ELECTRONIC MUSIC AND MIDI REVOLUTION

A SUMMER INTERN REPORT

Submitted by

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Rajiv Gandhi University of Knowledge Technologies
Basar, Adilabad (Dist), Andhra Pradesh

JULY 2014

ROCKSTUDIOS

Electronic Music Production

Kukatpally, Hyderabad

CERTIFICATE

Certified that the summer internship report on **"Industrial exposure to Industry of Electronic Music"** is the bonafide work of "**Kodiganti Akash,** Roll No: **B092195**" 3rd Year B.Tech in **Electronics & communication** Engineering of RGUKT Basar Campus of Rajiv Gandhi University of Knowledge Technologies (RGUKT), Andhra Pradesh carried out under my supervision during 26.5.2014 to 30.07.2014.

Place: Kphb, kukatpally
Date: 30-07-2014

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CERTIFICATE

Certified that the summer internship report on **"MATLAB Signal Processing as a part of Industrial exposure to Industry of Electronic Music"** is the bonafide work of "**Kodiganti Akash,** Roll No: **B092195**" 3rd Year B.Tech in **Electronics & communication** Engineering of RGUKT Basar Campus of Rajiv Gandhi University of Knowledge Technologies (RGUKT), Andhra Pradesh carried out under my supervision during 26.5.2014 to 30.07.2014.

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ABSTRACT

In a Musical Industry:

Signal Processing works are tremendously used in Musical industry for Digital Audio Production where recording, mixing and editing and many Audio FX are implemented so that the resultant audio file obtained is of a high quality with improved timbre (tonal accuracy) and other sound parameters.

An audio file of an instrumental format (not the commercial mp3 songs) consists melodies, harmonies, tunes of different instruments been played along with vocals (however vocal is not considered as of now). For a person who want to learn how a song (melody or tune ,the way the wave is varying w.r.t. time) can be played in his instrument, he must know what is the pitch and the scale (if necessary chords) .

A musician who has a good knowledge on frequencies of every note that is being played, can reproduce by trial and error methods to start from the Root note, then finding chords and scale then the exact tune. But for a beginner who wants to play the exact music might be a tough task.

Now, a software in which the required audio file is given , working out the processing steps inside it generates the frequencies (pitch) , next chords, later for scale then itself shows the notes with respect to time.

Adding audio effects to this project makes it look great for a musician to perform advanced signal processing steps. Channel separation, tempo changing etc., can be included in the later versions.

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REFERENCES

1. INTRODUCTION

MUSIC

Technically, the word Music refers to a complex amalgam of melody, harmony, rhythm, timbre and also includes the silence, in a particular structure.
Mathematically, it is the formulation of Trigonometric and Fourier analysis which includes complex matrices with complex computations.
Psychologically, it is unique in each person's life depending on how they perceive it. Music stimulates, changes mood and feelings also sometimes bounds to spiritual, mental therapy and reduce stress.
Scientifically, it is the harmony of waveforms that travel in space with an orderly sequence of arrangement so as to produce a unified and continuous propagation of waves.
Artistically, Music is the capability of brain for intelligence and style, heart for genuine and believable emotions, courage to do something creative/exciting.

Gramophone

Magnetic tape cassette

Digital Music

Mechanical Music: In earlier days, music could be created though the mechanical triggering of a musical device.

Tape based Music: When tape-based recording came along in the middle part of the last century, it became possible to edit two or more problematic performances together into a single, good take. *Magnetic recording* is a backbone technology of the electronic age. It is a fundamental way for permanently storing information.

Electronic or Digital Music:
However, when it came to the encoding of a musical passage and then faithfully playing it back while still being able to edit or alter the tempo, notes, and control variables of a performance.

MIDI ORIGIN

Keyboard synthesizers were commonly monophonic devices (capable of sounding only one note at a time) and often generated a thin sound quality. These limiting factors caused early manufacturers to look for ways to combine instruments together to create a thicker, richer sound texture.

Father of MIDI: Dave Smith, January 2013: Technical Grammy for the creation of MIDI.

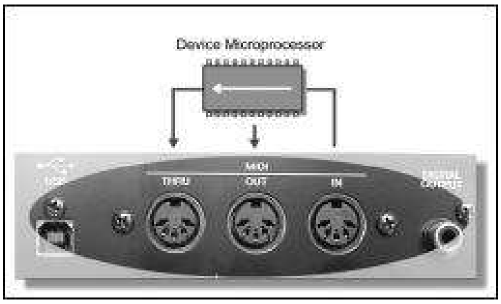
Dave Smith

Electronic Music Creator

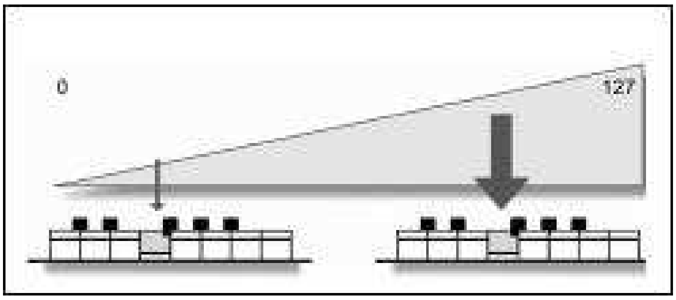
Synchronization resulted in developing MIDI:
Creating a digital electronic instrument protocol, which was named the Universal Synthesizer Interface (USI). As a result of this early protocol, equipment from different manufacturers could be able to communicate directly.
In the fall of 1981, USI was proposed to the Audio Engineering Society. During the following two years, a panel (which included representatives from the major electronic instrument manufacturers) modified this standard and adopted it under the name of Musical Instrument Digital Interface (MIDI).

MIDI REVOLUTION

The Musical Instrument Digital Interface is a digital communications protocol. That’s to say, it is a standardized control language and hardware specification that makes it possible for electronic instruments, processors, controllers, and other device types to communicate performance and control-related data in real time.

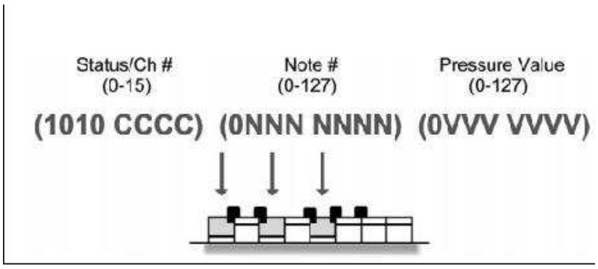


MIDI Device

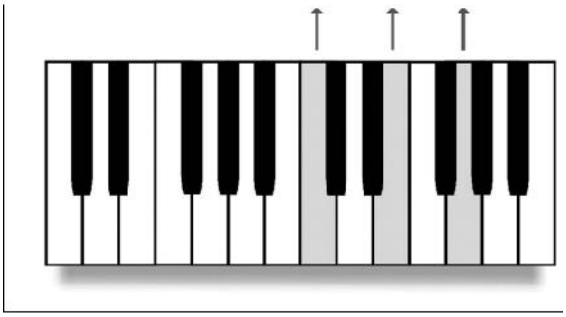


Pressure variation in pressing key

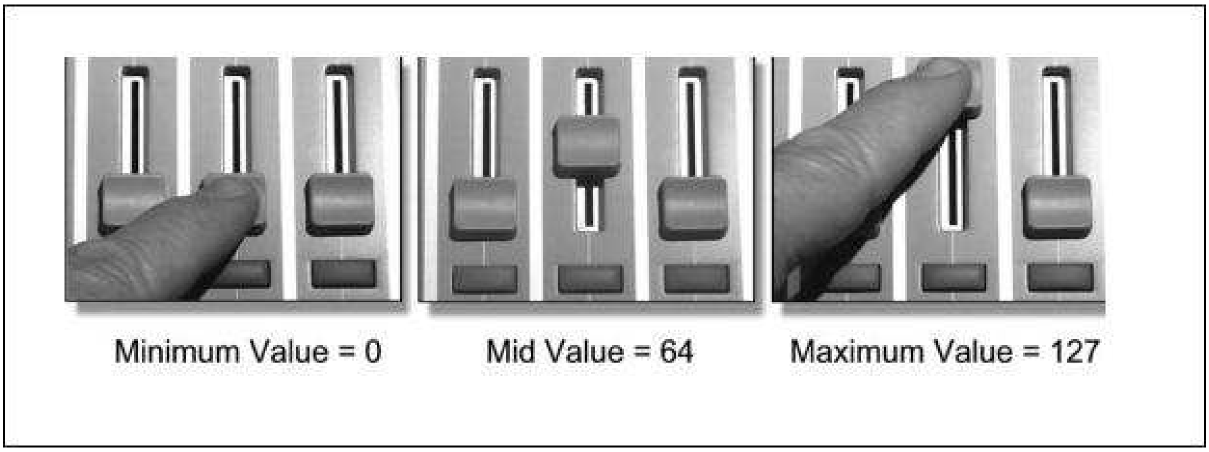
- The Digital Word format is the one that features at MIDI revolution.
- The MIDI Message is transmitted in either Serial or in parallel communication.
    Channel Pressure Messages
    Polyphonic Key Pressure
    Pitch Bend Messages
- MIDI data can only travel in one direction through a single MIDI cable.
- It is used in mobile ring-tone in past days, recording studios and much of multimedia applications are based on MIDI data.
- MIDI specification is of 2 bytes the status byte and the data byte.
    A status byte is used to identify the type of MIDI function.
    A data byte is used to associate a value to the event that's given by the accompanying status byte.
- Up to 16 discrete MIDI channels can be transmitted through a single MIDI cable or designated port.
- MIDI Modes - Poly/Mono
    Poly: An instrument that's set to respond to MIDI data polyphonically will be able to play more than one note at a time.
    Mono: Conversely, an instrument that's set to respond to MIDI data monophonically will only be able to play a single note at any one time.
- Controller Values
    Channel Volume
    Stereo Balance
    Sound Variation
    Sound Timbre



Polyphonic key pressing



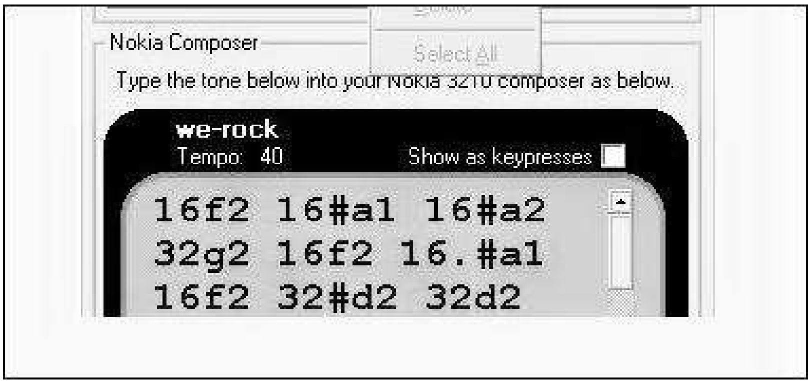
Chords on piano (C major)



Equalizer with dB values

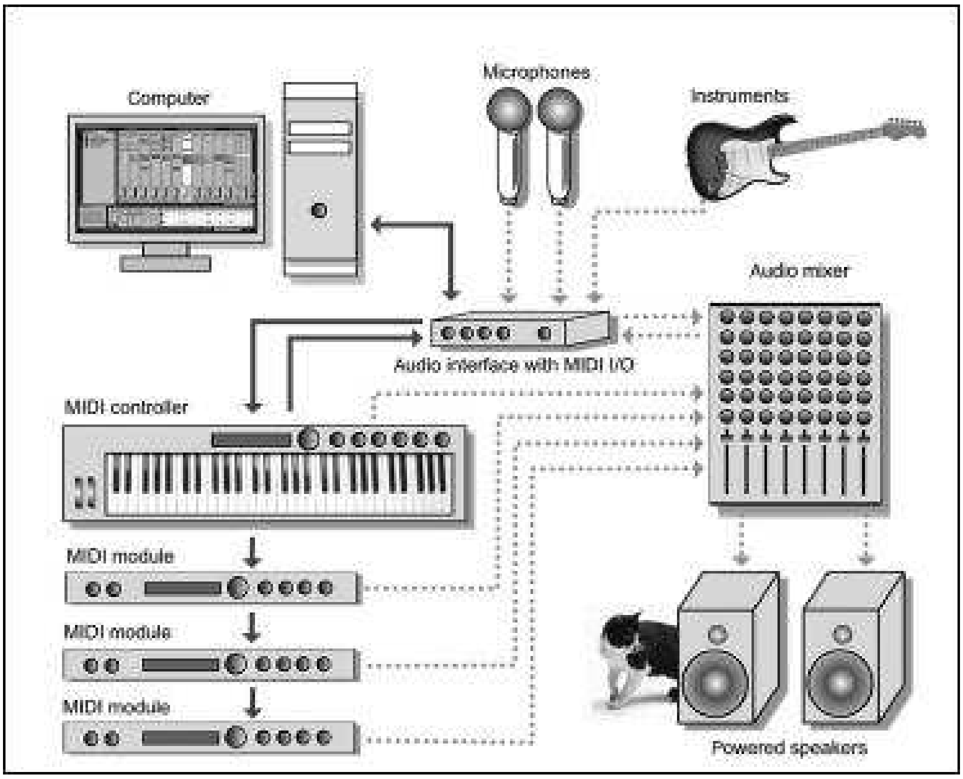
MIDI ON PHONE

With the integration of the General MIDI standard into various media devices, one of the fastest-growing MIDI applications, surprisingly the oldest mobile phones which comfortably resting in our pocket —the ring tone on our cell phones .MIDI helps us to reach out and touch someone through ring tones.



