SOUND AND ITS ATTRIBUTES

Sound is vibration, as distinguished by the sense of hearing. We generally hear vibrations that travel through air, but sound can also travel through gases, liquids and solids. It cannot travel through a vacuum (such as exists in outer space). When the vibrations arrive at our ears, they are converted into nerve impulses that are sent to our brains, allowing us to distinguish the sound. In more technical language, "sound" is an fluctuation in pressure, particle displacement, or particle velocity spread in an elastic material."

Following are the attributes of sound :

1. Sound - Frequency and wavelength
2. Sound - Amplitude

1. ***Sound - Frequency and wavelength***: The frequency is the number of air pressure oscillations per second at a fixed point engaged by a sound wave. One single oscillatory cycle per second corresponds to 1 Hz. The wavelength is the distance between two successive crests and is the distance that a wave complete a cycle.

   The frequency range of sound capable of being heard to humans is in the range of between 20 and 20,000 Hz. These ranges vary by individual and generally reduce in size with age. It is also an uneven curve - sounds near 3,500 Hz are often perceived as louder than a sound with the same amplitude at a much lower or higher frequency. Above and below this range are **ultrasound** and **infrasound**, respectively.

2. ***Sound - Amplitude*** : a sound also has amplitude, a property subjectively heard as loudness. The amplitude of a sound is the measure of the displacement of the air pressure wave from its mean or quiescent state.

Extracting Attributes of the Sound

Noises are uneven and disordered vibrations as well as all possible frequencies. Their wave diagram does not repeat in time. Noise is an aperiodic series of waves. Sounds that are sine waves with fixed frequency and amplitude are supposed as pure tones. While sound waves are generally visualised as sine waves, sound waves can have arbitrary shapes and frequency content, limited only by the equipment that generates them and the medium through which they travel.

The amount of noise in a sound varies independently of its amplitude; you can have a nice, clean, resonant sound, thumbp!, or you can have a wheezier sound, sssshbp. In the waveform display, you have got to know how to distinguish these: noise is irregular, pitched sounds are regular. But if the degree of noisiness could be computed, it could be displayed separately; for example, a clean sound could look like this:

and a noisy one with a similar envelope could look like this:

MONO VS STEREO SOUND

Now let us see "What are differences between mono and stereo sound?"

The terms stereo (Stereophonic) and mono (Monophonic) are frequently used when referring to amplifier connections. I think the sound is more natural and real. I think most people here would agree that true stereo sounds better than mono. I myself prefer stereo. Though, there are many types of stereo that aren't true, and the discuss is over which of those are less preferable than mono.

In the case of many early recordings that were imagined in mono and later remixed in stereo, the problem is two-fold:

First, any stereo remix is at difference with the original artist's/producer's/engineer's conception and is therefore inauthentic.

Second, many stereo remixes are not good stereo; that is, instruments and voices are panned seemingly arbitrarily, as if L-R spatial separation were the only important goal, and realistic sound staging of no effect. This is actually true of many "original" stereo mixes, as well, since stereo in its early days was not well understood by all those employing it.

Frequently, the recording itself is "multi-mono"- every sound recorded in mono and then placed somewhere in a stereo pan, as opposed to sounds being recorded in stereo with two (or more) mics. This makes a true stereo mix impossible, although one can accomplish a pretty decent simulation with variety of mixing techniques.

Generally a stereo amplifier has two independent channels, one left and one right. The left and right signals of the stereo signal are similar but not exactly the same. The two channels are used to give the audio a sense of depth. If one instrument or voice is only produced in the left channel, it will seem to originate from the left side of the listening area. If a particular sound is only slightly louder in one of the channels, that sound will seem to originate off center slightly toward the channel in which the sound is louder. If you have two speakers but supply mono signal to both of them, there will be no sense of separation or depth. If a mono signal fed to both channels of a stereo amplifier, with a speaker on each channel, the output will mono. If a stereo signal is fed to the same amp/speaker set up, the output will be stereo. If a speaker is bridged onto a stereo amplifier, the output of the speaker will be a mono output, even if the signal fed into the amplifier is a stereo signal. Even if two speakers are bridged onto the amplifier, the output will still be mono because the output from each speaker has the same content.

> In the following diagrams, 'X', 'Y', and 'Z' are the different sounds (instruments, vocals...) in the audio. The red letters are where the signal 'appears' to originate from and the yellow letters are where they are being reproduced.

Mono with one speaker:

In this case, the speaker is directly in front of the listening position and the audio appears to (and does) originate from the speaker.

Mono with 2 speakers:

In this case, you can see that the identical signal is reproduced by both speakers. While the signal content going to each speaker is exactly the same, this is a mono system. If the level of the signal is the same in both speakers, the signal will appear to originate exactly in the center of the speakers.

Below, you can see that the signal content from each speaker is the same but it is a little louder in the right channel. This means that it will give the impression to originate a little to the right of center.

Stereo audio:

In this diagram, you can observe that the 'x' part of the audio is reproduced just as in both channels and appears to originate in the center of the two speakers. The 'y' part of the audio is only in the left speaker and appears to originate from the left speaker's position. The 'z' part of the audio is only reproduced by the right speaker. This means that it will appear to originate from the right speaker's position.

Below, you can see that the 'y' part of the audio is produced in both channels but is at a reduced level in the right channel. This will cause the 'image' of the y part of the audio signal to appear to originate from left of center (not the far left or the center). This is how the audio 'stage' is reproduced with a stereo signal (different signals are recorded/reproduced at different levels in each of the speakers).

Sound Systems :

It's a common question in many church sound system projects, "Will our system be mono or stereo?" What usually follows is a lengthy discussion about the applicability of mono sound systems versus stereo systems, the difference between two channel sound systems and stereo sound systems, the benefits and limitations of each, the architectural constraints and program requirements that will affect the decision, and the cost implications of each.

It is very apparent that everyone has their own interpretation of the terms 'mono' and 'stereo', influenced by their own experiences and expectations. Translating one's experience with home audio systems or project studios into a large venue like a church or a theatre often takes an adjustment in conceptual thinking, so we always have to provide a reference point for the discussion of mono, stereo and two channel sound systems. Let's start off with mono.

Mono (Monophonic) Sound System :

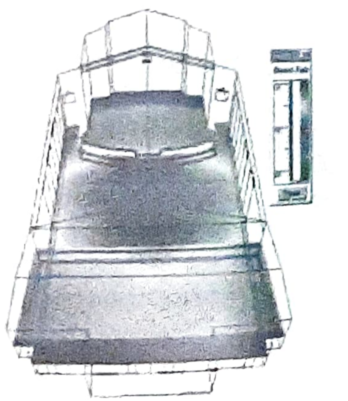
Mono or monophonic describes a system where *all the audio signals are mixed together and routed through a single audio channel.* Mono systems can have multiple loudspeakers, and even multiple widely separated loudspeakers. The key is that the signal contains no level and arrival time/phase information that would replicate or simulate directional cues. Common types of mono systems include single channel centre clusters; mono split cluster systems, and distributed loudspeaker systems with and without architectural delays. Mono systems can still be full-bandwidth and full-fidelity and are able to reinforce both voice and music effectively. The big advantage to mono is that everyone hears the very same signal, and, in properly designed systems, all listeners would hear the system at essentially the same sound level. This makes well-designed mono systems very well suited for speech reinforcement as they can provide excellent speech intelligibility.

Stereo (Stereophonic) Sound System :

True stereophonic sound systems have two independent *audio signal channels*, and the signals that are reproduced have a specific level and phase relationship to each other so that when played back through a suitable reproduction system, there will be an apparent image of the original sound source. Stereo would be a requirement if there is a need to replicate the aural perspective and localization of instruments on a stage or platform, a very common requirement in performing arts centres.

This also means that a mono signal that is panned somewhere between the two channels does not have the requisite phase information to be a true stereophonic signal, although there can be a level difference between the two channels that simulates a position difference, this is a simulation only.

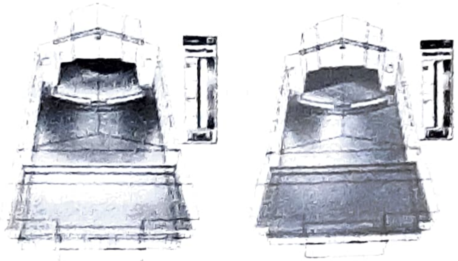
An additional requirement of the stereo playback system is that the entire listening area must have equal coverage of both the left and right channels, at essentially equal levels. This is why your home stereo system has a "sweet spot" between the two loudspeakers, where the level differences and arrival time differences are small enough that the stereo image and localization are both maintained. This sweet spot is limited to a fairly small area between the two loudspeakers and when a listener is outside that area, the image collapses and only one or the other channel is heard. Living with this sweet spot in your living room may be OK, since you can put your couch there, but in a larger venue, like a church sanctuary or theatre auditorium, that sweet spot might only include 1/3 the audience, leaving 2/3 of the audience wondering why they only hear half the program.



In addition a stereo playback system must have the correct absolute phase response input to output for both channels. This means that a signal with a positive pressure waveform at the input to the system must have the same positive pressure waveform at the output of the system. So a drum, for instance, when struck produces a positive pressure waveform at the microphone and should produce a positive pressure waveform in the listening room. If you don't believe that this makes a tremendous difference, try reversing the polarity of both your hifi loudspeakers some day and listening to a source that has a strong centre sound image like a solo voice. When the absolute polarity is flipped the wrong way, you won't find a stable centre channel image, it will wander around away from the centre, localizing out at both the loudspeakers.

Two Channel

This is what many people mistake for stereo sound systems, because there are two channels and a "stereo" console is connected to the front of the system, and stereo amplifiers and equalizers are used throughout the system. What is missing from most of these systems is uniform coverage of the entire listening area, and a minimal level and phase response difference for each channel's coverage of the listening area. To achieve proper loudspeaker coverage to replicate a stereo image in a large venue, it is necessary to have a loudspeaker system for each channel that can provide uniform coverage of the entire listening area while maintaining the directional cues. This is a very expensive, and sometimes impossible proposition.



Two channel systems usually suffer from having half the people in the listening area only hear half the audio program, which makes two channel systems a poor choice for music reinforcement. A large portion of the listeners hear a completely different music mix from other listeners. This is an all-too-common oversight in venues that are intended for music and entertainment, even high profile venues where they deserve better system designs. It tends to be a common misconception brought forward by people with a background in portable or live sound systems.

When a two channel system is used to reinforce a mono voice microphone, the seats either side of the room centreline, exactly between the two loudspeakers, also experience substantial variations in frequency response and uniformity of coverage due to interference and signal cancellation when identical signals arrive from the two channels at different times. This makes two channel systems especially ineffectual for speech reinforcement applications.

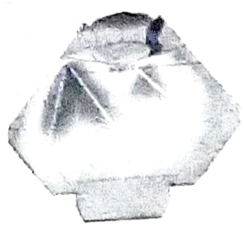
Left/Centre/Right

There are specialized applications for sound systems described as Left/Centre/Right configurations. This must combine the best of both worlds, right? Well it can, but the loudspeaker system must also be designed to the highest common denominator not the lowest. It is also important with an LCR system that the mix engineer understand which signals must be routed to which loudspeakers, and which signal routings will create problems. LCR systems also are not suitable for use in all room shapes and configurations.

LCR systems are common in drama theatres and large churches where there is a requirement for both mono speech reinforcement and music or sound effects cues to be localized or mixed with a particular perspective, with a stereo or stereo-like imaging. The three loudspeaker systems must each provide coverage of the entire seating area while maintaining level and directional cues, just like the mono and stereo systems described above. There are some clever "cheats" a system designer can use to achieve the extended listening area for stereo coverage, and that invloves using "stingers" or infill loudspeakers.

Stingers

A stinger is a speaker positioned to provide coverage of a floor area that may be inaccessible from the left or right cluster mounting position.



In the example above, the left and right stingers are separate from the centre cluster, and would be fed from a signal delay so that for people in the right side of the room, the signal from the left stinger would arrive at the same time as the sound from the left cluster would have if it could have reached that location. For people sitting near the front right, they would still hear the program from the left channel to maintain the integrity of the program content, even though the stereo image would be skewed to one side compared to the image perceived by the people sitting in the middle of the room. The speaker selection, as well as the level and delay setting of the stingers are all quite critical to their successful integration, if they are too loud, too late or too early they will detract from the image the rest of the audience hears. If you're going to try this at home study up on Haas Effect, setting levels and delays for precedence, as well as time domain measurement systems.

It is also possible to use some of the components of the centre cluster as the stingers, especially with the advent of lower cost DSP systems that allow a matrix of signal delays to be developed for each input to the unit. Even a few years ago this would have been exceedingly expensive and complicated (and nearly impossible to explain), now it can be programmed quickly on a computer screen and be made to work quickly and easily. This approach works best when the centre cluster has similar sounding loudspeaker components to the left/right clusters.

Which one's the best?

As with many questions about sound systems, there is no one right answer. A well designed mono system will satisfy more people than a poorly designed or implemented two channel sound system. The important thing to keep in mind is that the best loudspeaker design for any facility is the one that will work effectively with the, programmatic, architectural and acoustical constraints of the room, and that means (to paraphrase the Rolling Stones) "You can't always get the system that you want, but you find some times that you get the system that you need." If the facility design (or budget) won't support an effective stereo playback or reinforcement system, then it is important that the sound system be designed to be as effective as possible, even if that means giving up a desirable program requirement like stereo.

SOUND CHANNELS

Now let us see "What is sound channel in hardware context?"

For the duration of the pre-production stage, it will become necessary to imagine and define terms of basic equipment packages. The specific contents of each package type will vary from studio to studio, rental house to rental house, and even mixer to mixer.

Members of the industry use these terms for simplification only. Note that the use of the term "channel" is synonymous with "equipment package".

One-Mic Channel: The One-Mic Channel is the most basic of the generic sound recording packages (channels). It consists of a Nagra 4.2 sync recorder, headphones, one condenser "shotgun" microphone, a short mic cable.

Stage Channel: The Stage Channel is a complete sound recording package for theatrical style film making (such as feature films, commercials, and episodic television series). Contents usually include: one Nagra 4.2; a production mixing panel; soundcart; fishpole; three condenser mics; duplex mic cable; a few hundred feet of assorted single mic cables; etc. The word Stage Channel originated from the concept of filming on the soundstage. All of the basic sound recording tools are present; but replacement equipment and specialty items are not included.

Location Channel: The thought behind the Location Channel is that we have TWO Nagras, as well as a very full complement of microphones and other needed equipment. The magic word is redundancy. Location Channels are complained up Stage Channels, and feature two Nagras and plenty of equipment for major setups and contingencies. Radio mics are still extra, though.

Video Mic Channel: The Video Mic Channel is just a Stage Channel complete except that there is no Nagra recorder. It is for film-style video production, and includes a soundcart, mixing panel, fishpole, condenser mics, etc.

Now let us see "what is sound channel in programming context?"

In programming context a sound channel is a queue of sound commands that is managed by the Sound Manager, together with other information about the sounds to be played in that channel. The commands placed into the channel might originate from an application or from the Sound Manager itself. The commands in the queue are passed one by one, in a first-in, first-out (FIFO) manner, to the Sound Manager for interpretation and processing.

Your application can open several channels of sound for simultaneous output on the accessible audio hardware. The number and quality of concurrent channels of sound are limited only by the abilities of the machine, mainly by the speed of the CPU. The Sound Manager at present supports multiple channels of sound only on machines equipped with an Apple Sound Chip or equivalent hardware. To maintain maximum compatibility between machines for your applications, you should always check the operating environment to make sure that the ability to play multiple channels of sampled sound is present before attempting to do so.

There are eight available sound channels. You can use a Sound Channel object in script to access and modify any of the eight sound channels.

Note: You can modify only the first two sound channels in the Score of the Director user interface.

You can create a reference to a Sound Channel object by using the top level sound() method, the Player object's sound property, or the Sound object's channel() method. For example, you can reference sound channel 2 by doing the following:

Use the top level sound() method.

```
-- Lingo syntax
objSoundChannel = sound(2)

// JavaScript syntax
var objSoundChannel = sound(2);
```

Use the Player object's sound property.

```
-- Lingo syntax
objSoundChannel = _player.sound[2]

// JavaScript syntax
var objSoundChannel = _player.sound[2];
```

Use the Sound object's channel() method.

```
-- Lingo syntax
objSoundChannel = _sound.channel(2)

// JavaScript syntax
var objSoundChannel = _sound.channel(2);
```

SOUND AND ITS EFFECT IN MULTIMEDIA

There are mainly five (5) sound/audio effects available in all most audio editing software. These are the followings:

1. Amplitude effects

2. Delay effects
3. Time/Pitch effects
4. Reverse effect
5. Invert effect

1. Amplitude Effects:

The amplitude effect classified in the following eight (8) groups:

1) Amplify
2) Fade In / Fade Out
3) Normalize
4) Compressor
5) Expander
6) Envelope
7) Mute
8) Vibrato

Amplify: Amplify effect is used to increase or decrease the amplification of the sound in the media file. If you select a part of the file with the mouse, this effect will amplify or attenuate this exact part of the file. if not the sound of the entire file will be amplified or diminished.

