice the switching frequency for full wave rectification. During operation, the rectifier diodes can transition from forward bias (on) to reverse bias (off) in less than 35 nanoseconds, thus inducing a large, fast transient voltage impulse or spike in the highly inductive power supply secondary circuit. These transient events will increase the strength of any EMI from the first ten switching frequency harmonics. The related EMI components are coupled conductively to the mains terminal through power cable. The radiated noise is also coupled inductively to the chassis and the audio lines.

High speed signal traces on the chassis are another source for EMI. The communication between the system microcontroller and all other peripheral ICs like memory devices, video processor, audio processor etc., occurs at very high frequencies. These high frequency signal traces behave like antennas and radiate electromagnetic interference. The longer these traces, the higher the interference. This radiated electromagnetic fields enter the audio path through capacitive or inductive coupling.

Noise can also enter to the audio circuit due to different grounding of the chassis and the audio circuit. External noise sources can induce noise currents and electrostatic charge on TV chassis. Noise currents induced into the cable shields also flow through the chassis, since the shields terminate on the chassis. There is also coupling between the chassis and the internal circuitry, noise on the chassis can couple into the internal audio. This noise coupling can be decreased by connecting the signal ground to the chassis. This allows the entire grounding system to fluctuate with the noise.

Another source of noise heard from the speakers is the class D type audio amplifiers usually used in LCD television applications. A class-D amplifier or switching amplifier uses switching mode of transistor to regulate power delivery. The amplifier, therefore, features the high power efficiency (low energy losses), which additionally results in lower weight by eliminating the bulky heat sinks. If voltage conversion is necessary, the on-the-way high switching frequency allows to replace the bulky audio transformers by small inductors. Low pass LC-filtering smoothes the pulses out and restores the signal shape on the load. Class-D amplifiers also provide high output voltages. The input signal is converted to a sequence of pulses whose average value is directly proportional to the amplitude of the signal at that time. The frequency of the pulses is typically ten or more times the highest frequency of interest in the input signal. The final switching output consists of a train of pulses whose width is a function of the amplitude & frequency of the signal being amplified, and hence these amplifiers are also called PWM amplifiers. A switching controller is required to produce a high power replica of the comparator's PWM signal. The output contains, in addition to the required amplified signal, unwanted spectral components (i.e. the pulse frequency and its harmonics) that must be removed by a passive filter. The filter is usually made with (theoretically) lossless components like inductors and capacitors in order to maintain efficiency. Figure 4.6 shows the block diagram of a class-D amplifier.

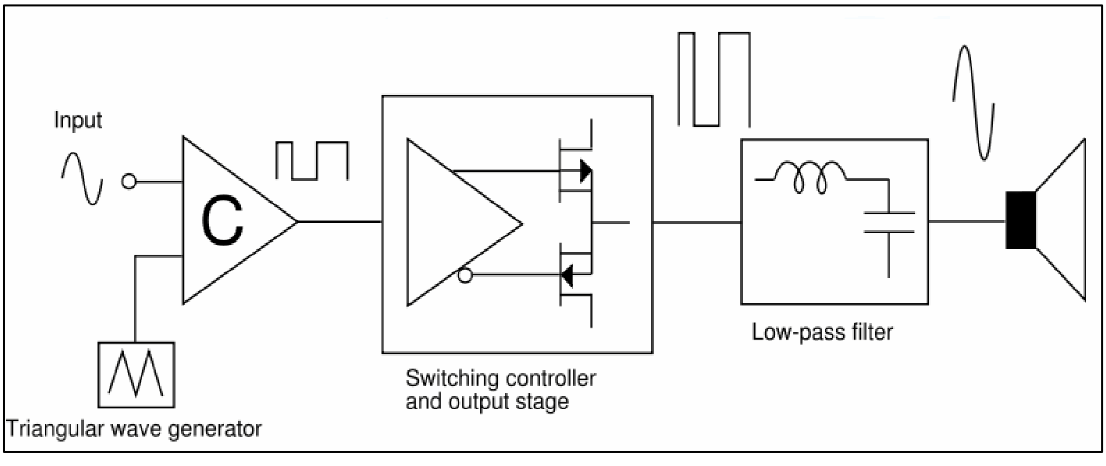


Figure 4.6 Block diagram of a basic switching (class-D) amplifier (Switching Amplifier, 2008)

Switch-mode amplifiers often include an output LC low pass filter. The LC filter prevents power from being wasted due to the switching artifacts. This is a formidable task because real LC filters loose their effectiveness above a certain frequency. All inductors have a parallel capacitance thus exhibiting self resonance. Capacitors also have both series parasitic resistance and inductance that limit the filter's attenuation at high frequencies.

Switch mode amplifiers tend to have very noisy grounds that will impose unwanted common mode potentials to the amplifier inputs. This can seriously limit the performance of the audio system. One way to solve this problem is to connect the amplifier ground to the chassis ground.

4.3 Audio Noise Test and Measurement Setup

Audio noise measurement is one of the performance tests implemented on an LCD television motherboard. In this thesis, the audio noise problem is investigated on a 32" LCD television with a 1366x768 panel resolution. A MP7722 class-D type stereo single ended audio amplifier is used on the TV chassis and provides 8W audio output power on 8 ohm speakers.

Attempts to measure noise in audio equipment as RMS voltage, using a simple level meter or voltmeter, do not produce useful results; a special noise-measuring instrument is required. This is because noise contains energy spread over a wide range of frequencies and levels, and different sources of noise have different spectral content. For measurements to allow fair comparison of different systems they must be made using a measuring instrument that responds in a way that corresponds to how we hear sounds. There are three requirements for proper audio noise measurements:

- An audio signal generator
- A bandwidth limiting filter
- An audio signal analyzer

The Audio Precision 2700 series computer controlled instruments are used for audio testing. 2700 series instruments include both signal genration and analysis capability for audio stimulus-response testing. The 2700 series instrument consists of two key components:

- The 2700 series hardware, which provides the physical platform and measurement electronics necessary for proper signal interface and precise signal generation and analysis. The hardware is mounted in a aluminium and steel chassis that can be installed in a standart 19" wide equipment rack using optional mounting hardware.
- The 2700 series control software, AP2700. AP2700 runs on a personal computer and provides the control, computation, display, report and automation functions for a 2700 series instrument. The control software on the PC communicates with the 2700 hardware via a proprietary bus interconnection called the Audio Precision Interface Bus, which requires a dedicated cable and PC-mounted interface card.

4.3.1 Connecting the Device Under Test (DUT)

The 32" LCD television is used as the DUT. It has both audio inputs and outputs. The test signal is applied from the external video input of the TV. The audio precision instrument generator is used to produce the audio test signal. 1kHz sinusoidal sound signal is used as the audio test signal. A colour bar pattern signal is used as the video test signal. Fluke TV pattern generator is used to produce the required video test signal.

For unweighted noise measurements, a bandpass filter is used to limit the signal bandwidth between 20Hz and 20kHz. The Audio Precision AUX-0025 filter accessory is used to meet the required filtering. It is a dual channel, passive, multi-stage LCR filter providing ± 0.05dB flatness over the 20kHz audio passband.

The DUT is measured at its power amplifier outputs(speaker connections), using the left and rigth channels. For this measurement, the loudspeakers are replaced with 8ohm power resistors as a terminating load. The power resistors are connected to the

AUX-0025 filter analog inputs. The measurement bandwidth is limited by the filter and the analog outputs of the filter are connected to the audio precision instrument analyzer inputs. The measured audio and noise signals can be monitored also with an oscillocope by using the analyzer signal monitor outputs. The measurement setup and the connections are as shown in Figure 4.7.