Nokia 1100 model **The MIDI tone - as a ringtone**

ELECTRONIC MUSIC



Electronic Music Production Setup

With the introduction of **electronic music production and MIDI**, a musical performance could be captured in the digital domain and then faithfully played back in a production-type environment that mimicked the traditional form and functions of multi-track recording. Basic tracks could be recorded one at a time, allowing a composition to be built up using various electronic instruments. MIDI finally made it possible for a performance track to be edited, layered, altered, spindled, mutilated, and improved with relative ease and under completely automated computer control. If you played a bad note, fix it. If you want to change the key or tempo of a piece, change it. If you want to change the expressive volume of a phrase in a song, just do it! Even its sonic character (timbre) can be changed. These capabilities just hint at the power of MIDI!

FIRST: The introduction of **Yamaha's popular DX-7** synthesizer in the winter of 1983 pioneered Electronic Music Production.

Yamaha DX-7 synthesizer **AKAI MPC500 sampler**

Over the course of time, new instruments came onto the market that offered improved sound and functional capabilities that led to the beginnings of software sound generators, samplers, and effects devices. With the eventual maturation of software instruments and systems that could emulate existing devices or create entirely new range of functions and sound, hardware controllers began to quickly spring onto the scene that made use of MIDI to communicate physical control movements into analogous moves in a program or plug-in software interface.

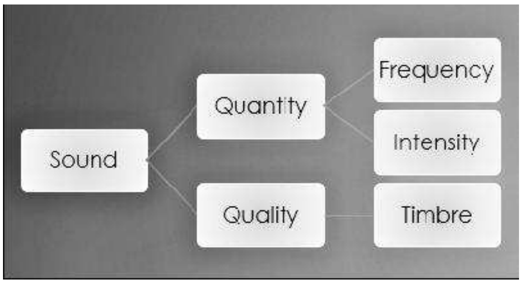
With the introduction of drum machines, modern day synths, samplers, and powerful hardware or software instruments, it is not only possible but also relatively easy to build up a composition using instrument voices that closely mimic virtually any instrument that can be imagined. In the early days, studio musicians spoke out against MIDI, saying that it would be the robot that would make them obsolete. Although there was a bit of truth to this, these same musicians are now using the power of MIDI to expand their own musical palate and create productions of their own. Today, MIDI is being used by many professional and nonprofessional musicians alike to perform an expanding range of production tasks, including music production, audio-for-video and film postproduction ,and stage production. Such is progress.

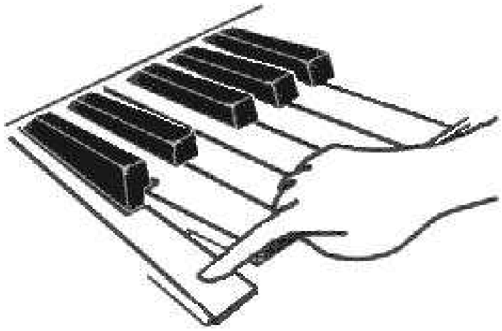
REQUIREMENTS TO ACHIEVE PROJECT GOAL

The aim of the project is to design an ".exe" file that generates Music signals and provides a better way of accomplishing the Music Industry requirements for Digital Audio Production.

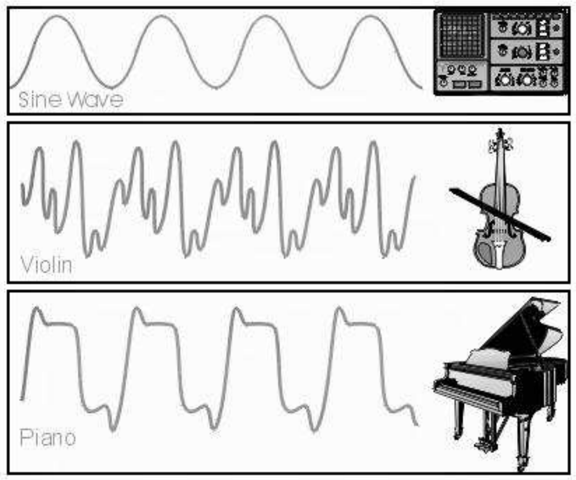
1. Physics involved in sound and acoustical resemblance
2. Mathematics and Fourier analysis of complex real world signals
3. Analog signal processing through analog electronic devices
4. Control system of particular musical instrument and corresponding waveform patterns
5. Electronic instrument interfacing with digital computers
6. How exactly Music production is done in studios and digital environment
7. Special effects and DSP effects in Music processing
8. Generating MIDI Sheet Music software with advanced C coding
9. MATLAB processing techniques on Signal Processing
10. Set of Rules in Music to be applied so as to minimize the complexity in finding the SCALE/NOTE
11. GUI design, a step behind building (user friendly) exe file

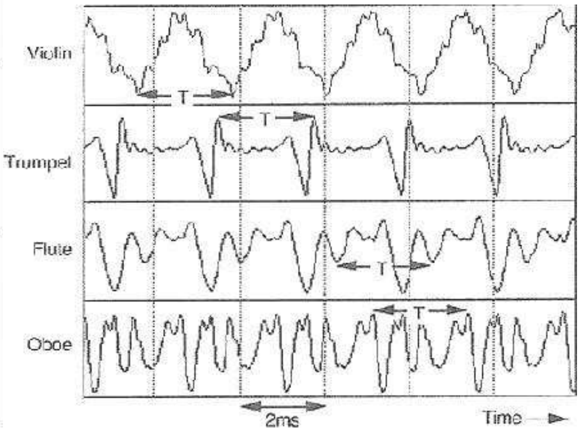
2. PHYSICS OF SOUND (AUDIO)





Sound parameters **a key is being pressed on a keyboard**





Electronic and Acoustic wave-forms **Different Wave-forms**

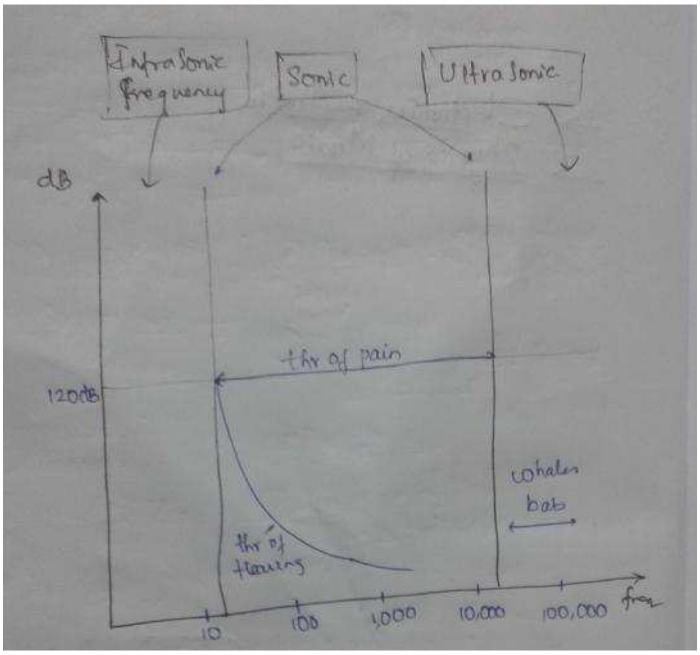
According to physics, Sound consists of three basic elements: frequency, intensity (loudness), and timbre (overtones).

The **frequency** of a sound corresponds to the vibrating frequency of the object that produced the sound. In terms of human physiology, the human ear can perceive frequencies from around 20 to 25,000 Hz; however, the ear is most sensitive to frequencies between 1000 and 2000 Hz.

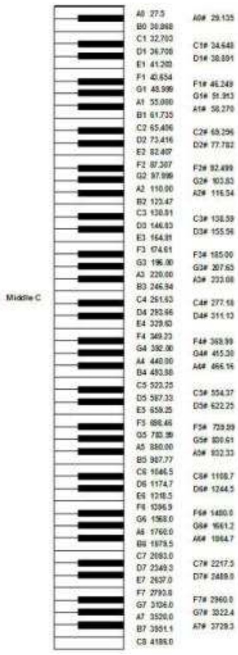
The **intensity** of a sound corresponds to the amount of sound energy transported across a unit area per second (or W/m2) and depends on the amplitude of oscillation of the vibrating object. As you move further away from the vibrating object, the intensity drops in proportion to one over the distance squared. The human ear can perceive an incredible range of intensities in terms of decibels is between 0 and 120 dB

Timbre (or tonal quality) represents the complex wave pattern that is generated when the overtones of an instrument, voice, etc., are present along with the fundamental frequency. The most intense frequency sounded is typically referred to as the fundamental frequency. The overtones of importance have frequencies up to nth harmonic and harmonics play important role in timbre characteristics.

Every instrument has its own unique tonal quality. The reason for an instrument's unique set of overtones depends on the construction of the instrument (where control systems is an undertaking part).



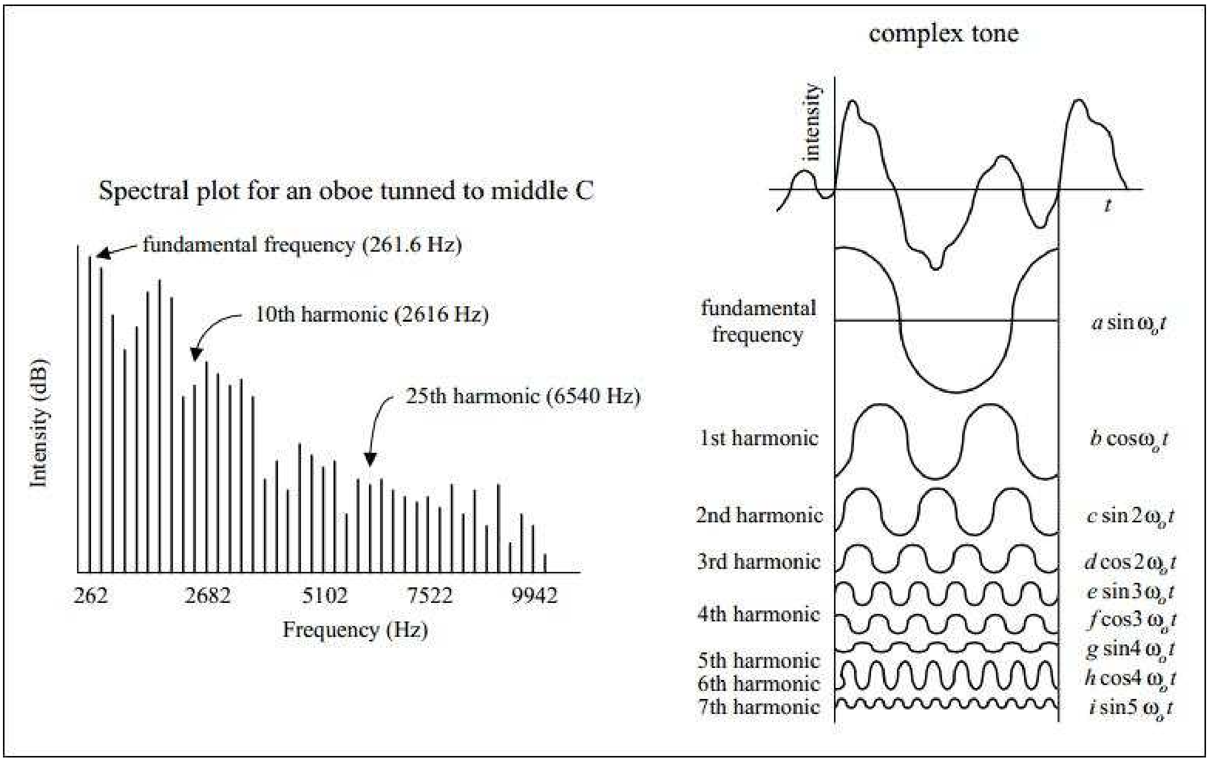
Range of Sonic Sounds



Sounds corresponding to Threshold of hearing

MAKING CIRCUITS:

The art of synthesizing sounds via electric circuits is fairly complex business. To accurately mimic an instrumental sound, train whistle, bird chirp, etc., you must *design circuits* that can generate complex waveforms that contain all the overtones and decay and rise time information. For this purpose, **special oscillator and modulator circuits** are needed.