Fade In and Fade Out:

Use the Fade In effect to fade in the sound in the media file. If you select a part of the file with the mouse, this effect will fade in the sound of this exact part of the file. Otherwise the sound of the beginning of the file will be faded in.

Use the Fade Out effect to fade out the sound in the media file. If you select a part of the file with the mouse, this effect will fade out the sound of this exact part of the file. Otherwise the sound of the end of the file will be faded out.

Normalize: Use this effect to achieve the maximum amount of amplification that will not result in clipping. If you select a part of the file with the mouse, this effect will amplify the highlighted selection to the percentage of the greatest level. if not the sound of the entire file will be normalized.

Compressor: Compressor effect is used to reduce the dynamic range of an audio signal. For example, compressors can be used to remove the variations in the peaks of an electric bass signal by clamping them to a constant level (thus providing an even, solid bass line.) Compressors can also be useful in compensating for the wide variations in the level of a signal produced by a singer who moves frequently or has an unreliable dynamic range.

Expander: Expander effect is used to expand the dynamic range of an audio signal. Expander boosts the high-level signals and satisfies low level signals. If you select a part of the file with the mouse, this effect will be applied to this exact part of the file. Otherwise the sound of the whole file will be changed.

Envelope: Envelope effect is used to change the audio file amplitude in accordance with the specified coordinates. It allows to absorb the sound and make it quiet slowly. It generally used to smooth beginning or ending of the sounds. Also this effect is useful for creating the audio loops and samples. If you select a part of the file with the mouse, this effect will be applied to this exact part of the file. Otherwise the sound of the whole file will be affected.

Mute: Mute effect is used to switch off the sound in the edited audio file. If you select a part of the file with the mouse, this effect will mute this exact part of the file. Otherwise the sound of the whole file will be muted.

Vibrato: Vibrato equals to a cyclical changing of a certain frequency of the input signal. If you select a part of the file with the mouse, the Vibrato effect will be applied to this exact part of the file. Otherwise whole file will be changed.

2. Delay Effects:

The Delay effect classified in the following five (5) groups:

1) Delay
2) Phaser
3) Flanger
4) Chorus
5) Reverb

Delay: This effect permits you to create an echo effect of your audio track by replaying the sounds of the selected audio portion after a certain period of time. Applying of this filter can bring life to dull mixes, widen and fill out your instrument's sound.

You can use this function to create single echoes, as well as a number of other effects. Delays of 35 milliseconds (ms) or more will be perceived as discrete echoes, while those falling within the 35-15 ms range can be used to create a simple chorus or flanging effect. (These effects will not be as effective as the actual Chorus or Flanger effects, as the delay settings will be fixed and will not change over time).

Phaser: The Phaser filter makes the selected portion of your audio thinner or fuller through mixing the automatically filtered and unfiltered audio signals. You can apply this filter to give a "synthesized" or electronic effect to natural sounds.

The Phaser achieves its distinctive sound by creating one or more notches in the frequency domain that eliminate sounds at the notch frequencies.

Phasing is very similar to flanging. If two signals that are identical, but out of phase, are added together, then the result is that they will cancel each other out. If, however, they are partially out of phase, then partial cancellations and partial enhancements occur. This leads to the phasing effect.

Flanger: The Flanger effect is one of the other elaborated audio effects that is created by mixing a signal with a slightly delayed copy of itself, where the length of the delay is constantly changing. With the Flanger filter you can "shape" the sound through controlling how much delayed signal is added to the original. Use it if you want to create the "whooshing" sound effect in some fragment of your audio track.

Flanger is a special case of the Chorus effect: it is created in the same way that Chorus is created. In days gone by, flanging used to be created by sound engineers who put their finger onto the tape reel's flange, thus slowing it down. Two identical recordings are played back simultaneously, and one is slowed down to give the flanging effect.

Flanger gives a "whooshing" sound, like the sound is pulsating. It is essentially an exaggerated Chorus.

Chorus: The Chorus effect allows you to make your audio sound fuller. It can make a single instrument sound like there are actually several instruments being played. It adds some thickness to the sound, and can be described as 'lush' or 'rich'.

The Chorus effect is so named because it makes the recording of a vocal track sound like it was sung by two or more people singing in chorus. This is achieved by adding a single delayed signal (echo) to the original input.

The Chorus differs from the Flanger in only a couple of ways. One difference is the amount of delay that is used. The delay times in a Chorus are larger than in a Flanger. This longer delay doesn't produce the characteristic sweeping sound of the Flanger. The Chorus also differs from the Flanger in that there is generally no feedback used.

Reverb: The Reverberation filter helps you apply the particular effect when the sound stops but the reflections continue, decreasing in amplitude, until they can no longer be heard.

You can use this function to set Reverb effect that is used to simulate audio space, and consists of both early reflections and echoes that are so closely spaced that they are perceived as a single fading sound. Reverb is different from the basic echo function in that the delays are not repeated at regularly spaced intervals. Reverb function can create a wide range of high-quality reverb effects.

It is the sound you hear in a room with hard surfaces where sound bounces around the room for a while after the initial sound stops. Reverb is used to simulate the acoustical effect of rooms and enclosed buildings. In a room, for instance, sound is reflected off the walls, the ceiling and the floor. The sound heard at any given time is the sum of the sound from the source, as well as the reflected sound.

3. Time/Pitch Effects:

The Time/Pitch effect classified in the following two (2) groups:

1) Time Stretch
2) Pitch Shift

Time Stretch: The Time Stretch effect permits to change the tempo (rhythm), but keep the pitch the same throughout. If you select a part of the file with the mouse, this effect will change the tempo of this exact part of the file. Otherwise the tempo of the whole file will be changed.

Pitch Shift: The Pitch Shift effect shifts the frequency spectrum of the input signal. It can be used to mask a person's voice, or make the voice sound like that of the "chipmunks", through to "Darth Vader". It is also used to create harmony in lead passages, although it is an "unintelligent" harmonizer.

4. Reverse effect:

With the help of this function you can make a selection play backwards by reversing the order of its samples. It is useful for creating special effects.

If you select a part of the file with the mouse, this effect will applied to this exact part of the file. Otherwise the sound of the whole file will be changed.

5. Invert effect:

With the help of this function you can simply invert the samples, so that all positive offsets are negative and all negative offsets are positive. Inverting does not produce an audible effect, but it can be useful in lining up amplitude curves when creating loops, or pasting. On stereo waveforms, both channels are inverted. If you select a part of the file with the mouse, this effect will be applied to this exact part of the file. Otherwise the sound of the whole file will be altered.

Using VST Effects:

Various Audio Editor supports *VST audio plug-ins* and effects on your computer. VST (*Virtual Studio Technology*) effects and filters are *native real-time plugins* developed by Propellerhead/Steinberg. They usually provide a custom graphical user interface, displaying controls similar to the physical switches and knobs on audio hardware. With the help of VST effects you can easily extend the range of powerful effects available in various Audio Editor. In case you have VST effects on your computer you will need to point out their directory to your Audio Editor program. VST effects are very easy to use. Most VST effects have their own settings dialog where you can configure the parameters of the effect. Some plug-ins may support real-time preview. But, of course, if you have some problems you will need to consult the documentation provided by the plug-in manufacturer.

ANALOG V/S DIGITAL SOUND

What does it mean to be digital and what does it mean to be analog?

Well, they are both ways of encoding information. Digital lends itself to computers and other electronic equipment by recording information into 1's and 0's. This data can then be read by electronic instruments and then produced into something familiar we can understand such as words, picture or sound.

Analog on the other hand is included of continuous and variable electrical waves that represent an infinite number of values. I know what you're saying : "Siv Siv Siv". Think of it like this, (and mind you this is a very rough analogy), If sound was recorded digitally, it is made into 0s and 1s right? Those 0s and 1s represent all the little bits of a sound. When you put them together you get a full sound. Analog records sound just as it hears it, it doesn't break it down into all these separate pieces...it's CONTINUOUS.

As we become more computerized, everything is going digital, and why not? Digital offers a lot of improvements over analog, so much so that we shouldn't even be having the conversation. Honestly, besides the whole sound issue, it's not even a competition.

Analog — It's All About Resolution

Analog has — and always will have — better resolution than digital, but it comes with the side effect of sound coloration. When an audio signal is passed through physical elements such as tubes and capacitors it will be affected in some way. Even if the processing filers are on bypass, the act of routing a signal through these components changes the signal. This can be a benefit or a detriment, depending on your sonic perspective and preference. Many people want what they call a "fat, warm sound." This is certainly a characteristic that analog equipment in mastering can communicate, but there are two other factors even more important in achieving this sound: (1) the skill of the mastering engineer choosing the equipment/turning the knobs, and (2) how the music was recorded, as the mastering engineer can only work with what an artist or producer delivers. If the original recording has the production qualities of the Backstreet Boys, analog mastering cannot give it the sonic characteristics of early Steely Dan. Likewise, if a mastering engineer over-processes a good recording, even the best analog gear can sound harsh or muddy. The bottom line: Simply having a piece of analog gear in a signal chain is no guarantee of "analog" sound.

Digital — It's All About Control

Digital — with its ability to apply (and undo) unique changes to a virtually limitless number of scenarios — has greater control than analog, but it comes with the side-effect of lower resolution. though, keep in mind that "lower resolution" is a relative term. The 24/96 platform that many hi-fi enthusiasts proclaimed to be "as good as analog" has already been eclipsed by much better resolution rates and technological concepts (DSD, for example). What we're talking about is theoretical resolution. Much of what is criticized as missing in digital recording and processing falls into the "unheard, but felt" category; overtones and undertones that are beyond the scope of current sampling standards. In theory, digital will never have the resolution of analog. At a certain point, though, it becomes unnoticeable to the human ear, especially when utilized by a skilled and knowledgeable audio engineer.

Therefore where does all this leave us? It depends on the application. In mastering, the ultimate goal is to apply changes to music that maximize its sound — punchier drums, clearer bass...or whatever is desired — without imparting unwanted coloration from the process itself, while at the same time maintaining the highest resolution. For this application, then, the perfect solution would probably contain the best elements of both analog and digital. Keep in mind, though, that either platform is just a means to accomplish work...not the work itself.

Analog versus digital sound in reference of Sampling

In analog sound recording, such as that on phonograph discs, audiocassettes, and standard audiotapes, an analog of the source audio waves is physically produced. Playback then requires an abrasive physical device to literally trace the recorded sound wave. Digital sound recording, such as that on compact discs, videodiscs, and CD-ROMs, instead involves taking multiple discrete measurements of the voltage levels of the continuous source audio waves, a process known as *sampling*. The most common sampling rate is 44.1 kilohertz (kHz), or 44,100 times per second, which guarantees at least two measurements of any humanly audible sound wave. (The typical sound range audible to a person is 20 Hz to 20 kHz.) The accuracy of the recorded voltage measurements depends critically on the number of binary digits (bits) used to record the measurements. additional bits enable finer distinctions to be made in audio voltage levels and, in turn, enable a closer approximation of the original sound wave. The industry standard of 16 bits is sufficient to produce an audibly smooth curve with very little distortion. For even greater fidelity, music studios sometimes use 24-bit encodings for their master tapes. Because the recorded bits are read from the internal reflective layer of the CD by a laser, the disc remains untouched by any physical object and thus does not degrade under normal use.

BASICS OF DIGITAL SOUND-SAMPLING

Now let us see "What is sound-sampling?"

Definition: "Sampling is the process of re-using portions of sound recordings in a piece."

In music, **sampling** is the work of taking a segment, or **sample**, of one sound recording and reusing it as an instrument or a different sound recording of a song. This is usually done with the help of a sampler, which can be a piece of hardware or a computer program on a digital personal computer. Sampling is also possible with tape loops or with vinyl records on a phonograph. People who sample are generally referred to as **producers** or beatmakers. Although beatmaking can be done by means of various live instruments and synthesisers, sampling is the process most enjoyed by beatmakers.

Now let us see "How Digital Sampling Works?"

All digital recordings—starting in the recording studio—are made by creating digital samples of the original sound. The way it works is that special software "*listens*" to the music, and takes a digital snapshot of the music at a particular point in time. The length of that snapshot (measured in bits) and the number of snapshots per second (called the sampling rate) determine the quality of the reproduction. The more samples per second, the more accurate the resulting "picture" of the original music.

Compact discs sample music at a 44.1kHz rate—in other words we can say that, the music is sampled, digitally, 44,100 times per second. Each sample is 16 bits long. When you multiply the sampling rate by the sample size and the number of channels (two for stereo), you end up with a bit rate. For CDs, you multiply 44,100 X 16 X 2 and end up with 1,400,000 bits per second—or 1,400Kbps.

All these bits are transformed into data that is then copied onto some sort of storage medium. In the case of CDs, the storage medium is the compact disc itself; you can also store this digital audio data on hard disk drives, or in computer memory.

The space taken up by these bits can add up quickly. If you take a typical three-minute song recorded at 44.1KHz, you end up using 32MB of disk space. While that song can easily fit on a 650MB CD, it is much too large to download over a standard Internet connection, or to store on a portable music player.

This is where audio compression comes in. By taking selected bits out of the original audio file, the file size is compressed. If the right bits are excised, you will never miss them.

The Specifics of Sampling

An analog system allows a continuous variation in voltages which corresponds analogously to a continuous variation in air pressure. A digital system must make discreet measurements that approximate this continuum of values. The greater the number of measurements, the more accurate the result. The decision, therefore, is between accuracy and storage space.

Sample Rate

A *sample* is a single measurement of *amplitude*. The sample rate is the number of these measurements taken every second. In order to accurately represent all of the frequencies in a recording that fall within the range of human observation, generally accepted as 20Hz-20KHz, we must make sure that we choose a sample rate high enough to represent all of these frequencies. At first consideration, one might choose a sample rate of 20KHz since this is identical to the highest frequency. This will not work, however, because every cycle of a waveform has both a positive and negative amplitude and it is the rate of alternation between positive and negative amplitudes that determines frequency. for that reason, we need at least two samples for every cycle resulting in a sample rate of at least 40KHz. This principle is known as the *Nyquist Theorem*. It is usually stated as follows:

sample rate = 2 x highest frequency

or in another version:

N = 1/2(sample rate)

The Nyquist frequency (shown here as "N") is therefore defined as one half of the sample rate and is the highest frequency that can be accurately represented. So given some standard sample rates, we can easily find the Nyquist frequency:

Media	Sample Rate	Nyquist Frequency
CD Standard	44.1KHz	22.05KHz
DAT (alternate)	48KHz	24KHz
DVD	96KHz	43KHz

In each case, the Nyquist frequency is above the highest frequency in the range of human hearing.

Sample Resolution

Every sample can only be calculated to a certain degree of *accuracy*. The accuracy is dependent on the number of bits used to represent the amplitude, which is also well-known as the *sample resolution*. The sample resolution determines the best possible signal to noise ratio of the digital medium in question as follows:

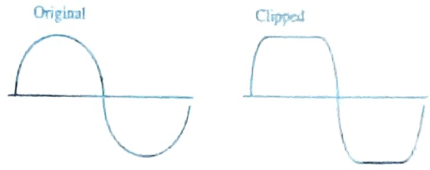
8-bit (X 6) = 48db SNR
12-bit (X 6) = 72db SNR
16-bit (X 6) = 96db SNR
24-bit (X 6) = 144db SNR

To put these numbers in perspective, consider the following: the dynamic range of human hearing from the threshold of perception to the threshold of pain is between 130 and 140db. In contrast to that, the dynamic range of high quality audio tape is around 70db and the dynamic range of CDs (16 bit) is 96db. DVD, on the other hand, has a sample resolution of 24 bits, allowing it theoretically to capture the full dynamic range of acoustic music.

The other point to consider is that every *six decibels* adds one bit to the sample resolution. Why is this important? Because, if the hottest signal in a 16-bit recording is at -6db (with 0db the loudest signal the system can represent) the result is the same as a 15-bit recording with the hottest signal at 0db. In fact, if you normalized this file (raising the highest amplitude to 0db), the result would be the same to a 15-bit recording.