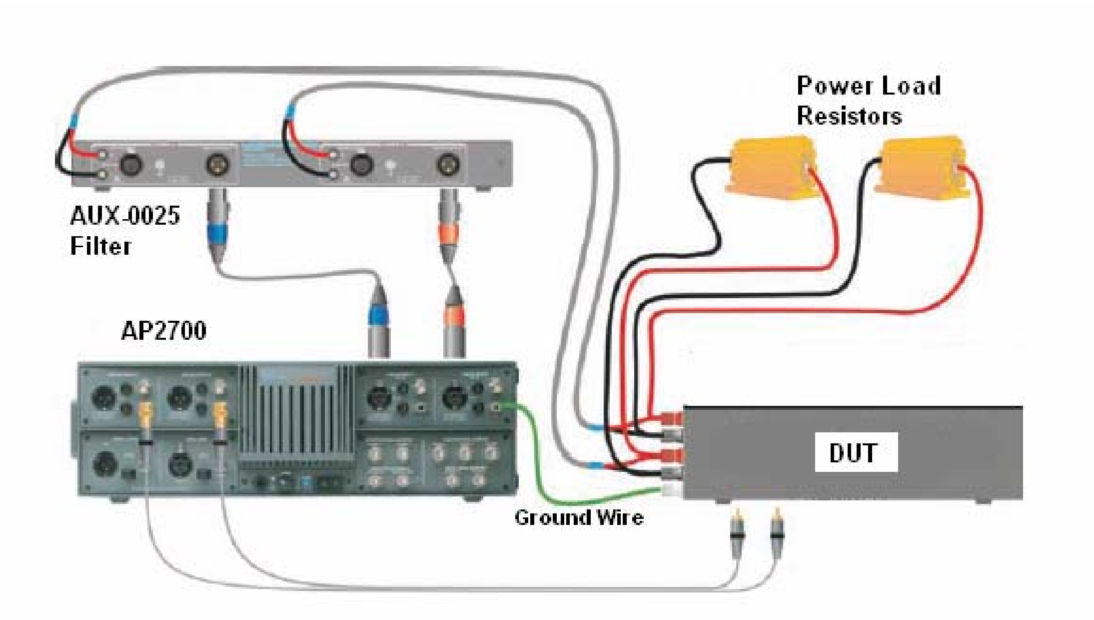


Figure 4.7 Test setup for audio noise measurement

4.3.2 Measuring the Audio Noise

Audio noise measurement requires two measurements to be made successively. Firstly, the signal level is measured at a specified volume level. Then, the noise level is measured by turning off the generator. The ratio between the two measurements gives the audio noise. The measurement steps are as following:

- The DUT volume level is adjusted to maximum for audio noise measurement.
- In the AP2700 control software window, the analog generator and analyzer settings are adjusted. 1kHz, 500mVrms sinusoidal signal is selected in the generator window.
- The Amplitude function is selected on the analyzer window.
- For convenience, the measured results are stored as dBrA and dBrB for both channels.

- In the BW filter fields, the measurement bandwidth is defined between 22Hz and 22kHz.

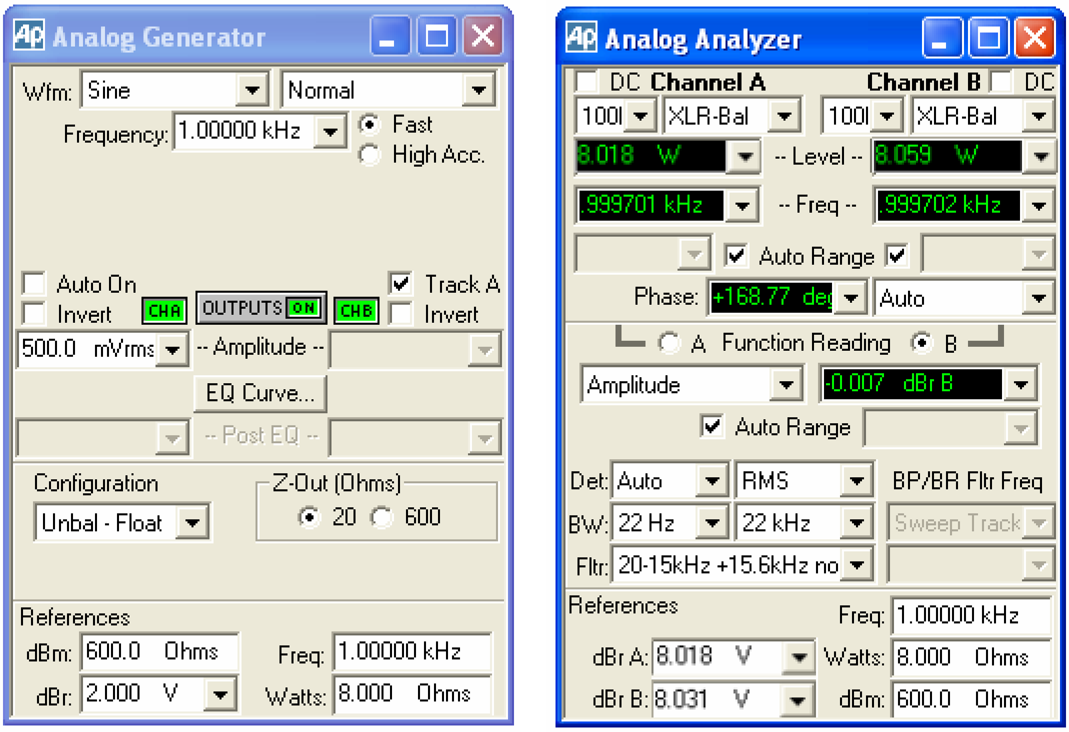


Figure 4.8 Analog analyzer and generator windows

- The measured level on the analyzer is captured as the reference by pressing F4 button.

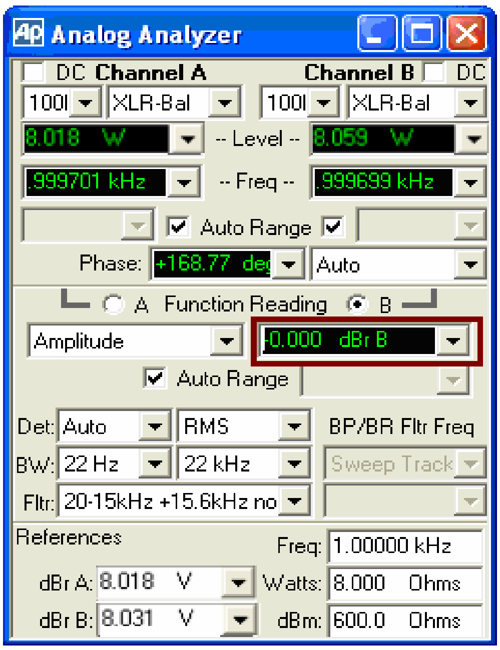


Figure 4.9 Reference level reading

- The generator is turned off and the reading in the function meter gives the ratio between the signal and the noise levels.

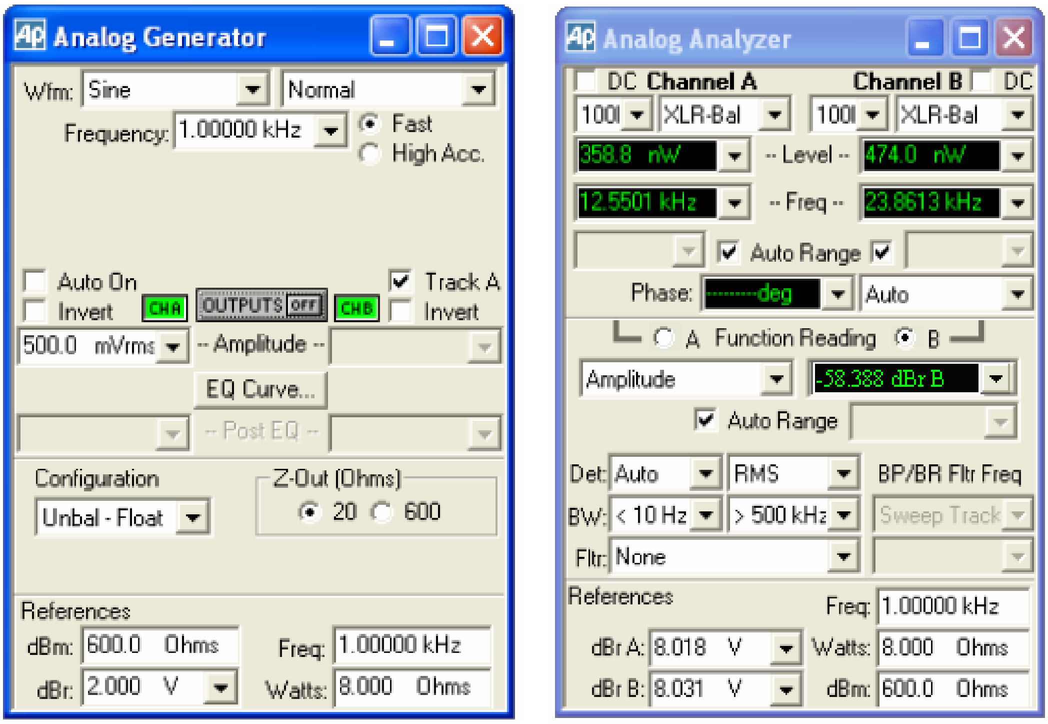


Figure 4.10 Analog generator and analyzer windows when the source is off

The noise reading is -58dBr for both channels. To achieve a good noise performance, the measured noise level should be below the specified limits. Audio noise in LCD television for maximum audio output power should be less than -60dBr.