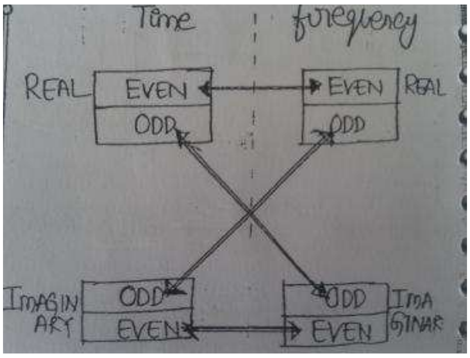


Fourier series representation of a single tone

Signal = (asin t+bcos t) + (csin2 t +dcos2 t) + (esin3 t +fcos3 t) +.....
The above mathematical formulation constitutes to the origin of Fourier series.

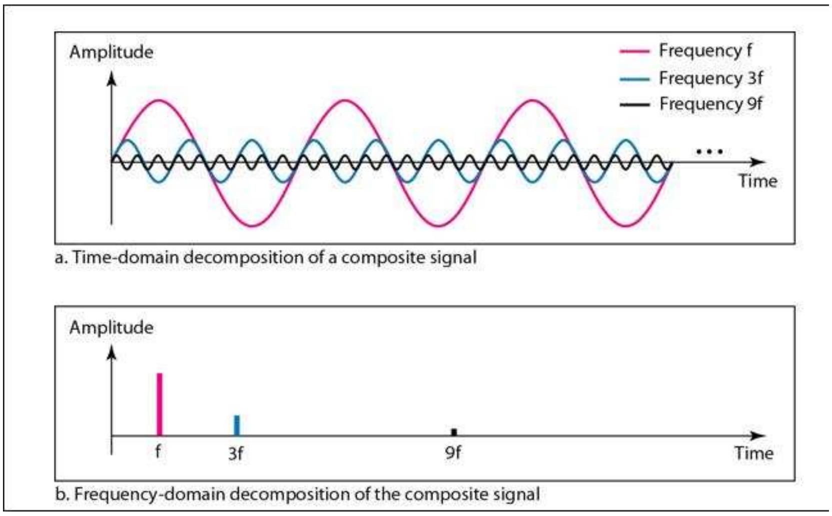
Above figure shows how the sound is perceived by the ear, a frequency of 261.6 Hertz, plus harmonics at 523.2, 784.8, 1046.4 Hertz, etc. If this note were played on another instrument, the waveform would look different. However, the ear would still hear a frequency of 261.6 Hertz plus the harmonics. Since the two instruments produce the same fundamental frequency for this note, they sound similar, and are said to have identical pitch. Since the relative amplitude of the harmonics is different, they will not sound identical, and will be said to have different timbre.

COMPLEX SIGNAL ANALYSIS

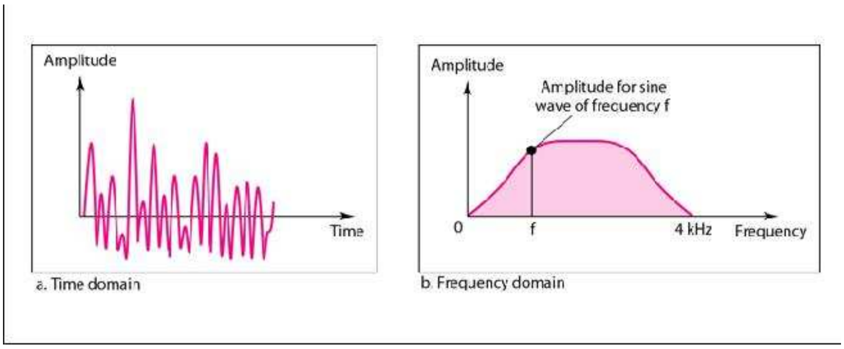


FOURIER ANALYSIS

- ✓ If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies.



- ✓ If the composite signal is non-periodic, the decomposition gives a combination of sine waves with continuous frequencies.



Speech signals generally refers to a non-periodic waveform in which the frequency is not provided in discrete levels instead it is shown by a bandwidth of continuous band of frequencies ranging from 300 to 3500 Hz.

Unlike speech signals the Music signals corresponds to a dynamic nature, as when beats are taken into account, they are periodic in nature which is contrary to vocals.
Hence, a keen observation on frequency spectrum in necessary for audio signals.

PIANO

The frequencies of the keys on a piano. Note that the piano keys are arranged in groups of 12. Each set of 12 keys spans an octave which is the doubling of frequency. For example the frequency of A_N is 2^N A_0 or N octaves higher than A_0, e.g. $A_7=2^7\times27.5=3520$Hz. Black keys correspond to sharp notes.

Notes	Prime		Second		Third	Fourth		Fifth		Sixth		Seventh
Natural	C		D		E	F		G		A		B
Sharp		C#		D#			F#		G#		A#	
Flat		Db		Eb			Gb		Ab		Bb	
Latin	Do		Re		Mi	Fa		Sol		La		Si
Frequency (Hz)	261.63	277.18	293.66	311.13	329.63	349.23	369.99	392.00	415.30	440.00	466.16	493.88
Scale step	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$	$\sqrt[12]{2}$
Distance in semitones	1	1	1	1	1	1	1	1	1	1	1	1
Distance in cents	100	100	100	100	100	100	100	100	100	100	100	100
	|– – – – – – – – – – – – 1 Octave = 12 Semitones = 1200 Cents – – – – – – – – – – – –|											

Different parameters corresponding to Piano (electronic Keyboard)

The most commonly accepted pitch standard is the note $A_4=440$Hz, also known as the concert pitch. In *equal tempered chromatic scales* each successive pitch (e.g. piano key) is related to the previous pitch by a factor of the twelfth root of 2=1.05946309436 known as a half-tone or a semi-tone (forms an exponential curve).

Major Scales

This is one of the diatonic scales consisting of five whole steps (each whole step is two semitones) and two half steps between the third and fourth and seventh and eighth notes. It is often considered to be made up of seven notes (eight if one includes the first note of the next octave of the scale). A prominent example of a major scale is the **C-major** (C, D, E, F, G, A, B, C) and reproduced here together with indications of the distances between the notes

Tone-tone-semitone-tone-tone-tone-semitone

Minor Scales

Similarly a natural minor interval pattern would be:

Tone-semitone-tone-tone-semitone-tone-tone

For example, in the key of ‘A’ minor, the natural minor scale is ‘ABCDEFGA’

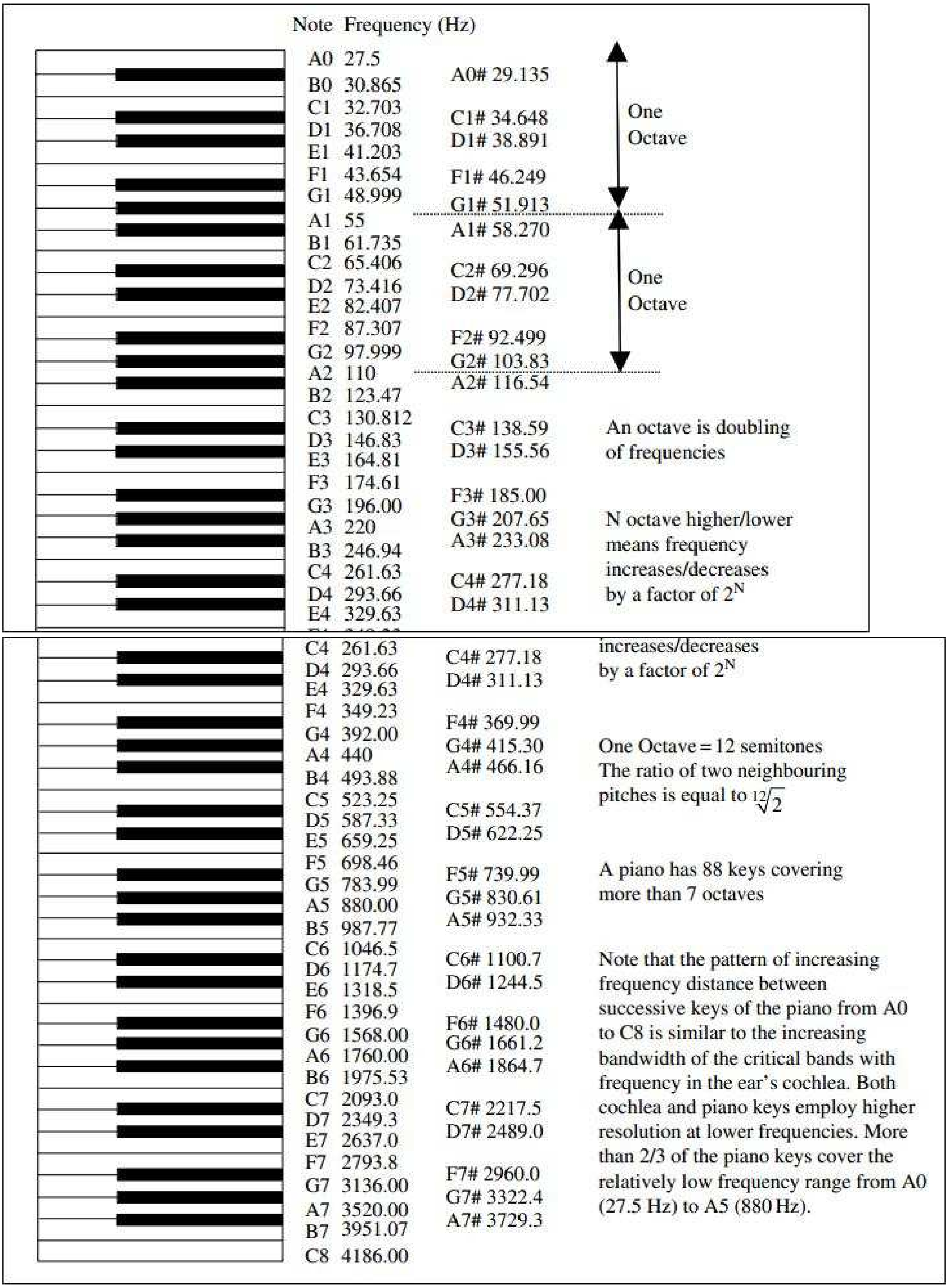
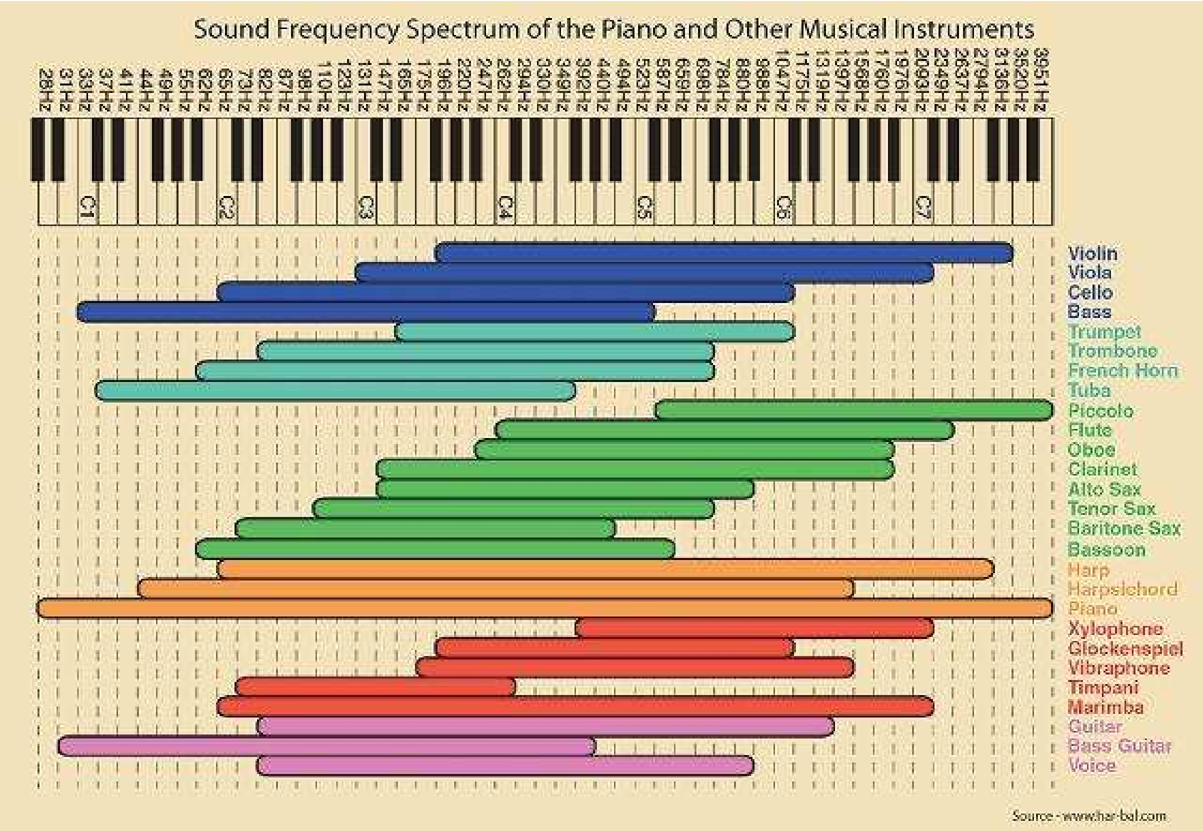


Figure showing octaves and keys with pitch values

A NOTE ON TIMBRE

Timbre is more complicated, being determined by the harmonic content of the signal. Hearing is based on the amplitude of the frequencies, and is very insensitive to their phase. The shape of the time domain waveform is only indirectly related to hearing, and usually not considered in audio systems.