Clipping

Both analog and digital media have an upper limit beyond which they can no longer accurately represent amplitude. Analog clipping (or overdrive or distortion) varies in quality depending on the medium. A tube amplifier, for example, has a much warmer distortion than a solid state amplifier. In each case the upper amplitudes are being altered, distorting the waveform and changing the timbre, but the alterations are slightly different. Digital clipping, in contrast, is always the same. Once an amplitude of 1111111111111111 (the maximum value in a 16 bit resolution) is reached, no higher amplitudes can be represented. The result is not the smooth, rounded flattening of analog clipping, but a harsh slicing of off the top of the waveform, and an unpleasant timbral result.



Now let us see example to understand sound- sampling

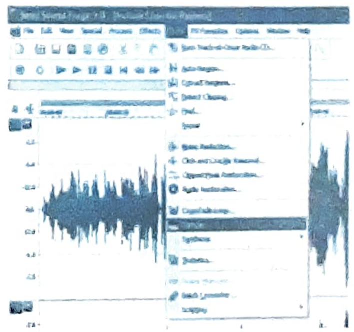
In Sound Forge software with the help of Sampler Tool, we can use the powerful and easy-to-use sample-editing capabilities to create and edit samples and then transfer them to our external and internal samplers.

Using Sound Forge Software with a Sampler

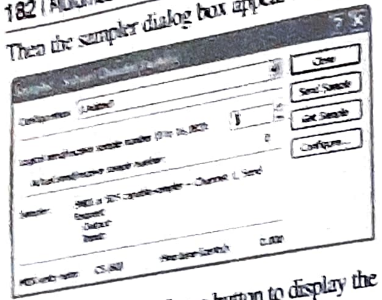
Choose Sampler option from the Tools menu, to send data to and receive data from a sampler.

Set up an external sampler

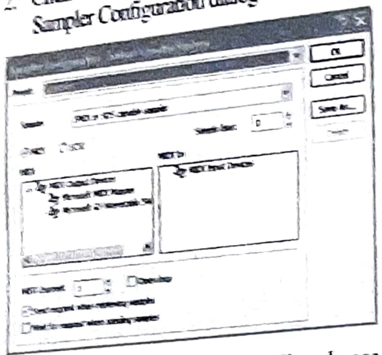
1. Choose Sampler from the Tools menu to display the Sampler dialog as figure.



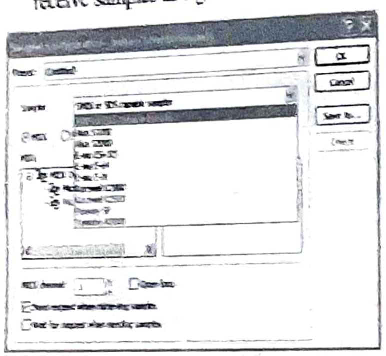
Then the sampler dialog box appear as figure.



2. Click on the Configure button to display the Sampler Configuration dialog box as figure.



3. From the *Sampler* drop-down list, choose the sampler to which you wish to send and receive samples as figure.



If your sampler is not listed here, choose SMDI or SDS capable sampler.

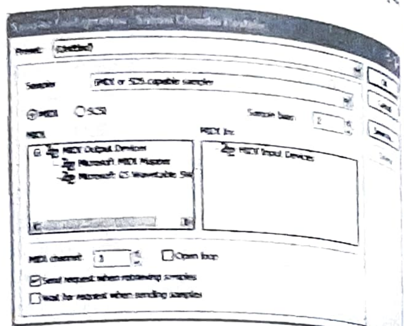
4. Specify input/output settings for the sampler.

■ If your sampler uses MIDI/SDS transfer, click the MIDI radio button and choose input and output ports in the MIDI Out and MIDI In boxes.

■ If your sampler uses SCSI/SMDI trans-fer, click the SCSI radio button and select your sampler in the Sampler dialog box.

The Sampler dialog box will list all devices connected to the selected SCSI host, including devices that are not samplers.

5. In the *Sample bias* box, specify a number-ing offset that allows you to add or subtract a value to the Logical send/receive sample number to compensate for differing sample storage schemes as figure.



It is often easiest to set the Sample bias so that a Logical send/receive sample number of zero corresponds to the first available sample storage number in your sampler.

Perform steps 6 through 9 only if you selected the MIDI radio button in Step 4.

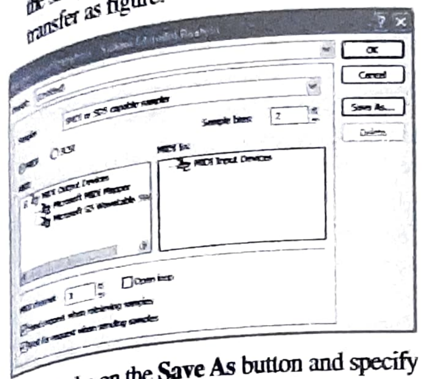
6. In the *MIDI channel box*, specify the MIDI channel through which sample data will be sent when using SDS.

7. Select the *Open loop* check box if you want to send SDS sample data immediately upon clicking the Send Sample button. This is an unconditional transfer of sample data (no handshake).

When this check box is selected, the sampler's MIDI output does not have to be connected to your computer's MIDI input. However, Open loop decreases sample data speed and is more prone to transfer errors.

8. Select the *Send request when retrieving samples* check box if you wish for the Sampler tool to send a request (handshake) to the sampler before beginning an SDS sample transfer.

9. Select the *Wait for request when sending samples* check box if you want the Sampler tool to wait for a request (handshake) from the sampler before beginning an SDS sample transfer as figure.



10. Click the on the **Save As** button and specify a preset name to save your settings.

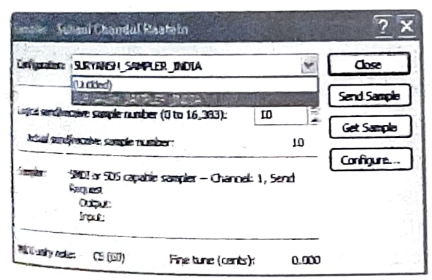
Send or receive samples

Sending a sample will always send the entire contents of the active data window. Any selection contained in the waveform display is ignored. When receiving a sample, the entire contents of the active data window are replaced with the new sample data (a warning will be displayed before any data is replaced).

1. From the **Tools** menu, choose **Sampler**.

2. Choose a sampler from the Configuration drop-down list.

3. In the *Logical send/receive sample number* box, specify the number your sampler uses as its location reference for samples sent or received AS FIGURE . This number can be biased for specific samplers with the Sample bias setting in the Sampler Configuration dialog.



The Actual send/receive sample number is displayed below the Logical send/receive sample number. The actual value is the sum of the logical value and the specified sample bias.

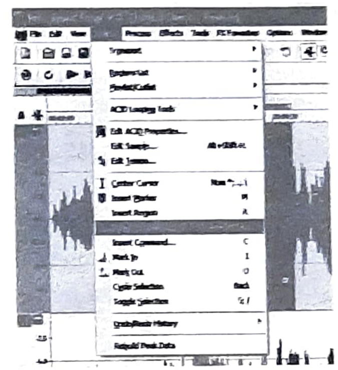
4. Click the **Send sample** or **Get sample** button to start the data transfer.

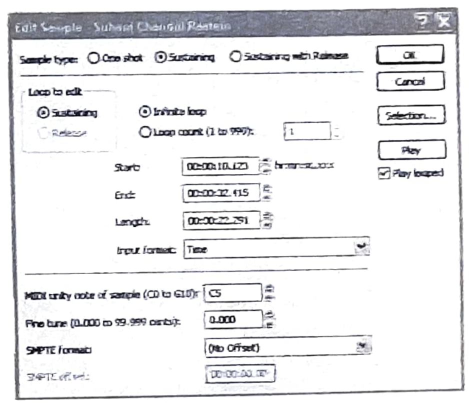
Creating or Editing Sample Information

The Edit Sample dialog allows us to create or edit samples and loops.

Perform any of the following actions to create a sample:

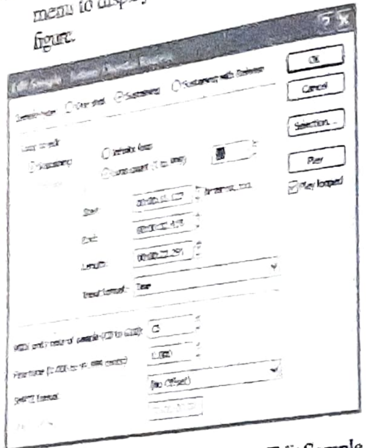
■ Choose Insert Sample Loop from the Special menu to create a sample loop



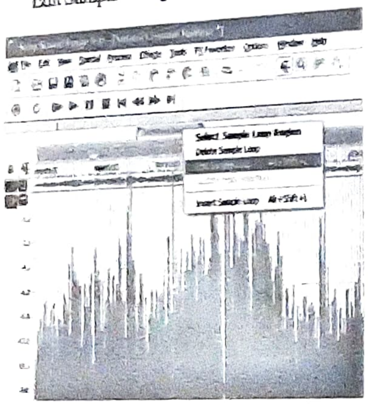


■ Press Alt+L key from the keyboard when a range of data is selected to create a sample loop without displaying the Edit Sample dialog.

- Choose Edit Sample Loop from the Special menu to display the Edit Sample dialog as figure.



- Right-click a loop tag and choose Edit Sample Loop from the shortcut menu to display the Edit Sample dialog box as figure.



BASIC OF DIGITAL SOUNDS

(i) Sampling and Sampling Rate :

Digitizing is the process of converting analog sound into digital sound. Analog sound is represented with the help of continuous waveform or sound waves. To digitize such a sound wave, the wave is divided into a series of snapshots or samples at uniform intervals. This is known as *sampling*. The pictorial representation of the sampling of a sound wave is as follows :

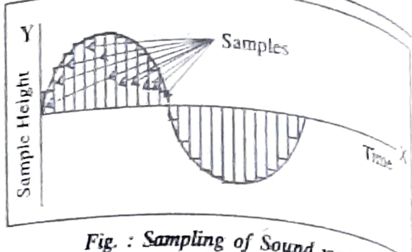


Fig. : Sampling of Sound wave

In this figure, each vertical bar is a smple. The height of the bar represent the amplitude the bar represent the amplitude of the wave at that point. *The sampling rate refers to the number of samples recorded per second from the source audio.* A higher sampling rate leads to better sound quality. Sampli rate determine how often the analog signal is digitized. A higher sampling rate means that more processing power would be required while recording and playback. This is bacause the computer has to repeat a set of operations for every sample within the audio. The more number of sample used to represent each second of audio, the more will be the number of operations that the computer must process. Sampling rate is measured in *Kilohertz (KHz)*. The sampling rate of a sound waves is different for different delivery media. For example the sampling rate is 44.1 KHz for a CD and around 11 KHz for sound that is to be broadcast over radio.

(ii) Bit Depth :

Bit depth refers to the number of bits used to represent one sample of an audio signal. The higher the number of bits used to represent a sample, the more accurate the sound produced. A high bit depth also means that more memory would be required to store each sample. We can improve quality of sound by using extra bits to represent sample. The standard bit depths used are 8, 16 and 24. Today, professional sound recording systems use 32 bits as well.

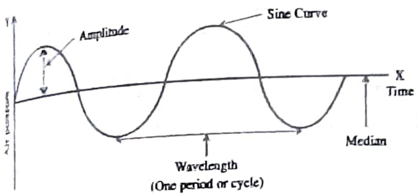
(iii) Pitch :

We hear sounds when our eardrums detect fluctuations in air pressure. Sound waves of different frequencies produce different sound. We can detect waves with frequencies ranging from 20 *Hz to 20,000 Hz*. The human ear can distinguish should having different frequencies by the sensation that are produced by these sounds. These sensations are referred to as the **pitch** of sound. A high pitch would correspond to a high frequence sound and a lower pitch would signify a low frequency.

Therefore we can say that the human ear can distinguish frequencies of sound through the pitch of the sound.

(iv) Median :

The central line passing through the length of the sine wave is called the *median*. Fig.. displays the median of a sine wave. Parts of the wave close to the median signify low amplitude or less vibration of particles. The farther the wave is from the median, the greater is the amplitude.



Sound Wave

> ◎ 1 Hertz (Hz) is equal to one cycle per second (CPs)
>
> 1 Hz = 1 cycle/sound
>
> or
>
> 1 Hz = 1 cps
>
> Hertz is the measurement unit of frequency.

(v) Wavelength

"The wavelength of a sound wave is the distance covered by the wave in completing one cycle."

Wavelength can be measured from its waveforms as the distance between the two end points of a complete cycle. Wavelength is measured in maters (m).

> ◎ The amplitude of a sound wave is the maximum displacement of the vibrating particles from their original positions. Amplitude is the measurment of air pressure. The amplitude of sound is measured in decibels (db).

(vi) Quantization

Now let us see "What is Quantization ?"

Quantization determines the digital measured value of the analog waveform at the sampling time.

The quantization process consists of converting a sampled signal into a signal that can take only a limited number of value. Quantization is measured in terms of number of bits and its typical value are 8, 12 and 16 bits. An 8-bit quantization provide 256 (2^8) possible values. A 16-bit quantization in CD quality results in more than 65, 536 possible values.

(vii) Number of Channels

It refers to the number of channels used for audio recording. For mono (mono) system the typical value of number of channels are 1 and for stereo (stero) systems are 2 (two).

SOUND ON PC

To play sound on PC the following hardware devices are generally required in a computer system:

(i) Sound board
(ii) Input devices such as micro phone.
(iii) Output devices such as speakers or head-phones.
(iv) MIDI devices may also be used as both input and output devices for sound/audio data.
(v) Synthesized sound can alos be generated on a computer using keyboard and sound sequencer software.
(vi) Sound mixers are used to combine multiple channels of sound with controls like synchronization points.

To play sound on PC we required the following software :

(i) **Sound Recording** s/w such as - Mixcraft4, h-Track studio, Overdub!, Blaze Media

Pro, Audio Terminator, All Editor, MP3 AudioMixer, Wave Creater, RiffWorks T4.

(ii) **Sound Editing** s/w such as - Wavesaur, Wave Editor, SoX, Jokosher, AVS Audio Editor, Sound Forge, MP3DirectCut, Power Sound Editor, Sound Engine, DJ Audio Editor etc.

(iii) **Audio clips :** This is a library of audio clips (pre-made sound, music and narration). We can easily select and directly import an audio clip and use it from this library.

(iv) **Text to speech conversion s/w :** We can use this s/w to convert written text into corresponding sound.

(v) **Speech-to-text conversion s/w :** We can use this s/w to convert speech into corresponding text.

(vi) **Audio file importing :** We can multimedi application by using imported audio files in some standard formats such as .wav (Windows files), .MID (MIDI files), .voc (voice file) and .AIF (Audio Interchange file) etc.

(vii) **Voice Recognition S/w :** The voice recognition s/w is used to identify the speaker of a given voice by matching the voice with a set of voices stored in a database along with the details of the speaker of each voice.

Introduction of Sounds and Audio Devices

With the help **Sounds and Audio Devices** applet in Control Panel, we can allocate sounds to some system events. Some examples of system events are a computer program performing a job or having a problem performing a job, **minimizing** or **maximizing** a program window, or trying to copy a file to a floppy disk without a disk in the floppy disk drive.

Sounds vary from a simple beep to a brief piece of music. We can assign these sounds to system events according to our first choice. For instance, we can allot a specific sound that Windows plays every time you obtain a new e-mail message or print operation complete. We can also save all of our sound assignments as a sound scheme. Soon after, we can assign a completely different set of sounds to system events, save this scheme under a new name, and switch between the old and new schemes without losing our settings.

Common tasks

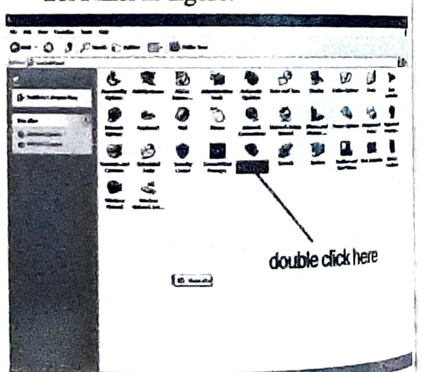
To customize system sounds you can frequently perform the following tasks:

1. **Assign sounds to system events:** We can assign sounds to a variety of system events. For example, you can configure Windows to play a special sound every time you get a new e-mail message.
2. **Create a sound scheme:** We can modify which sounds are played for a variety of system events and save the event sound links as a sound scheme.
3. **Change the system sound volume:** We can modify the computer's volume and choose whether to display the volume control icon in the notification area.
4. **Adjust the volume for multimedia recording devices:** We can adjust the volume for multimedia recording devices, such as a microphone's input volume.
5. **Adjust the volume for multimedia playback devices:** We can adjust the volume for multimedia playback devices, such as speakers.
6. **Adjust speaker volume:** We can adjust the speaker output for each speaker on our computer.