CHAPTER FIVE

DESIGN AND IMPLEMENTATION OF AUDIO NOISE ELIMINATION CIRCUIT

The noise signal rides on the clear audio signal and is amplified by audio amplifier. So, even the noise has low level, it becomes noticeable at the speaker outputs after amplification. The new method proposed in this thesis is based on detecting the existence of the signal at the input of the amplifier and its level. If the amplitude of the audio signal at the amplifier input is below the specified level, then the signal will be evaluated as noise and the audio amplifier will be disabled by the circuit. If the amplitude of the audio signal is above the specified level, then the signal can not be evaluated as noise and the circuit will enable the amplifier to give output. Investigations showed that signals which are coming from audio processor with an amplitude lower than 20mV can not be easily detected by the user in normal volume level. If the signal level at the input of the audio amplifier is below the specified level, the circuit gives an output with a logic high and if the input signal level is higher than the specified level, the circuit gives an output with a logic low. The output of the circuit is connected to one of the input pins of the system microcontroller. According to the state of this input pin microcontroller controls the audio amplifier. The signal level which will cause a state change at the output of the circuit can also be adjusted easily by the resistor values at the input stage of the circuit. By this way, only the signals which have an amplitude above the specified limit will be amplified and heard from the speakers. The unwanted noise signal has a small amplitude so, it will be cancelled by the circuit and a clear sound will be heard from the speakers.

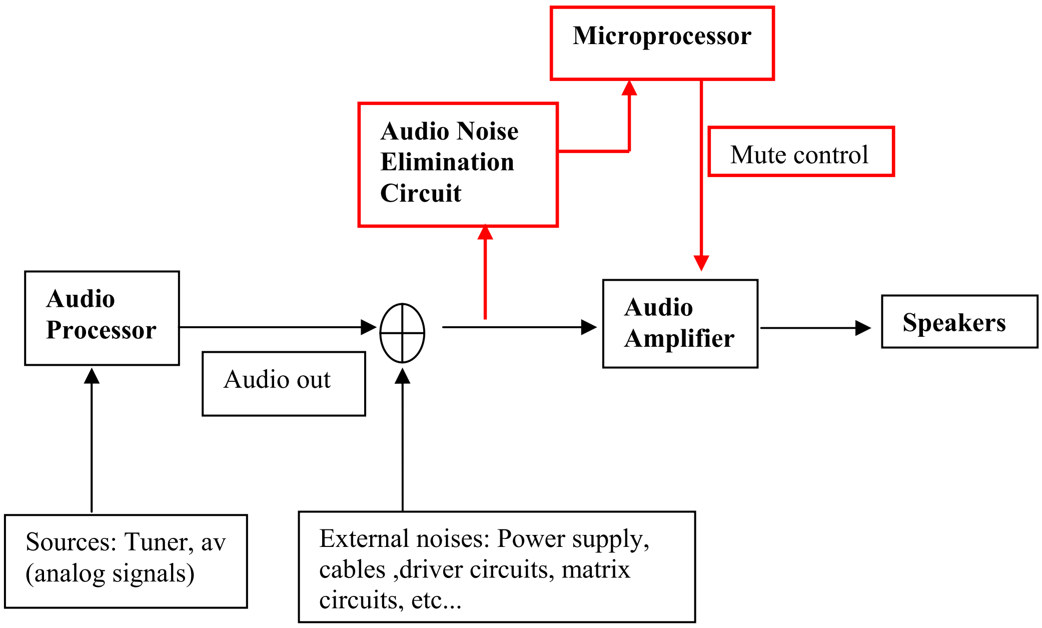


Figure 5.1 Block diagram of the new system with noise elimination circuit

5.1 The Working Principle of the Circuit

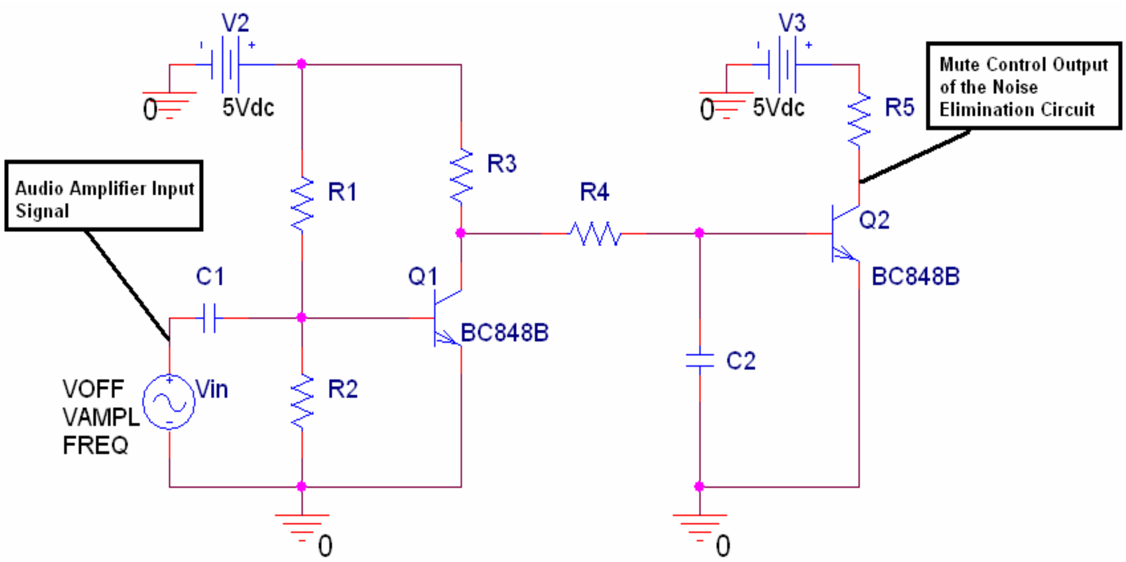


Figure 5.2 Schematic of the circuit

As can be seen from the schematic of the circuit, there are two npn type bipolar junction transistors(BJT) in the circuit. The working of the circuit is based on the modes of operation of the two transistors. Depending on the bias conditions of the emitter-base junction(EBJ) and the collector-base junction(CBJ), different modes of operation of the BJT are obtained. If both the EBJ and CBJ are reverse biased, the transistor operates in the cutoff mode and if both of the junctions are forward biased, the transistor operates in saturation mode. If EBJ is forward biased and the CBJ is reverse biased, the transistor operates in the active mode.

The main purpose of the circuit is to detect the signal levels below a specified level and to give an output with high logic level. If the signal level is above the specified level, then the output of the circuit should change its state from high to low. The input is taken from the audio amplifier input which is amplified and heard from the speakers. The output of the circuit is used to control the audio amplifier output. The circuit detects the existence of input signal and its level. If input signal exists and its level is high enough, than the system lets the audio amplifier to give output. If the signal level is below the specified limit, than the system mutes the output of the audio amplifier. In this implementation, the limit is setted as 10mV by choosing proper resistor values. BC848 npn type SMD(Surface Mount Device) transistors are used in the circuit. The resistors are also chosen as SMD types.

In the circuit, the first transistor is used as a common emitter amplifier and the second one is used as a switch. So, the cutoff and saturation modes of operation are utilized. And the first transistor should operate in the active mode for only a very small range of input voltages. Therefore, a voltage divider bias arrangement is used for transistor Q1 to obtain a very sharp exponential relationship between the collector current i_C and the base-emitter voltage v_{BE}. With this arrangement the transistor is biased by fixing V_{BE}, so any small and inevitable differences in V_{BE} from the desired value will result in large changes in I_C and in V_{CE}, which is the desired condition in our case.