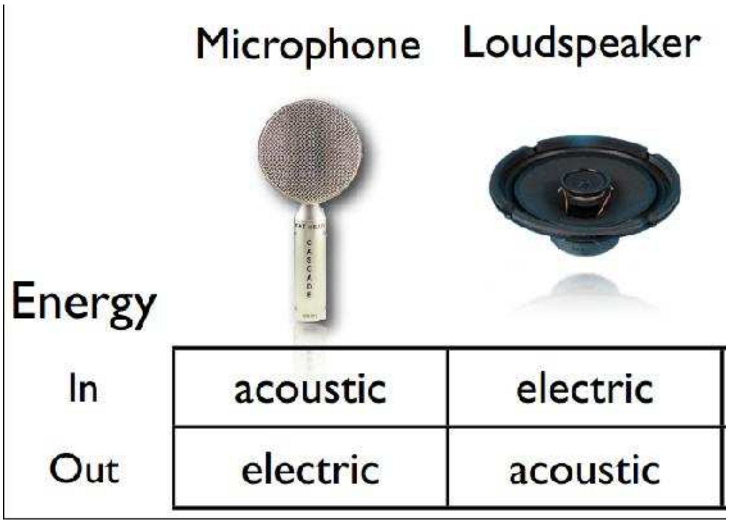
Phase detection of the human ear. The human ear is very insensitive to the relative phase of the component sinusoids. For example, two waveforms would sound identical, because the amplitudes of their components are the same, even though their relative phases are different.

The ear's insensitivity to phase can be understood by examining how sound propagates through the environment. Suppose you are listening to a person speaking across a small room. Much of the sound reaching your ears is *reflected* from the walls, ceiling and floor. Since sound propagation depends on frequency (such as: attenuation, reflection, and resonance), different frequencies will reach your ear through different paths. This means that **the relative phase of each frequency will change as you move about the room.** Since the ear disregards these phase variations, you perceive the voice as unchanging as you move position. From a physics standpoint, the phase of an audio signal becomes randomized as it propagates through a complex environment. Put another way, the ear is insensitive to phase because it contains little useful information.

However, it cannot be said that the ear is completely deaf to the phase .This is because a phase change can rearrange the time sequence of an audio signal. An example is the chirp system that changes an impulse into a much longer duration signal. Although they differ only in their phase, the ear can distinguish between the two sounds because of their difference in duration.
For the most part, this is just a curiosity, not something that happens in the normal listening environment. Suppose that we ask a violinist to play a note, say, the A below middle C. When the waveform is displayed on an oscilloscope, it appear much as the saw tooth.
Since octaves are based on doubling the frequency every fixed number of keys, they are a logarithmic representation of frequency. This is important because audio information is generally distributed in this same way. For example, as much audio information is carried in the octave between 50 hertz and 100 hertz, as in the octave between 10 kHz and 20 kHz. Even though the piano only covers about 20% of the frequencies that humans can hear (4 kHz out of 20 kHz), it can produce more than 70% of the audio information that humans can perceive (7 out of 10 octaves). Likewise, the highest frequency a human can detect drops from about 20 kHz to 10 kHz over the course of an adult's lifetime. However, this is only a loss of about 10% of the hearing ability (one octave out of ten). As shown next, this logarithmic distribution of information directly affects the required sampling rate of audio signals.

3. ROLE OF ANALOG ELECTRONICS

Audio electronics generally deals with *converting sound signals into electrical signals and vice-versa* .This conversion process typically is accomplished by means of a microphone and loudspeaker. Once the sound is converted to electrical form playing is quite simple i.e., we can amplify the signal, filter out certain frequencies from the signal, combine (mix) the signal with other signals, transform the signal into a digitally encoded signal that can be stored in memory, modulate the signal for the purpose of radio-wave transmission, use the signal to trigger a switch (e.g., transistor or relay) etc..



Transducers

Microphone and loud speaker specs:

Frequency response: The range of sound wave frequencies a microphone can transduce effectively.

Directional characteristics: The direction(s) in space that a microphone is most sensitive.

Signal to Noise Ratio: How much greater the audio signals is compared to the noise inherent in the microphone system. (Larger numbers are better)

Audio Amplifiers:

Electrical signals within audio circuits often require amplification to effectively drive other circuit elements or devices. Perhaps the easiest and most efficient way to amplify a signal is to use an op amp.

Audio amplifiers have high slew rates, high gain-bandwidth products, high input impedances, low distortion, high voltage/power operation, and very low input noise.

Audio Amplifier for guitar amplification

Inverting Amplifier:

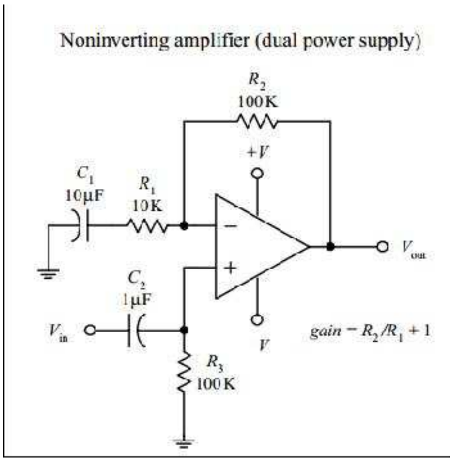
The following two circuits act as inverting amplifiers. The gain for both circuits is determined by $-R_2/R_1$, while the input impedance is approximately equal to R1.

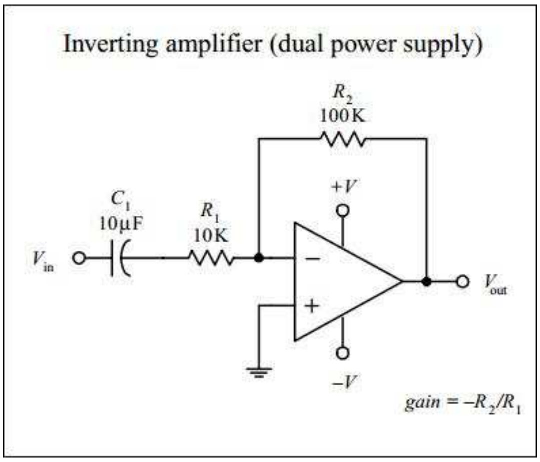
In both amplifier circuits, C_1acts as an AC coupling capacitor—it acts to pass ac signals while preventing unwanted dc signals from passing from the previous stage. Without C_1, dc levels would be present at the op amp's output, which in turn could lead to amplifier saturation and distortion as the ac portion of the input signal is amplified.C_1 also helps prevent low-frequency noise from reaching the amplifier's input.

Non-inverting Amplifier:

The preceding inverting amplifier works fine for many applications, but its input impedance is not incredibly large. To achieve a larger input impedance (useful when bridging a high-impedance source to the input of an amplifier), you can use one of the following non inverting amplifiers. The left amplifier circuit uses a dual power supply, whereas the right amplifier circuit uses a single power supply. The gain for both circuits is equal to R_2/R_1+1.

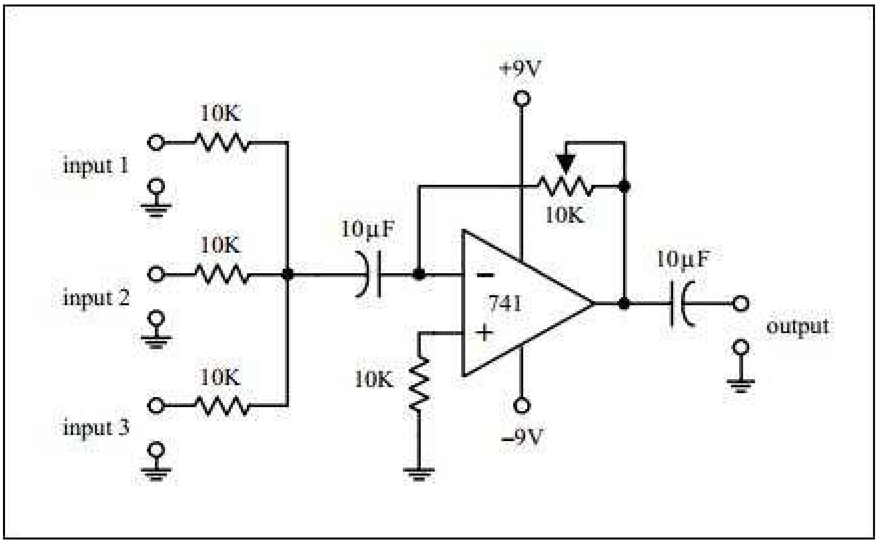
Components R_1, C_1, R_2, and the biasing resistors serve the same function as was seen in the inverting amplifier circuits. The non-inverting input offers an exceptionally high input impedance and can be matched to the source impedance more readily by adjusting C_2 and R_3. The input impedance is approximately equal to R_3.





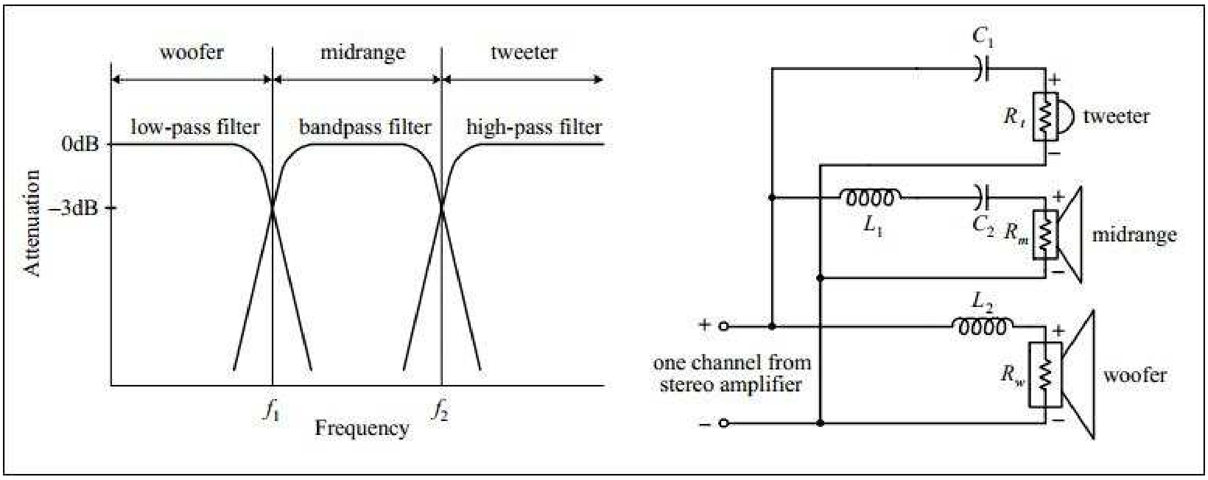
Mixer Circuits:

Audio mixers are basically summing amplifiers—they add a number of different input signals together to form a single superimposed output signal. The circuit below is a simple audio mixer circuit which uses an op amp. The potentiometer is used as an independent input volume controller.



Crossover Networks:

To design a decent speaker system, it would be best to incorporate a woofer, midrange speaker, and tweeter together so that you get good sound response over the entire audio spectrum (20 to 20,000 Hz). However, simply connecting these speakers in parallel will not work because each speaker will be receiving frequencies outside its natural frequency-response range. What you need is a filter network that can divert high-frequency signals to the tweeter, low-frequency signals to the woofer, and midrange-frequency signals to the midrange speaker. The filter network that is used for this sort of application is called a crossover network.



To determine the component values needed to get the desired response, use the following: $C_1=1/(2\ \ f_2R_t)$, $L_1=R_m/(2\ \ f_2)$, $C_2=1/(2\ \ f_1R_m)$, and $L_2=R_w/2\ \ f_1$, where f_1 and f_2 represent the 3-dB points shown in the graph.