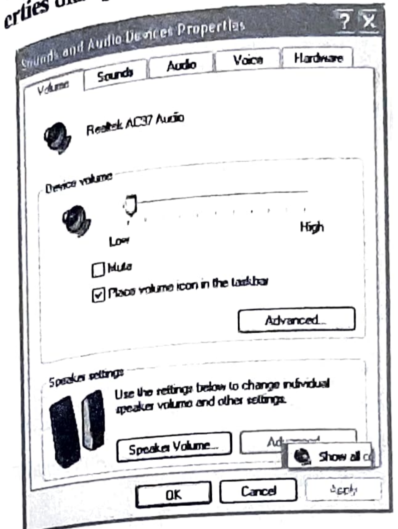
1-Assign sounds to program events:

To assign sounds to program events follow the following steps:

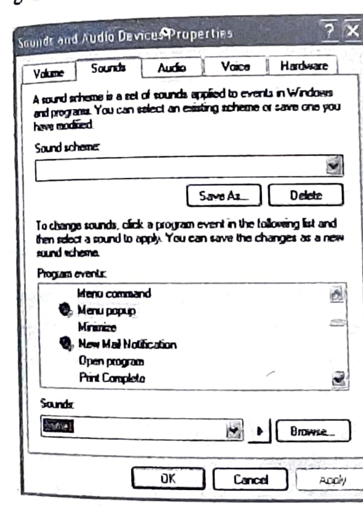
1. Open **Sounds and Audio Devices** in Control Panel as figure.



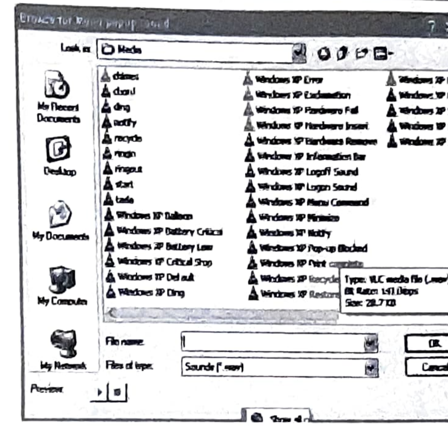
2. Then the **Sound and Audio device Properties** dialog box appears as figure.



3. Under the **Sounds** tab, in the **Program events** list, click the program **event** to which you want to assign a sound.
4. In the **Sounds** list box, click the sound you want to play whenever the selected program event occurs.



5. If the sound you want to use is not listed, click **Browse** to search for it as figure.



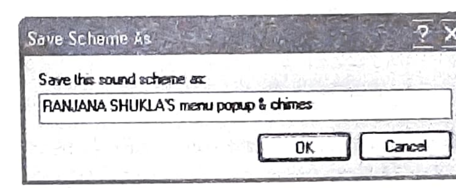
> ◎ To open Sounds and Audio Devices, click **Start**, click **Control Panel**, and then double-click **Sounds and Audio Devices**.

To test a sound, in the **Sounds** box, click the sound you want to test, and then click **Play sound**. To stop the sound, click **Stop** button.

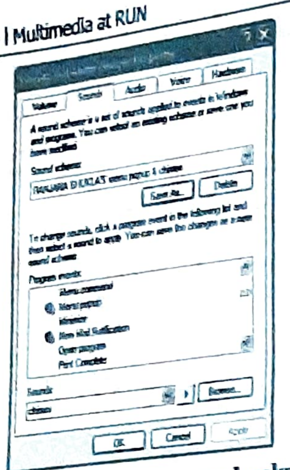
2-Creating a sound scheme:

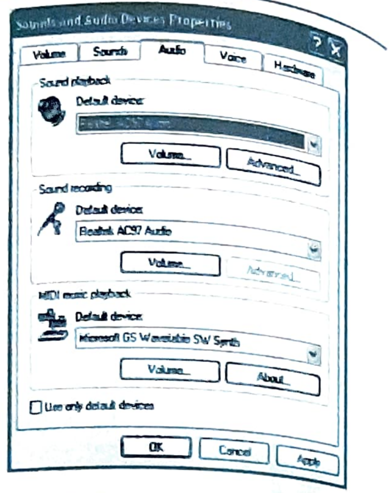
To create a sound scheme follows the following steps:

1. Click on the **Sounds and Audio Devices** applet in Control Panel.
2. Under the **Sounds** tab, in the **Program events** list, assign a sound for each **event** you want to save in a new sound scheme.
3. Under **Sound scheme**, click **Save As**.
4. In the **Save Scheme As** dialog box, type a name for the new sound scheme as figure.



5. The new scheme is displayed in the **Sound scheme** box as figure.

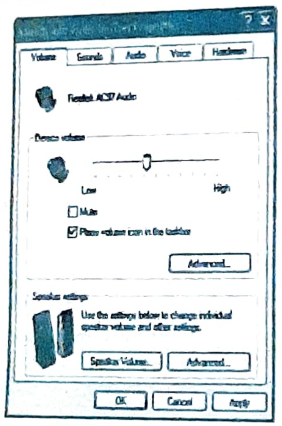




3-Changing the system sound volume:

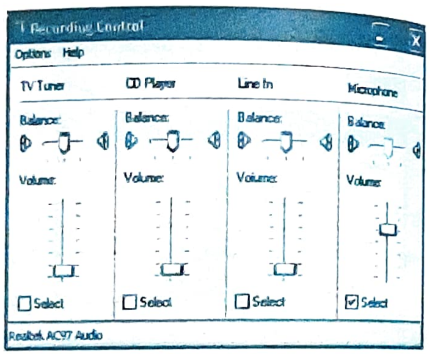
To change the system sound volume follows the following steps:

1. Click on the **Sounds and Audio Devices** applet in Control Panel.
2. Click on **Volume** tab.
3. On the **Volume** tab, under **Device volume**, drag the slider right or left to increase or to decrease the system volume as figure.



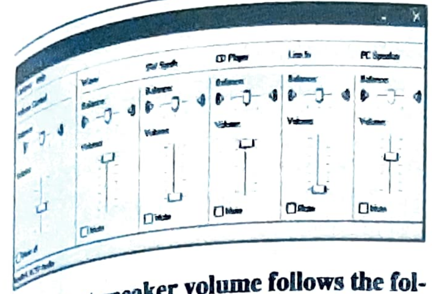
4-To adjust the volume for multimedia recording devices

1. Click on the **Sounds and Audio Devices** applet in Control Panel.
2. Click on the **Audio** tab as figure.
3. On the **Audio** tab, under **Sound recording**, click **Volume**.
4. In the **Recording Control** dialog box, drag the **Volume** slider for the appropriate device up or down to increase or decrease the input volume as figure.



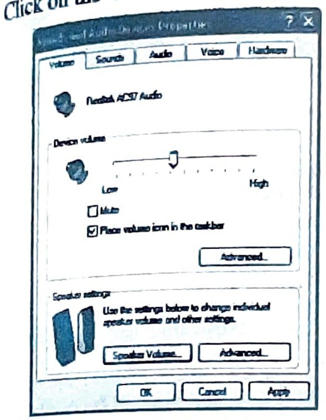
5-To adjust the volume for multimedia playback devices follows the following steps:

1. Click on the **Sounds and Audio Devices** applet in Control Panel.
2. On the **Audio** tab, under **Sound playback**, click **Volume**.
3. In the **Master Out** dialog box, drag the **Volume** slider for the appropriate device up or down to increase or decrease the output volume.

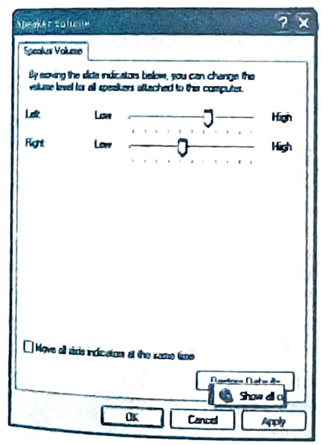


6-To adjust speaker volume follows the following steps:

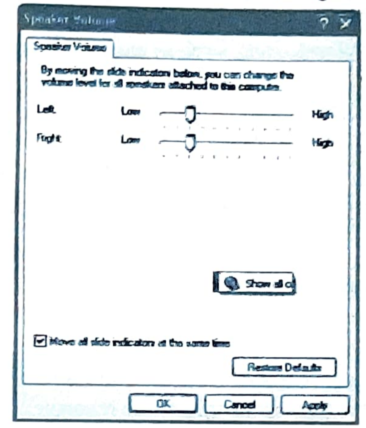
1. Click on the **Sounds and Audio Devices** applet in Control Panel.
2. Click on the **Volume** tab as figure.



3. On the **Volume** tab, under **Speaker settings**, click **Speaker Volume**.
4. Then the **Speaker Volume** dialog box appears as figure.



5. Move the slider controls to the *right* or *left* to increase or decrease the volume for each available speaker.
6. To move each slider control at the same time while saving their relative positions, select the **Move all slide indicators at the same time** check box as figure.



7. To return the speaker volume to the default positions, click **Restore Defaults**.

Speech overview

Speech capabilities for a personal computer refers to the ability to play back text in a spoken voice (that is referred to as text-to-speech or TTS), or to convert a spoken voice into electronic text (that is referred to as speech recognition or SR). The two capabilities are independent of each other. Many PC will have only *TTS*. SR may be installed later either by loading a speech package, or more commonly, through an application which has incorporated speech into it. For example, a new word processor or office tool suite may contain speech and it will be loaded at that time.

In general, speech- allowed applications will use Speech properties in Control Panel to access and control features. In this way, speech may be modified for your personal first choice or a exact office location. While TTS and SR may be loaded at different times, it is possible that not all the Help will be applicable. In cases where SR is not available, references to SR help and SR procedures may be unobserved.

Overview of Speech setup

For speech systems to act properly or for best outcome, the components require to be set up properly. Speech capabilities have been designed to work with system defaults so that a least effort is necessary on your part. Apart from physically connecting speakers and microphones, all other aspects are planned to work automatically. As well, some systems are ready with built in devices and it is possible that no configuration is required.

The first step for proper installation, or double-checking an existing installation, is following the set up guidelines presented here.

To set up a microphone

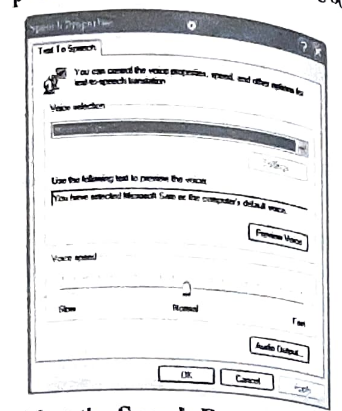
Microphones differ very much in design and purpose. They will carry on to develop and become more specific.

1. Locate the sound connections and connect the microphone jack to the computer. Most computers use an internal sound card and often the connections will be in the back of the system.
2. These will be a chain of connections the same size and diameter as the microphone jack. One will be labeled as the microphone connection, either with a small icon that looks like a microphone or explicitly labeled as such.
3. Plug the microphone into that connection.

To test the connection follows the following steps:

1. In control panel click on **Speech applet as figure.**

2. Then the Speech properties dialog box appears as figure.



3. Select the **Speech Recognition** tab.
4. Speak directly into the microphone. The sound level should register in the Microphone Level indicator.

> To open a Control Panel item, click **Start**, point to **Settings**, click **Control Panel**, and then double-click the appropriate icon.

To set up microphone options follows the following steps :

The audio input line may be selected.

1. In control panel click on **Speech applet**.
2. On the **Speech Recognition** tab, click **Audio Input**, and then click **Properties**.
3. Select either **Use automatically chosen line** or **Use this audio input line.**

   Use automatically chosen line sets the input line to a default that is determined by the speech system. Because of differences in drivers, capabilities and languages used, there might be variances in the selected default option. The selected default may not work with all options. If the device line does not work properly, you should manually select a new line using **Use this audio input line**.

   Use this audio input line allows you to select another line for audio input. The options present all audio line possibilities for the system. Not all audio lines are supported for speech.

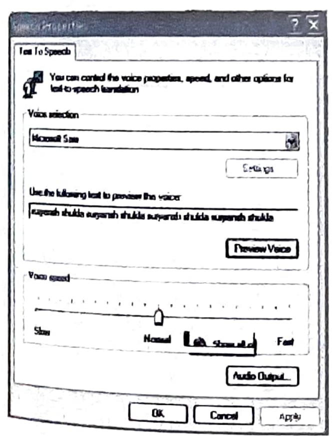
To set up speakers

Speakers differ very much in design and purpose. They will continue to diverge and become more specialized. Locate the sound connections and connect the speaker jack to the computer. Most computers use an internal sound card and frequently the connections will be in the back of the system. These will be a chain of connections the similar size and diameter as the speaker jack. In many cases there will be two sounds out connections.

1. One will be labeled as a line-out connection. Nearly all speakers requiring a separate power supply (such as an electrical adapter or batteries) should use this connection. It is as well used to export amplified sound to recording devices including recordable CDs and tape cassette systems. The other connection is for the non-powered speakers. while the signal is boosted by the computer, powered speakers may be spoiled if connected.
2. Plug the speaker into the proper connection.

To test the connection follows the following steps:

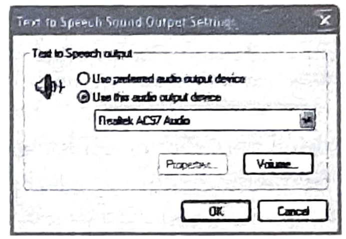
1. **In control panel click on Speech applet** as figure.
2. On the **Text-to-Speech** tab, click **Preview Voice** to hear the currently selected voice; the text will be spoken, highlighting the words as they are spoken. If the speakers are working properly, you will hear the spoken words as figure.



To set up speaker options

The audio output line may be selected. By default, this option is disabled. Though, other speech engines may include advanced properties for audio line out options. If so, **Audio Output** will be available. Follow instructions on the screen or those documented separately for the specific engine.

1. In control panel click on **Speech applet.**
2. On the **Text-to-Speech** tab, click **Audio Output.**
3. Then the **Text-to-Speech sound output setting** dialog box appears as figure.



4. Follow instructions presented on the screen.

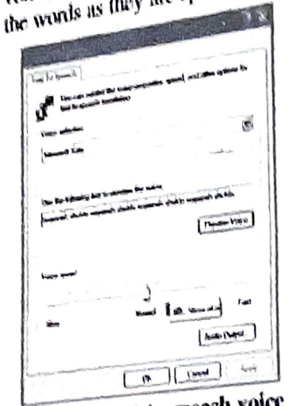
Overview of Text-to-speech:

Text-to-speech (TTS) is the capability of the operating system to have fun back printed text as spoken words. An internal driver, called a **TTS engine**, recognizes the text and using a synthesized voice selected from numerous pre-generated voices, speaks the written text. A *TTS engine* is installed with the operating system. Additional engines are also available through third party manufacturers. These engines often use a certain jargon or vocabulary; for example, a vocabulary specializing in medical or legal terminology. They can also use dissimilar voices allowing for regional accents such as British English, or speak a different language altogether such as German, French or Russian.

To determine the selected text-to-speech voice follows the following steps:

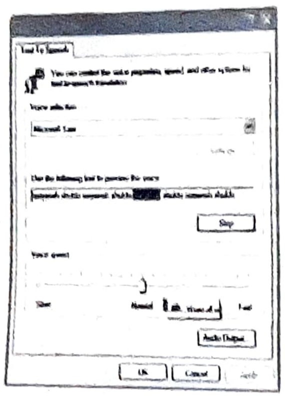
1. In control panel click on **Speech applet.**
2. On the **Text-to-Speech** tab, the displayed name in the Voice selection drop-down list box is the current active voice.

3. Click **Preview Voice** to hear the active voice; the text will be spoken, highlighting the words as they are spoken as figure.



To preview the text-to-speech voice follows the following steps:

1. In control panel click on **Speech applet**.
2. On the **Text-to-Speech** tab, the displayed name in the Voice selection drop-down list box is the active voice.
3. Click **Preview Voice** to hear the presently selected voice; the text will be spoken, highlighting the words as they are spoken. During playback, **Preview Voice** will change to **Stop**. Click **Stop** to interrupt the voice playback as figure.