For transistor Q1, the biasing is achieved by using resistors R1 and R2. The total input voltage v_I (bias+signal) is applied between the base and emitter. In order to operate this transistor in saturation mode, the bias voltage should be setted to a dc value, which is above the treshold level of the base-emitter junction. This dc voltage value is adjusted to 1V with resistors R1 and R2 and a small amplitude ac signal rides on this dc voltage. The total output voltage v_O (bias+signal) is taken between collector and ground. Resistor R3 has two functions: to establish a desired dc bias voltage at the collector and to convert the collector signal current to an output voltage. Since the base-emitter voltage $V_{BE_{on}}$ is 0.68V for a BC848 type npn transistor, it will be effectively cutoff for v_I<0.68V and the transistor will begin to conduct for v_I>0.68V. At the fixed bias voltage of 1V, the transistor will conduct and a small resistance path will exist between the collector and emitter. The desired condition for the circuit is to detect the signals with amplitude levels smaller than 10mV. If the amplitude of the ac signal v_{sig} riding on 1V is smaller than 10mV, v_I is always higher than 0.68V and the transistor is always in conducting state(saturation mode). As v_{sig} is increased, i_B will increase, i_C will correspondingly increase and v_C will decrease. Eventually, v_C will become lower than v_B by 0.4V, at which point the transistor leaves the saturation region and enters the active region. In order to detect the signals with an amplitude of 10mV, the total input voltage v_I should be 0.01V(10mV) lower than the dc bias voltage of 1V at edge of saturation point. At saturation, the value of $V_{CE(sat)}$ is fixed at 0.09V for BC848 type transistor. The following steps are used to select the component values:

- The base-emitter voltage V_{BE} is fixed at 1V by selecting R1=10k and R2=2.5k.
- The Thevenin equivalent circuit for Q1 is

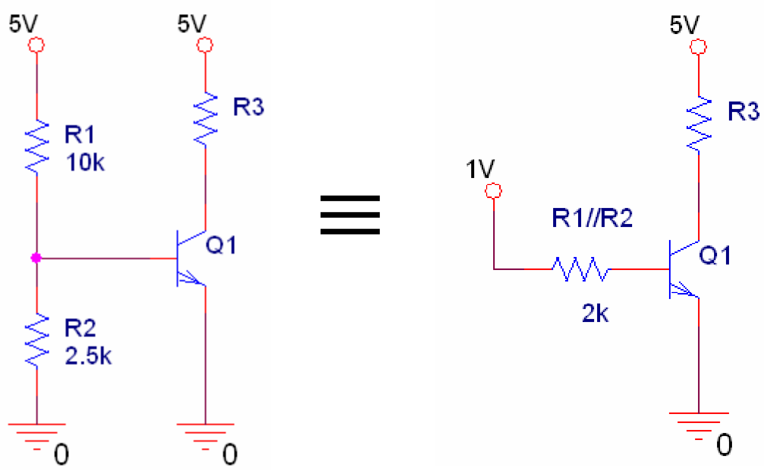


Figure 5.3 Voltage divider bias arrangement and its Thevenin equivalent

- The collector resistance R3 is calculated by using the following formulas at edge-of-saturation point(EOS).

$$v_{I(EOS)} = i_{B(EOS)} * R_B + V_{BE} \Rightarrow i_{B(EOS)} = \frac{v_{I(EOS)} - V_{BE}}{R_B} \tag{5.1}$$

At edge of saturation point v_I should be 0.99V which is 1V-0.01V. The base-emitter junction treshold voltage is 0.68V for BC848 type transistor. R_B is the parallel combination of resistances R1 and R2 in the Thevenin equivalent circuit. Substituting the required values in the above equation gives

$$i_{B(EOS)} = \frac{0.99 - 0.68}{2k} = 0.15mA \tag{5.2}$$

For a BC848 type transistor the typical value of current gain β is 20 at saturation. So, the collector current is,

$$i_{C(EOS)} = i_{E(EOS)} * \beta = 3mA \tag{5.3}$$

And the collector resistance is given by,

$$R3 = \frac{V_{CC} - V_{CE(sat)}}{i_{C(EOS)}} = \frac{5 - 0.09}{3mA} = 1.6k \tag{5.4}$$

The R4 and C2 combination forms a lowpass filter with a very low cutoff frequency. The main purpose of using this circuit is to filter out the dc component of the amplified signal at the collector of Q1. The cutoff frequency can be calculated as,

$$f_c = \frac{1}{2\Pi RC} \ . \tag{5.5}$$

Substituting 64k and 4uF in the above formula for R and C respectively gives f_c as 0.6Hz, which means that only the DC level of the collector voltage appears at the base of Q2. With the calculated component values above, if the input signal level is lower than 10mV, Q1 stays in saturation mode because the necassary biasing voltage is achieved by using the voltage divider bias arrangement. The low voltage level at the collector of Q1 causes Q2 to operate in cutoff mode. Thus, the collector current of Q2 will be negligibly small and the collector voltage will be equal to Vcc(5V). As the amplitude of the input voltage is increased above 10mV, Q1 stays in saturation mode at positive intervals. At negative intervals of input signal, Q1 is in active mode. So, it amplifies the input signal and drives the lowpass filter circuit, which produces a high dc level at the base of Q2. This high DC level takes Q2 to on mode(the switch is on) and the output voltage of the circuit will be approximately equal to 0V. So, if the amplitude of the input signal is above 10mV, the circuit gives an output with low logic level.

The figures below show the input voltage, Q1 collector voltage and the output of the lowpass filter for 5mV and 30mV input signal amplitude levels.