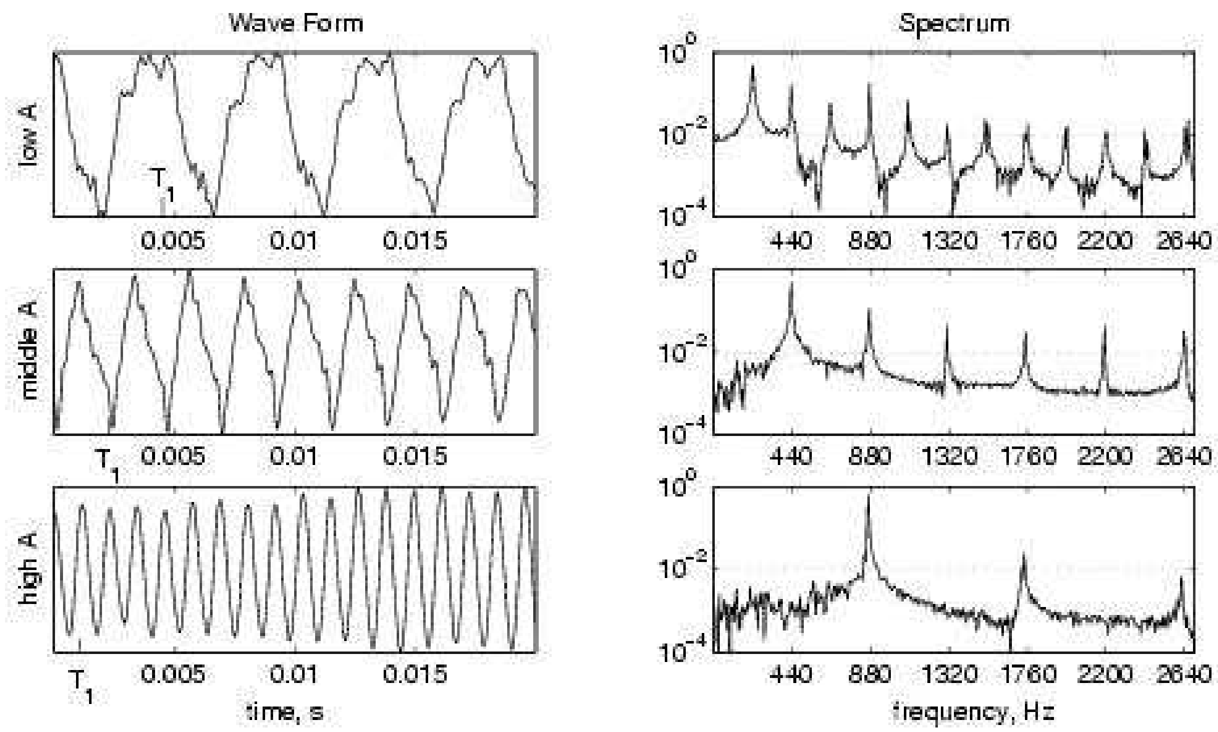
4. ROLE OF CONTROL SYSTEM IN MUSICAL INSTRUMENTS:

In practice the contribution of any signal can be analyzed both in time domain and frequency domain analysis. Generally, the notes played in an instruments corresponds to a case of **underdamped** system. The parameters such as delay time, rise time, peak time, settling time can be obtained from the plot and we can estimate the values for damping ratio, damped frequency of oscillation, damping factor etc., then making the polar plot or any other frequency plot results in minimizing the complexity of analysis.

Transfer function: In designing the filter response in MATLAB, the transfer function makes easy to accomplish the task. The cut-off frequency and 3dB bandwidth also the Gain bandwidth product analysis makes the task much easier.

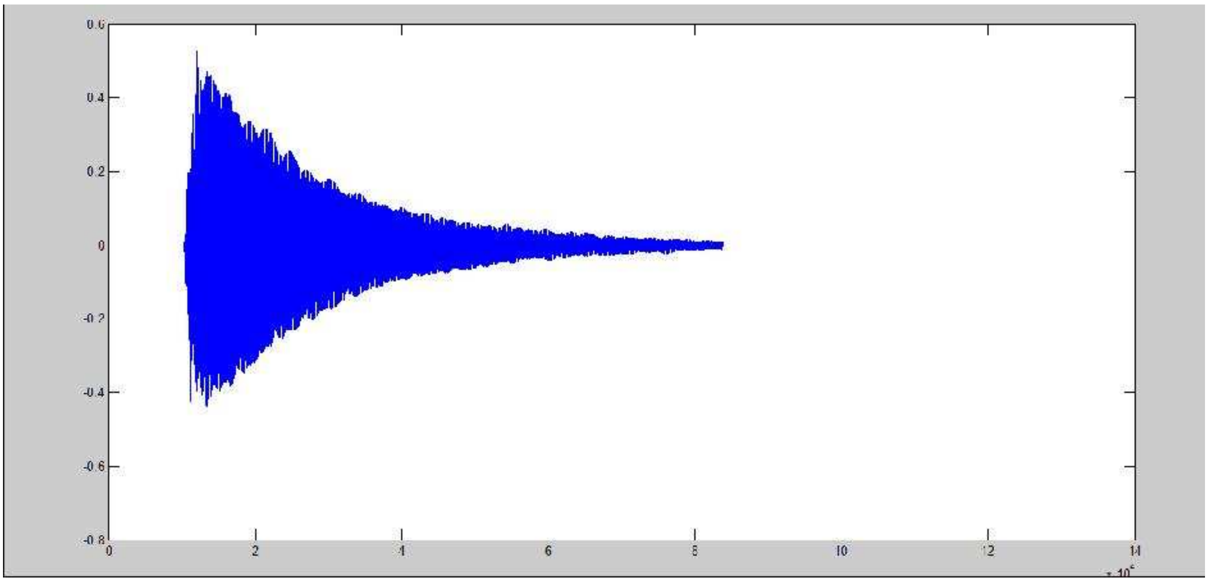
Piano tone generation: Each tone has its own characteristics in either domains (i.e., time domain and frequency domain), here the presence of harmonics (overtones) decides the timbre of any note.

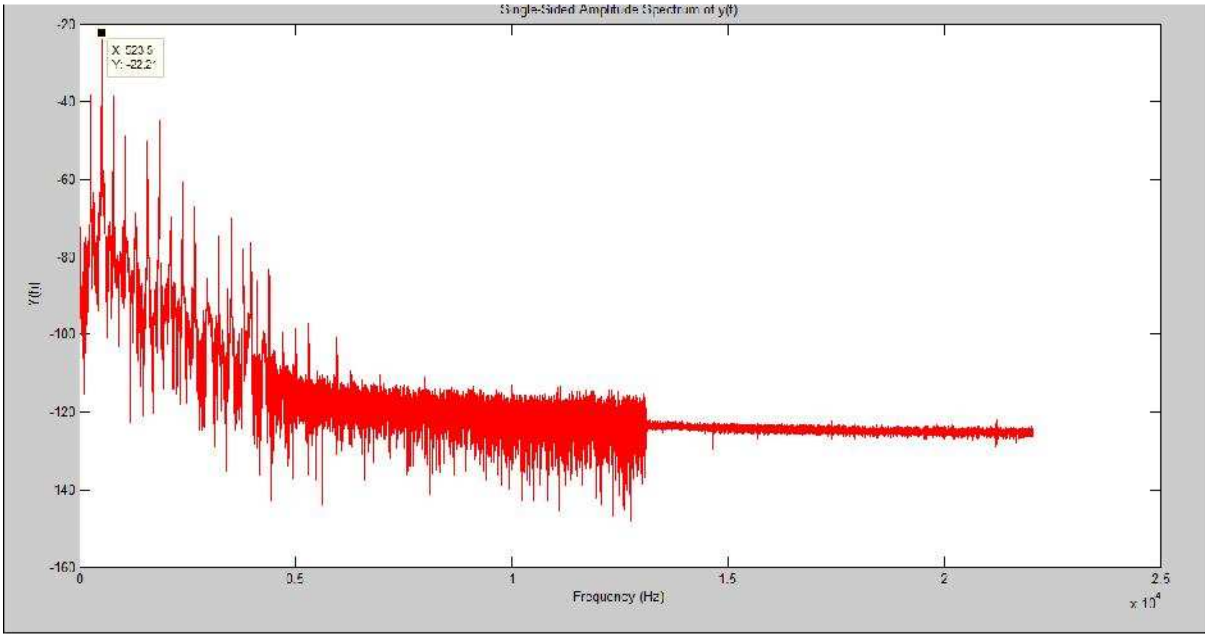
Stability concept: Piano (keyboard) tones (frequencies) are a function of exponentially increasing one, and each tone is a decreasing function of time.



Waveforms and corresponding frequency response plot for different octaves

Middle C note time and frequency domain plots in MATLAB:





5. HARDWARE AND SOFTWARE USED

Hardware:

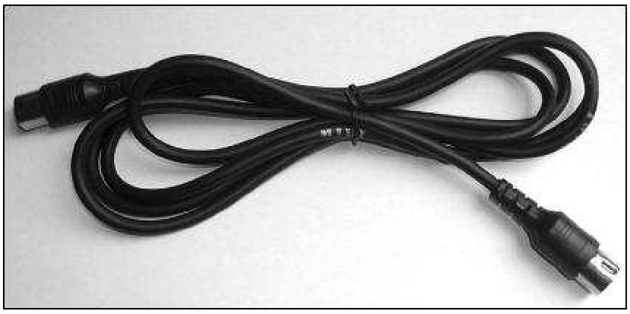
In addition to the huge number of electronic MIDI Instruments that are currently on the market, a vast array of supporting MIDI hardware systems also exists for the purpose of connecting, interfacing, distributing, processing, and diagnosing MIDI data. These systems are used to integrate all of the individual tools and toys into a working environment that will hopefully be designed to be powerful, cost effective and easy to use.

5.1 SYSTEM INTERCONNECTIONS

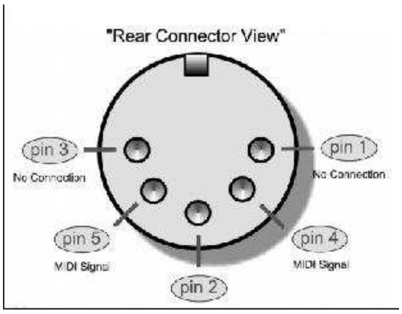
- Universal Serial Bus (USB)
- FireWire
- Wi-Fi or IEEE 802.11
- Networking

- **Universal Serial Bus (USB):**
  USB 3.1 (up to 10 G bits/second)-For even higher bandwidth requirements, 10 times faster than version 2.0. (Full duplex)

MIDI Cable:



MIDI Cable



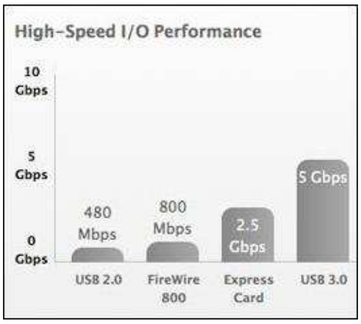
Rear view

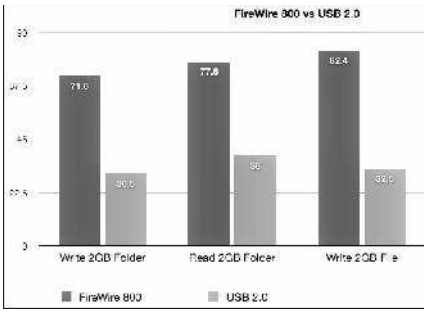
✓ Now a days MIDI cables are replaced by USB ports so as to standardize.

- **FireWire:**

The FireWire protocol is similar to the USB standard in that it uses a twisted-pair wiring to communicate bidirectional, serial data within a hot-swappable, connected chain.

✓ Unlike USB (which can handle up to 127 devices per bus), up to 63 devices can be connected within a connected FireWire chain.



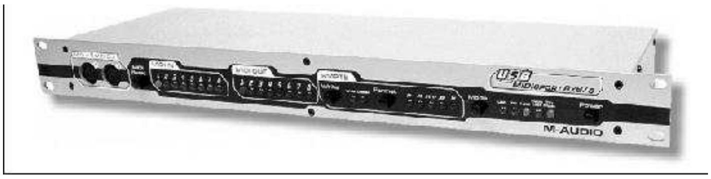


Firewire to USB interfacing **Plots for USB vs. Fire wire**

✓ Unlike USB, compatibility between the two modes is mildly problematic, as FireWire 800 ports are configured differently from their earlier predecessor and therefore require adapter cables to ensure compatibility.

- **Wi-Fi or IEEE 802.11:**

It is a 2 way radio communication at a rate of radio frequency i.e., 11Mbps (2.4 GHz).