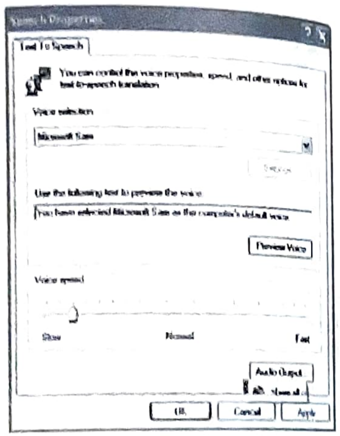


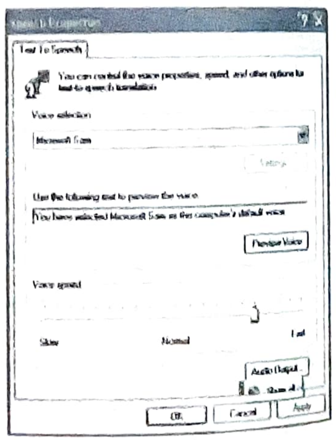
To change the text-to-speech voice or engine follows the following steps:

1. In control panel click on **Speech applet**.
2. On the **Text-to-Speech** tab, the name displayed in the Voice selection drop-down list box is the active voice.
3. Click the active voice inside the drop-down list box, or use the arrow to display a list of available voices.
4. Click a new voice to select it.
5. The newly selected voice will speak using the text in **Preview Voice** box.
6. Click **OK** or **Apply** to accept the new voice.

To change the text-to-speech voice rate follows the following steps:

1. In control panel click on **Speech applet**.
2. Select the **Text-to-Speech** tab.
3. Move the **Voice speed** slider to change the rate of the text-to-speech voice. By default it is set to normal.

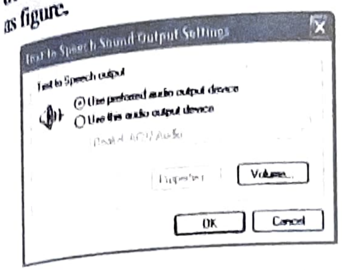




4. Click **Preview Voice** to hear the presently selected voice at the new rate; the text will be spoken, highlighting the words as they are spoken.

To select an audio output device follows the following steps:

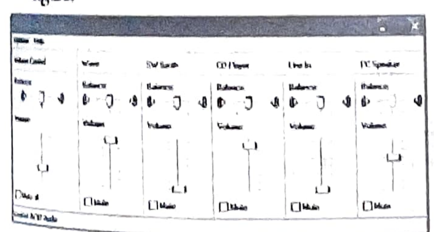
1. In control panel click on **Speech applet**.
2. On the **Text-to-Speech** tab, click **Audio Output**.
3. Select either **Use preferred audio output device** or **Use this audio output device** as figure.



To change the text-to-speech volume follows the following steps:

To adjust the volume output levels, follow the procedure below. Hoyouver, not all devices support this option in the same way. Some devices will not support volume control and the Volume button will be unavailable. Other devices may use their own display. In those cases, follow the instructions presented on the screen or documented separately with the engine.

1. In control panel click on **Speech applet**.
2. On the **Text-to-Speech** tab, click **Audio Output**, and then click **Volume**.
3. A volume control mixer will be displayed as figure.



4. Adjust the appropriate device to the required level.

Speech recognition overview:

Speech recognition (SR) is the ability of the operating system to convert spoken words to written text. An internal driver, called an SR engine, recognizes words and converts them to text. The SR engine may be installed with the operating system or at a later time with other software. During the installation process, speech-enabled packages such as word processors and youb browsers, may install their own engines or they can use existing ones. Additional engines are also available through third party manufacturers. These engines often use a certain jargon or vocabulary; for example, a vocabulary specializing in medical or legal terminology. They can also use different voices allowing for regional accents such as British English, or speak a different language altogether such as German, French or Russian.

You need a microphone or some other sound input device to receive the sound. In general, the microphone should be a high quality device with noise filters built in. The speech recognition rate is directly related to the quality of the input. The recognition rate will be significantly loyour or perhaps even unacceptable with a poor microphone. The Microsoft Speech Recognition Training Wizard (Voice Training Wizard) guides you through the process and recommends the best position to place the microphone allowing you to test it for optimal results.

Once you have installed the system and it is working, it is important to train it for your environment and speaking style. On the Speech **Recognition** tab, click **Train Profile** and use the Voice Training Wizard to train the system to recognize background noises such as a fan, the hum of air conditioning, or other office sounds. It adapts to your speaking style including accents, pronunciations and even idiomatic phrases.

Speech Recognition Tips

Speech recognition is not designed for completely hands-free operation; you'll get the best results if you use a combination of your voice and the mouse or keyboard. Also a consistent quality of speech results in the best results. When speaking to others, you usually understand from the context and environment even when whispered, shouted, or talking quickly or slowly. However, speech recognition understands words best when spoken in a more predictable manner.

- Speak in a consistent, level tone. Speaking too loudly or too softly makes it difficult for the computer to recognize what you've said.
- Use a consistent rate, without speeding up and slowing down.
- Speak without pausing betyouen words; a phrase is easier for the computer to interpret than just one word. For example, the computer has a hard time understanding phrases such as, "This (pause) is (pause) another (pause) example (pause) sentence."
- Start by working in a quiet environment so that the computer hears you instead of the sounds around you, and use a good quality microphone. Keep the microphone in the same position; try not to move it around once it is adjusted.
- Train your computer to recognize your voice by reading aloud the prepared training text in the Voice Training Wizard. Additional training increases speech recognition accuracy.
- As you dictate, do not be concerned if you do not immediately see your words on the screen. Continue speaking and pause at the end of your thought. The computer will display the recognized text on the screen after it finishes processing your voice.
- Pronounce words clearly, but do not separate each syllable in a word. For example, sounding out each syllable in "e-nun-ci-ate," will make it harder for the computer to recognize what you've said.

Speech recognition requirements

To use speech recognition, you need the following hardware and software components:

- A high-quality close-talk (headset) microphone. A Universal Serial Bus (USB) microphone is recommended.
- A 400 megahertz (MHz) or faster computer with 128 megabytes (MB) of memory.
- Microsoft Internet Explorer 5.0 or later.
- The speech recognition engine must be installed. It is available with Microsoft Office XP, but might not have been installed. For more information, click **Related Topics**.

Important

- Speech recognition engines are language-specific. The first three Microsoft speech engines that are available are Simplified Chinese, U.S. English, and Japanese. Engines for other languages will become available.

> ◎ If you are using a tablet computer, you can use Tablet PC Input Panel for handwriting and speech tasks. For more information about Input Panel, click **Related Topics**.

Using the Microsoft speech recognition engine

The Microsoft speech recognition engine enables you to insert text into a document using specific programs. You can dictate text in any Microsoft Office XP program, in Internet Explorer 5 or later, and in Outlook Express 5.0 or later. Other software programs might eventually support the Microsoft speech recognition engine. You cannot dictate text in Notepad at this time.

Along with being language-specific, some speech engines may be region-specific. For example, the Microsoft English ASR Version 5 engine is intended for speakers of U.S. English. British, Australian, and other non-U.S. English speakers may have difficulty using it due to variations in accent.

To use speech recognition, you should have a high-quality close-talk (headset) microphone and a sound card or USB port. You should be able to position the microphone so it is close to your mouth. Use the Microphone Wizard to configure your microphone.

It is important to train the speech recognition engine to understand your voice. As you read the training text aloud, the engine looks for patterns in the way you speak to help it interpret the words you say. Training creates a speech profile for the individual speaker.

Speech recognition is not designed for completely hands-free operation; you'll get the best results if you use a combination of your voice and the mouse and keyboard, or your voice and a handwriting input device.

SOUND STANDARDS ON PC

Todays most multimedia applications use audio in the form of music and/or speech. MIDI stands for *Musical Instrument digital interface*. MIDI is an international standard for music (sound) developed in the early 1980. This sound standard determines the cabling, hardware and communications protocol needed to connect a computer with electronic musical instruments and recording equipment. In otherwords we can say that the MIDI standard define how to code all the elements of musical scores, such as sequences of notes, timing conditions and the instrument to play each note. The MIDI language in binary form. Each word describing an action of a musical performance is assigned a specific binary code. Microsoft introduce a set of MIDI standards with its windows 3.1 operating system that have ultimately become the industry norms and are known as "The General MIDI standards". There are mainly two (2) different components of a MIDI interface. These are as follows :

(i) **Hardware to connect the equipment :** The physical connection of musical instruments are specified by MIDI hardware. It adds a MIDI ports to an instrument, it specifies a MIDI cable, and processes electrical signals received over the cable.

(ii) **A data format that encodes information to be processed by the hardware :** The MIDI data format does not include the encoding of individual sampling values such as audio data formats. Instead, MIDI uses a specific data format for each instrument, describing things like the start and end of score, the basis frequency and loudness, in addition to the instrument itself.

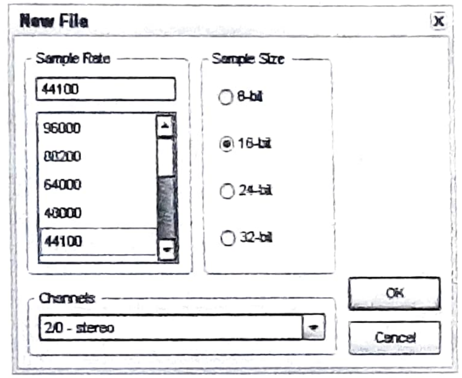
CAPTURING AND EDITING SOUND ON PC

Operations with Files

When you use **AVS Audio Editor** you will need to select some files, load them into the program, view their information and save them into different formats.

Creating a New File

Select the **New...** option from the **Main Menu** or the Create New File button of the Shortcut Bar. The following dialog box will be displayed:



Select the Sample Rate (8'000 - 96'000 Herz), Sample Size (8-bit - 32-bit) and the Number of Channels (1/0 - mono -- 3/4+LFE 7.1 DolbyProLogicIIx) and click the OK button to speedily create a file of the selected format.

> ◎ AVS Audio Editor creates all files using a temporary audio file format and saves all the changes to the edited file only when you select to save them using one of the ways.

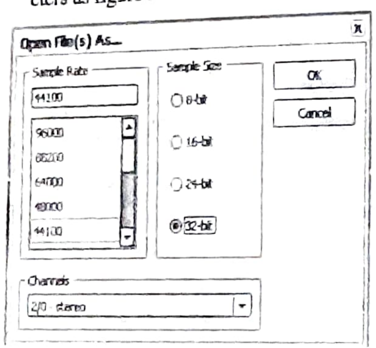
Opening Files

To start editing a file, you require to load it to AVS Audio Editor. There are several ways to do it:

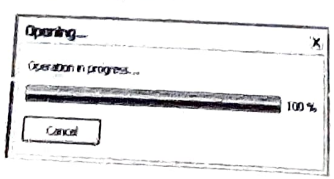
1. Select the Open... option from the Main Menu or the Open File button of the Shortcut Bar. The following Open file window will be displayed:

You can select a file or several files with your mouse or keyboard and click the Open button.

2. Select the Open as... option from the Main Menu. This option will allow you open the file as if it were of a different format than it actually is (therefore called "quick-and-easy" conversion method). After you select a file or several files in the window above, you will be offered to set the following parameters as figure :



3. Select the Sample Rate (8'000 - 96'000 Herz), Sample Size (8-bit - 32-bit) and the Number of Channels (1/0 - mono -- 3/4+LFE 7.1 DolbyProLogicIIx) and click the OK button to quickly convert the edited file into the selected format. Then the following dialog box appeared:



> The original file will not be converted unless you decide to save it. AVS Audio Editor opens all files into a temporary format and saves all the changes to the original file only when you select to Save them using one of the ways.

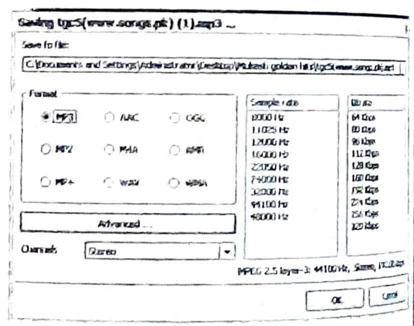
4. Select the Open and Append option of the Main Menu. This option will let you open the file and append it to the already opened one making one file out of two. The files will be merged into one with the second opened file appended to the end of the first opened one.

AVS Audio Editor supports all major audio formats. When you load a file for editing, AVS Audio Editor converts the waveform type to its own internal temporary file type for faster editing and better handling of large file sizes.

Saving Files

You can change the format of the edited file when you save it. It can be done to reduce the size of the audio file, to change the number of the channels or to make it possible to playback the file using mobile devices, such as mobile phones, portable players, etc.

To change the file format and save it using the selected format, you can press the Save File as... button of the Shortcut Bar or select the Save as... item of the Main Menu. Then the following dialog box appeared:



Here you can select the output file format and specify the output file format parameters such as Sample rate (Frequency), Bitrate and the number of Channels.

> Some formats, such as AAC, M4A, WMA and WAV allow you to select more than two channels. You can set up to eight channels depending on your desires and the devices that will be used to playback the resulting audio files.

> It is possible to specify Advanced MP3 parameters clicking the appropriate button. You can find the detailed information about these settings in the Appendix section.

After you select all the parameters, click the Save button to accept the changes and save the audio file or Close to discard the changes and close this window.

If you loaded several files into AVS Audio Editor and would like to save them all, you can use the Save All option of the Main Menu or the Save all button of the Shortcut Bar. It is also possible to save not the whole file but only the currently selected part of it using the Save Selection as... option of the Main Menu.

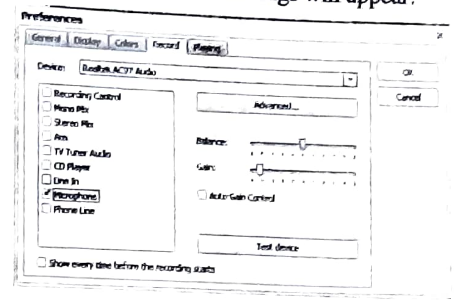
Playing and Recording:

In AVS Audio Editor you can operate Play and Record as well as other transport functions like you do on "real-world" tape recorders. Audio Player Toolbar is situated in the left bottom corner of the Main Window.

- Click Play to play the portion of the waveform that is currently being viewed, or the portion that is highlighted.
- Click Pause to pause the playback.
- Click Stop to end waveform playback.
- Click Record on the Audio Player Toolbar to start recording at the current insertion point. Any waveform data after that will be recorded over.

> The Record button will change into Stop - when you start the recording process. Just press it to stop the recording and return to the editing.

You can configure your input devices right from the AVS Audio Editor, without need to go to the Windows Control Panel. To do that you can select the Preferences... item in the Main Menu or the Preferences / Recording Settings buttons of the Shortcut Bar. Select the Record tab of the Preferences window. The form that will let you change the device settings will appear:

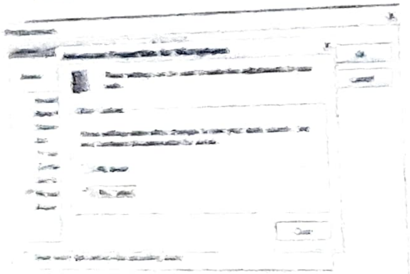


Here you can:

1. select the device that will be used for sound recording. Press the drop-down combo-box and select the necessary device from the list, if you have more than one input device installed on your computer;
2. select the input jack active on the device. The following input jacks might be available depending on your input device configuration:
   - **Mono Mix** - allows you to record the sound from a program player or a hardware tuner connected to your personal computer in mono mode;
   - **Stereo Mix** - allows you to record the sound from a program player or a hardware tuner connected to your personal computer in stereo mode;
   - **SPDIF** - allows you to record the sound from any external device connected to the digital input jack of your computer sound card;
   - **Aux** - allows you to record the sound from any external device connected to the Aux (auxiliary) input of your computer sound card;
   - **CD Player** - allows you to record the sound from a laser audio disc in you computer CD/DVD-ROM drive;
   - **Line In** - allows you to record the sound from any external device connected to the Line In input of your computer sound card;

- **Microphone** - allows you to record the sound from a microphone connected to the Microphone input of your computer sound card;
- **Phone Line** - allows you to record the sound from an external device connected to the Phone Line input of your computer sound card;

3. **Set the device Balance** - the difference of the sound volume between the right and the left channel.
4. **Set the input device Gain** - the loudness of the device input.
5. **Test the device pressing** - the Test button to make sure that the device is in working order and the Balance and Gain are set correctly.
6. For some devices it is also possible to change some advanced settings clicking the Advanced... button. The Advanced Properties window will pop up to let you configure the advanced device settings. You should consult your hardware documentation for more details on these settings:



After you select all the settings for your input device you can click the OK button to accept the changes made and go on recording the sound from the selected and configured device.