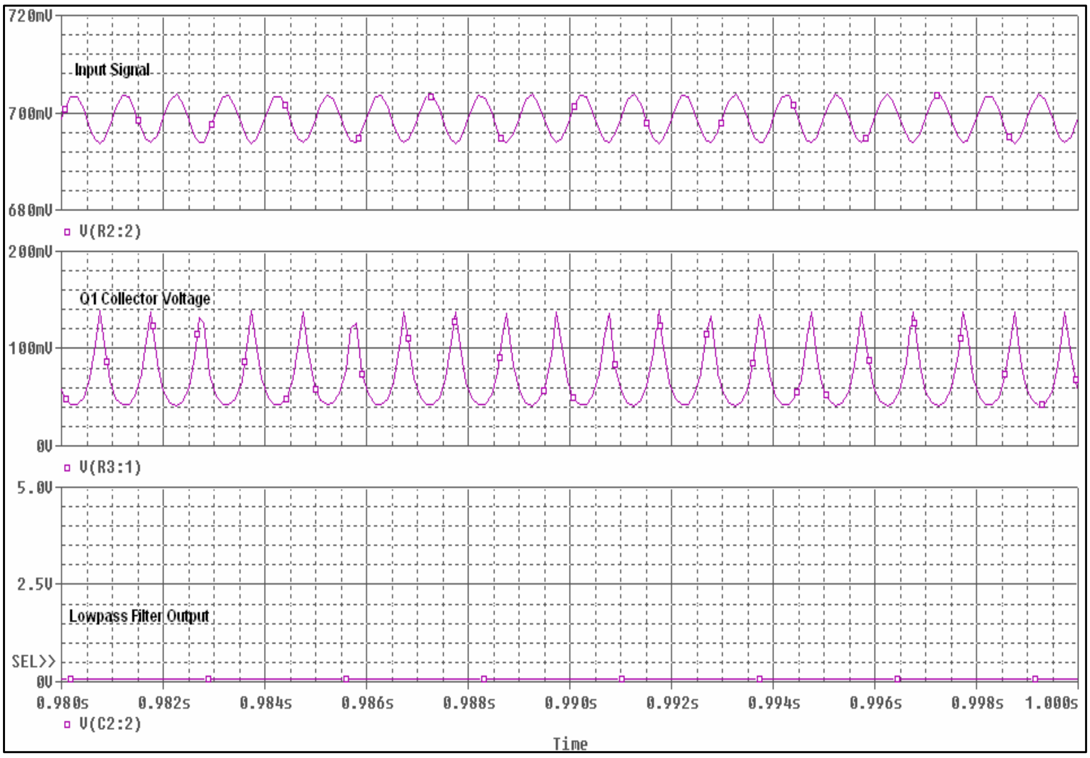


Figure 5.4 Orcad simulation result for 5mV input amplitude level

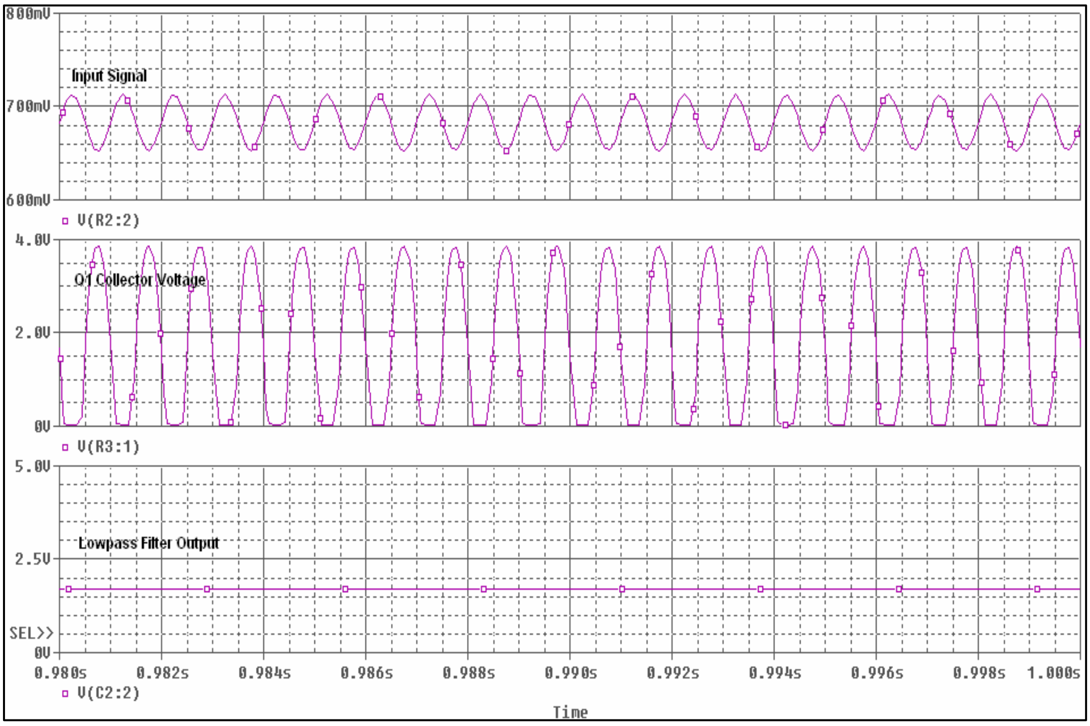


Figure 5.5 Orcad simulation result for 30mV input amplitude level

As can be seen from the results, if the amplitude of the input signal is smaller than 10mV, such as in Figure 5.4, a very low voltage is obtained at the output of the lowpass filter, which causes Q2 to operate in cutoff mode and the circuit gives an output with high logic level. If the amplitude of the input signal is higher than 10mV, such as in Figure 5.5, 1.7V is obtained at the output of the lowpass filter, which causes Q2 to operate in saturation mode and the circuit gives an output with low logic level.

The figures below show the mute control output for 9mV and 12mV input signal amplitude levels.

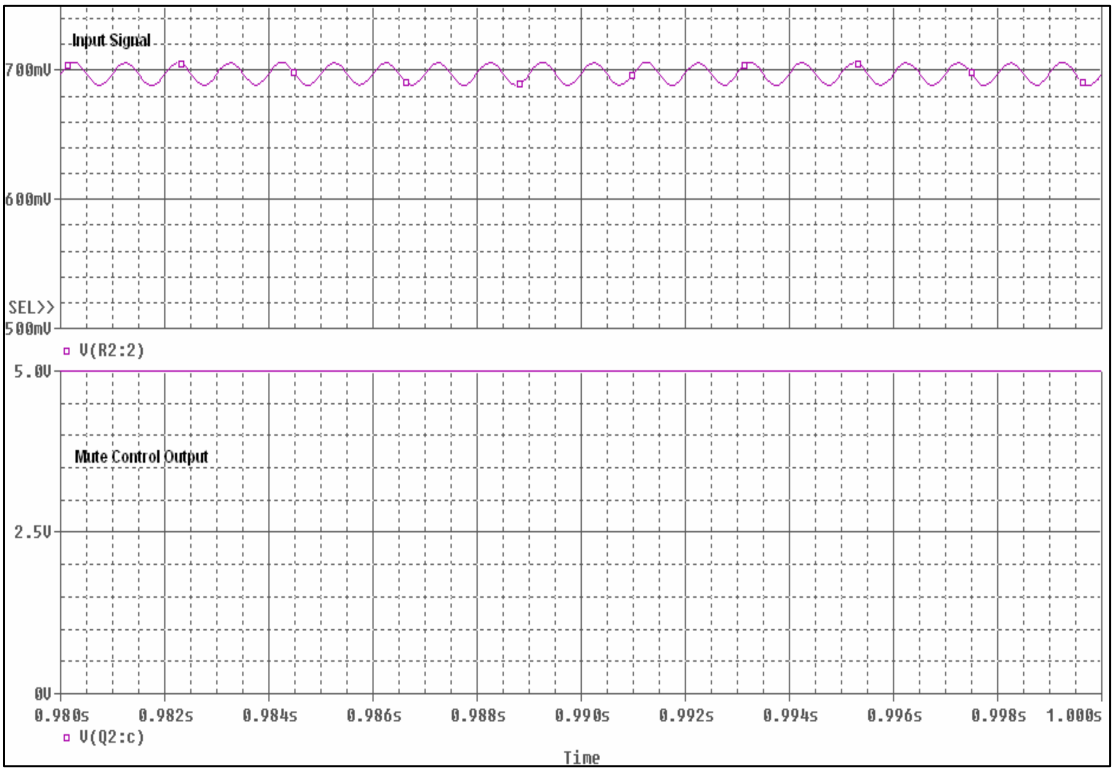


Figure 5.6 Mute control output for 9mV input signal

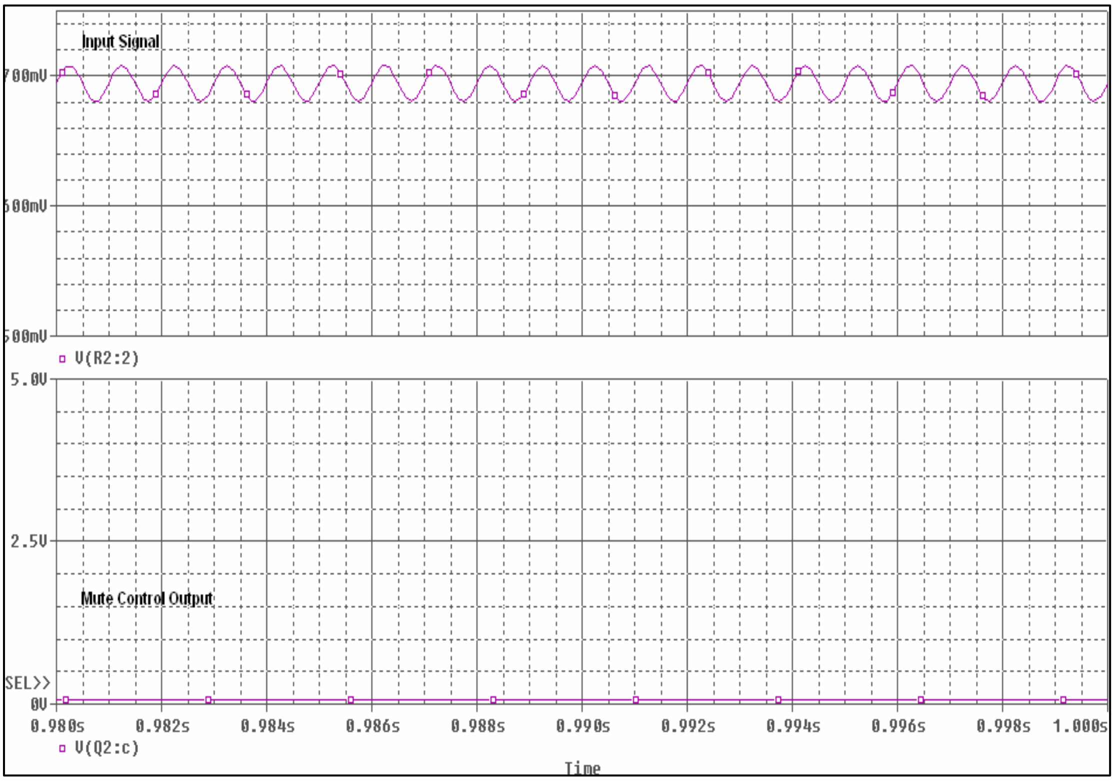


Figure 5.7 Mute control output for 12mV input signal

The same circuit should be used for left and right channels to detect the signal level at both channels. The audio amplifier should be muted only if the signal level is lower than 10mV at both channels. If the signal level is higher than 10mV at one of the channels or at both of the channels, the circuit output is low and the amplifier is unmuted. This arrangement can be used for stereo applications.