Wireless MIDI Device **USB multiport with 8 by8 MIDI ports**

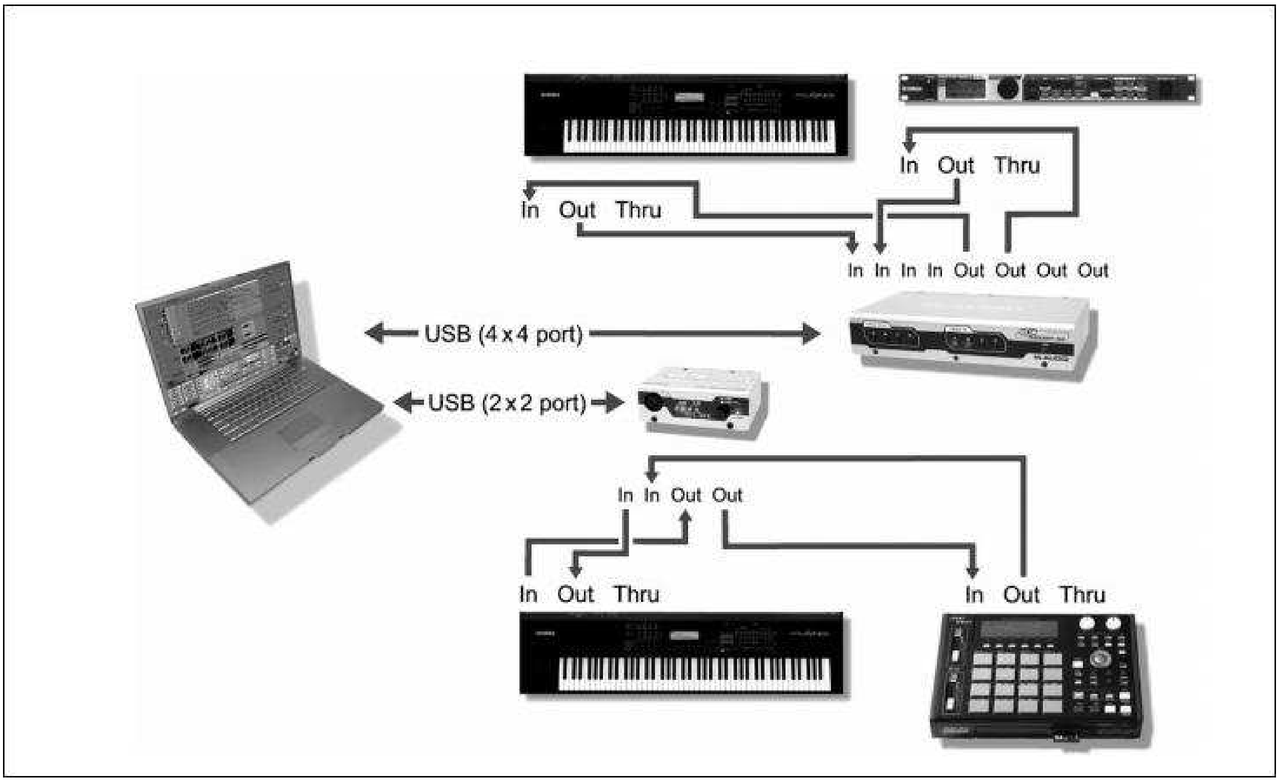
- **Networking:**

Local area network (LAN) connections

✓ Data may be shared between independent computers in a home or workplace LAN environment.

✓ Computer terminals may be connected to a centralized server, allowing data to be stored, shared, and distributed from a central location.

5.2 ELECTRONIC INSTRUMENTS



Connecting Electronic instruments, synth and Soundcard with Laptop (PC)

ELECTRONIC KEYBOARDS AND GUITARS

The most common instruments that in almost any MIDI production facility will probably belong to the keyboard family. This is due to the fact that keyboards were the first electronic music devices to gain wide acceptance initially developed to record and control many of their performance and control parameters.

Electronic Guitar **Electronic Keyboard**

The MIDI keyboard controller is a keyboard device that's expressly designed to control hard/software synths, samplers, modules, and other devices within a connected MIDI production system.

THE SYNTHESIZER

A synthesizer (or synth) is an electronic instrument that uses multiple sound generators, filters and oscillator blocks to create complex waveforms that can be combined into countless sonic variations.
Synthesizers generate sounds using a number of different technologies or program algorithms. The earliest synthesizers were analog in nature and generated sounds using a technology known as frequency modulation (FM) synthesis.
The sample-based systems are often called wavetable synthesizers to control a sample's overall sound character such as sample mixing, envelope, pitch, volume, pan, and modulation.

YAMAHA MOTIF XF Synthesizer

Additive synthesis: It makes use of combined waveforms that are generated, mixed, and varied in level over time to create new timbres that are composed of multiple and complex harmonics that, like the waveforms, vary over time.
Subtractive synthesis: It makes extensive use of filtering to alter and subtract overtones from a generated waveform (or series of waveforms).
For example, a device could start with a square or saw tooth waveform that, with the use of filters, could be altered to approximate an acoustic instrument. These generated sounds can also be filtered and changed in level over time to more closely approximate a desired sound.

YAMAHA MOTIF XF connected to MAC (Desktop) and Soundcard (M-Audio)

THE SAMPLER

A sampler is a device that can convert audio into a digital form that is then imported into internal random access memory (RAM). Once audio has been sampled or loaded into RAM (from disk, disc, or diskette), segments of sampled audio can then be edited, transposed, processed, and played in a polyphonic, musical fashion. In short, a sampler can be thought of as a digital audio memory device that lets you record, edit, looped, modulated, filtered, and amplified and reload samples into RAM.
Signal processing capabilities, such as basic editing, looping, gain changing, reverse, sample-rate conversion, pitch change, and digital mixing can also be easily applied to change the sounds in an almost infinite number of ways.

For example, a single key might be layered so that pressing the key lightly would reproduce a softly recorded sample, while pressing it harder would produce a louder sample with a sharp, percussive attack.
Most samplers have extensive edit capabilities that allow the sounds to be modified in much the same way as a synthesizer, using such modifiers as:

- ✓ Velocity
- ✓ Panning
- ✓ Expression (modulation and user control variations)
- ✓ Low-frequency oscillation (LFO)
- ✓ Attack, delay, sustain, and release (ADSR) and other envelope processing parameters
- ✓ Keyboard scaling
- ✓ After touch

Many sampling systems will often include such features as integrated signal processing, multiple outputs (offering isolated channel outputs for added live mixing and signal processing power or for recording individual voices to a multi-track recording system), and integrated MIDI sequencing capabilities.

5.3 ALTERNATIVE MIDI CONTROLLER DEVICES

Nontraditional controller types also can be used in place of a keyboard or to augment a keyboard's functionality.

The Drum Machine:
In its most basic form, the drum machine uses ROM-based, prerecorded waveform samples to reproduce high-quality drum sounds from its internal memory.
Sound files can be loaded into a DAW to build up rhythm tracks, or to lift percussion loops from loop sources and libraries that are available on CD and DVD.

MIDI Drum Controllers:
MIDI drum controllers are used to translate the voicing and expressiveness of a percussion performance into MIDI data. These devices are great for capturing the feel of a live performance, also provides flexibility of automating or sequencing a live event.

The Keyboard as a Percussion Controller:
Since drum machines respond to external MIDI data, probably the most commonly used device for triggering percussion and drum voices is a standard MIDI keyboard controller. One advantage of playing percussion sounds from a keyboard is that sounds can be triggered more quickly because the playing surface is designed for fast finger movements and doesn't require full hand/wrist motions. Another advantage is its ability to express velocity over the entire range of possible values (0-127), instead of the limited number of velocity steps that are available on certain drum pad models.

Most drum machines allow drum and percussion voices be manually assigned to a particular MIDI note value. As the percussion sounds might not be related to any musical interval, you're free to assign a drum voices to any keyboard note and range that you'd like.

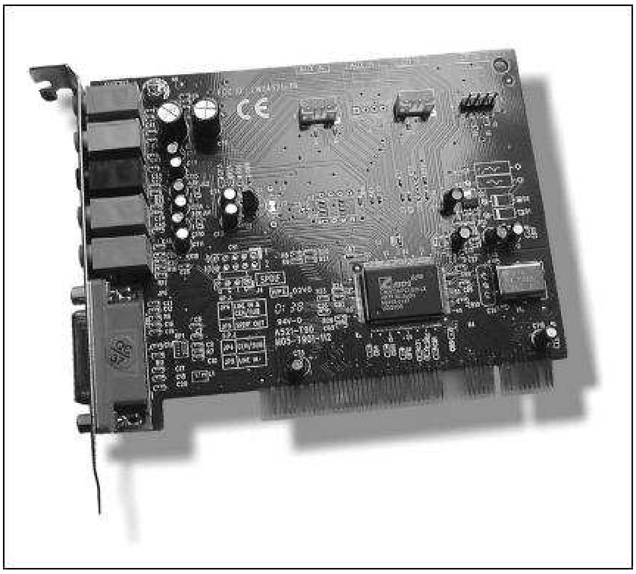
MIDI Guitars and Basses:
Guitar players often work at stretching the vocabulary of their instruments beyond the norm. They love doing nontraditional gymnastics using such tools of the trade as distortion, phasing, echo, feedback, etc.
Due to advances in guitar pickup and microprocessor technology, it's now possible for the notes and minute inflections of guitar strings to be accurately translated into MIDI data. With this innovation, many of the capabilities that MIDI has to offer are now available to the electric (and electronic) guitarist.

MIDI Wind Controllers:
MIDI wind controllers differ from keyboard and drum controllers because they're expressly designed to bring the breath and key articulation of a woodwind or brass instrument into the world of the MIDI performance. These controller types are used because many of the dynamic- and pitch-related expressions (such as breath and controlled pitch glide) simply can't be communicated from a standard music keyboard. In these situations, wind controllers can often help create a dynamic feel that's more in keeping with their acoustic counterparts by using an interface that provides special touch-sensitive keys, glide- and pitch-slider controls, and sensors for outputting real-time breath control over dynamics.

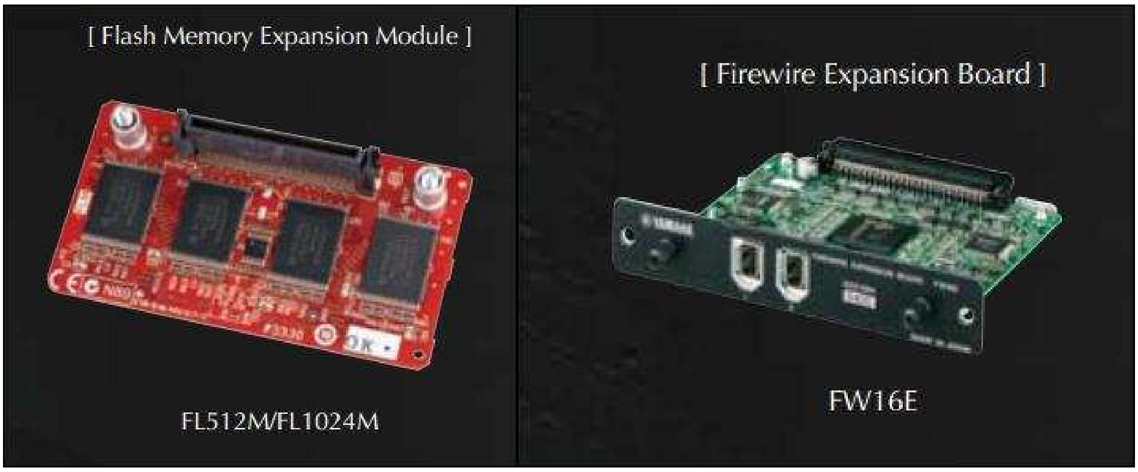
Electronic Drums (Percussion) **AKAI Sampler**

SOUND CARD

The Sound card must translate between Sound waves and bits, which can be found in almost every home, generate sounds using a simple form of digitally controlled FM synthesis. Both the soft-and hardware synth systems almost always conform to the General MIDI specs, which has universally defined the overall patch and drum-sound structure so that a MIDI file will be uniformly played by all such synths with the correct instrument voicing and levels.



Sound card for PC



6. DIGITAL AUDIO PRODUCTION

Due to the fact that MIDI is a digital medium and as such can easily be interfaced with devices that output or control digital audio. Devices such as samplers, digital audio workstations, hard disk recorders, and digital audio are commonly used to record, reproduce, and transfer sound within such an environment.

In recent years, the way that electronic musicians store, manipulate, and transmit digital audio has changed dramatically. As with most other media, these changes have been brought about by the integration of the personal computer into the modern-day project studio environment. In addition to sequencing MIDI data and controlling production-related devices in a MIDI system, newer generations of computers and their hardware peripherals have been integrated into the MIDI environment to receive, edit, manipulate, and reproduce digital audio with astonishing ease.

Top Panel

1. Pitch Bend wheel
2. Modulation wheel
3. Ribbon Controller
4. [SELECTED PART CONTROL] button
5. [MULTI PART CONTROL] button
6. ASSIGNABLE FUNCTION buttons
7. MASTER VOLUME
8. Assignable Knobs
9. Assignable Control Sliders
10. [REMOTE ON/OFF] button
11. [ARPEGGIO ON/OFF] button
12. EFFECT buttons
13. [MASTER EFFECT] button
14. OCTAVE [UP] and [DOWN] buttons
15. SEQ TRANSPORT buttons
16. LC-Display
17. Function buttons
18. Sub Function buttons
19. Data dial
20. [INC/YES] button
21. [DEC/NO] button
22. Cursor buttons
23. [EXIT] button
24. [ENTER] button
25. MODE buttons
26. Bank buttons
27. Group [A] – [H] buttons
28. Number [1] – [16] buttons
29. [COMMON EDIT] button
30. [PROGRAM] button
31. [CATEGORY SEARCH] button
32. [PERFORMANCE CONTROL] button
33. [TRACK] button
34. [MUTE] button
35. [SOLO] button

Rear Panel

1. Power switch
2. AC power cord socket
3. ETHERNET connector
4. USB connectors
5. LCD Contrast Control
6. FireWire expansion board (FW16E) cover
7. DIGITAL OUT connector
8. MIDI IN/OUT/THRU connectors
9. FOOT SWITCH jacks
10. FOOT CONTROLLER jacks
11. ASSIGNABLE OUT L and R jacks
12. OUTPUT L/MONO and R jacks
13. PHONES jack
14. A/D INPUT jacks
15. GAIN knob

DIGITAL AUDIO RECORDING

The encoding and decoding phases of the digitization process center around two processes:

- ✓ Sampling
- ✓ Quantization

Sampling is a process that effects the overall bandwidth that can be encoded within a sound file, while quantization refers to the resolution (overall quality and distortion characteristics) of an encoded signal compared to the original analog signal at its input.