Editing Options

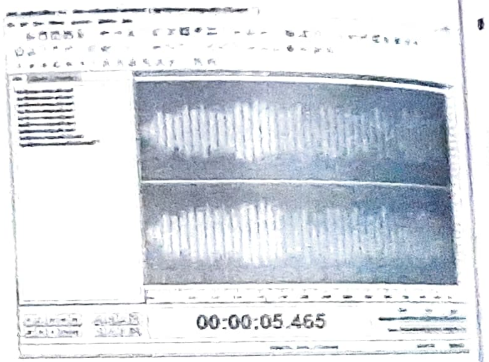
AVS Audio Editor lets the user perform simple editing operations - copy, paste and delete - with audio files. You can also add markers to your audio file to mark the important moments and simplify the navigation through the file and convert sample type of your audio file, changing its Sample Rate, Sample Size and the Number of Channels. See the appropriate chapters for more detail.

Copying

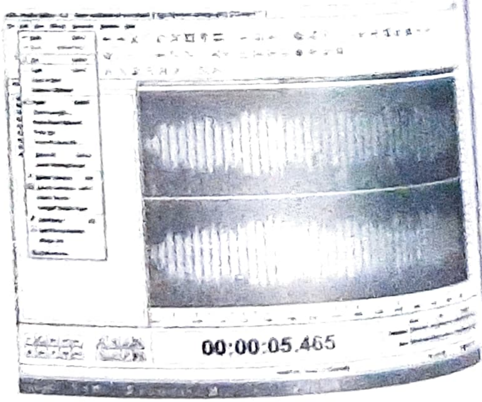
We can perform the following Copying operation in AVS Audio Editor:

- **Cut** - this optionis used to cut the currently selected part of the edited audio and copy it to the program internal clipboard. This option only allows you to cut the selected part of the audio, thus if no part of the audio file has been selected nothing will be cut and copied. When you cut a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in **AVS Audio Editor** using one of the **Paste** options.

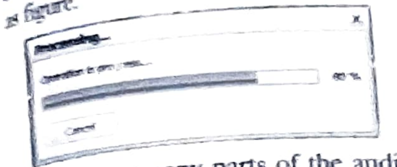
1. To copy any desired parts of any audio file first of all select that part as following figure.



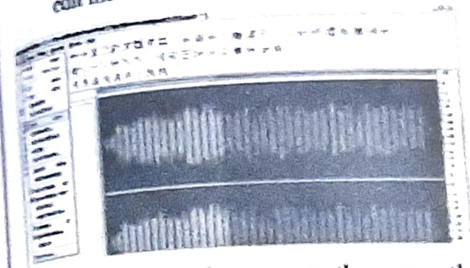
2. Then click on the cut option from the edit menu or Ctrl+x as figure.



3. After clicking on the cut option the processing... message dialog box appear as figure.

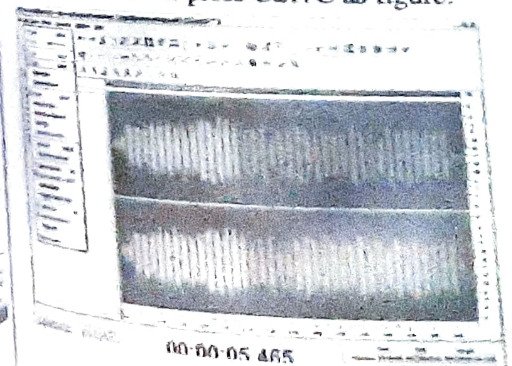


4. To paste your copy parts of the audio file, click on the paste option from the edit menu or press Ctrl+V as figure.

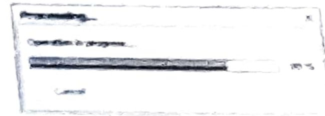


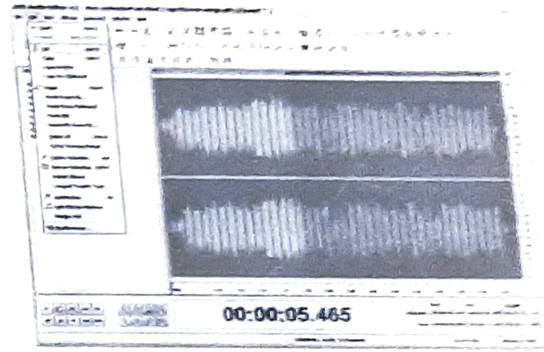
- **Copy** - use this option to copy the currently selected part of the edited audio to the program internal clipboard. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied. When you copy a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in **AVS Audio Editor** using one of the **Paste** options.

1. To copy any desired parts of any audio file first of all select that part as following figure.
2. Then click on the copy option from the edit menu or press Ctrl+C as figure.

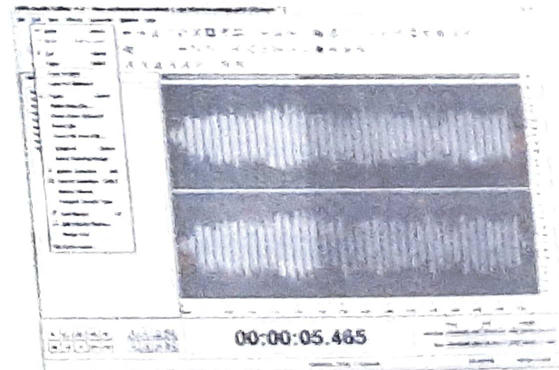


3. After clicking on the cut option the processing... message dialog box appear as figure.

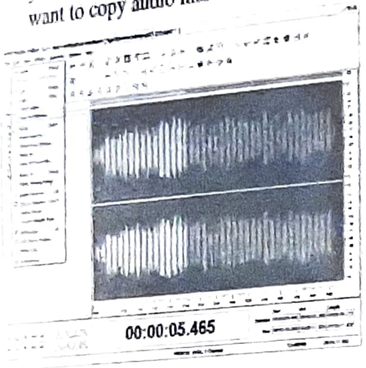




- **Copy to New** - use this option to copy the currently selected part of the edited audio to a new file. The new file will be created automatically and will have the same parameters (sample rate, sample size and the number of channels) as the currently edited file. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied.



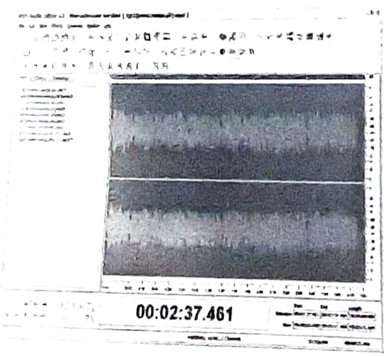
- **Copy to Clipboard** - use this option to copy the currently selected part of the edited audio to the common windows clipboard. This option only allows you to copy the selected part of the audio, thus if no part of the audio file has been selected nothing will be copied. When you copy a part of the audio you can insert it afterwards to some other place of the same audio or to a different audio, opened in AVS Audio Editor using one of the Paste options. This option is particularly useful when there are several instances of AVS Audio Editor opened and you would like to copy some audio between them or if you additionally use some other editor and want to copy audio into it.

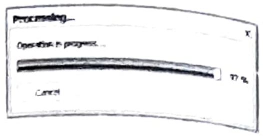


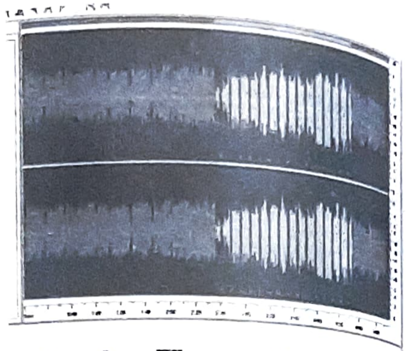
Pasting

The following Pasting operation are available in AVS Audio Editor:

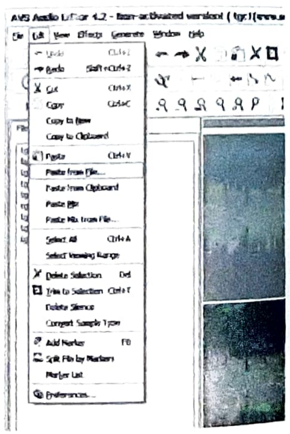
- **Paste** - use this option to paste (insert) the previously copied audio fragment to the currently edited audio file. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.

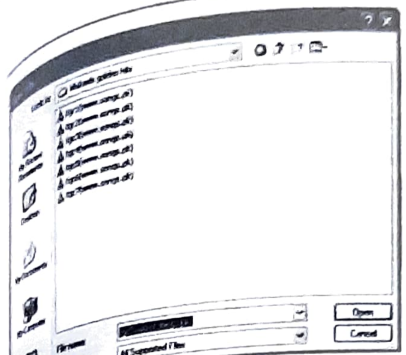


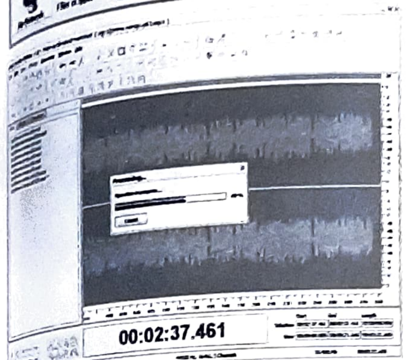


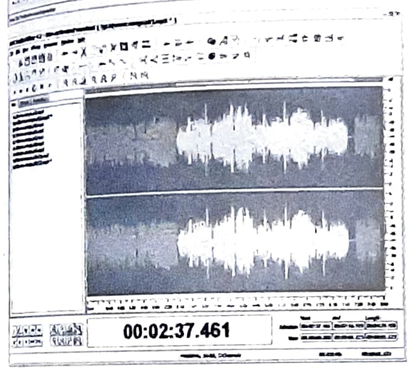


- **Paste from File** - - use this option to paste (insert) an audio file to the currently edited audio file. After you select this option the Open File window will be opened to let you select an audio file. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste an audio of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.





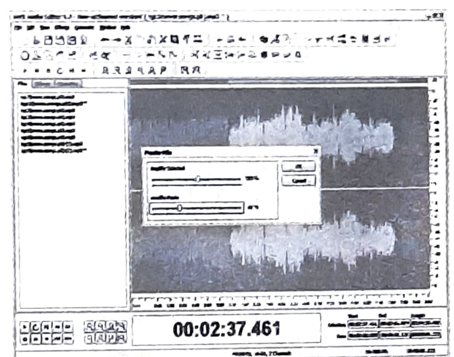


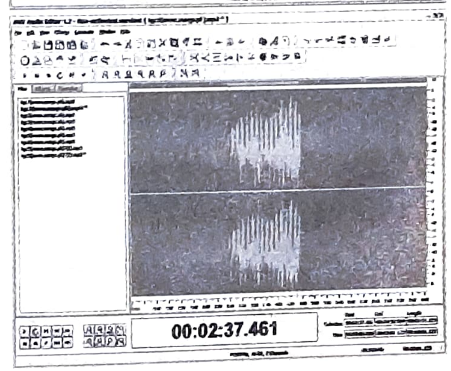


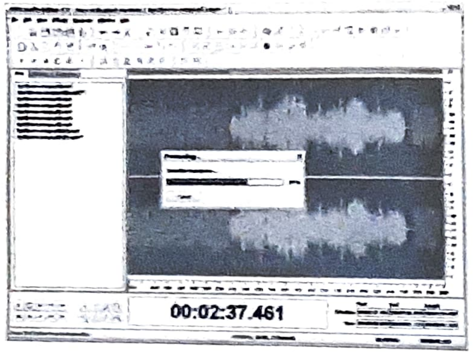
- **Paste from Clipboard** - - use this option to paste (insert) the previously copied audio fragment to the currently edited audio file. This option is particularly useful when there are several instances of AVS Audio Editor opened and you would like to copy and paste some audio between them or if you additionally use some other editor and want to paste audio from it. It will be inserted right from the cursor position expanding the file to make it fit the inserted fragment. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be 3 minutes 20 seconds. If some part of the file is selected prior to the paste operation, it will be overwritten.

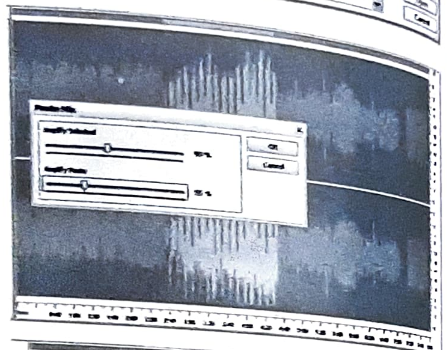
> to use this option with more than one instance of **AVS Audio Editor** you will need to use the **Copy to Clipboard** in one of them and Paste to Clipboard in the other, so that the selected fragment could be available to all the instances of the program.

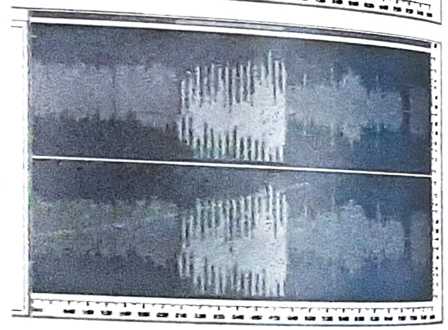
- **Paste Mix** - use this option to paste (insert) the previously copied audio fragment to the currently edited audio file. It will be inserted right from the cursor position overwriting the original audio and not changing its duration. For instance, if you have a three-minute file and paste a copied fragment of 20 seconds, the resulting audio will be still 3 minutes. Before you paste some audio, you can change the following parameters:





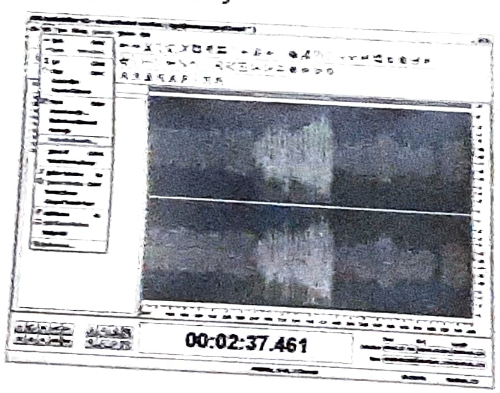






Set Amplify Selected option to change the current audio file or selection amplification in relation to the rest of the audio and Amplify Paste to set the loudness of the audio in the inserted mix. If some part of the file is selected prior to the paste operation, the Paste Mix operation will be applied to it only.

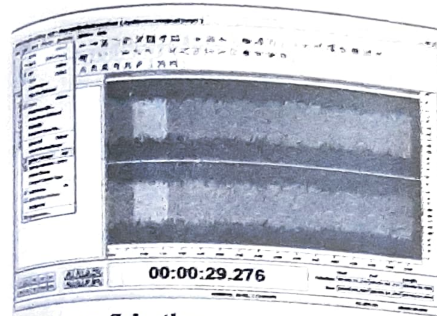
- **Paste Mix from File** - use this option to paste (insert) an audio file to the currently edited audio file. After you select this option the Open File window will be opened to let you select an audio file. It will be inserted right from the cursor position overwriting the original audio and not changing its duration. For instance, if you have a three-minute file and paste an audio of 20 seconds, the resulting audio will be still 3 minutes. If some part of the file is selected prior to the paste operation, the Paste Mix operation will be applied to it only.



Deleting and Trimming

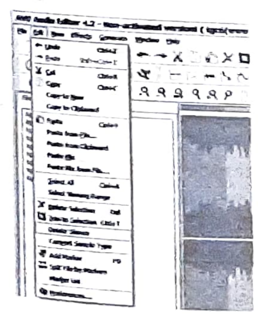
The following Deleting and Trimming operations are available in AVS Audio Editor:

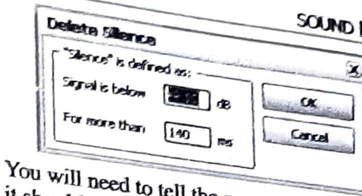
- **Delete Selection** - use this option to delete the currently selected fragment of the audio file. This option only allows you to delete the selected part of the audio, thus if no part of the audio file has been selected nothing will be deleted.



- **Trim to Selection** - use this option to delete the all the audio in the file except the currently selected fragment. It is the opposite to the simple Delete Selection option. This option only allows you to delete some part of the audio if there is a selected part, thus if no part of the audio file has been selected nothing will be deleted.

- **Delete Silence** - use this option to delete all the moments of silence in the audio. When you select this option the following window will be opened:

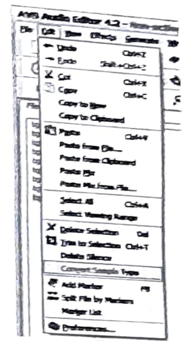


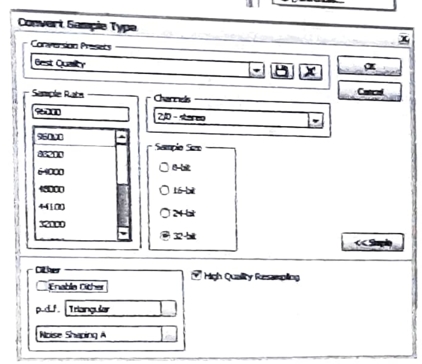


You will need to tell the program what exactly it should consider as moments of silence - you need to set the strength of the signal in decibels (for silence usually negative values are used) and its duration (in milliseconds). If some part of the audio is selected, the instances of silence will be deleted only in this selection.