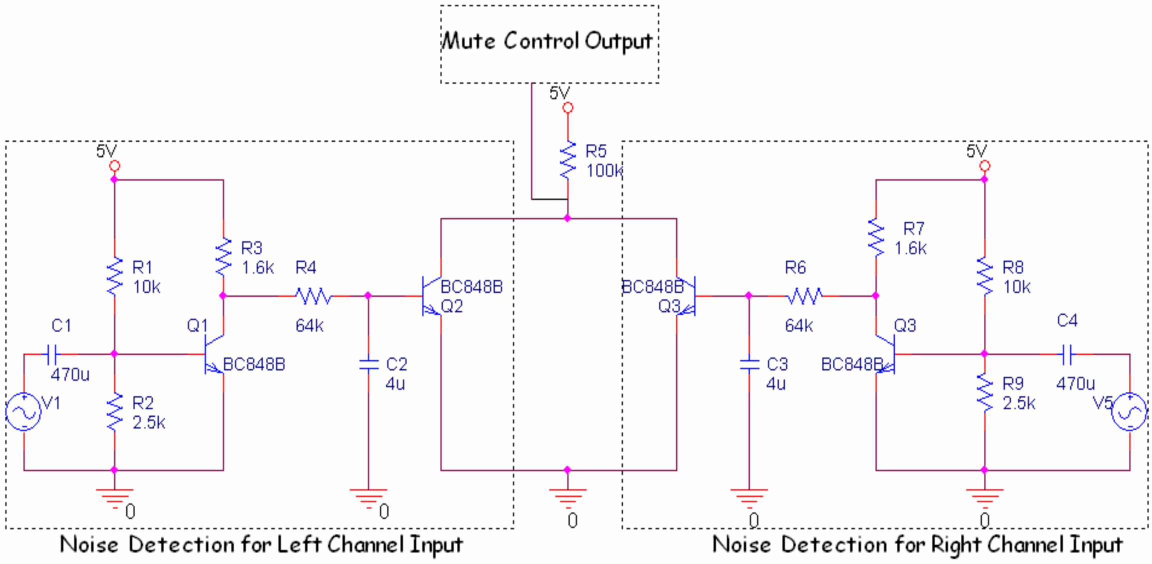
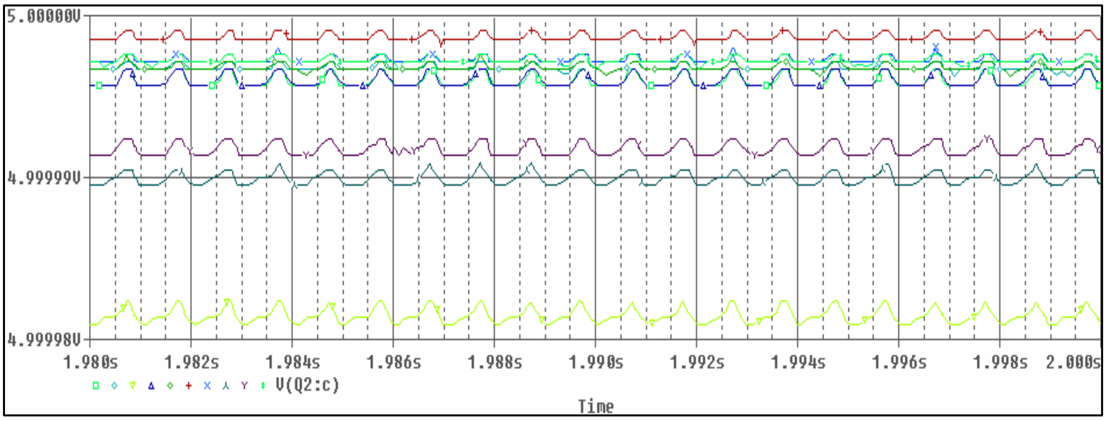
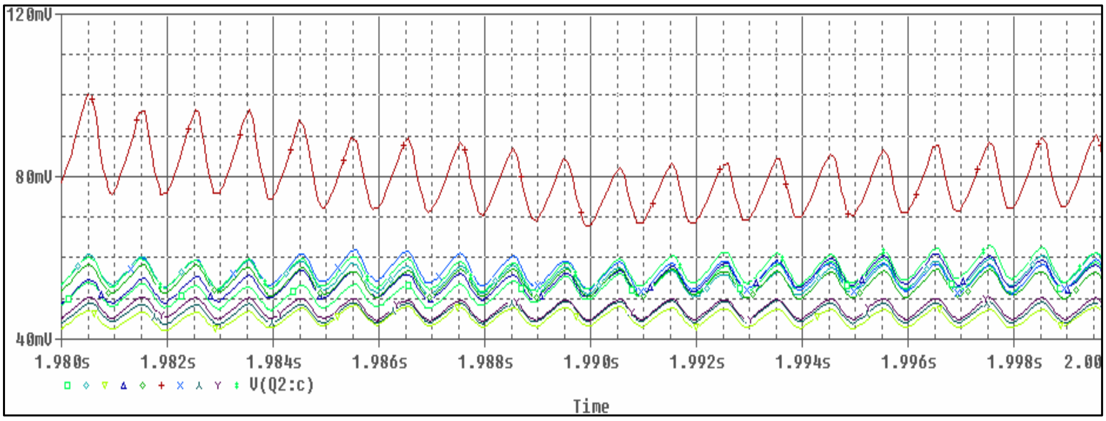


Figure 5.8 Audio noise elimination circuit drawn for stereo applications

In this application the component values are calculated in order to operate the circuit to be able to detect the signals below and above 10mV amplitudes. This level can be changed by selecting different values for R1 and R2, because the fixed dc base voltage for Q1 is adjusted with R1 and R2. If the base voltage is adjusted to a higher dc level, then the circuit can detect higher signal levels. Therefore, the operation of the circuit is sensitive on the characteristics of the transistors and the tolerances of the resistors R1 and R2. The Monte Carlo simulation results below show how the tolerances of the resistors will affect the performance of the circuit. The tolerances are setted to %5 while the amplitude of the sinusoidal input voltage is at 8mV and 15mV.



(a)



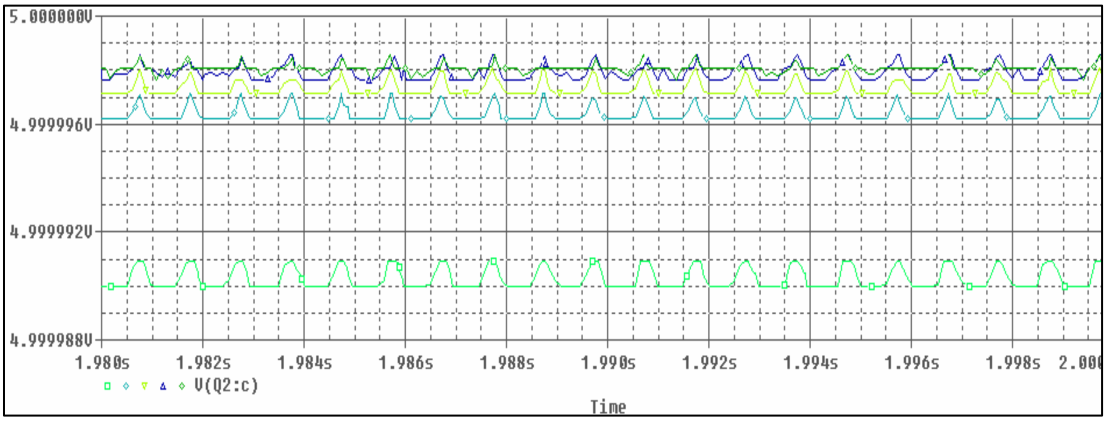
(b)

Figure 5.9 Mute control outputs for Monte Carlo simulation results with %5 resistor tolerances:

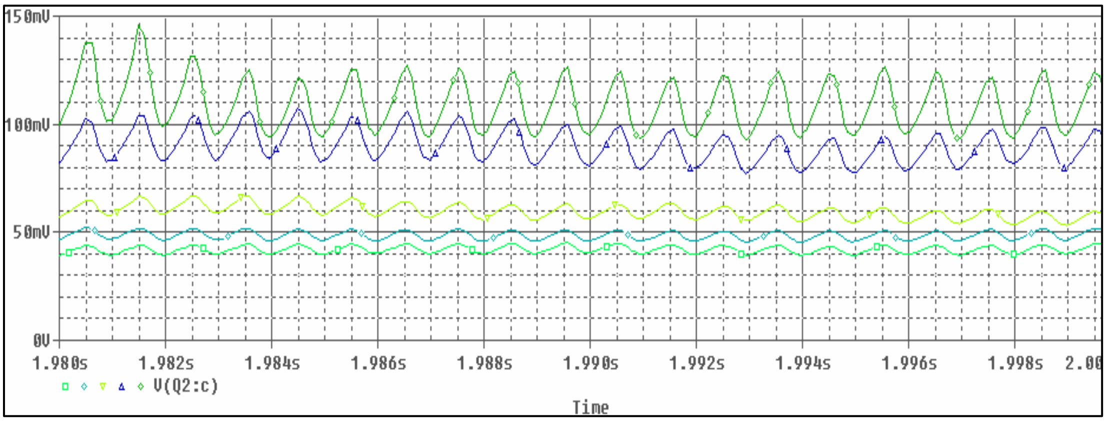
a) 8mV sinusoidal input signal

b) 15mV sinusoidal input signal

During the operation, the temperature inside the television may reach to high values. The designed circuit should also operate properly at these high temperature values. The figure below shows the simulation result obtained for mute control output at different temperatures.



(a)



(b)

Figure 5.10 Mute control outputs for temperature sweep simulation results with temperature values at 15, 25, 35, 45 and 50 ^0C a) 8mV sinusoidal input signal b) 15mV sinusoidal input signal

5.2 The Implementation of the Circuit

The designed circuit is implemented on the same 32" LCD television as in chapter 4. The amplifier input is connected to the input of the circuit. The output of the circuit is connected to one of the input pins of the system microcontroller. The enable pin of the amplifier is connected to one of the output pins of the microcontroller. By a simple software control depending on the state of the input pin, microcontroller changes the state of the output pin and controls the amplifier. And the measurement result showed that the applied circuit has improved the noise performance of the TV by decreasing the audio noise level to -64.871 dBr for channel A and to -63.486 dBr for channel B.

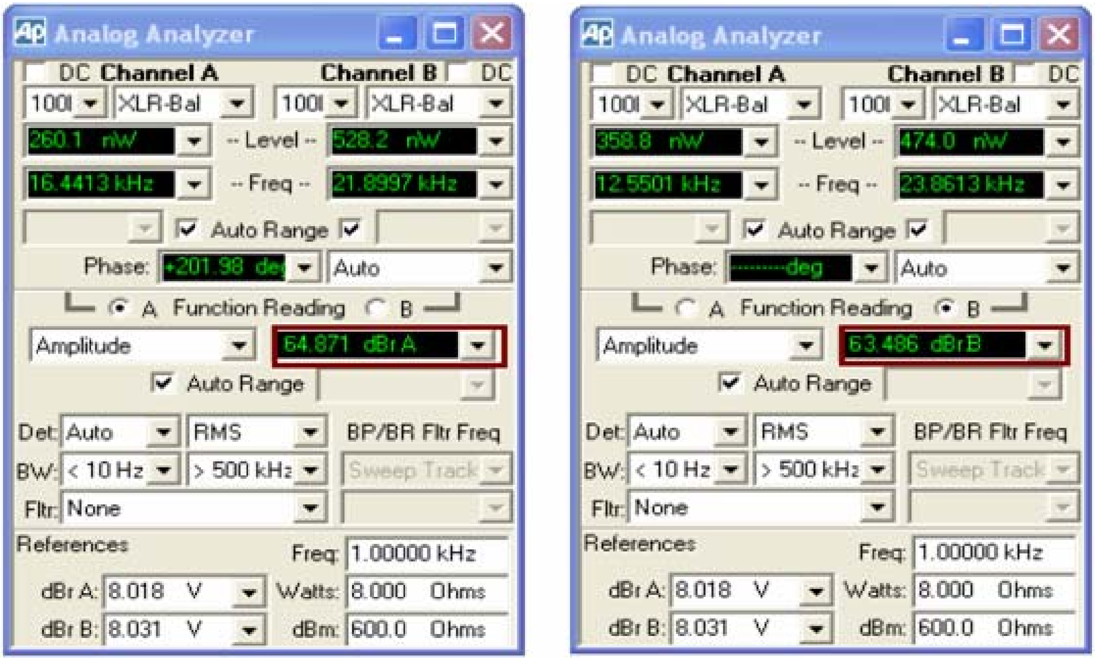


Figure 5.11 Audio noise measurement for 32” TV

During operation the circuit detects the audio signal level at the input of the amplifier. And if the signal level is below the specified limit, the amplifier is muted, if it is above the specified limit, the amplifier is unmuted. The TDA8933 class-D type audio amplifier has been used on the chassis of the television where the circuit is implemented. Because the audio signal is a continuous signal with changing amplitude levels, the amplifier is muted or unmuted very often. Therefore, the time interval between enabling or disabling the amplifier and the output of the amplifier becomes an important parameter that will effect the performance of the circuit. The oscilloscope printouts below show the amplifier enable and the output signals.

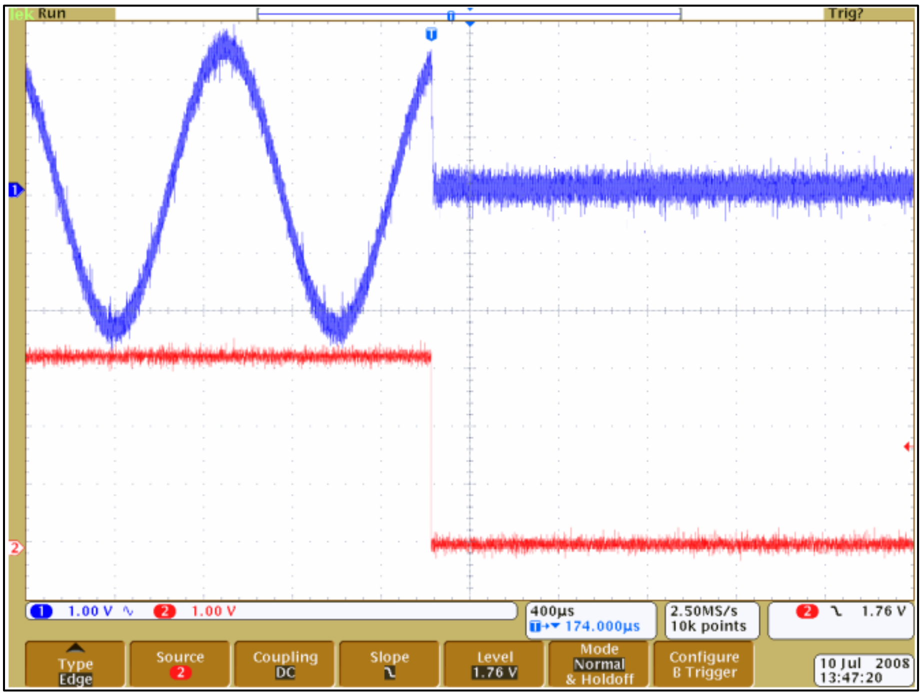


Figure 5.12 Mute state for TDA8933 with red showing amplifier enable and blue amplifier output signals