- ✓ Encode
- ✓ Decode

Encoding and Decoding are the processes to convert the quantized values to binary coding and the reverse

- ✓ PCM or PWM
- ✓ Delta Modulation

These are the modulation techniques used to process the data within short distance i.e. Baseband communication. Generally the audio formats are converted to Pulse Code Modulation (PCM) or DPCM (a differential case) then followed by playing with those samples.

- ✓ Programmed algorithms

For DSP processors C codes in CCStudio, for MATLAB the MATLAB codes are used for signal processing.

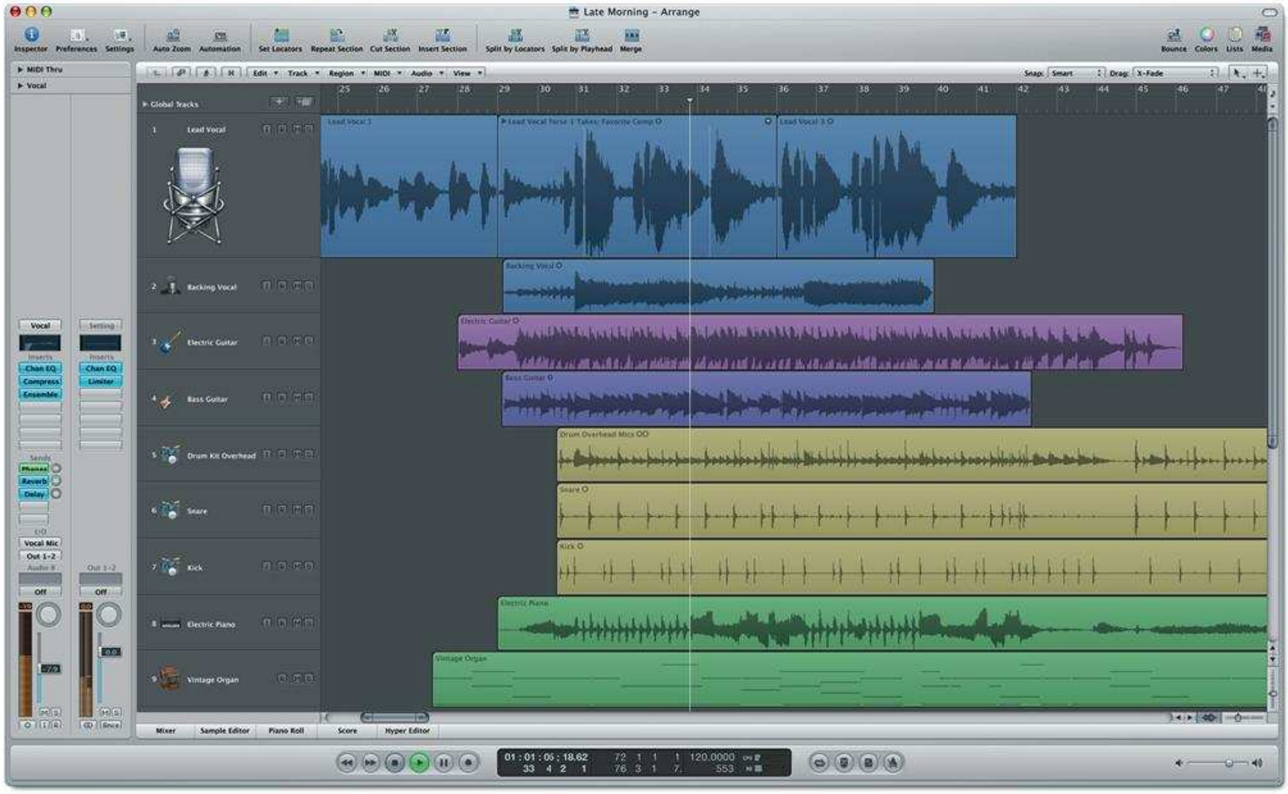
THE DAW

In recent years, the term digital audio workstation (DAW) has increasingly come to signify an integrated, computer-based, hard-disk recording system that commonly offers such features as:

- ✓ Advanced multi-track recording, editing, and mix down capabilities
- ✓ MIDI sequencing, edit, and score capabilities
- ✓ Integrated and plug-in signal processing support
- ✓ Support for integrating software plug-in instruments (VSTs) and peripheral music programs
- ✓ Integration of peripheral hardware devices such as controllers and audio and MIDI interface devices

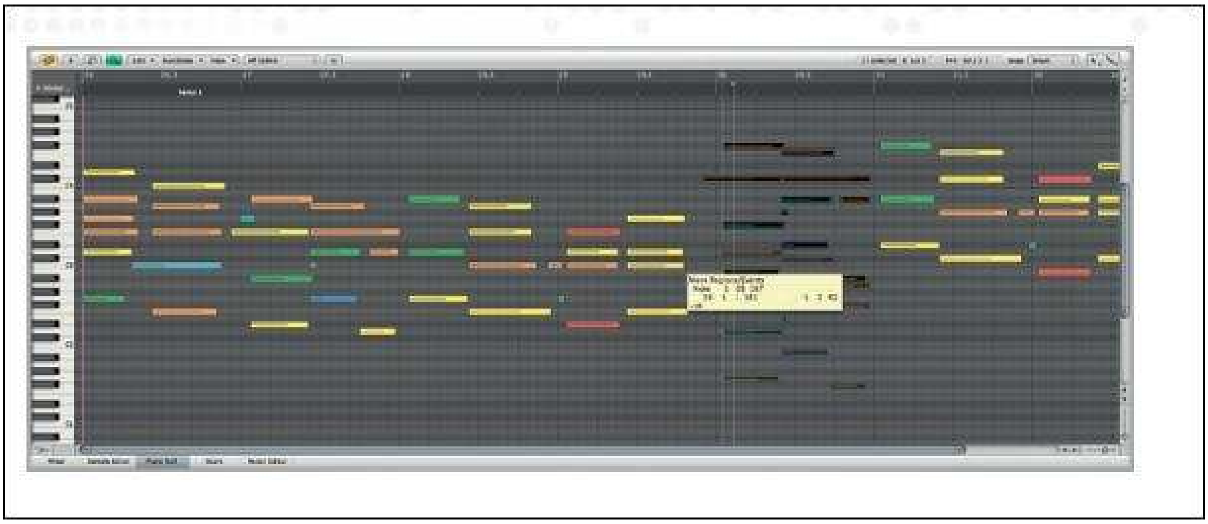
Truth of the matter is, by offering a staggering amount of production power for these software-based programs and their peripherally connected devices have revolutionized the faces of professional, project, and personal studios in a way that touches almost every life within the audio and music production communities.

6.2 ELECTRONIC MUSIC PRODUCTION



Logic Pro 9 -An advanced music production software overview

A complete system for music and audio production for multi-track recording and editing, a full suite of virtual instruments and effects, VSTs, DSP effects and many more



A modern way of representing the midi data for manipulation and also it is user friendly.

Showing VSTs Equalizer, ADSR parameters, Stereo Speed, Clip distortion, Amplifier

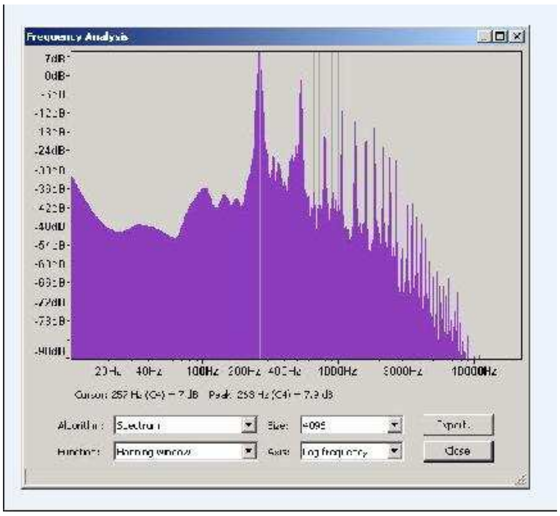
VSTs (Virtual Synthetic Technologies) plays a major role in specifying the mood of the sound i.e., it has relevant buttons on which different sets of knobs corresponding to different modulations and effects.

Channel Equalizer **Waveform editing**

Beats, a concept of looping the original (small) track for the entire track period helps a musician to perform within no time, instead of playing the instrument for whole period.

6.3 MATLAB PROCESSING

MATLAB is a matrix programming language which provides command for working with transforms, such as the Laplace and Fourier transforms and Signal Processing tools. Transforms are used in science and engineering as a tool for simplifying analysis and look at data from another angle.

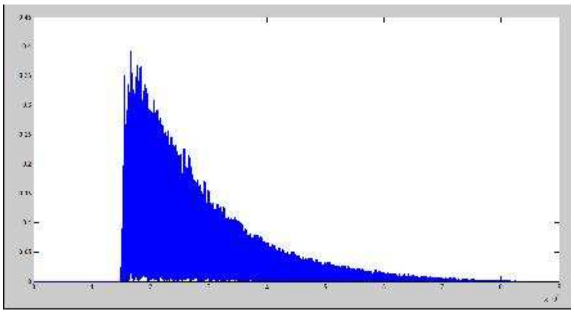


Frequency spectrum for C4 (261.63Hz)

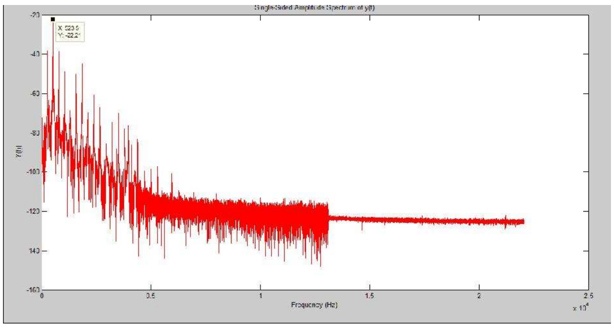
Note Frequency Chart

	Octave 0	Octave 1	Octave 2	Octave 3	Octave 4	Octave 5	Octave 6	Octave 7	Octave 8
C	16.35	32.70	65.41	130.81	261.63	523.25	1046.50	2093.00	4186.01
C#	17.32	34.65	69.30	138.59	277.18	554.37	1108.73	2217.46	4434.92
D	18.35	36.71	73.42	146.83	293.66	587.33	1174.66	2349.32	4698.64
D#	19.45	38.89	77.78	155.56	311.13	622.25	1244.51	2489.02	4978.03
E	20.60	41.20	82.41	164.81	329.63	659.26	1318.51	2637.02	5274.04
F	21.83	43.65	87.31	174.61	349.23	698.46	1396.91	2793.83	5587.65
F#	23.12	46.25	92.50	185.00	369.99	739.99	1479.98	2959.96	5919.91
G	24.50	49.00	98.00	196.00	392.00	783.99	1567.98	3135.96	6271.93
G#	25.96	51.91	103.83	207.65	415.30	830.61	1661.22	3322.44	6644.88
A	27.50	55.00	110.00	220.00	440.00	880.00	1760.00	3520.00	7040.00
A#	29.14	58.27	116.54	233.08	466.16	932.33	1864.66	3729.31	7458.62
B	30.87	61.74	123.47	246.94	493.88	987.77	1975.53	3951.07	7902.13