Convert Sample Type

To convert the sample-rate, bit resolution, and channel format of an audio file that is currently loaded in **AVS Audio Editor** into a new format type (such as 44KHz/16-bit/stereo to 22KHz/8-bit/mono) use the **Convert Sample Type** option of the **Main Menu.**





Convert Sample Type directly processes samples within the file, or re-samples the data, so that the audio retains the same frequency and duration as the original file.

Every aspect of conversion can be adjusted to suit your needs. You can set the necessary **Sample Rate** (8'000 - 96'000 Herz), **Sample Size** (8-bit - 32-bit) and the **Number of Channels** (1/0 - mono -- 3/4+LFE 7.1 DolbyProLogicIIx).

You can also use the advanced options of the resampling clicking the Advanced >> button (it will change into << Simple once it is pressed). After you open the advanced settings panel of the Convert Sample Type window you will be able to set the following parameters:

- **Enable Dither** - use this option to apply an intentional form of noise, used to randomize the resampling errors thus preventing audio distortions. When you choose to enable dither, you can select the type of Probability Density Function (p.d.f.) used - Triangular, Gaussian, Uniform or Colored differing in type of randomizing in noise creation. It is also possible to change the Noise Shaping type - A or B - which is not actually dither, but rather a feedback process that has dither within it although it is used for the same purposes.
- **High Quality Resampling** - use this option to enable the high quality resampling, that will allow you to maximally avoid distortions. It will take more time and processor power than the sample type conversion without this option enabled.

OVERVIEW AND USING SOME SOUND RECORDING

Sound Recorder overview

Using Sound Recorder, you can record, mix, play, and edit sounds. You can also link or insert sounds into another document. You can modify an uncompressed sound file by:

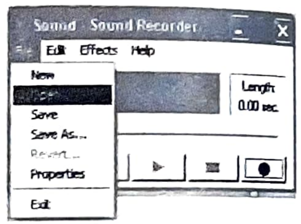
- Adding sounds to a file.
- Deleting part of the sound file.
- Changing the playback speed.
- Changing the playback volume.
- Changing the playback direction.
- Changing or converting the sound file type.
- Adding an echo.

- To record sounds, your computer must be equipped with a microphone.
- Recorded sounds are saved as wave form (.wav) files.

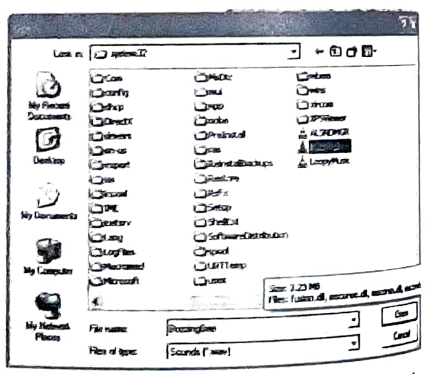
Using some sound recording

To record a sound into a sound file follow the following steps

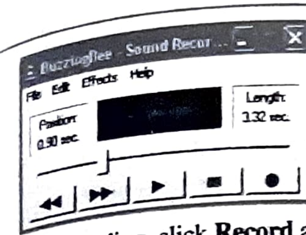
1. Make sure you have an **audio input device** connected to your computer.
2. Click on the **Open** option from the File menu **as figure**.



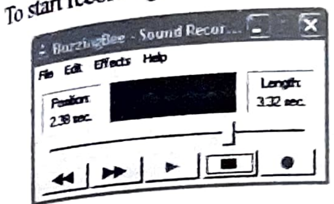
3. Then the open dialog box appears as figure.



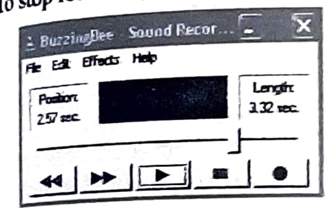
4. In the **Open** dialog box, double-click on the **sound file** you want to modify.
5. Move the slider to the place in the file where you want to record sound as figure.



6. To start recording, click **Record as figure**.

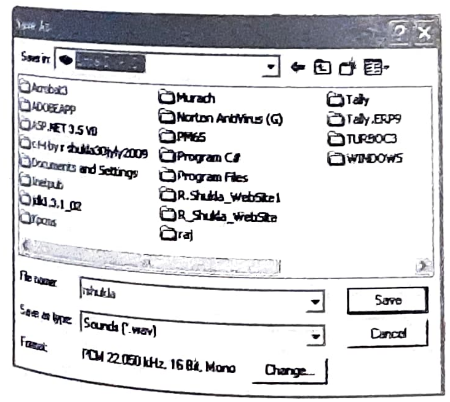


7. To stop recording, click **Stop**



To save these recorded fragments of the file follow the following steps:

1. Click on the **Save As** option from the file menu.
2. Then **Save As** dialog box appears as figure.
3. Select your desired drive to save your file from Save In drive list box.
4. Type your sound file name in File name text bos (rshukla) as figure.



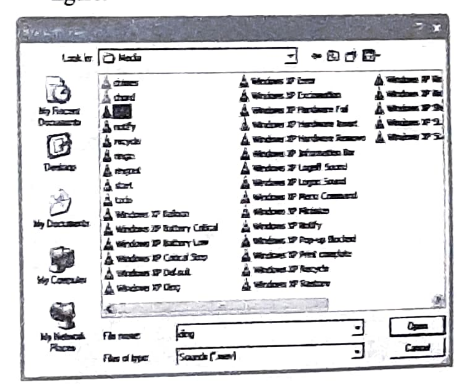
5. Finally click on the Save button to save your file.

To play a sound follows the following steps:

1. Click on the Open option from the **File** menu.
2. In the **Open** dialog box, double-click the sound file you want to play.
3. Click **Play** to start playing the sound.
4. Click **Stop** to stop playing the sound.

To overlay (mix) sound files follow the following steps:

1. Click on the Open option from the **File** menu.
2. In the **Open** dialog box, double-click the **sound file** you want to modify.
3. Move the slider to the place in the file where you want to overlay the sound file.
4. On the **Edit** menu, click **Mix with File**. Then Mix With File dialog box appears as figure.



5. Double-click the name of the file you want to mix.

To insert a sound file into a document

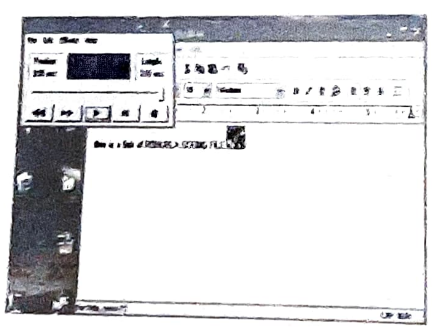
1. Click on the Open option from the **File** menu.

2. In the **Open** dialog box, double-click the **sound file** you want to insert.
3. On the **Edit** menu, click **Copy**.
4. Using a word processing program such as WordPad, open the document into which you want to copy the sound, and then click where you want to insert the sound.
5. On the **Edit** menu, click **Paste**.

To link a sound file to a document

1. Click on the Open option from the File menu.
2. In the **Open** dialog box, double-click the **sound file** you want to **link**.
3. On the **Edit** menu, click **Copy**.

4. Using a word processing program such as WordPad, open the document into which you want the link to the sound, and then click where you want to insert the sound.
5. On the **Edit** menu, click Paste Special.
6. In the **Paste Special** dialog box, click **Paste Link**, and then click **OK**.



SOUND EDITING SOFTWARE

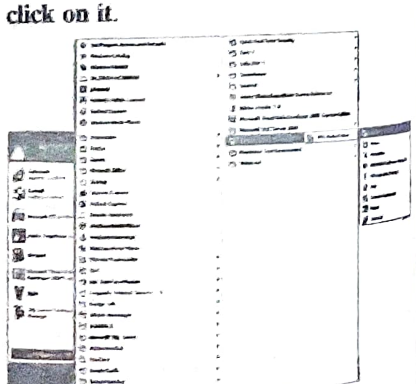
Following are the list of sound editing software.

1. AVS Audio Editor 2. Sound Forge

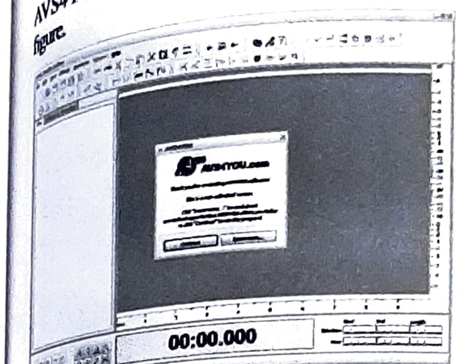
1. AVS Audio Editor:

AVS Audio Editor is a powerful, full-featured and easy to use digital audio editor. AVS Audio Editor will be of interest to professionals and amateurs, it is very easy to use, and it lets you perform a lot of operations without any difficulties. Once you get started you will be surprised to see the range of possibilities this program offers to you. Therefore what are the main functions of **AVS Audio Editor**? Assume you have recorded some audio data, at this time you can do whatever you want with it: cut it, copy, paste, and move- the same operations that you can do with the text in a word processor. In case you make a mistake, press the **Undo** button, and try again. Friendly interface allows you to perform lots of different operations in an easy way. You can use it to record your own music, voice or other audio material, edit it, mix it with other audio or musical parts, add various effects to it, and master it so that you can burn it onto a CD, post it on the World Wide Web (WWW), or e-mail it. **AVS Audio Editor** supports all major audio file formats. **AVS Audio Editor** has a great variety of audio effects and tools: **Delay**, **Flanger**, **Reverb**, **Phaser**, **Amplify** and lots of others.

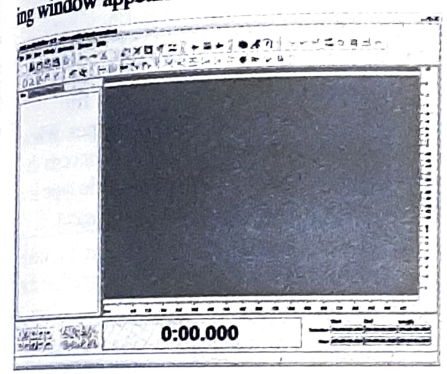
To start AVS **Audio Editor** clicks on Start menu and choose **All Programs** → AVS4YOU → **Audio** → **AVS Audio Editor** as figure and **click on it.**



Then AVS Audio Editor Application window with AVS4YOU dialog box will be appeared as figure.

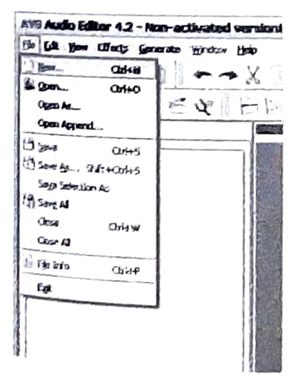


After clicking on the continue button the following window appears:

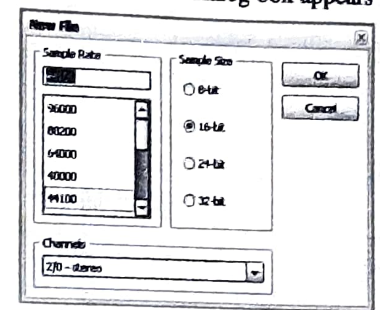


To create a New File follows the following steps:

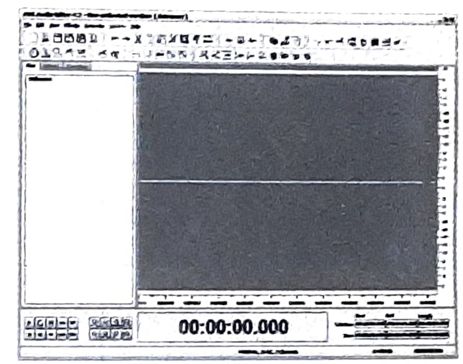
1. Go to the file menu and select new option or press Ctrl+N from the keyboard as following figure.



2. Then the New File dialog box appears as:



3. Click on the ok button to create a new file that has a default name **UnKnown as figure.** All the newly created files will have the **Unknown** name until you save them and assign a name to them.



Main Window (PROGRAM INTERFACE)

AVS Audio Editor main window contain the following SIX (6) parts:

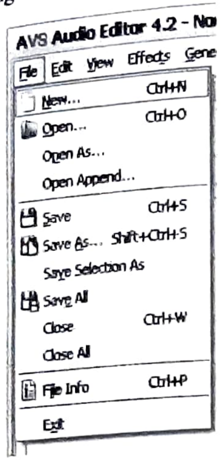
1. Main Menu (MENU BAR)
2. Shortcut Bar (TOOL BAR)
3. Effects and Filters Panel
4. Waveform Editing Space
5. Bottom Toolbar
6. Status Bar

1. Main Menu

AVS Audio Editor can be operated with the help of **Main Menu** elements. Flexible system of menu elements is a perfect tool for navigating and operating the application, controlling all the processes.

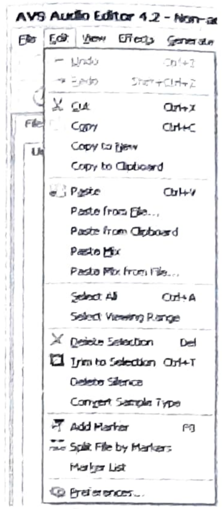
The Main Menu has the following structure:

1. File menu section



Option/Command	Shortcut	Functions
New	Ctrl+N	This option is use to create a new audio file.
Open	Ctrl+O	This option is use to open an existing audio file. AVS Audio Editor supports a wide variety of data types. When you load a file for editing, AVS Audio Editor converts the audio file type to its own internal temporary file type for faster editing and better handling of larger file sizes.
Open as...		This option is use to open an existing audio file as if it were a file of some other format. See the Open as section for more detail on it.
Open and Append...		This option is use to open an existing audio file and append it to the end of the previously opened file. The newly created file will consist of two opened files at once, the second one starting were the first one ends.
Save	Ctrl+S	This option is use to save the active audio file with its current file name and location.
Save as...	Shift+Ctrl+S	This option is use to save the active audio file with a different file name and/or location, in a different file format.
Save Selection as...		This option is use to save just the highlighted selection to a file.
Save All		This option is use to save all the changes to all the opened edited files.
Close	Ctrl+W	This option is use to close the currently opened edited file.
Close All		This option is use to close all the opened edited files.
File Info	Ctrl+P	This option is use to view the available audio file info.
Exit		This option is use to terminate the program and exit.

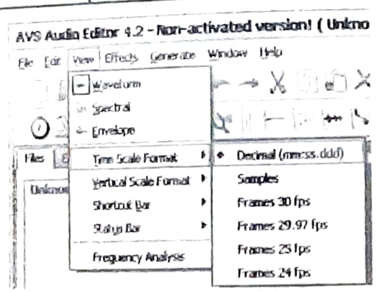
2. Edit menu section



Option/Command	Shortcut	Functions
Undo	Ctrl+Z	This option is use to reverse the last command, such as deletion, transforms, etc. If this option is not available, it means there is no action to undo. AVS Audio Editor allows virtually unlimited (limited only by hard drive space) levels of Undo.
Redo	Shift+Ctrl+Z	This option is use to repeat your last command or action.
Cut	Ctrl+X	This option is use to remove the selection from the active file and place it to the program internal clipboard.
Copy	Ctrl+C	This option is use to copy the selection into the program internal clipboard. It is used when you work with one file within one copy of AVS Audio Editor.
Copy to New		This option is use to copy the selection or the whole audio file into a new one. A new file will be automatically created and opened.
Copy to Clipboard		This option is use to copy the selection into the windows clipboard so that it would be available to the other opened programs. It is used when you work with two or more copies of AVS Audio Editor and want to copy/paste from one program to the other.
Paste	Ctrl+V	This option is use to insert the contents of the internal clipboard at the insertion point or to replace any selection.
Paste from File...		This option is use to insert the contents of an audio file at the insertion point or to replace any selection.
Paste from Clipboard		This option is use to insert the contents of the windows clipboard at the insertion point or to replace any selection.