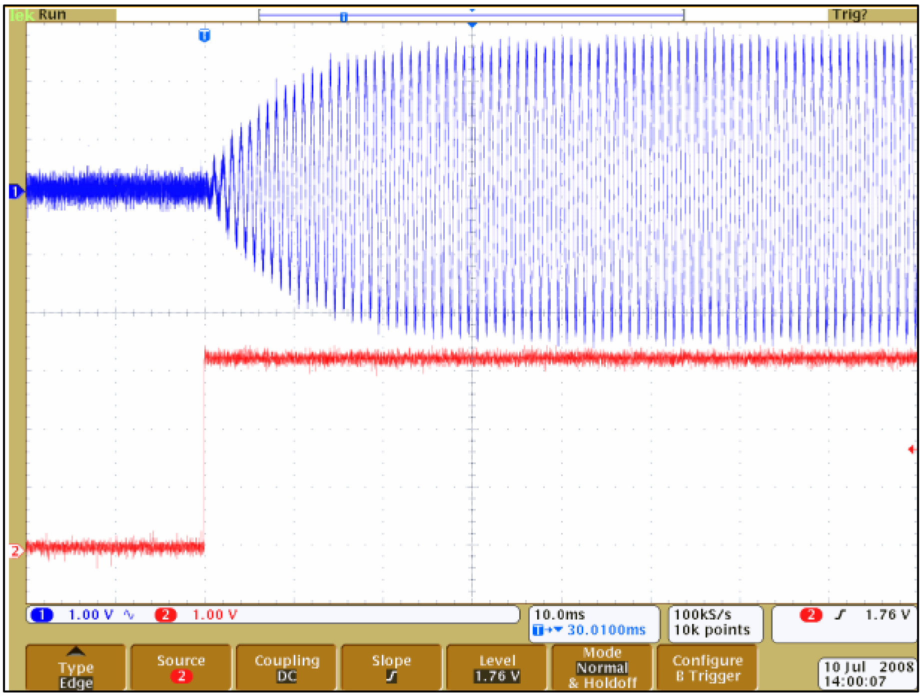


Figure 5.13 Unmute state for TDA8933 with red showing amplifier enable and blue amplifier output signals

As can be seen from the figures above, there is not a time difference between the input and the output of the TDA8933, so the circuit can operate properly as intended. The amplifier can switch between the mute and unmute states very fast. The muting and unmuting of the amplifier is achieved with an internal circuitry consisting of a transistor for switching which provides a long life time for the amplifier.

5.3 The Advantages of Audio Noise Elimination Circuit

The causes of audio noise problem and the design steps of the proposed circuit are explained in the previous sections. The performance of the circuit is analyzed by using the simulation programs and also by connecting the physical circuit on the 32" LCD TV chassis. The measurement results show that the circuit has decreased the audio noise level and has improved the performance of the TV. The following items list the major advantages of using the designed circuit:

- It is possible to use in every system including audio processor and audio amplifier.
- The effect of other circuits on audio quality can be reduced.
- In silent scenes of the programs, noise coming from internal circuits will be cleared.
- If no input exists at external sources while the volume level is high, the user will not hear any noise.
- The circuit can be implemented only by using SMD components and can be integrated easily on the motherboard of electronic equipment.
- The circuit consists of commonly available components and the total cost is very low.

The same operation to reduce the audio noise can also be implemented in different ways such as using an operational amplifier as a comparator and connecting the audio input voltage and the treshold voltage to its inputs. A standart op-amp operating without negative feedback can be used as a comparator. When the voltage difference between the noninverting and the inverting inputs is positive, the high gain of the opamp causes its output to switch to the positive supply. When the voltage difference is negative, opamp output switches to the negative supply. In order to sense the signal amplitudes below and above 10mV with an opamp comparator, 10mV dc signal should be produced on the chassis and this voltage should be applied to the noninverting input of the opamp. The effective value of the signal at the audio processor output should be connected to the inverting input of opamp. The opamp output changes its state depending on the difference between the inputs and this state change can be used as mute control of the amplifier. But, converting the ac signal at the output of the audio processor to dc by finding its effective value requires another circuit to be connected between audio processor output and the comparator input. So, the whole system will be more complicated than the proposed circuit in this thesis. It will also require a larger area on the PCB. The total cost will be also much higher than the proposed circuitry.

Another approach to reduce the audio noise is using hum eliminators produced by different companies. But these devices are large in size and are not suitable for placing on the equipment chassis. They are placed on the audio path between different devices and can only be used by connecting different audio instruments together such as in home theater systems or in TV studios.

CHAPTER SIX
CONCLUSION

In this thesis, the causes of noise on analog audio signals are investigated and a new solution is proposed to reduce the audio noise. The proposed solution consists of a circuit which is mounted on the motherboard of the electronic equipment. Because the new circuit is implemented on an LCD TV chassis, the basic building blocks of an LCD television and the audio noise measurement procedure are also explained.

The most important reason of audio noise is the grounding problems on the PCB of the equipment or between instruments connected together. The purpose of grounding is to provide safe, reliable and cost efficient power distribution. In the case of sensitive electronic systems, such as audio, video and computer systems it is also necessary that the grounding system provide a stable and low impedance connection to earth. The main problem related with grounding is the generation of ground loops while connecting different grounding points. These ground loops act as loop antennas, where the beginning and end of the loop are connected. The ground loops tend to pick up 50Hz AC signal being broadcast by a building’s electrical wiring with all of the harmonics and electrical noise from other components on the PCB and other instruments. All of these AC signals are heard as hum or buzz like signals from the speakers of the audio equipment and degrades the performance of the specified equipment. The isolated star ground is the most popular implementation of technical grounding to overcome different grounding problems. This approach has been in practice for many years, while it is not a perfect solution, it is the solution that presents the least compromises.

Another cause of audio noise is the electromagnetic interference. Electromagnetic interference is a disturbance that affects an electrical circuit due to electromagnetic radiation from an external source. These unwanted electromagnetic signals make the operation of the circuits difficult or impossible. The most significant contributers to the radiated electromagnetic signals are the systems operating at high frequencies like SMPS and memory devices. The most common applications to reduce the noise caused by the EMI is to use shielded cables or to provide ground planes near high frequency signal traces. By this way, the noise signal can be bypassed to the ground through a very small resistance path. Also ferrites can be used for input and output lines.

The new proposed circuit in this thesis can be integrated on the PCB of any audio equipment. It consists of only SMD components and provides a very cost effective solution. The circuit is tested on an LCD television chassis and provides a clear audio signal. The measurement results showed that the circuit reduces the noise level in audio signal. The circuit accomplishes this by eliminating all of the signals below a specified level regardless of the source generating the noise. This level can also be adjusted by selecting different component values for different applications. The audio noise problem is one of major performance problems that can affect the productibility of the specified equipment therefore, special care should be taken during the design. The new solution provides a high degree of audio performance and also reduces the design time.

REFERENCES

Atkinson, C. & Giddings, P. (1995). Grounding Systems and Their Implementation. *Journal of the Audio Engineering Society, 43* (6). Retrieved March, 2008, from http://www.engineeringharmonics.com/papers/ground_systems.pdf

Broeders, J. H., Meywes, M., & Baker, B. (2008). *Noise and Interference.* Retrieved May, 2008 from http://www.nalanda.nitc.ac.in/industry/AppNotes/BurrBrown/appnotes/DesignSem3.pdf

Colours of Noise. (2008). Retrieved May, 2008 from http://en.wikipedia.org/wiki/Colours_of_noise

Cortina, E. (November 2, 2007). *Signals and Noise.* Retrieved February, 2008, from http://dpnc.unige.ch/tp/elect/doc/04-Signal_and_Noise.pdf

Electromagnetic Interference – Considerations in Structured Cabling Systems. (2002). Retrieved April, 2008 from http://www.siemon.com/us/white_papers/archive.asp

Electronic Noise. (n.d). Retrieved February, 2008, from http://www.ceid.upatras.gr/faculty/alexiou/ahts/presentations/02_Noise.pdf

EMI Control Applications Notes For Typical Computer Subsystems. (2008). Retrieved May, 2008 from http://www.steward.com/techinfo/subsystems.html

Hofer, B. (October, 2003). *Measuring Switch-mode Power Amplifiers.* Retrieved May, 2008 from http://ap.com/library/whitepapers.htm

Najmabadi, F. (2006). *BJT AmplifierCircuits.* Retrieved May, 2008 from http://aries.ucsd.edu/NAJMABADI/CLASS/ECE65/06-W/NOTES/BJT3.pdf

Noise in Analog Circuits. (2005). Retrieved April, 2008, from http://hamers.chem.wisc.edu/chem628_fall2005/noise/analog_circuit_noise.pdf

Noise. (n.d). Retrieved January, 2008, from http://en.wikipedia.org/wiki/Noise

PCB Grounding System and FAN2001/2011 High Performance DC-DC Converters. (May 5, 2005). Retrieved March, 2008 from http://www.fairchildsemi.com/an/AN/AN-42036.pdf

RF, RFIC&Microwave Theory, Design. (n.d). Retrieved February, 2008, from http://www.odyseus.nildram.co.uk/RFIC_Theory_Page.htm

Switching Amplifier. (2008). Retrieved May, 2008 from http://en.wikipedia.org/wiki/Switching_amplifier

TVS Diode Application Note. (January 2, 2002). Retrieved April, 2008, from http://www.semtech.com/pc/downloadDocument.do?id=497