Frequency component list



Time Domain

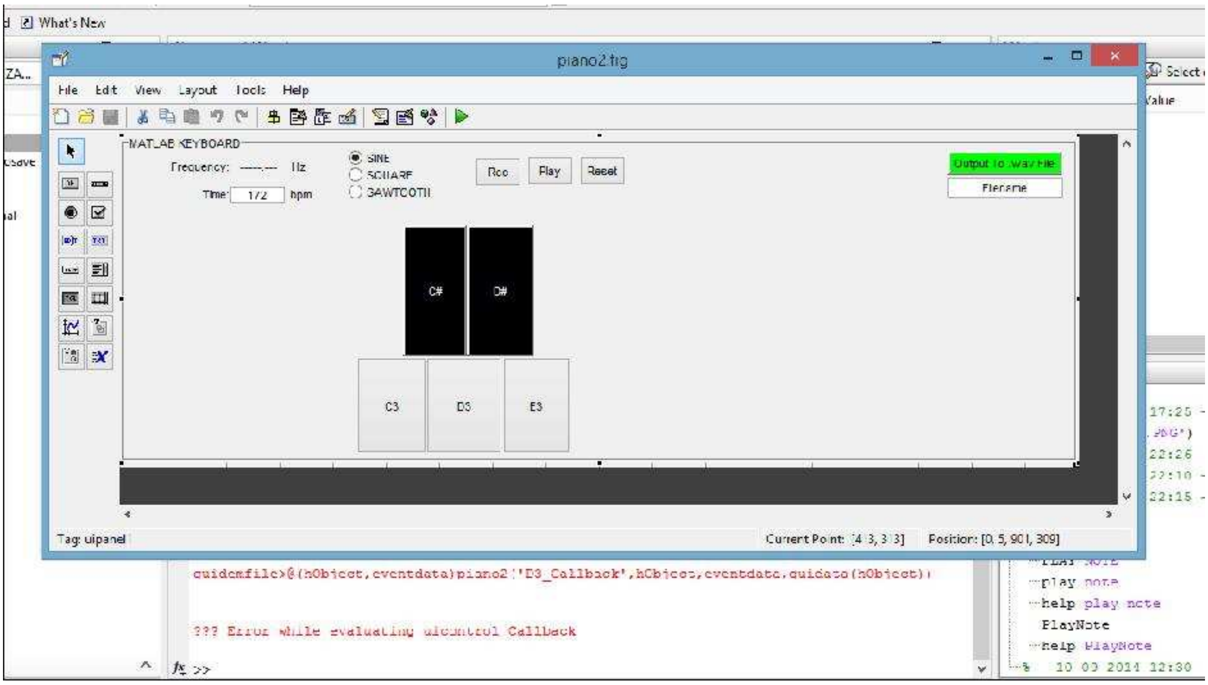


Frequency Domain

For example, the Fourier transform allows us to convert a signal represented as a function of time to a function of frequency. Laplace transform allows us to convert a differential equation to an algebraic equation.

MATLAB provides FFT algorithm for DFT spectrum known as Fourier and Fast Fourier transform.

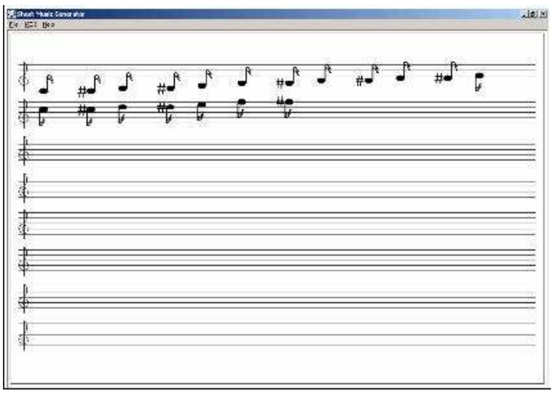
GUI DESIGN

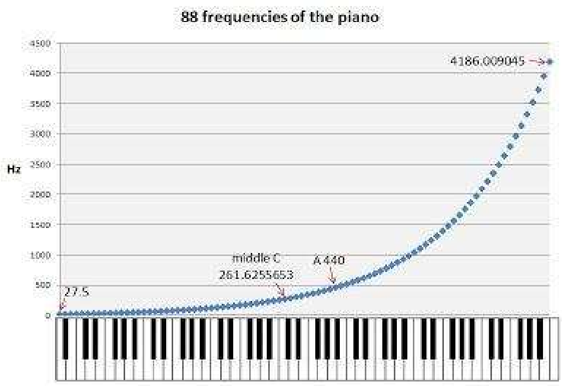


MATLAB provides an interesting way of generating user friendly pop ups for easy way of accessing. This is done in GUI (Graphical User Interface) where different buttons, checkbox, edit block etc., are used for making the output to be user friendly.

MIDI SHEET MUSIC GENERATOR

A fully functional interface between a standard Musical Instrument Digital Interface (MIDI) device and a personal computer, including a software application for processing and display of MIDI data. The hardware interface uses an (Atmel Mega32) microcontroller to facilitate communications between a MIDI device and computer. The microcontroller receives MIDI data through a standard MIDI cable, filters and encodes the data, then sends packets to the PC via a serial UART connection.



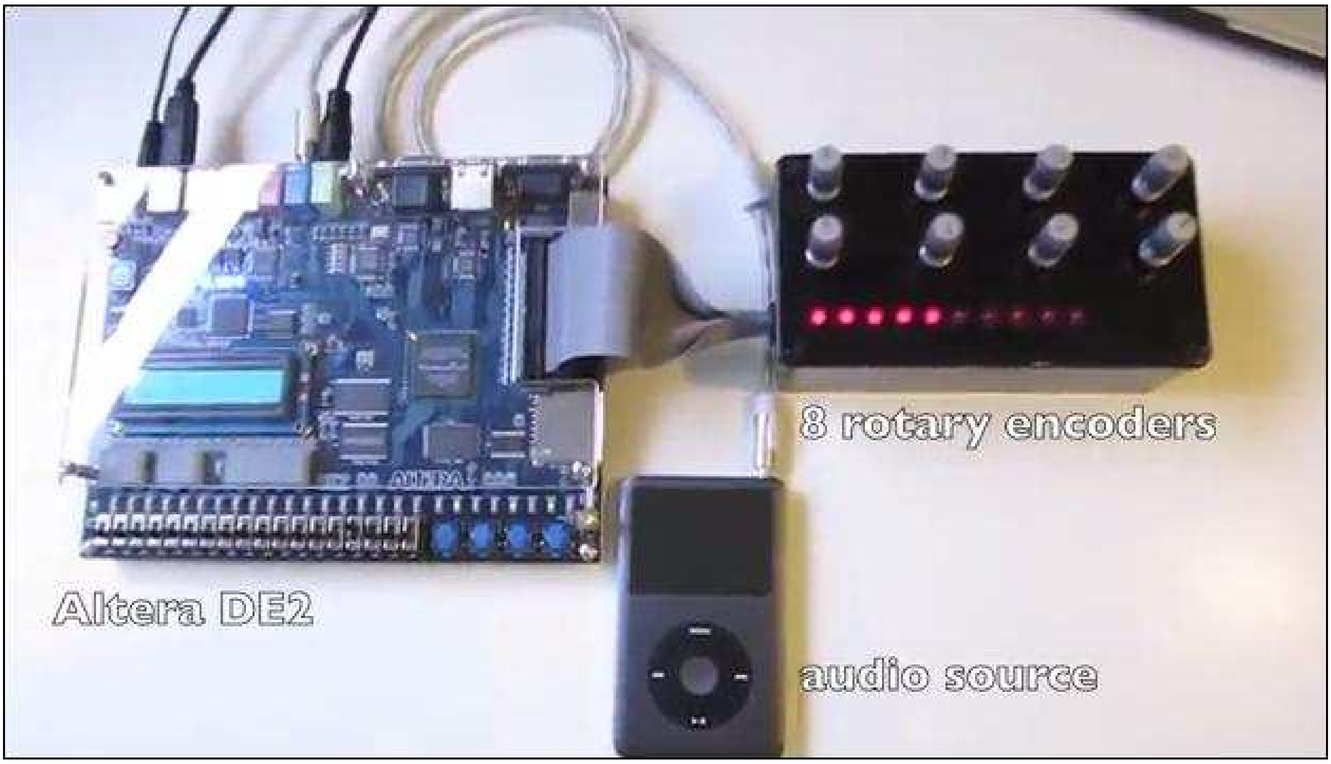


Sheet Music Generator program **Exponential rise in frequency components**

7. DIGITAL SIGNAL PROCESSING (DSP) - AUDIO EFFECTS

- **Basic filters**
    - LP, HP, BP, BS
- **Equalizers**
    - Shelving and peak filters
- **Advanced filters**
    - Time varying filters
        - Wah-Wah Phasor
    - Delay filters
        - Vibrato
        - Flanger
        - Chorus
        - Echo
    - Modulators
        - Ring modulation
        - Tremolo (AM)
        - Vibrato (FM)
    - Nonlinear Processing
        - Compression
        - Limiters
        - Distortion
        - Enhancers/Exciters
    - Special Effects
        - Panning
        - Reverb
        - Surround sound

ADVANCEMENT THROUGH FPGA



The DSP effects can be easily implemented through FPGA (Field Programmable Gate Arrays) which also helps in handling different type of signals easily and effectively with high speed.

Software synthesizers have great flexibility in the connection of the basic components of many synthesis methods, and can generate any kinds of sounds of any synthesis methods. However, the performance of the current microprocessors are not still enough for generating many sounds at a time. Speedup of the computation by some simple hardware is expected especially for real time performance. With Field Programmable Gate Arrays (FPGAs) along with hardware accelerators, we can realize any kinds of combinations of different sampling rates and data width of sounds in generating many sounds (350 sounds on a desktop computer and 110 sounds on a note computer) at a time. This performance can be achieved by fully utilizing the flexibility of FPGAs and DSPs.

8. CONCLUSION

Musical signal processing has a wide range of applications including digital coding of music for efficient storage and transmission on mobile phones and portable music players, modeling and reproduction of the acoustics of music instruments and music halls, digital music synthesizers, digital audio editors, digital audio mixers, spatial-temporal sound effects for home entertainment and cinemas, music content classification and indexing and music search engines for the Internet.

- ✓ Speech processing details from IIIT HYD – TTS (text to speech conversion)
- ✓ Vocal courses I & II
- ✓ Keyboard and computer music production
- ✓ Variable power supply with 8051 microcontroller interfaced with LCD (a project from Elevouge-Mumbai)
- ✓ Robotics-ARK techno solutions

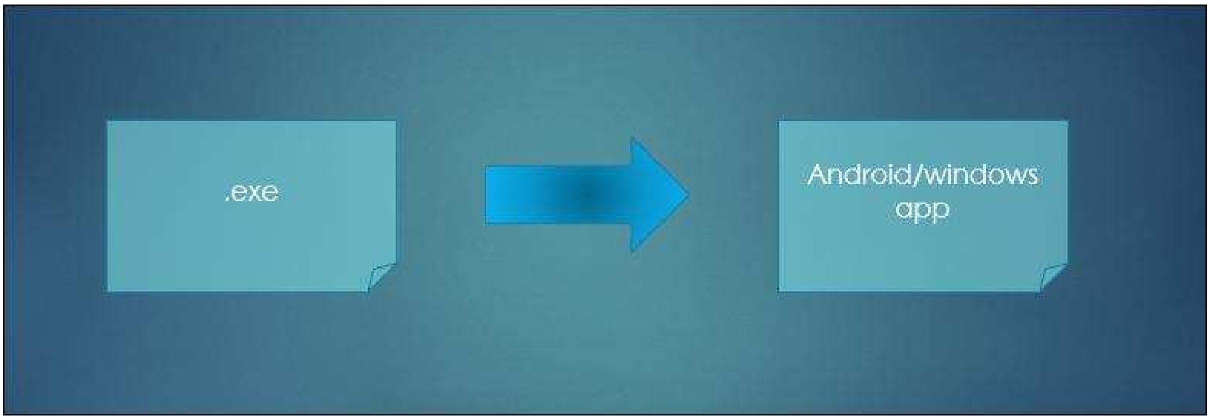
Demonstration of 'microcontroller based robots' and 'TTS setup'

YAMAHA E333 Electronic Keyboard

Interfacing Keyboard with Laptop

SCOPE

From the above knowledge of microcontrollers and music interfaces, it is possible to develop a software for sheet music generation and Developing GUI for generating piano frequency Note in MATLAB (with DSP FX), then processing the application to exe file then to an Android/windows app.



REFERENCES:

Books referred:

- ✓ "The MIDI Manual A Practical Guide to MIDI in the Project Studio" by David Miles
- ✓ Cardiff University contents
- ✓ "Digital Signal Processing" by SK.Mitra
- ✓ "Multimedia Signal Processing - Theory and application in Speech, Music and Communication" by Saeed Vaseghi
- ✓ "Practical Electronics for Inventors" by Paul Scherz

Websites:

- ✓ www.mathworks.in
- ✓ www.tutorialspoint.com/matlab.htm
- ✓ www.electronics.howstuffworks.com/gadgets/audio-music/cassette.htm
- ✓ www.davesmithinstruments.com/
- ✓ www.yamaha.com/products/music-production/synthesizers/motif_xf/motif_xf6/
- ✓ www.steinberg.net/en/products/vst/rnd_portico_plug_ins/vcm.html
- ✓ www.youtube.com/watch?v=ff_-lhLLjLU (AR Rahman playing finger board)
- ✓ http://amath.colorado.edu/pub/matlab/music/

Other Materials:

- ✓ Sheet Music Generator -MIDI PC interface (A Cornell University project)

  By- Meg Walraed Sullivan
- ✓ Subjects like Control Systems, DEC, FM modulation, SS (along with the help of Mr.Narasimham and Mr.Kiran Faculty at ACE academy)