Paste Mix		This option is use to insert the contents of the internal clipboard at the insertion point mixing the audio tracks.
Paste Mix from File...		This option is use to insert the contents of the audio file at the insertion point mixing the audio tracks.
Select All	Ctrl+A	This option is use to make the selection of the whole audio track.
Select Viewing Range		This option is use to make the selection of the visible area of the audio track only (in case the waveform is zoomed in, only the area within the visible area will be selected).
Delete Selection	Del	This option is use to remove the current selection. The deleted portion is not copied to the clipboard, and can only be retrieved through Undo.
Trim to Selection	Ctrl+T	This option is use to delete everything except the selected portion (the exact opposite of Delete selection).
Delete Silence		This option is use to remove periods of silence between words or other sounds.
Convert Sample Type		This option is use to convert your opened audio to another format or change bitrate, sample rate, number of channels and so forth.
Add Marker	F8	This option is use to add a marker at the current position of the cursor in the audio file on the timeline.
Split File by Markers		This option is use to split the audio file into several separate files using the current markers at the timeline as the start and end points for these files.
Markers List		This option is use to open and manage the list of the set markers.
Preferences...		This option is use to view and edit the program preferences.

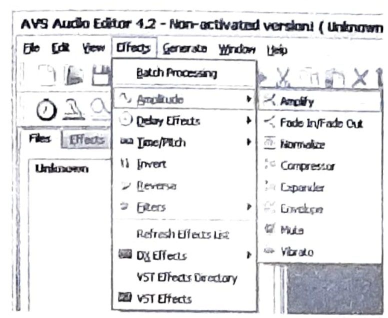
3. View menu section



Option/Command	Functions
Waveform	This option is use to choose a **Waveform View** mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time

Spectral	This option is use to choose a **Spectral View** mode for viewing data. It displays a waveform by its frequency components, where x-axis (horizontal) is time, and y-axis (vertical) is frequency. This allows you to analyze your audio data to see which frequencies are most prevalent.
Envelope	This option is use to choose a **Envelope View** mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time. It is the same as the **Waveform View** but its boundaries are smoothed.
Time Scale Format	This option is use to change the format of the horizontal scale which represents the timeline of the audio file. The possible values include: **Decimal (mm:ss.ddd), Samples, Frames 30 fps, Frames 29.97 fps, Frames 25 fps** and **Frames 24 fps.**
Vertical Scale Format	This option is use to change the format of the vertical scale which represents the changes of the audio in the file within time. The possible values include: **Sample Values, Percentage** and **Decibels.**
Shortcut Bar	This option is use to show or hide the elements of the **Shortcut Bar**. You can select to show or hide all the elements of the **Shortcut Bar** using **the Show** option, or select each element separately.
Status Bar	This option is use to show or hide the elements of the **Status Bar**. The possible items include: **Sample Format, File Size** and **Duration (mm:ss.ddd).**
Frequency Analysis	This option is use to show or hide **Frequency Analysis Window**. The **Frequency Analysis Window** contains a graph of the frequencies at the insertion point or at the center of a selection.

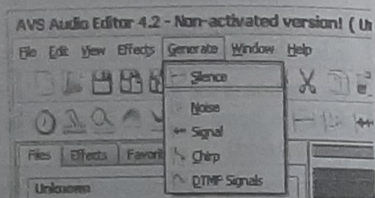
4. Effects menu section



Option/Command	Functions
Batch Processing	This option is use to apply several effects to one or several audio files at once.
Amplitude	This option is use to apply the effects included into the *Amplitude* effects group. They are: *Amplify, Fade In/Fade Out, Normalize, Compressor, Expander, Envelope, Mute and Vibrato.*
Delay Effects	This option is use to apply the effects included into the *Delay Effects* group. They are: *Delay, Phaser, Flanger, Chorus and Reverb.*

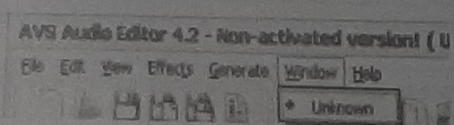
Time/Pitch	This option is use to apply the effects included into the Time/Pitch effects group. They are: *Time Stretch* and *Pitch Shift*.
Invert	This option is use to apply the *Invert* effect to the audio.
Reverse	This option is use to apply the *Reverse* effect to the audio.
Filters	This option is use to apply the effects included into the Filters group. They are: *Low/High/Band Pass, Notch Filter, Low/High Shelf, Peak EQ Filter, Noise Remover, Equalizer, FFT Filter and FIR Filter.*
Refresh Effects List	This option is use to refresh the effects list in case some effects became available. It is usually applicable to newly added *DirectX filters*.
DX Effects	This option is use to list DirectX filters installed on your system and apply them to the audio.
VST Effects Directory	This option is use to add or remove the VST effects loaded to AVS Audio Editor and available for audio editing.
VST Effects	This option is use to list VST effects installed on your system and apply them to the audio.

5. Generate menu section



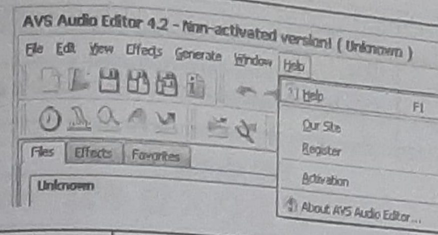
Option/Command	Functions
Silence	This option is use to create a period of silence of a certain length within the audio.
Noise	This option is use to create a noise signal of a certain type and length within the audio.
Signal	This option is use to create a signal with certain parameters within the audio.
Chirp	This option is use to create a chirp signal with certain parameters within the audio.
DTMF Signals	This option is use to create DTMF signals with certain parameters within the audio.

6. Window Menu



This menu contain list of all open file.

7. Help menu section

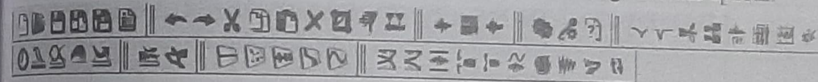


Option/Command	Shortcut	Functions
Help file	F1	This option is use to open the Help File window.
Our Site		This option is use to see the links to AVS4YOU.com on the web.
Register		This option is use to go online to the AVS4YOU.com web site and use the online pay system to buy our software.
Activation		This option is use to activate the program.
About AVS Audio Editor...		This option is use to open the window that shows the information about AVS Audio Editor.

- The Window menu section will let you switch between all the opened windows with the audio files loaded into AVS Audio Editor.

2. Shortcut Bar

The Shortcut Toolbar includes the following buttons for quick accessing of menus options:



In the table below you will find the functions you can perform with the help of each button:

File section

Symbol of Button	Name of Button	Functions
	Create New File	This button is used to create a new audio file.
	Open File	This button is used to open an existing audio file. AVS Audio Editor supports a wide variety of data types. When you load a file for editing, AVS Audio Editor converts the audio file type to its own internal temporary file type for faster editing and better handling of larger file sizes.
	Save File	This button is used to save the active audio file with its current file name and location.
	Save File as...	This button is used to save the active audio file with a different file name and/or location, in a different file format.

	Save All Files	This button is used to save all the changes to all the opened edited files.
	File Info	This button is used to view the available audio file info.

Edit section

Symbol of Button	Name of Button	Functions
	Undo	This button is used to reverse the last command, such as deletion, transformation, etc. If this option is not available, it means there is no action to undo. AVS Audio Editor allows virtually unlimited (limited only by hard drive space) levels of Undo.
	Redo	This button is used to repeat your last command or action.
	Cut	This button is used to remove the selection from the active file and place it to the program internal clipboard.
	Copy	This button is used to copy the selection into the program internal clipboard. It is used when you work with one file within one copy of AVS Audio Editor.
	Paste	This button is used to insert the contents of the internal clipboard at the insertion point or to replace any selection.
	Delete Selection	This button is used to remove the current selection. The deleted portion is not copied to the clipboard, and can only be retrieved through Undo.
	Trim to Selection	This button is used to delete everything except the selected portion (the exact opposite of Delete selection).
	Add Marker	This button is used to add a marker to the current cursor position.
	Split by Markers	This button is used to split the audio file into several separate files using the current markers at the timeline as the start and end points for these files.

Display section

Symbol of Button	Name of Button	Functions
	Waveform View	This button is used to choose a Waveform View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time.

	Spectral View	This button is used to choose a Spectral View mode for viewing data. It displays a waveform by its frequency components, where x-axis (horizontal) is time, and y-axis (vertical) is frequency. This allows you to analyze your audio data to see which frequencies are most prevalent.
	Envelope View	This button is used to choose a Envelope View mode for viewing data. It displays audio data in the familiar sound wave format, where y-axis (vertical) represents amplitude and x-axis (horizontal) represents time. It is the same as the Waveform View but its boundaries are smoothed.

Options section

Symbol of Button	Name of Button	Functions
	Preferences	This button is used to view and edit the program preferences.
	Recording Settings	This button is used to view and change the current recording device and its settings.
	Show Help	This button is used to open the Help File window.

Amplitude Effects section

Symbol of Button	Name of Button	Functions
	Amplify	This button is used to apply the Amplify effect to the audio.
	Fade In/Fade Out	This button is used to apply the Fade In/Fade Out effects to the audio.
	Normalize	This button is used to apply the Normalize effect to the audio.
	Compressor	This button is used to apply the Compressor effect to the audio.
	Expander	This button is used to apply the Expander effect to the audio.
	Envelope	This button is used to apply the Envelope effect to the audio.
	Mute	This button is used to apply the Mute effect to the audio.
	Vibrato	This button is used to apply the Vibrato effect to the audio.
	Reverse	This button is used to apply the Reverse effect to the audio.
	Invert	This button is used to apply the Invert effect to the audio.

Delay Effects section

Symbol of Button	Name of Button	Functions
	Delay	This button is used to apply the Delay effect to the audio.
	Phaser	This button is used to apply the Phaser effect to the audio.
	Flanger	This button is used to apply the Flanger effect to the audio.
	Chorus	This button is used to apply the Chorus effect to the audio.
	Reverb	This button is used to apply the Reverb effect to the audio.

Time/Pitch Effects section

Symbol of Button	Name of Button	Functions
	Time Stretch	This button is used to apply the Time Stretch effect to the audio.
	Pitch Shift	This button is used to apply the Pitch Shift effect to the audio.

Filters section

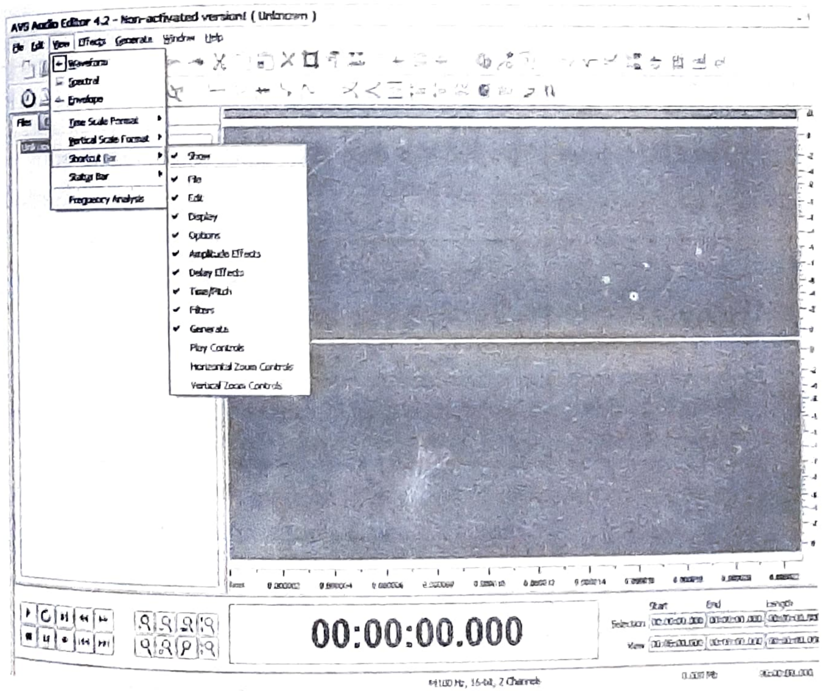
Symbol of Button	Name of Button	Functions
	Low/High/Band Pass	This button is used to apply the Low/High/Band Pass filter to the audio.
	Notch Filter	This button is used to apply the Notch filter to the audio.
	Low/High Shelf	This button is used to apply the Low/High Shelf filter to the audio.
	FFT Filter	This button is used to apply the FFT filter to the audio.
	FIR Filter	This button is used to apply the FIR filter to the audio.
	Equalizer	This button is used to apply the Equalizer filter to the audio.
	Peak EQ Filter	This button is used to apply the Peak EQ filter to the audio.
	Noise Remover	This button is used to apply the Noise Remover filter to the audio.

Generate section

Symbol of Button	Name of Button	Functions
	Generate Silence	This button is used to create a period of silence of a certain length within the audio.
	Generate Noise	This button is used to create a noise signal of a certain type and length within the audio.
	Generate Signal	This button is used to create a signal with certain parameters within the audio.
	Generate Chirp	This button is used to create a chirp signal with certain parameters within the audio.
	Generate DTMF Signals	This button is used to create DTMF signals with certain parameters within the audio.

Play Controls section

The play control section does not appear by default. If you want to add this Shortcut bar in Shortcut Toolbar then go to View ' Shortcut Bar 'Play Control option as following figure and click on it.



Symbol of Button	Name of Button	Functions
	Play	This button is used to start the playback of the current selection, or of the current audio file from either the left edge of the selection, or from the beginning of the file, to either the right edge of the selection or to the end of the file.
	Pause	This button is used to pause the playback.
	Stop	This button is used to stop the playback.
	Play Looped	This button is used to play the current audio file or selection repeatedly, looping it until the Stop button is pressed.
	Play to End	This button is used to start the playback of the current selection, or of the current audio file from either the left edge of the selection, or from the beginning of the file, to the end of the file.
	Record	This button is used to start the recording from the source selected in the AVS Audio Editor Parameters dialog.

Horizontal Zoom section

The Horizontal Zoom section does not appear by default. If you want to add this Shortcut bar in Shortcut Toolbar then go to View ' Shortcut Bar ' Horizontal Zoom controls option and click on it.

Symbol of Button	Name of Button	Functions
	Zoom In	This button is used to zoom in on the center of the current audio file window. After zooming, use the Time Ruler to scroll to the desired location.
	Zoom Out	This button is used to zoom out from the current location.
	Full Zoom	This button is used to zoom all the way out to fit the entire waveform or session in the display window.
	Zoom in to Left Edge of Selection	This button is used to zoom in to the left edge of the current selection.
	Zoom to Selection	This button is used to zoom to the current selection. If no selection is made, this button zooms in on the cursor location.
	Zoom in to Right Edge of Selection	This button is used to zoom in to the right edge of the current selection.

Vertical Zoom section

The Vertical Zoom section does not appear by default. If you want to add this Shortcut bar in Shortcut Toolbar then go to View ' Shortcut Bar ' Vertical Zoom Controls option and click on it

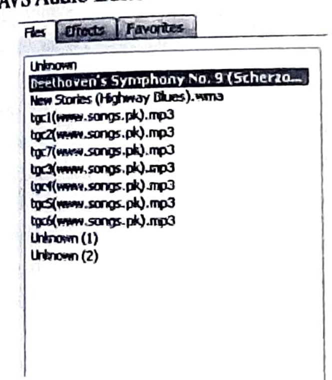
Symbol of Button	Name of Button	Functions
	Vertical Zoom in	This button is used to increase the vertical resolution scale of the waveform and Vertical Scale.
	Vertical Zoom Out	This button is used to decrease the vertical resolution scale of the waveform and Vertical Scale.

You can select to show or hide the Shortcut Bar using the View section of the Main Menu or clicking the right mouse button within the Shortcut Bar area and selecting the appropriate panels.

Effects and Filters Panel

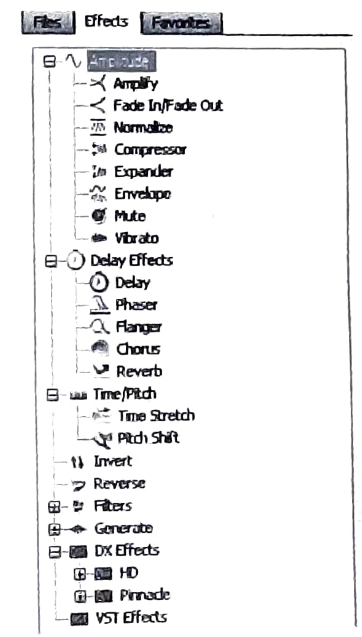
Effects and Filters Panel is a universal easy-to-use tool to perform many different operations with audio files. Using this panel you can add, remove files and apply various effects and filters to the opened files.

If you select the Files tab of the panel, you will be able to see the list of the files currently loaded into AVS Audio Editor:



All the newly created files will have the Unknown name until you save them and assign a name to them. You can add new files to the list, remove the unwanted ones from the list or create new files using the Main Menu or Shortcut Bar.

To start editing the files loaded into AVS Audio Editor you will need to select the Effects tab:



You can create lots of presets for every Effects and Filters Panel option or use the already existing ones.

The Effects tab menu consists of the following submenus:

(i) Amplitude Effects
(ii) Delay Effects
(iii) Time/Pitch Effects